Electronics for Analog Signal Processing - I
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Lecture - 18
Frequency Limitations of an Amplifier

In the last class, we were exposed to the frequency limitation of amplifiers. Let us consider this example now, which illustrates how the so called gain characteristic of the amplifier gets limited.

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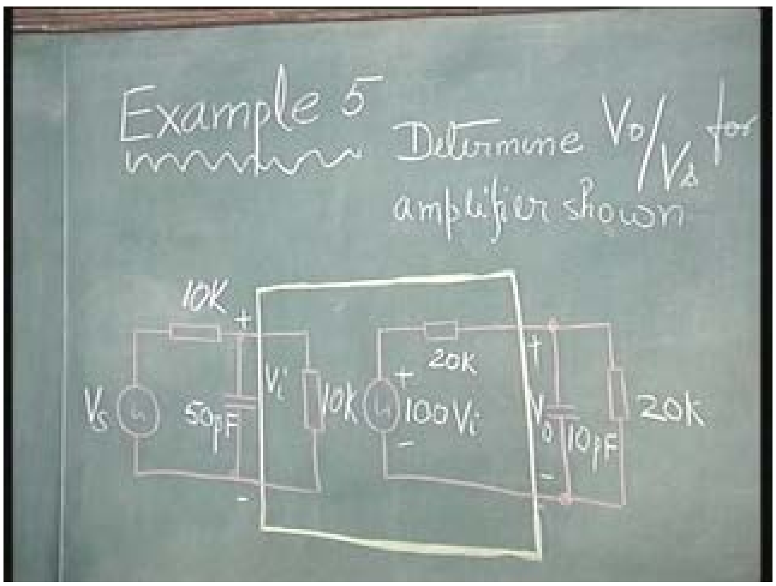

Determine V naught over V s, V naught over V s, that is, the amplifier gain for the amplifier shown, wherein, apart from the usual input impedance, source impedance, output impedance and the load impedance, we have the effect of stray capacitance; one at the output, 10 picofarad, one at the input, 50 picofarad. The input capacitor, total together, comes out to be 50, let us say; and the output capacitor comes out to be 10 picofarads. This is the typical order of the amplifier capacitors.

Now, under this circumstance, I would like to know V naught over V s. Now, you can write down the different group equations, etcetera and solve it; but, I would like to solve this by just observation.

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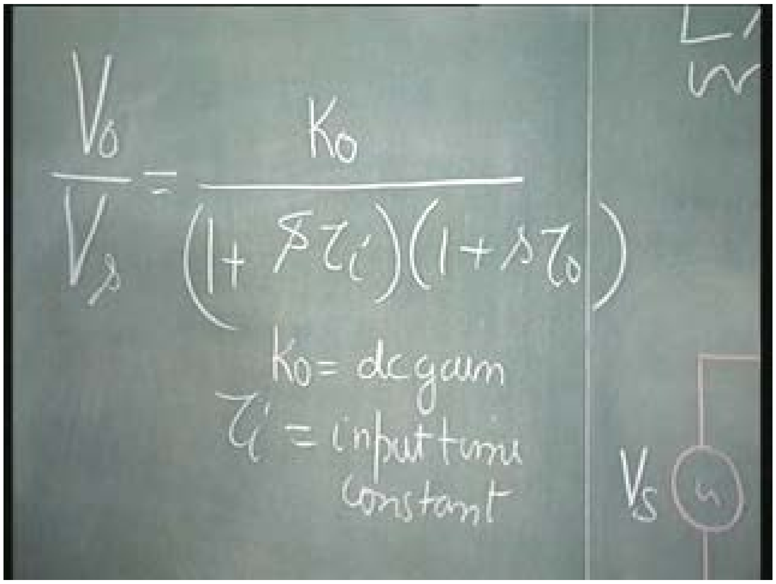

In all these cases, I have told you that the amplifier gain comes out as K zero, which is independent of frequency, divided by 1 plus s tau i, 1 plus s tau naught. This is the input time constant, tau i is the input time constant; K naught is the dc gain. I discussed, independent of frequency, dc gain; tau i is the input time constant, and tau naught is nothing but the output time constant.

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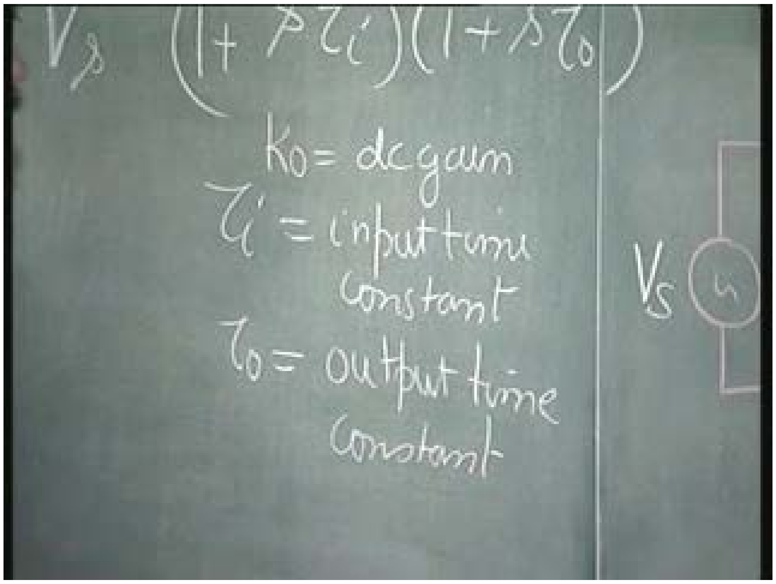

So, for any amplifier, this can be written this way. If there are 2 time constants, if there are 3 time constants, let us say, you have one amplifier cascaded to another amplifier; then, there will be an input time constant, an intermediate time constant and an output time constant. If we have, instead of 2 stages, 3 stages; then correspondingly, there will be 4 time constants.

If there is a single stage, there will be 2 time constants, at least. 2 stages - 3 time constants: input, intermediate, output; 3 stages - 4 time constants; n stages - at least, n plus 1 time constants.

So, this this is something that you can remember. That, depending upon the number of stages, we can say what is the order of the system. This is called first stage; if it has 2 time constants, it is called a second order system, second order. This is a third order; this is a fourth order, amplifier system.

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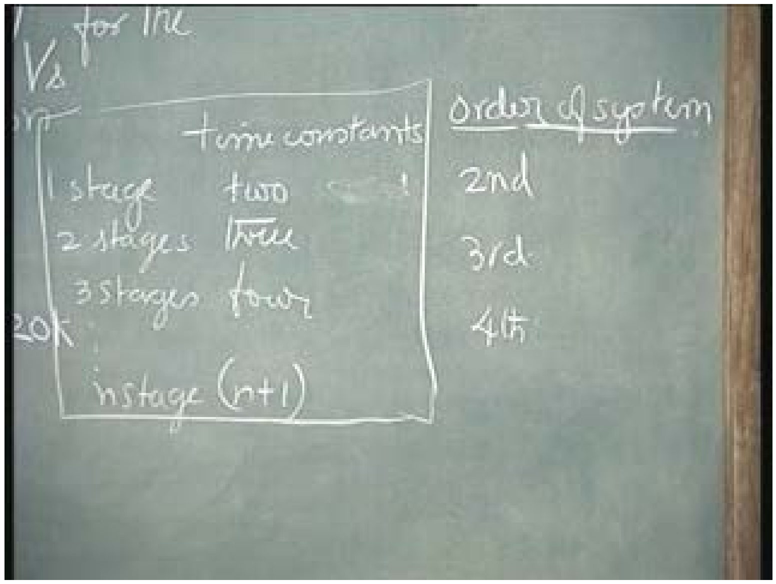

The order of the system is dependent upon the number of time constants. The first stage is already a second order system. There is nothing like first order at all. Second stage corresponds to third order, minimum. Third stage corresponds to fourth order system. So, order depends upon the number of poles. These are called the poles of the system. These are called the poles. There are no zeros here. These concepts, you have learned in your Control Engineering. These are called poles of the system.

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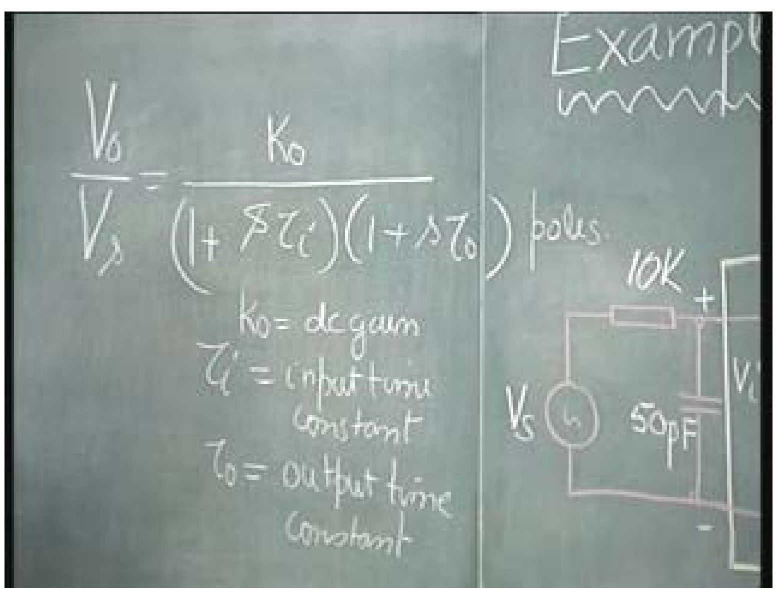

The poles are located at omega equal to, or actually, s equal to minus 1 over tau i; s 1 2 and 1 over tau naught. Equate this to zero; you will get the pole. s is equal to minus 1 over tau i is one pole; and minus 1 over tau naught is the other pole.

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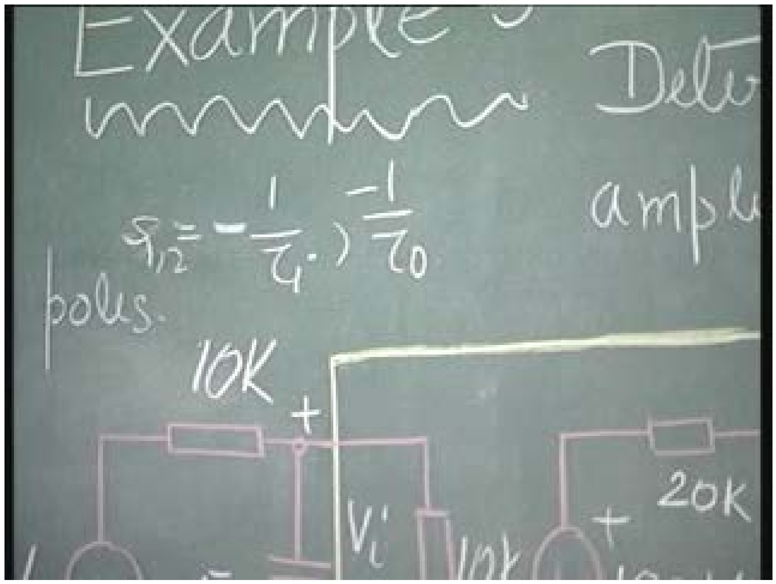

The poles of this system always rely on the negative real axis. The poles of this... such systems will always lie on the negative real axis. This corresponds to negative real axis. So, of the s domain; s domain has real part and imaginary part; in that, it will lie on the negative real axis. So, this is a two pole system or a second order system. There are 2 time constants connected with this system and we have obtained the Bode's plot in the last class.

We have seen that a Bode's plot for the magnitude of V naught over V s versus omega will look like... these are asymptotic; K naught which is independent of frequencies at low frequencies. At frequencies much lower than 1 over tau i and 1 over tau naught, the constant equal to K naught; and then, it will decrease at... we have discussed this - 20 decibels per decade, 20 decibels per decade; and then, the next time constant will bring about 40 decibels per decade attenuation. All these things we have understood.

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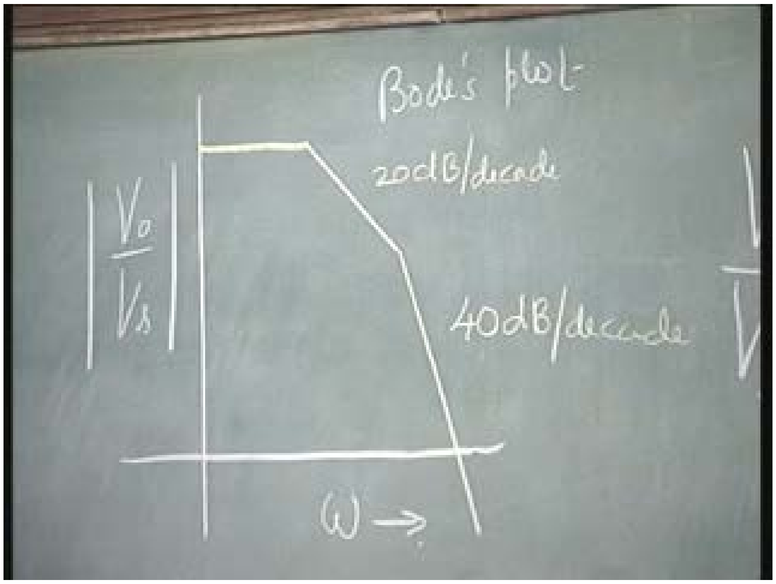

So, assuming that this is the dominant time constant and this is less dominant; so, we can say that, for example, if tau i is much greater than tau naught, then this corresponds to omega equal to 1 over tau i and this corresponds to omega equal to 1 over tau naught. These are called corner frequencies. In the last class, we have understood. These are called corner frequencies. This fall off is a 20 decibels per decade; this is at 40 decibels per decade; or 6 decibels per octave, 12 decibels per octave.

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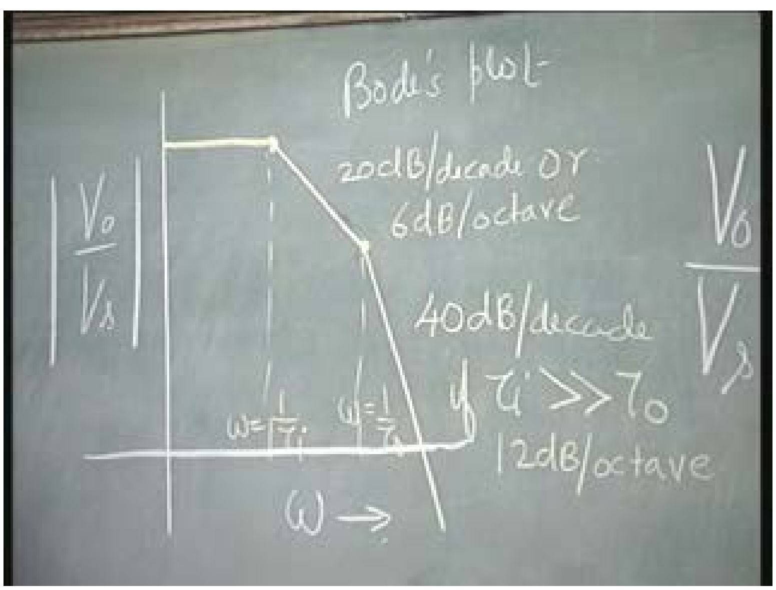

So, this is what we have seen. We have understood how, therefore, the frequency response characteristic of any amplifier will change. Now, let us take this for the example that is chosen; how much it is going to be. What is K naught? How do you evaluate K naught?

Now, I would not like to write down any equation. Look at this circuit. K naught... you forget about the existence of the capacitor.

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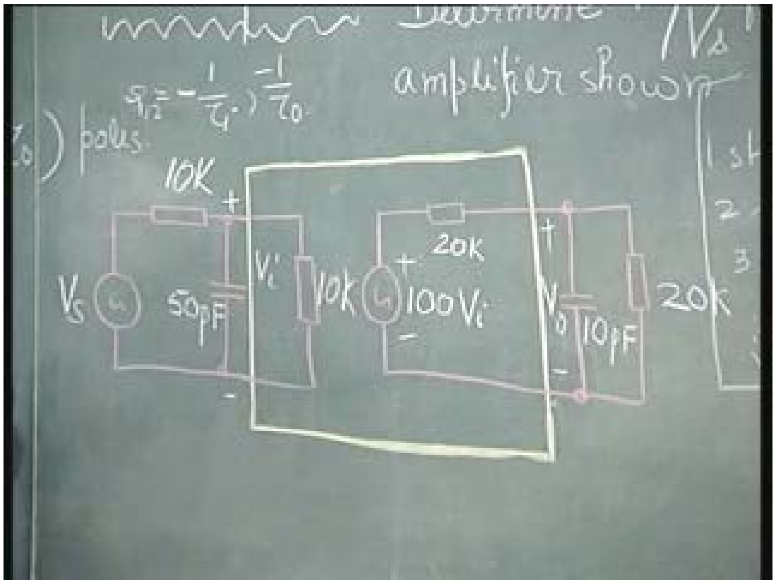

It is ((frequency... Refer Slide Time: 10:10)). So, V i is going to be half of V s because you have 10 K here, 10 K here; V i is going to be half of V s. Again, that half of V s is going to be multiplied by 100; so, 50 times V s appears here. That 50 times V s again is going to be halved because you have 20 K, 20 K; so, we have 25 times V s. So, K naught is 25.

Again, let us go through this. V s is going to be halved and appears here as V s by 2. So, V i is equal to V s by 2. So, V i equal to V s by 2, for the example. So, 100 times V i becomes 50 V s. This 50 V s is again divided equally between 20 K, 20 K; appears as 25 V s. So, K naught is equal to 25.

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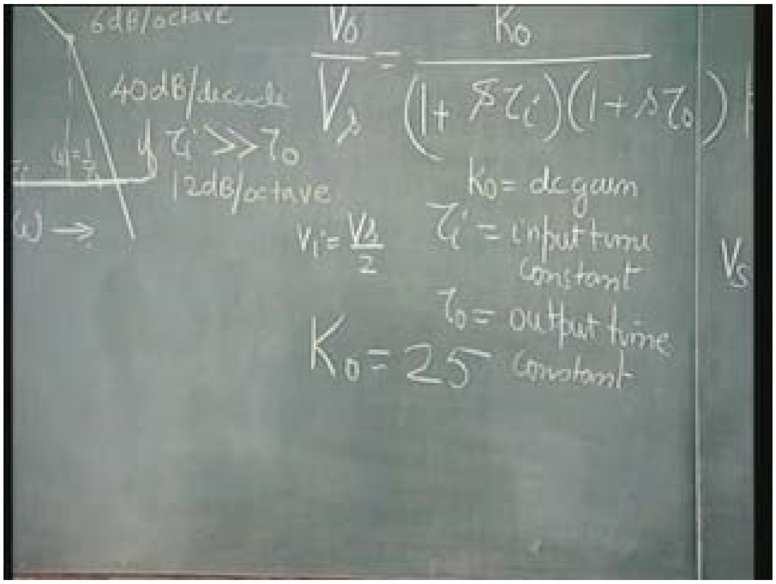

So, just by observation, we are now able to write down the value of K naught, for this example. Next, what is tau i? I have told you, tau i is equal to C i, this capacitor, into effective resistance across it, which is 10 K; you short this; and 10 K, which is, 10 K parallel 10 K is 5 K. So, you have C i into 5 K, which is 50 picofarads into 5 K; which is nothing but 250, 250 into 10 to power minus 9. Or, 250 nanoseconds. That is the time constant of the input.

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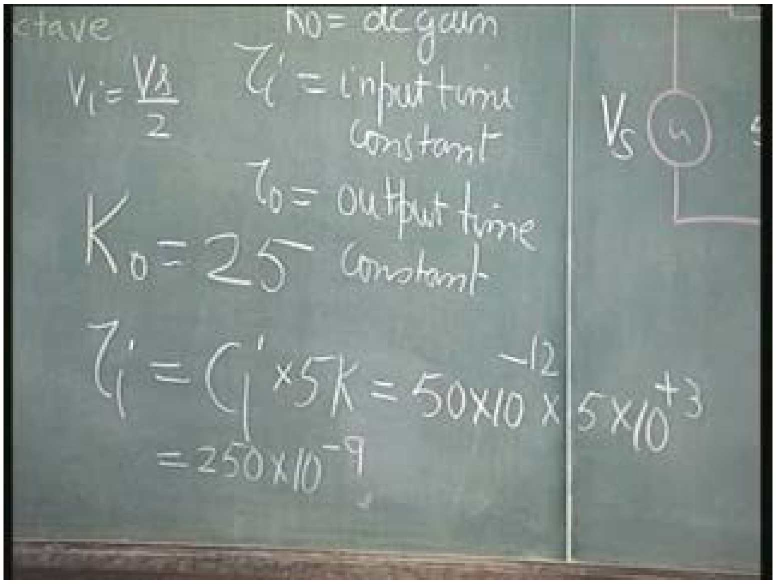

So, tau i is known. tau naught similarly is nothing but C naught, which in this case is, 10 picofarads. And again, short the sources. So, 20 K parallel 20 K is the effective resistance across the capacitance, which is 10 K. 20 K parallel 10, 20 K is 10 K; so, which is, 10 picofarads. I would take again; this is equal to 100 nanoseconds, 100 nanoseconds.

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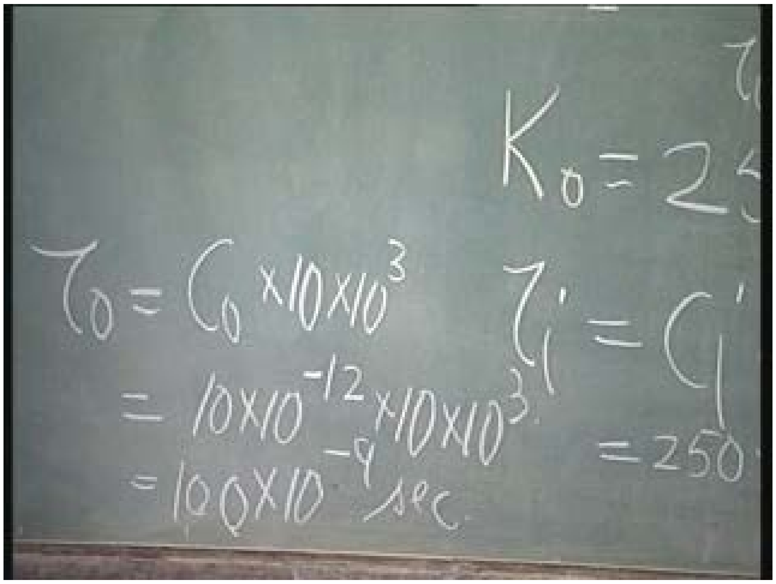

So, we have solved the problem. So, I can write down the transfer function V naught over V s as, V naught over V s as, 25 divided by 1 plus s into 250 nanoseconds, 1 plus s into 100 nanoseconds. That is the answer.

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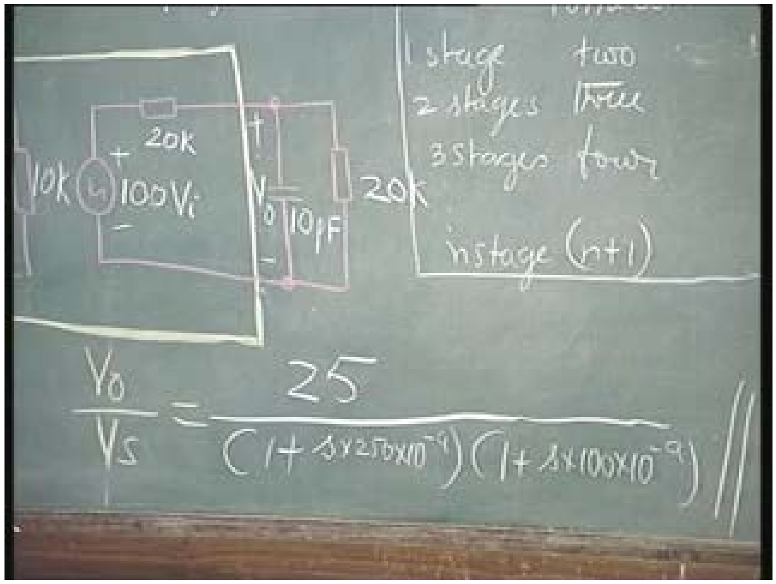

Now, I would like to expand here. Suppose we plot the Bode's plot. Then, this corresponds to gain of 25, remaining constant. If you convert it into decibels, it will be 20 log to the base 10 of 25. That is the decibels. 20 log to the base 10 of 20; whatever it is. That is the gain, constant gain, at low frequencies.

And then, at one corner frequency corresponds to 1 over 250 into 10 to power minus 9 or omega 1, we will call it omega i; corresponds to... this is omega i, this is omega naught. So, 1 over 250 nanoseconds. Or, this is equal to, omega i equals, therefore, 1000 divided by 250, that is, 4 into 10 to power 6 radians per second; or, 4 mega radians per second. 1000 divided by 250 is 4, into 10 to power 6 radians per second. That is omega i.

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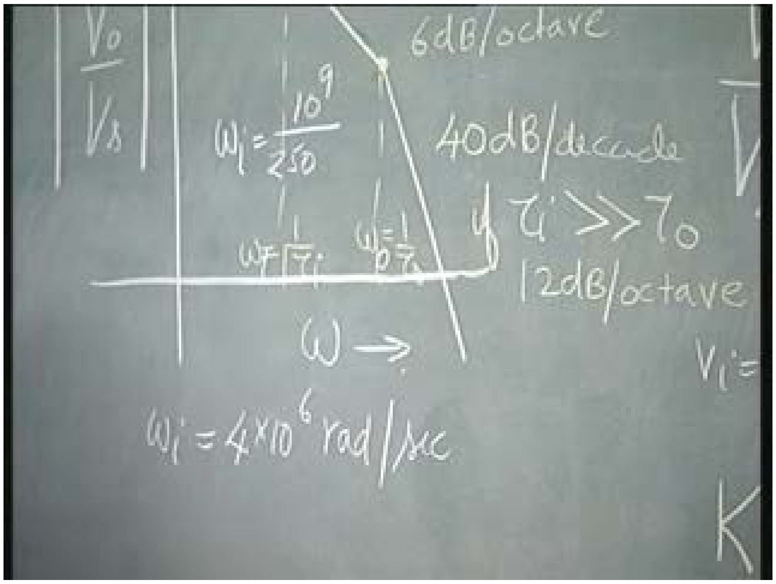

Similarly, omega naught is going to be, is going to be, 1 over 100 nanoseconds; or, 10 into 10 to the power 6 radians per second. So, we have got the corner frequencies also for this so called Bode's plot; and this is the complete frequency response of the circuit.

4 is comparable to 10. So, suppose I ask you to determine the upper cut-off frequency of this. Upper cut-off frequency or upper 3 dB frequency is going to be defined... I told you clearly. If this corner frequency is far away from this, then, this itself is the upper cut-off frequency, because this does not influence this. But now, in this case, 4 and 10 are pretty close to one another; so, upper cut-off frequency actually is going to be less than 4 mega radians per second, because at this particular point, apart from the influence of this, this also is going to influence in reducing the gain. So, you have to evaluate the upper cut-off frequency for this.

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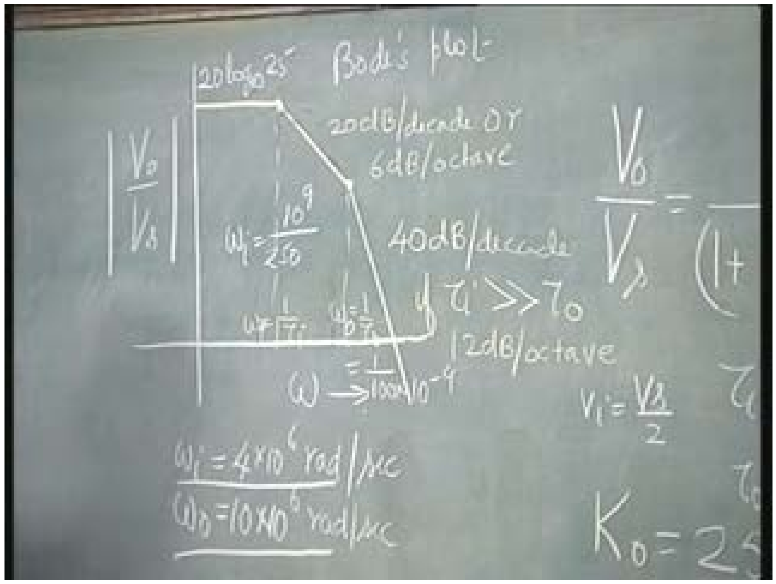

How do you evaluate this? This is very simple. Upper cut-off frequency, we have also said, is nothing but 3 dB point, upper 3 dB point, where the gain falls to 1 over root 2 times the maximum, the low frequency gain. So, if the gain of this is 25 by this, then, when the gain falls to 25 by root 2, that is the frequency corresponding to upper cut-off frequency.

So, we will see that, if omega i and omega naught are comparable, then, 1, 25 by 1 plus, let us say, j omega s, j omega by omega i, 1 plus j omega by omega naught is the actual gain. The magnitude of this is going to be 25 by 1 plus omega by omega i square under the root; again, under the root, 1 plus omega by omega naught square. So, when this becomes equal to 25 by root 2, that frequency corresponds to upper cut-off frequency. So, 25 25 gets cancelled; that means, then, this whole factor becomes equal to root 2.

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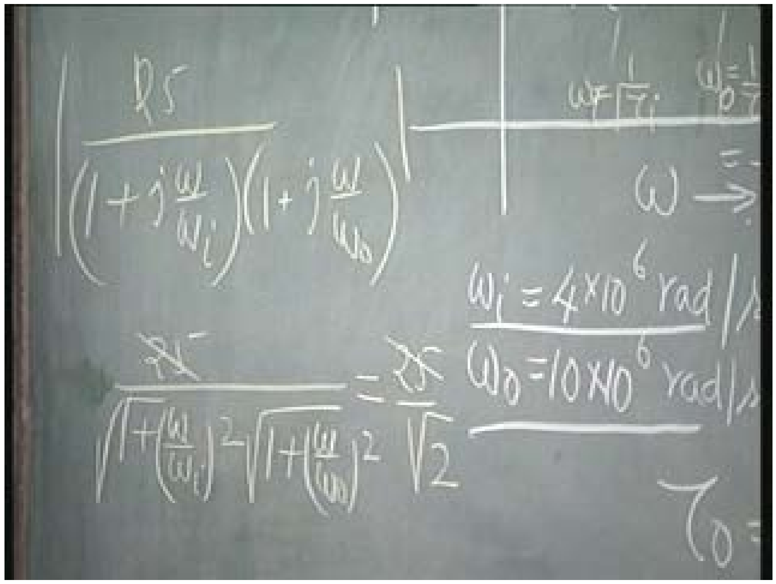

So, we can square this. So, 2 becomes equal to 1 plus omega by omega i whole square into 1 plus omega by omega naught... So, this is going to be a quadratic equation, which if you solve, will give you the upper cut-off frequency. So, I would like you to work this out; solve this quadratic equation. This is a quadratic equation in omega by omega naught, omega square; to get omega to the power 4 and omega square. We can solve for omega square and get the upper cut-off frequency here.

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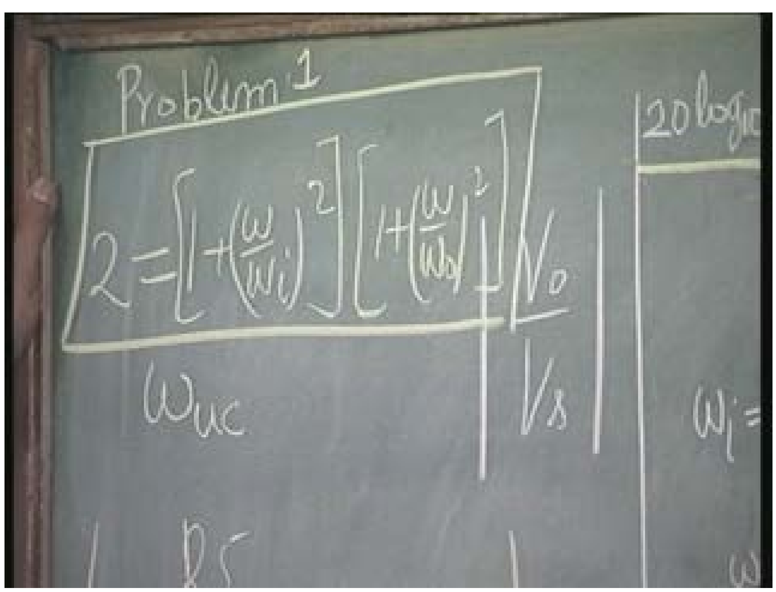

So please, as a homework problem complete this. That is not part of the example problem. Obtain the cupper cut-off frequency for the example. Obtain the upper cut-off frequency for the amplifier. So, you will notice that the upper cut-off frequency is going to be less than the lowest of the corner frequencies.

So, let us consider the Example 6. An amplifier has V naught over V s given as 100 by 1 plus s by pi into 10,000; that whole thing multiplied by 1 plus s by pi into 10 to the power 6. So, this is the characteristics of the amplifier. You are asked to find out its bandwidth.

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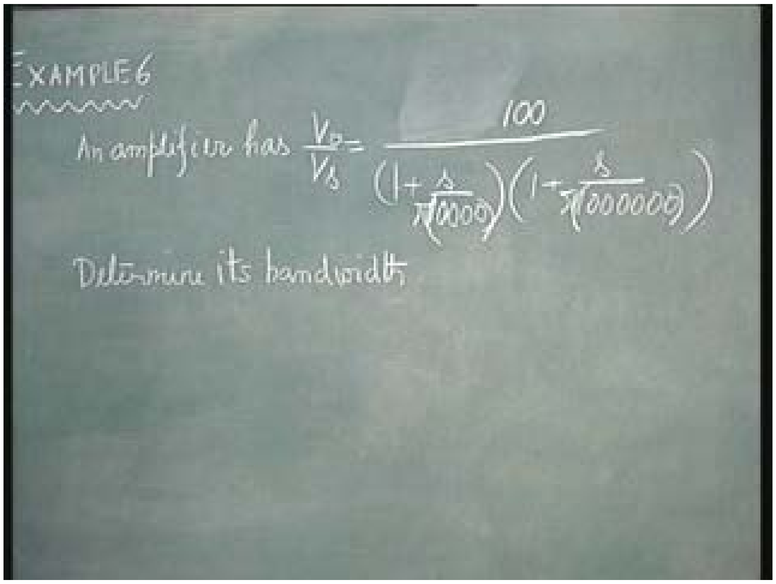

Now, I have told you earlier that upper cut-off frequency is the same as upper 3 dB frequency; also same as half power point, upper half power point. Also known as upper half power point, upper 3 dB frequency, upper cut-off frequency; all are one and the same.

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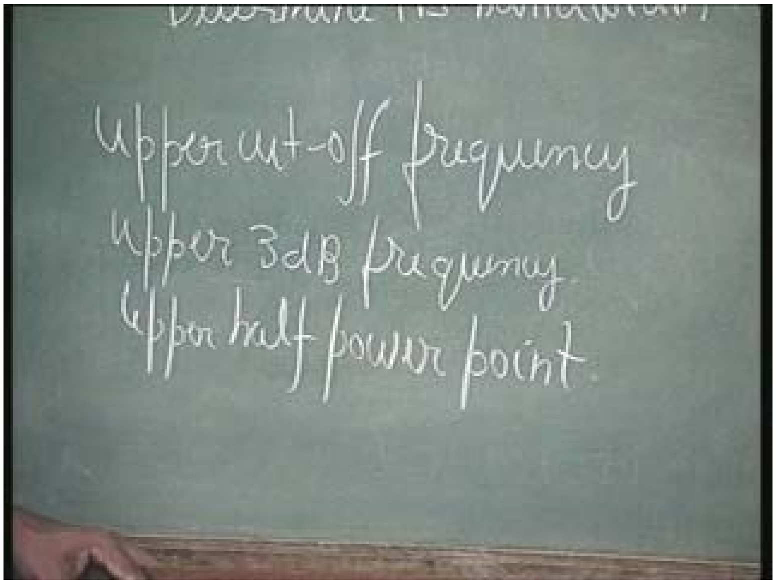

Now, in a situation where this particular frequency is dominant compared to this; this, its influence is far away. This is the first frequency which comes into picture. So, this dominates the attenuation characteristic of the amplifier; and this becomes of low significance. Therefore, even though this is a second order system, this can be equated for practical purposes; within the range of interest, it is only 100 divided by 1 plus s by pi into 10 to the power 4. So, this is half, we can approximate.

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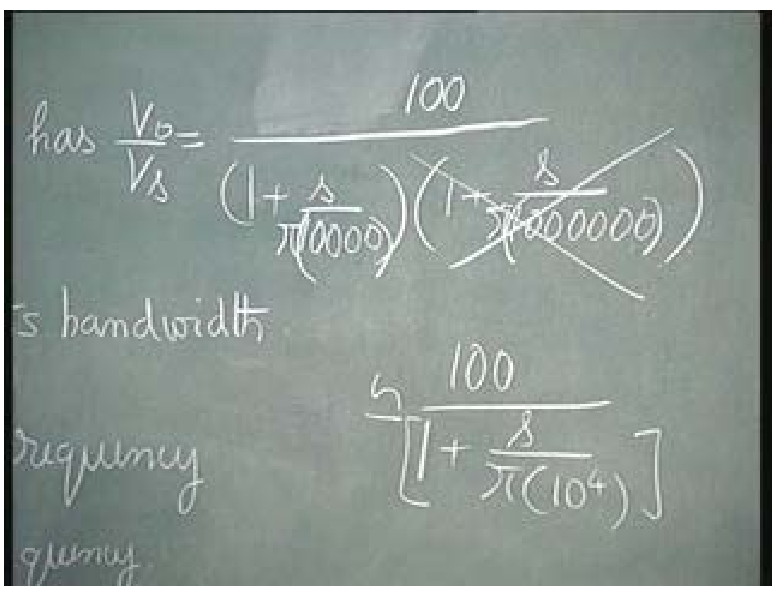

This is 10 to power 4, this is 10 to power 6; there is a 100 magnitude difference. So, we can ignore the effect of this, within the useful range of the amplifier. So, even though this is a second order system, in effect, we can, as an amplifier, we can consider it, approximate it, as a first order. That means, when the next corner frequency is an order of magnitude different, that is, 10 times higher, then, you can approximate the system as one order less, first order. So, please remember that.

So, if the frequencies, corner frequencies, are far apart from one another such that they are 10 times... then, it can be considered that the other one, far away one, can be ignored within the... So, 100 divided by this. And then, the bandwidth becomes... because s equal to j omega, I substitute; and this whole quantity should become equal to 1. That means that is the bandwidth.

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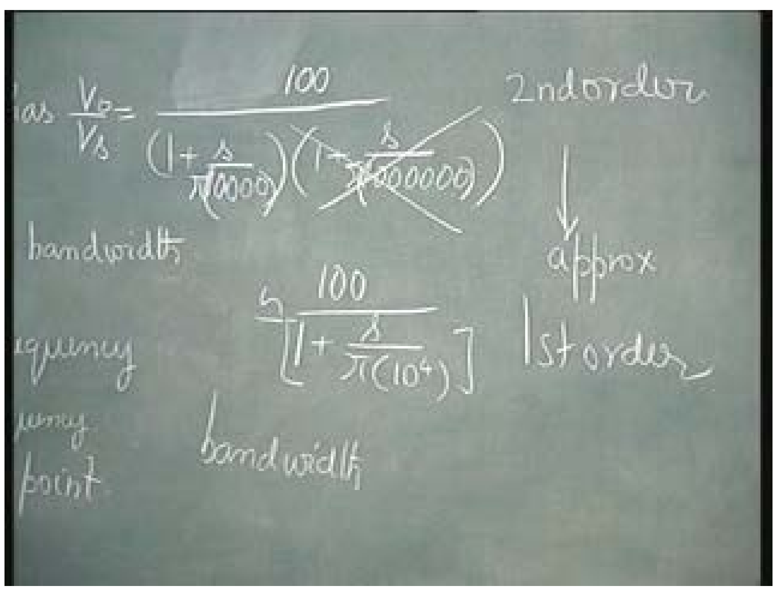

So, bandwidth becomes equal to this itself. So, if you want to convert this into frequency, radians per second, to hertz, then, you have to divide it by... So, in terms of hertz... So, this is going to be pi into 10 to power 4 divided by 2 pi or this is equal to... pi, pi, gets cancelled, 5 into 10 to power 3.

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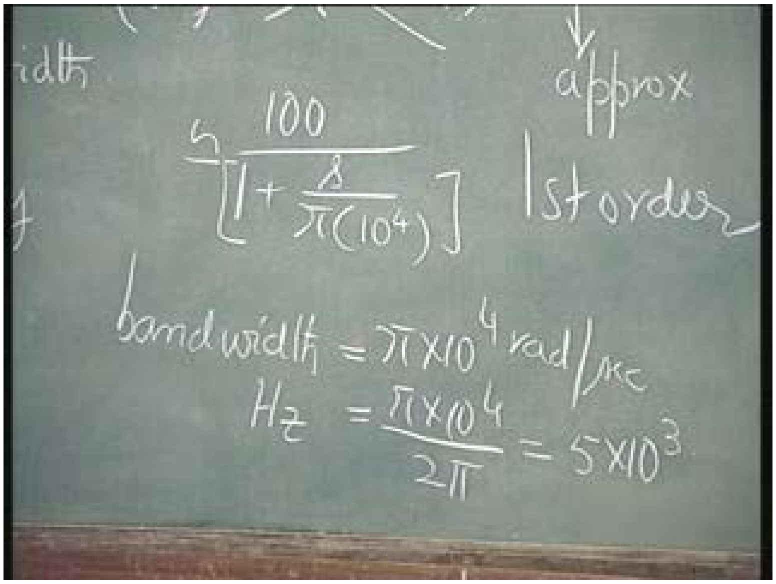

Or, answer is, bandwidth is equal to 5 Kilohertz.

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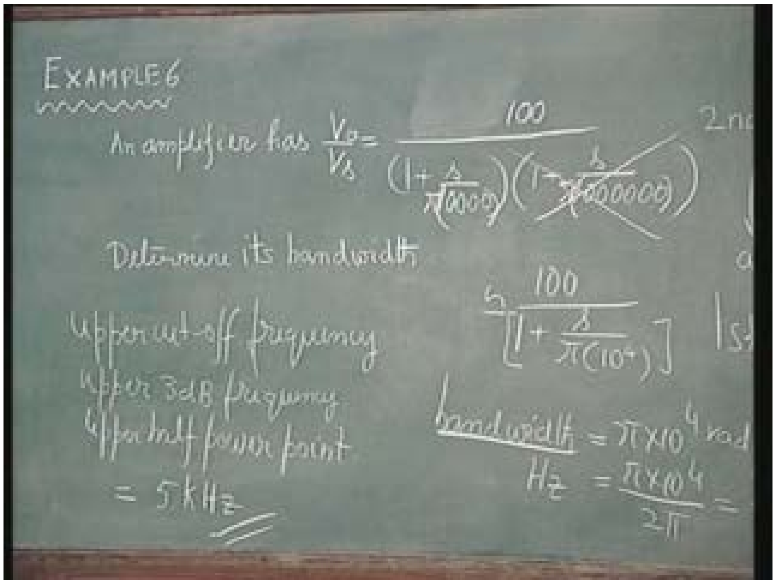

So, bandwidth; this is an important parameter associated with an amplifier, is essentially the first corner frequency, if other corner frequencies are far away from the first corner frequency. Otherwise, as will be given out in your example problem corresponding to Example 5, it will be dependent upon the other corner frequency as well.

In normal cases, the first corner frequency is the dominant time constant which is going to therefore fix up the bandwidth primarily this way. So, 5 Kilohertz in this case is the bandwidth of the amplifier. That means, the useful range, wherein the signal is not going to be much getting affected due to frequency; that range corresponds to 5 Kilohertz.

So, this is an important parameter associated with amplifiers. What does it mean? At 5 Kilohertz, already attenuation is going to be by 3 dB. At 5 Kilohertz, if you see, this becomes ,j 1 plus j; so attenuation becomes 3 decibels. It goes to 1 over root 2 - half power. So, beyond that, it is of no interest to us.

So, that means, all frequencies beyond this will be subjected to more attenuation and therefore will get distorted. This is called frequency distortion. What is it? That means, for frequencies much less than 5 Kilohertz, for the example, all frequency components in the example, the amplification factor is equal to 5, 100 in this case; remains constant, 100. Whereas, frequencies around 5 Kilohertz and much beyond 5 Kilohertz is going to be different for different frequencies.

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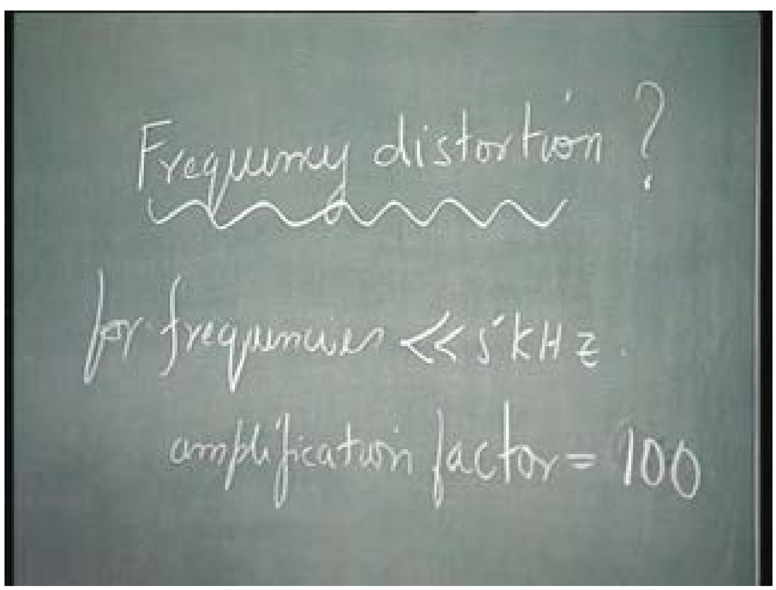

So, the input signal, if it is comprising of any frequencies with certain amplitude, the same amplitude ratio is not going to be maintained at the output. That is what is called distortion. Let us say, we have, say 100 Hertz signal and 100 Kilohertz signal at the input. 100 Hertz signal is definitely going to be amplified by 100. But, 100 Kilohertz signal is going to be attenuated and its amplification factor is going to be much less. So, the ratio of the amplitudes of 100 Hertz and 100 Kilohertz will not be maintained at the output; which means, what is going to be available at the input will be distorted at the output. Its shape is going to change.

Not only that, this is amplitude distortion. Further, these different frequency components will be subjected to phase shift also. As I told you, at 5 Kilohertz, the phase shift suffered is going to be 45 degrees; 1 plus j corresponds to 45 degrees. Already, there is a phase shift of 45 degrees. So, different frequency components will be suffering different phase shifts. So, this phase shift also will cause the signal to be distorted.

So, both amplitude distortion and phase distortion will occur because of the frequency dependence of gain. So, it is necessary that we must make sure that when we amplify signals, in this particular case, for this example, the signal frequency component should not be beyond 5 Kilohertz. Its useful range is only about 5 Kilohertz. So otherwise, if you have signals to be amplified which contain frequency components which are higher than this, we should have a upper cut-off frequency of bandwidth of the amplifier, which is much beyond whatever signal frequency, maximum frequency signal has.

For example, video amplifier; if it is that that we are designing, then the frequency components will be up to about megahertz. That means, upper cut-off frequency of that amplifier should be corresponding to tens of megahertz so that no distortion occurs in the signal.

If it is an audio amplifier, may be, the signal frequency components of interest will be only about 20 Kilohertz, the best audio amplifier, let us say. And therefore, the upper cut-off frequency need to be only about 20 Kilohertz. So, this is the essence of frequency distortion. Because of frequency dependence of amplifier gain, we have distortion, if we use amplifier for amplifying signals beyond its capability. So, we should always restrict amplifiers' use within its bandwidth.

Now, this frequency distortion will be causing amplitude distortion. What is amplitude distortion? Different frequency components will be subjected to different attenuations; that is why amplitude distortion is caused. Then, there is phase distortion. What is phase distortion? Different frequency components will be subjected to different phase shifts; that is going to cause phase distortion. So, both these cause frequency distortion.

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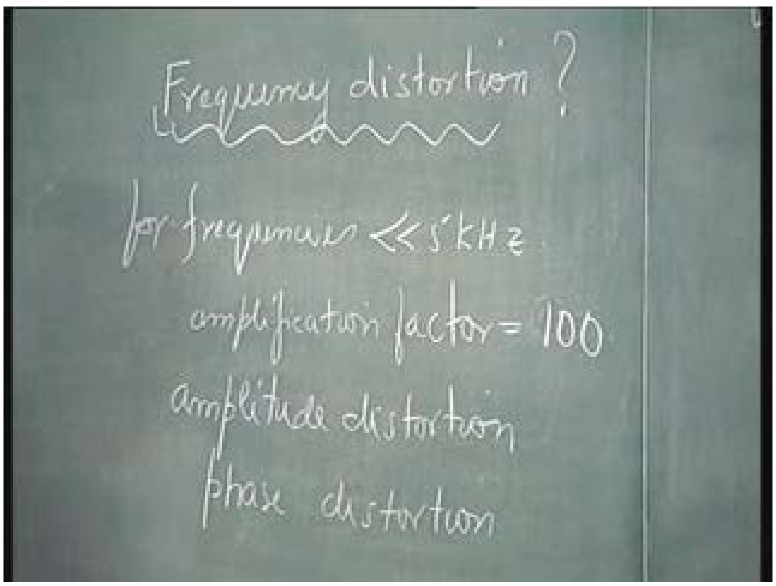

Apart from this, we have now some other aspect that we have to touch upon as far as amplifier performance is concerned. This is different from what we have been discussing so far. What is this? Consider an amplifier. Amplifier is a block. So, output is going to be related to input. Let us consider V naught, a voltage amplifier, we will consider. It could be current amplifier. I could as well consider I naught or anything. It is a function of input voltage, let us say. I am considering, let us say, voltage amplifier.

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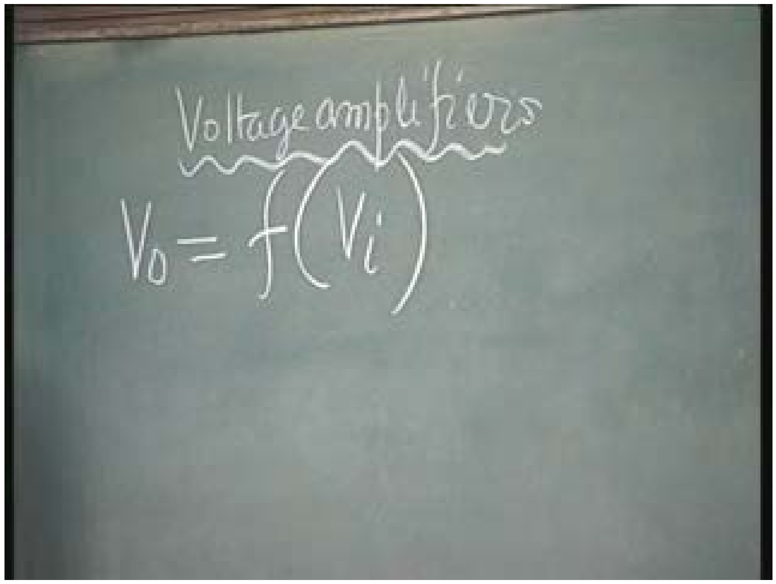

Strictly speaking, in general, any device will give me an output which is a function of V i; not just necessarily proportional to V i. So, if it is a general function of V i, then, that function can be split up as something, let us call it as a voltage, which is going to be called offset. This is independent of the input voltage. This is a voltage which is independent of V i. That voltage is called, as you have already seen, it is called offset voltage.

In an ideal amplifier, we do not want to have this kind of offset voltage. In an actual amplifier, this might also be called quiescent voltage. You have to get rid of it by decoupling it by means of a capacitor. So, this offset voltage is either called quiescent point voltage or offset voltage, depending upon your design of amplifier. This is independent of V i. Then, something that is going to be K naught into V i. This is what we have been calling as voltage gain. This K naught is the factor that we have been calling as voltage gain.

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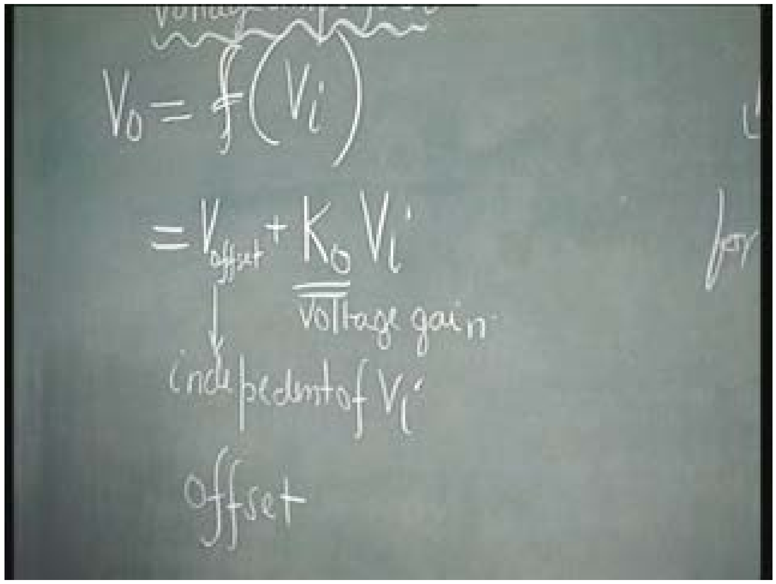

Linear dependence upon V i. Then, something that is going to be K 1 times V i square; then, K 2 times V i cube, so on... These terms are called nonlinearities.

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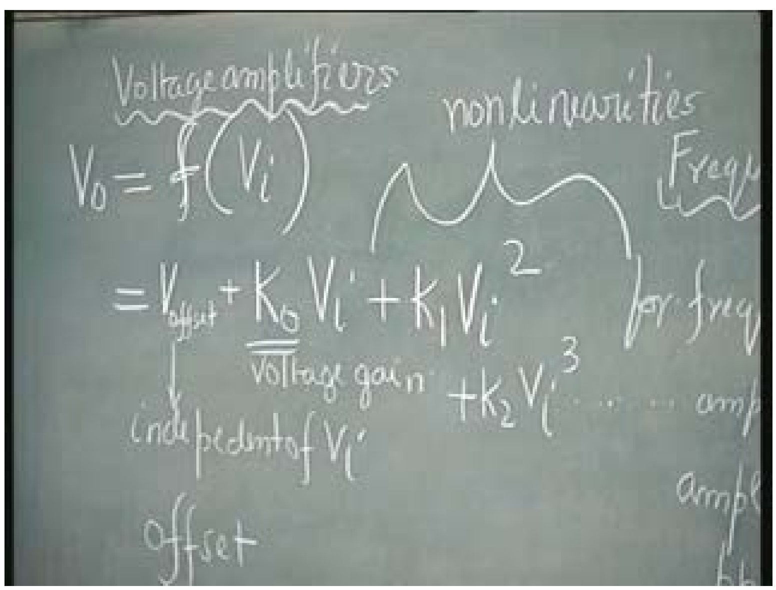

These are the nonlinearities. It is supposed to be only dependent upon V i; but the block that you have got, the active device you have got, is something that simply gives you a function of V i. So, in general, any amplifier that is designed by anybody will only give you output voltage or output current as a function of input voltage or input current. That can only have an offset voltage which is independent of V i; then something that is linearly dependent upon V i; that we call as voltage gain, plus, terms which are non-linear terms; it will be V i square, V i cube, V i...

Now, in a good amplifier, what should happen? The offset should be as small as possible. It should go towards zero and then K 1 should be zero. K 1, K 2, etcetera, all these things should be going towards zero; and K naught should be as constant as possible, as stable as possible.

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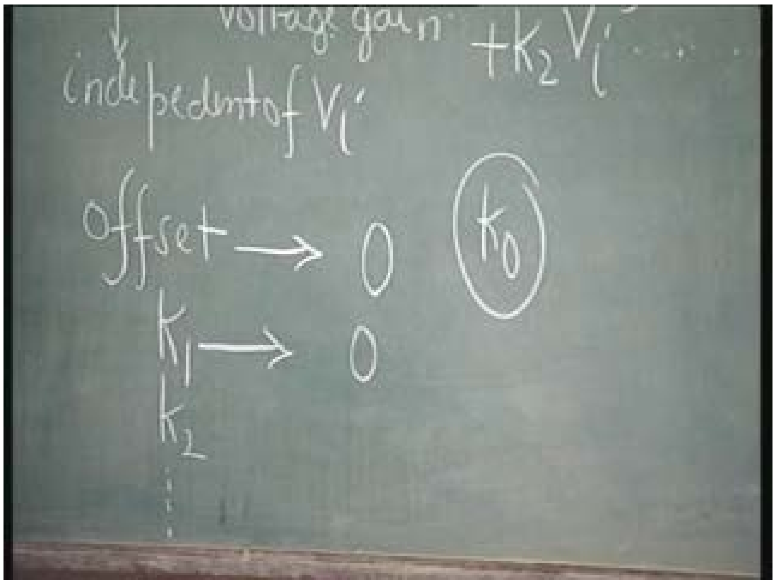

So, K naught is something that should be as stable as possible. Offset should go towards zero; K 1 should go towards zero; K 2 should go towards zero; all the nonlinearities should go towards zero. So, amplifier nonlinearities cause, again, distortion. This has nothing to do with frequency. This has only to do with amplitude of the input signal. So, this distortion, this depends upon where exactly we are operating in the amplifier, so called amplifier.

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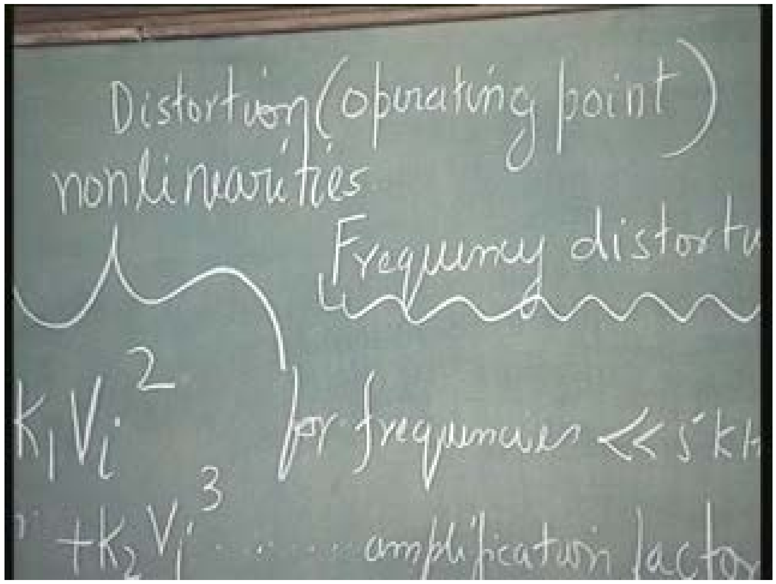

We can therefore select the operating point of the amplifier. We will select the operating point of the amplifier in such a manner that, the offset is zero; K 1, K 2, K n are as small as possible. Or, we will select the device for designing the amplifier in such a manner that this is valid.

That means, after selecting the device, then, we will select the operating point for the device. That means, biasing of the amplifier is done in order to make the device work in a region where the linearity is dominant, K naught is dominant. This should be maximized. K naught should be maximized and K 1, K 2 and K n should be minimized. This, it indicates this should be minimized; this should be maximized.

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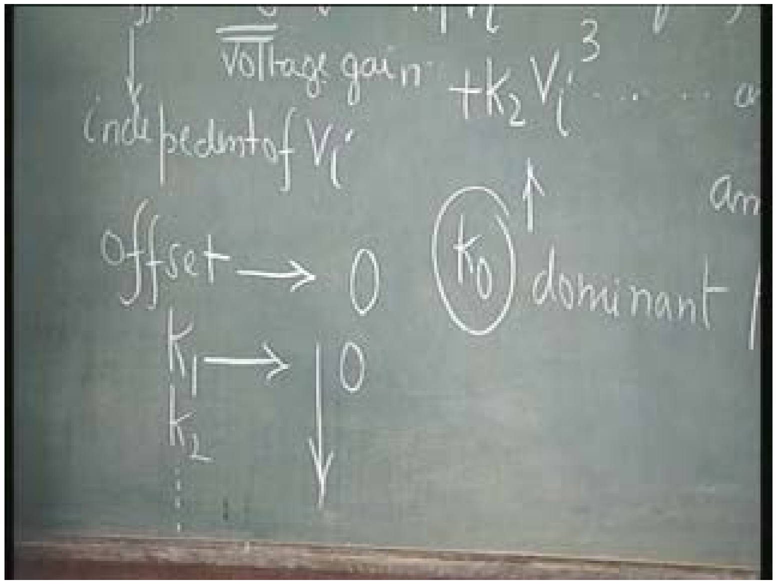

So, we will bias the active device; or, we will select the operating point of the device; or, we will design the circuit, the complex circuit, containing large number of devices, active devices, in such a manner that this whole thing is true. So, design of an amplifier involves selection of the operating point and the bias in such a way as to make the nonlinearities the minimum, linearity the maximum and offset voltage the minimum.

If you have the nonlinearities coming into picture, that will cause distortion. So, if V i is a sinusoid for example, V i square will become sine square. So, it will have harmonic components. At the input there was no harmonic; it is now generating a harmonic at the output. V i cubed will generate a third harmonic. V i to the power 4 will generate the fourth harmonic, like that...

So, that means all these nonlinearities will create unwanted components at the output. So, that causes distortion. If it is a sine wave at the input, output will now no longer be a sine wave. It will have all the harmonic components also present. That is what is going to be causing this distortion.

So, this part of the design is going to take up lot of time for us; after identifying the device, how to bias the device; how to range these different circuits in such a manner as to make the offset zero; and make the nonlinearity minimum and maximize the linearity. That will be the purpose of most of the design lectures of the future.

So, this is the case for an amplifier with input port and output port. The input port may be the two terminals of a differential stage. That means, you can also call the input port of the amplifier, instead of always referring the voltage with respect to this, you can call this as e 1 and this as e 2; and e 2 minus e 1 as being equal to V i.

So, that means now, it becomes a differential input operation. That means voltage reference can be taken with any point, arbitrary point; and these two voltages can be taken as absolute values from that arbitrary point. Earlier, we had taken the voltage reference as from here, and for this, from here. So, if you take these voltage references as from an arbitrary point, then, this becomes a differential input.

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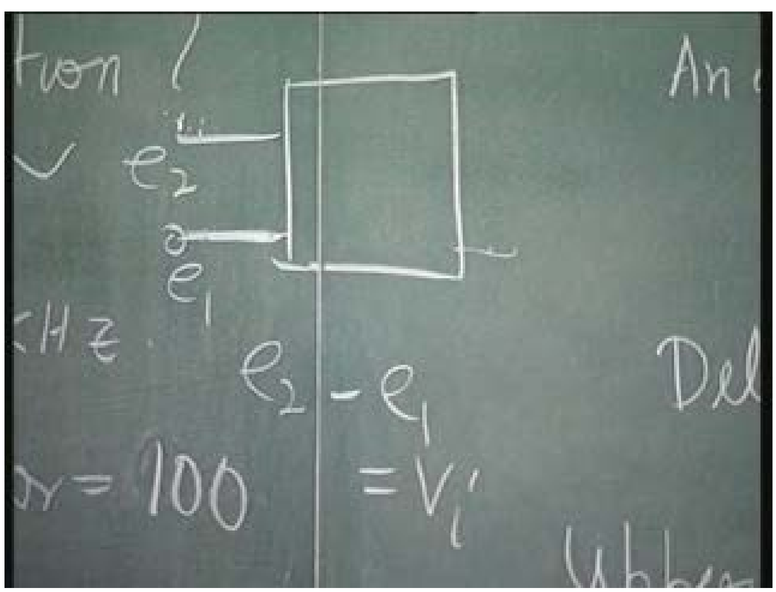

Similarly, if you take this from an arbitrary point, the same arbitrary point, this will become differential output. So, this becomes a differential input, differential output, amplifier, if the voltage reference, this is only a concept, is shifted somewhere else.

So, in a two port amplifier, this thing, you can treat it as differential input and differential output amplifier. So, what all we have said is valid whether it is differential input and differential output amplifier or just one port, with this as common terminal. That means, actually this becomes 4 terminal, becomes 3 terminal now because, we are connecting these two together and using this as a reference.

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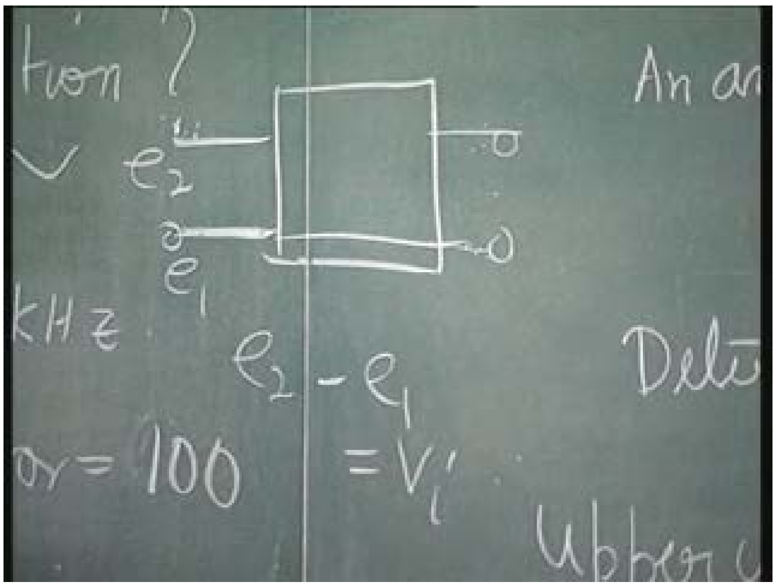

So, this becomes a 3 terminal amplifier. If this is not there, then it is a 4 terminal in 2 port amplifier.

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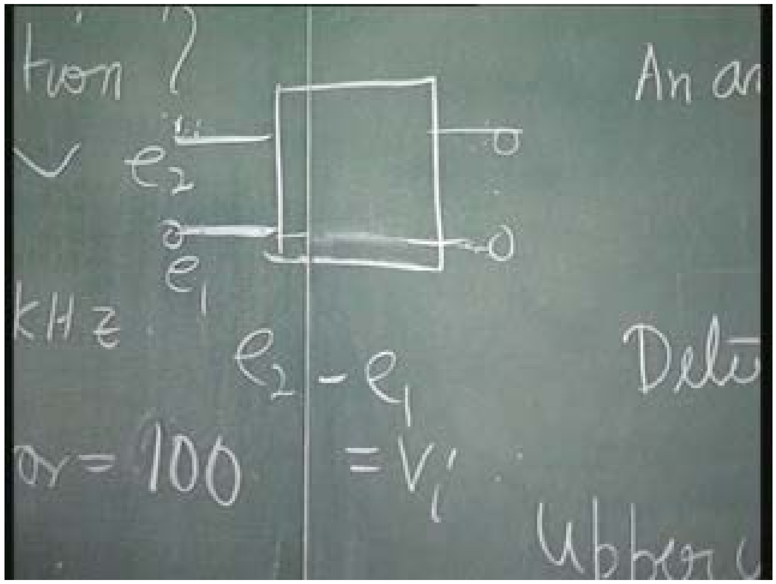

So, these discussions that we have had is going to be perfectly general and it is valid for three terminal as well as four terminal amplifiers. These are only conceptual references. Now, if you take arbitrary reference here, this will become differential input and differential output amplifier. So, even e 2, let us say, e 3, e 4; so, e 3 minus e 4 is what we have been calling as V naught.

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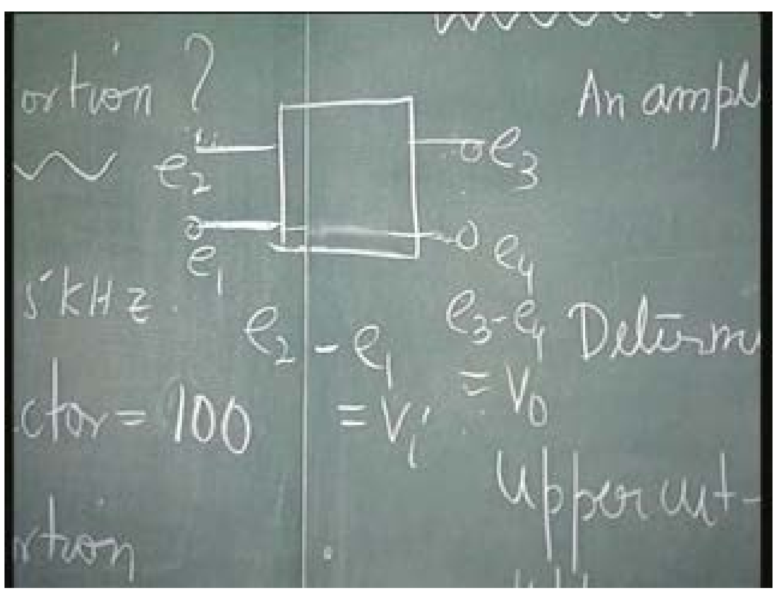

So finally, I would like to indicate that the amplifier works with power delivered from a separate source which is always called the dc power source. So, for biasing, we require a dc supply. So, this dc supply delivers the power required for biasing the amplifier.

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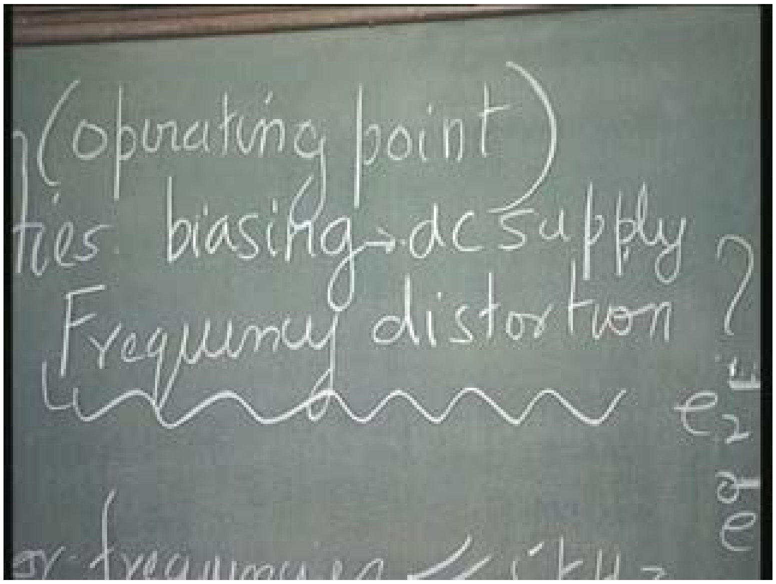

So, this dc supply, let us say, is plus V s and minus V s; then it is called dual supply.

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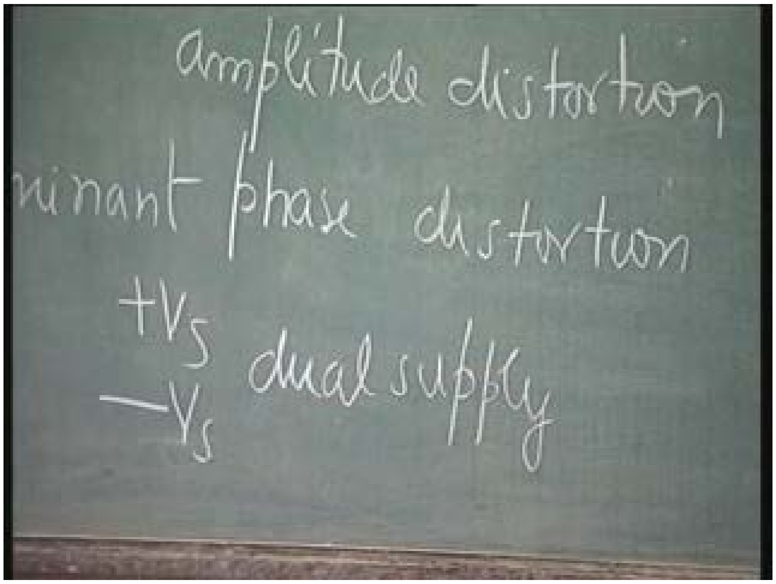

Why do you go for dual supply, plus V s and minus V s? Suppose this signal could be positive or negative, V i could be positive or negative. Then, in order to obtain this, we have to have both plus and minus supplies. The operating point should be such that output should be zero when the input is zero; so, which means, necessarily, in order that the offset of that is zero, you must operate the output at a voltage which is equal to zero. With plus V s and minus V s, the quiescent operating point for this should be zero. So, that is what requires a dual supply.

We can have single supply; then it will be V s and ground, zero; or, minus V s and zero. Then obviously, the operating point, in order to have maximum range for the output signal, output signal can be, minimum equal to zero, maximum equal to V s. Here also, output signal could be maximum equal to plus V s, minimum equal to minus V s.

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This is what is called the range for the output signal. Beyond this point, the output is said to go to saturation. This is another limitation. So, apart from these limitations, we also have what is called saturation.

Let me again explain this in terms of figures. This is V naught, this is V i. This is the output input characteristic of this amplifier, let us say. It should be, ideally speaking, linear. This is linearity. V naught and V i should be linearly related. That means, V naught is equal to K naught times V i. This is...this line passing through the origin. If it does not pass through the origin, it will be this; or, it could be the other one.

So, if it does not pass through the origin, this voltage is called the offset voltage; this or this. When V i is equal to zero, there is an output voltage. This is called a quiescent point. So, if it does not pass through the origin, then we are said to have an offset voltage. Now, that is clear, what is offset voltage.

There could be nonlinearity; that means, it is not going on being linear, it is going to a nonlinear region. The delta V naught by delta V i, the slope here, is the same as the slope here; it is same as the slope here. But here, the slope is less. So, this is called the nonlinearity. This is the linearity. This is the offset. Then ultimately, it will go to what is called saturation. Let us say, this is the maximum voltage your output of the amplifier can give; this is the minimum voltage the output of the amplifier can give; this is called saturation.

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Beyond this, output is not going to change. Beyond that, output is not going to change further. Even if the input is changed by any amount, output remains constant. Output is limited. So, output gets limited.

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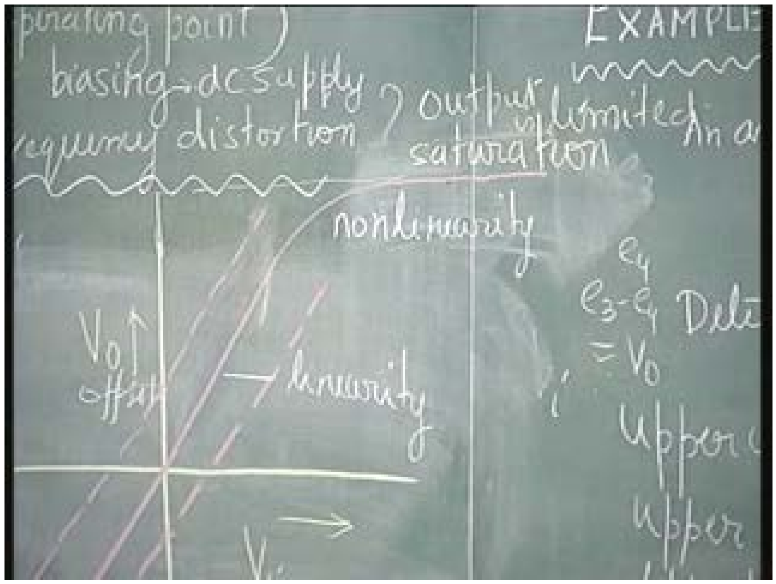

So, if power supplies are used for biasing, normally, these are the limits: plus V s and minus V s. It is made symmetric so that the quiescent point could be zero.

So, we have learnt a lot about distortion. That means, distortion occurs because of V i increasing; and then, further increase in the output voltage is not the same as the earlier increase for the same amount of V i change. So, that is called nonlinearity. Then, there is a region beyond which, further increase in V i does not cause any change in V naught; that is called saturation.

That can occur both on the higher side as well as the lower side. So, saturation nonlinearity, the nonlinearity because of the device, this non linearity; these are this kind of nonlinearities. In fact, the nonlinearity that I have put down here; it is symmetric here. That simply means that K 1 is zero. You can test it out.

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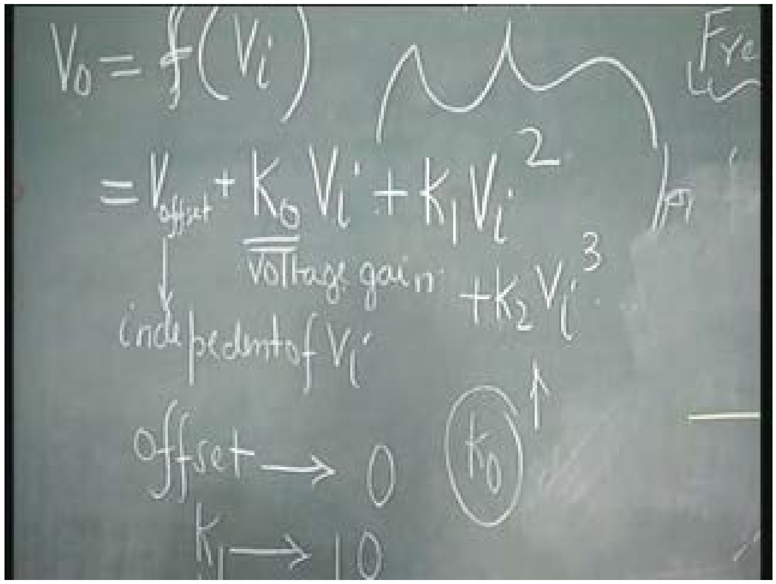

K 1 is zero; K 2 is present. This is going to have this kind of nonlinearity which is symmetric both for positive going and negative going signal, has K 1, K 3, etcetera, going to zero. But, K 2, K 4, etcetera, existing. So, this is one type of nonlinearity; so that depending upon the device that you select, the nonlinearities will change.

So, in the next class, we will learn more about these aspects: distortion, what does distortion cause, how distortion can be measured quantitatively, etcetera. This distortion that we have talked about for the second time is different from the frequency distortion, which comes about because of frequency dependence of the gain.