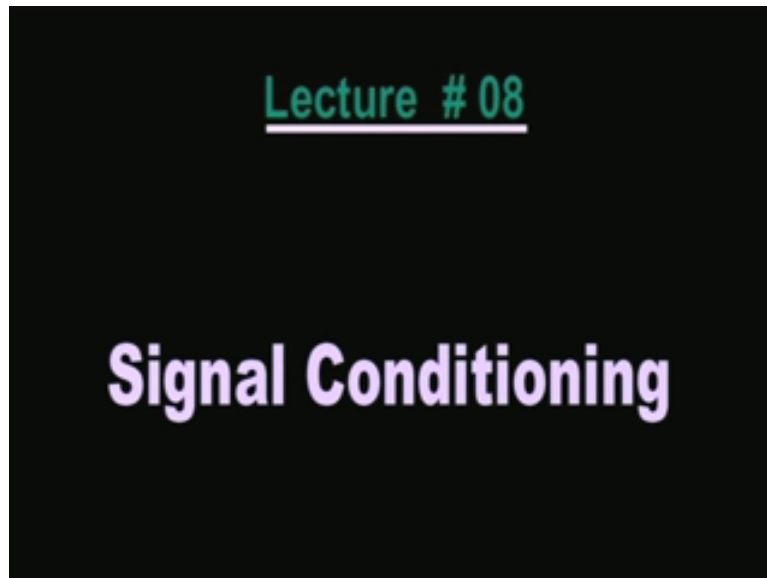


Industrial Automation and Control
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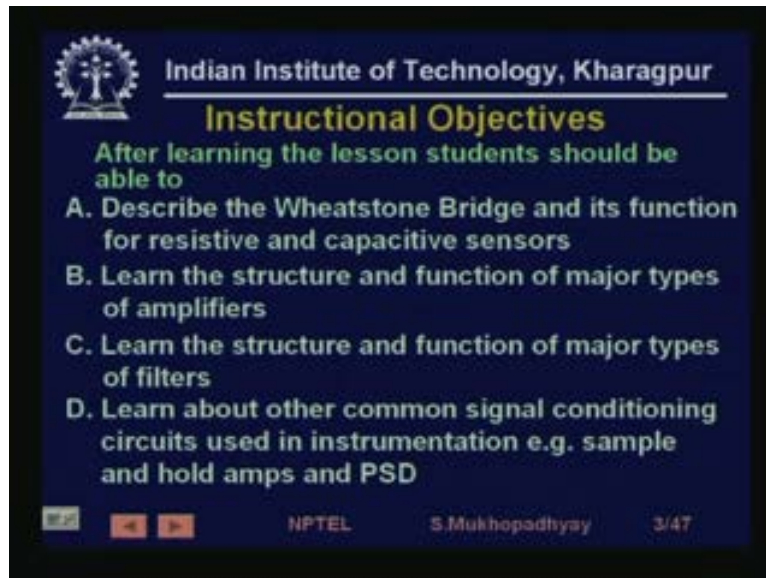
Lecture - 8
Signal Conditioning

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Welcome to lesson 9 of this course on industrial automation and control. Today, we will be talking about signal conditioning. As we have discussed right in the beginning that, the basic signal from the sensor, needs to be changed needs to be processed needs to be amplified, needs to be stripped of undesirable elements like filtered. So, several you know operations need to be done on this signal from the sensor to be able to finally, extract the value of the process variable accurately over time. So, all this is under signal conditioning.

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Instructional Objectives

After learning the lesson students should be able to

- Describe the Wheatstone Bridge and its function for resistive and capacitive sensors
- Learn the structure and function of major types of amplifiers
- Learn the structure and function of major types of filters
- Learn about other common signal conditioning circuits used in instrumentation e.g. sample and hold amps and PSD

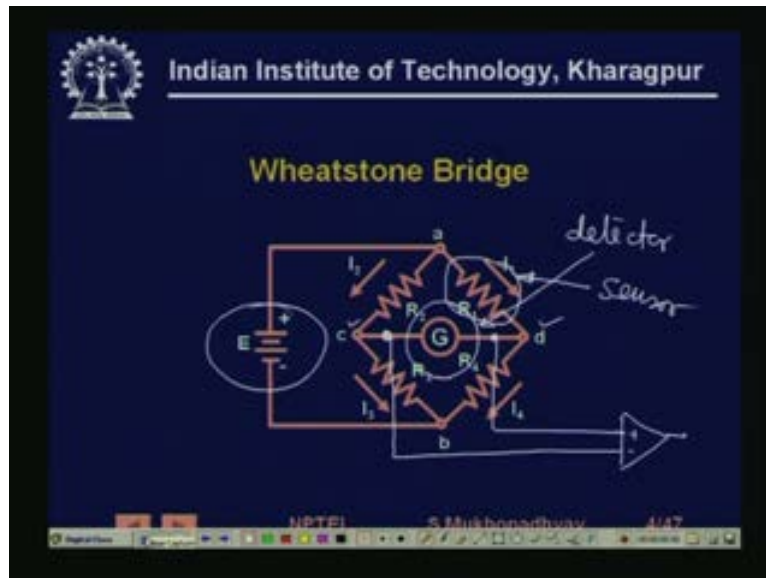
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After this going through this course, one should be able to describe a very important signal conditioning circuit called the Wheatstone bridge, which is widely used for resistive and capacitive sensors also for inductive sensors in some cases. So, this typically gives you a voltage signal and this voltage signal generally low level. So, this needs to be amplified and there are various types of amplifiers, which you which one can use. So, we will go through the various kind of amplifiers and learn the structure and function of them.

Then, we also need to use various kinds of filters for typically for noise filtering and also sometimes, you know separating out data. I mean the part of parts of signals, which are unnecessary, but which were required for let us say modulation we will see that. Lastly we will learn about some apart from amplifiers and filters. We will also learn about some you know miscellaneous types of circuit which are also very important.

We have chosen two of them. There are actually a wide variety of circuit which can be used, all of them cannot be covered. So, we have chosen two classes, one is that sample ((Refer Time: 03:11)) and the others are face sensitive detectors, which are very important in instrumentation. So, we go in to the lecture Wheatstone bridge.

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The Wheatstone bridge structure as we are seeing. In this case the bridge is drawn for 3-4 resistive elements, so this is the source. There are 4 resistances, this is the so called detector and in the case of an instrumentation setting generally, what is done is that, this bridge is operated in the unbalanced condition. As we can easily realize that, the voltage between this c and d points depend on the resistance ratios.

So, when the resistance ratios are equal that is R_2 by R_3 is equal to R_1 by R_4 . In that situation the voltage between c and d are is going to be 0. The instrumentation situation suppose, one of the sensors suppose this is the sensor whose, resistor varies with some process variable that we want to measure, suppose it is temperature. So, then we design the resistance in the other resistances in such a manner that at some reference temperature, this resistance is going to be such that the resistance ratios will be equal and the bridge output will be 0.

Now, as the temperature will change from that reference temperature, then this resistance is going to vary. So, the resistance ratio will no longer remain equal and therefore, some voltage will appear between this c and d points, which we will typically feed to some amplifier and the some downstream signal processing circuit. So, we essentially process this unbalanced voltage of the bridge. Now if you see the behavior of the bridge.

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Analysis of Resistive Bridges

Assuming detector resistance to be high,

$$V_o \leftarrow V_s \left[\frac{1}{1 + \frac{R_3}{R_1}} - \frac{1}{1 + \frac{R_2}{R_1}} \right]$$

For a single strain gage bridge

$$E_{th} \rightarrow \frac{V_s}{4} G \left(\frac{\Delta R}{R} \right) \left(\text{Strain} \right)$$

So, if you see an analysis then; obviously, from the previous diagram you can make out that the voltage between the c and d points, which is typically fed to the amplifier or which is the output of the bridge. So, we can call this the V output for the bridge that is given. It is easily found out that, it is given by $\frac{1}{1 + R_3/R_1} - \frac{1}{1 + R_2/R_1}$. Now, suppose R_1 is the resistance, which is varying. So, then this R_1 will be changing and therefore, the output voltage will be varying.

See in this case, you can find out that the output voltage is actually a non-linear function of R_1 , but if R_1 is typically very small, then it turns out that the output voltage becomes proportional to the change in R_1 . So, it is linear over a certain range. For example, and this kind of Wheatstone bridge is actually used in various configuration in instrumentation. For example, you can connect a single strain gauge into the bridge.

So, then the strain gauge forms one arm of the bridge and with applied force or pressure strain is created. And therefore, the resistance of the strain gauge will change. As we have learned in our previous lectures and therefore, the output voltage will change. Now, this is typically they are is mentioned as E_{th} , because or rather the rather E_{oc} or the open circuit voltage between the V and $c d$ points.

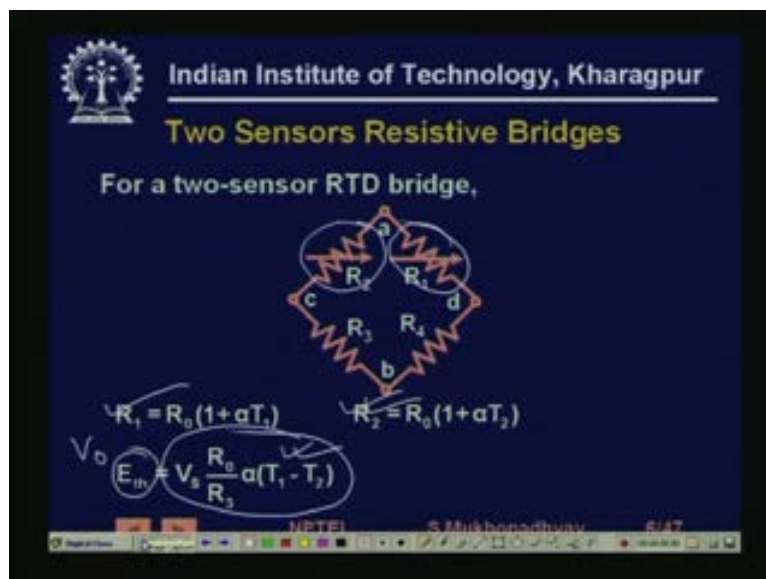
Now, it is generally considered open circuit because the impedance that is generally connected that is, these two terminals c and d are actually connected into an amplifier whose, input impedance is generally quite high, much-much higher compared to the resistance value

themselves. Therefore the voltage between c and d can be considered as the open circuit voltage or the Thevenin voltage, so that is why it is mentioned as E Thevenin.

It turns out that this E Thevenin becomes proportional to this G is actually you know delta R by R of the strain gauge. So, it will be proportional to delta R by R of the strain gauge. And since delta R is actually proportional to strains so therefore, eventually the output voltage is actually going to be proportional to the strain. So, this is how the strain gauge incorporated into a Wheat stones bridge produces an output voltage which is proportional to the strain.

Now similarly, if you have another kind of typical another sensor, which is also used in with a Wheat stones bridge is the resistance temperature detector. So, if you have a single resistance sensor temperature detector, then we are interested in measuring temperature and again the delta R value that is the change in resistance for a given change in temperature is given by this $R_2 = R_0(1 + \alpha \Delta T)$. So, therefore, you will find that again the output voltage is actually proportional to the change in temperature. So, it measures the temperature.

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Now, in certain cases we can even put two sensors in a bridge, so in such a case it happens that you see this is suppose, we are interested in measuring the temperature difference between any two points in a process. So, at one point we put one RTD and at another point we put another RTD and those two RTD's have been incorporated in the bridge in these two arms. So, then the resistance of first RTD varies like this $R_1 = R_0(1 + \alpha T_1)$ and similarly $R_2 = R_0(1 + \alpha T_2)$.

So, therefore, it will turn out that the output voltage or the E Thevenin is given by this. So, you can see that the voltage is now become proportional to T 1 minus T 2. So, we can create a voltage, which will become proportional to the temperature difference which we want to measure. So, this is possible.

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Four Sensors Resistive Bridges

For a four active strain gage bridge,

$$\frac{V_s}{4} = \frac{\Delta R}{R}$$

$$E_{th} = V_s \left[\frac{R_2(1+G_s)}{R_2(1+G_s) + R_3(1-G_s)} - \frac{R_1(1-G_s)}{R_1(1+G_s) + R_4(1-G_s)} \right]$$

$$= V_s \left[\frac{(1+G_s)}{2} - \frac{(1-G_s)}{2} \right]$$

$$= V_s G_s$$

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Similarly, If we can also put four sensors into a resistive bridge. So, this is for example, typically used as we have already seen in strain gauge load cells. So, if you put all the 4, only thing is that we have to ensure one thing, that is we have to ensure you see that if these 4 resistance. How is the voltage produced? The voltage is actually produced by creating by unbalancing the ratio of the two arms.

So, as the resistances change we must connect the resistances in such a manner that the ratio actually changes. So, it should not be that that R 1 R 4 R 2 R 3 are all changing in the same direction in the same ratio. If that happens, then we are not going to get any output voltage. So, typically as we have seen in the case of strain gauges, we put the strain gauges in such a manner that if R 1 increases R 4 will decrease and even we increase the sensitivity, we make put in such a manner that R 2 will increase and R 3 will decrease.

So, what will happen is that these bridge ratio is going to go up because R 2 is increasing R 3 is decreasing and this bridge ratio will go down, because this is decreasing and this is increasing. So, therefore the voltage difference between the c and d points is going to be equal to the sum of these resistance changes.

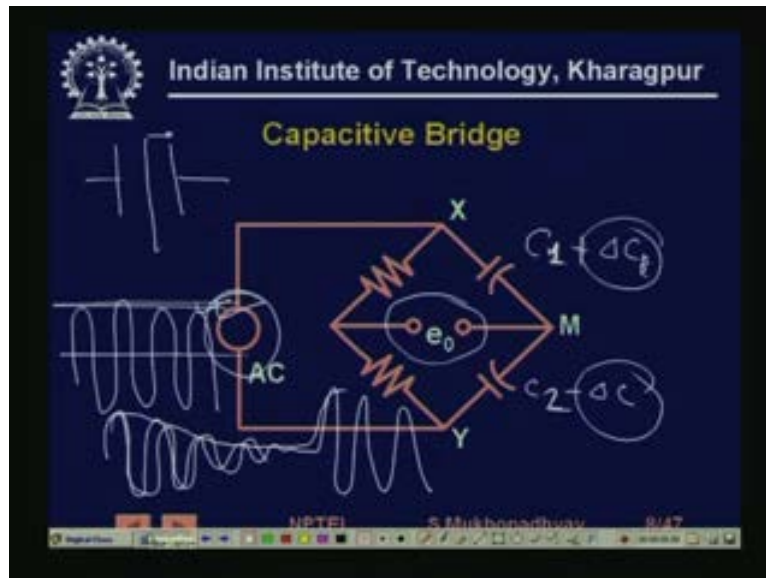
So, we set up this resistance in a clever manner and how we set up that we have seen in the case of the other sensor related lectures that we set it up in such a manner that all the resistance changes add up to the output voltage. And therefore, we get an increased sensitivity. So, we get a higher value of output voltage from the bridge.

So, it turns out that in this case that the output voltage seen in the single sensor case you had an output voltage, which is V_s by 4, the sensitivity into ΔR by R . Here, you are getting V_s into ΔR by R because the individual all the resistances are actually contributing to this resistance imbalance. So, you have got a 4 times sensitivity. So, typically by using a two arm bridge, there are two advantages one is that the sensitivity is doubled, second is that non-linearity are typically canceled. So, therefore we can operate the bridge linearly over a much larger of resistance variation.

Similarly, we see if we have four resistances, then the sensitivity is further increased. Also you can see that this is for example, in a strain gauge the resistance changes due to two things, one is the strain itself which we want to measure. The second thing is that, it can also change due to ambient temperature variation, but interestingly the resistance change due to ambient temperature will be in such a way that all these resistances will actually increase.

So, the resistance changes due to the temperature is not going to effect the bridge ratio and therefore, we can make our bridge output voltage completely insensitive to the variation in the temperature. This is also another advantage of having a push full configuration on a 2 or 4 arm bridge. So, this is why the strain gauge is actually very popular and I mean extensively used in instrumentation.

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Similarly, it is not only that resistive sensor can be incorporate in to strain gauge bridges. Actually, this this theory that the output voltage is depending on the bridge ratio is true for general impedances. So, therefore we can we can instead of resistances, we can also have you know capacitive sensor where you have capacitance C_1 and you have capacitance C_2 and therefore, this is C_1 and this C_2 .

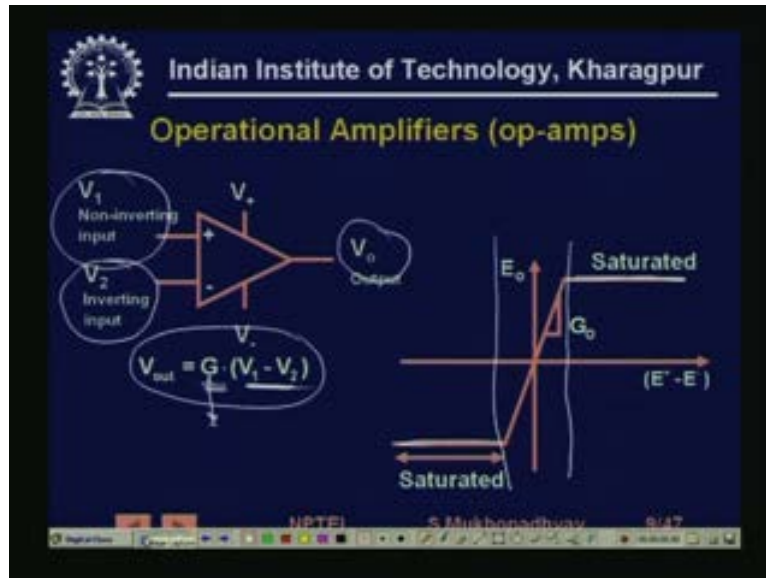
So, typically for capacitive sensors we often setup as we have seen in the case of displacement measurement that we set it up like a parallel plate capacitance. So, this plate is moving. So, what will happen is that C_1 will becomes C_1 plus delta C_1 and C_2 will becomes C_2 rather C_1 plus delta C and C_2 will becomes C_2 minus delta C .

So, typically sensor configurations are such that there is an equal and opposite change in the resistance. So, naturally the bridge voltage is is going to be proportional to this delta c and it is going to be considerably linear. Only thing we have to remember is that, in such cases we have to use an AC source, fact number 1. Fact number 2 is that, the output voltage is actually also going to be an AC.

So, it is not going to be a DC. So, we have to extract the amplitude of this wave somehow. And not only that, if we are going to measure a time varying signal, then this amplitude of the wave will actually vary in some ways, sometimes may be decreasing, sometimes may be increasing. So, if you have to get the actual signal out, then we have to track this variation of the amplitude and actually reject this sinusoidal variation, which is purely due to the

excitation source, we will have to reject. So, we will see in this lesson towards the end how that is rejected. So, apart from this, the bridge function exactly like a resistor like a resistive bridge.

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Now, then we come to the second element, which is an extremely important element in analog, a signal conditioning which is an operational amplifier. So, the operational amplifier is actually a circuit made of transistors, which behaves in a nice manner. It actually behaves like very close to an ideal amplifier.

So, it has two input terminals this one is called the non-inverting input, another is called the inverting input marked by plus and minus and the output is actually proportional to the difference between non-inverting and inverting input. Obviously, all voltages in circuits are limited by their power supply voltages and this gain is also very high. So therefore, it is not that we are going to get hundreds of voltages output from here. What is going to happen is that, within a small range this will behave like this equation. So, it will be like a straight line after that, when it reaches close to the power supply voltage then it will get saturated. So, this is a real operational amplifier characteristic.

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Op Amps	
Op Amp Characteristics	
Ideal	Reality
Infinite	10^4 to 10^6
High Open Loop Gain	
Infinite	300 K Ω to 1000 G Ω
High Input Impedance	
0	10 Ω to 5 K Ω (150 - 200 typical)
Low Output Impedance	

Implications:

- Used seldom in open-loop mode:
- Almost exclusively used in feedback mode.

So, if you see actually you know ideally operational amplifiers have some characteristic, but real operational amplifiers actually have characteristic, which are very close to the ideal characteristic that is why operation amplifiers are so extensively used. So for example, the input impedance of the amplifier is ideally infinite. Ideally it has an extremely high rather the open loop gain is infinite, that is the value of G is very, very large.

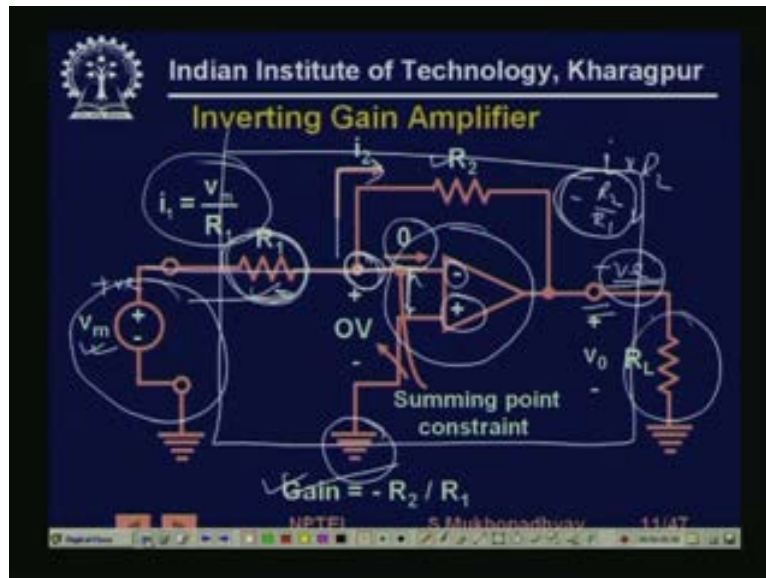
Typically, you will get of the order of 10 to the power 6. Actually, this 10 to the power 4 is the rather low value. So, in modern operational amplifiers we have 10 to the power 6, 10 to the power 7, even higher ranges sometimes. Then you have the input impedance is actually very high, so it does not draw any current, does not load the source and you can get I mean for some kinds of amplifiers, you get even thousands of giga ohms resistance.

The output impedance is typically very low, it is ideally suppose to be 0, but typically it can be depending on the current capability of the amplifier it can be between 10 ohms. You know 5 kilo ohm is actually of high value you get something like, you know 100 ohms around around 100- 200 ohms all the time.

So, the implications is that naturally you see with such a high input gain, you cannot really use it in open loop because then you have to apply very small voltages, especially in analog signal conditioning. So, the operation amplifier is typically is used in open loop only in the case of when it is used as a comparator that, but not otherwise. It is generally used with feedback. So, when you connect a high gain amplifier in feedback, then it is gain becomes

very stable and only determined by the external components. So, this property is used which will presently see some of these.

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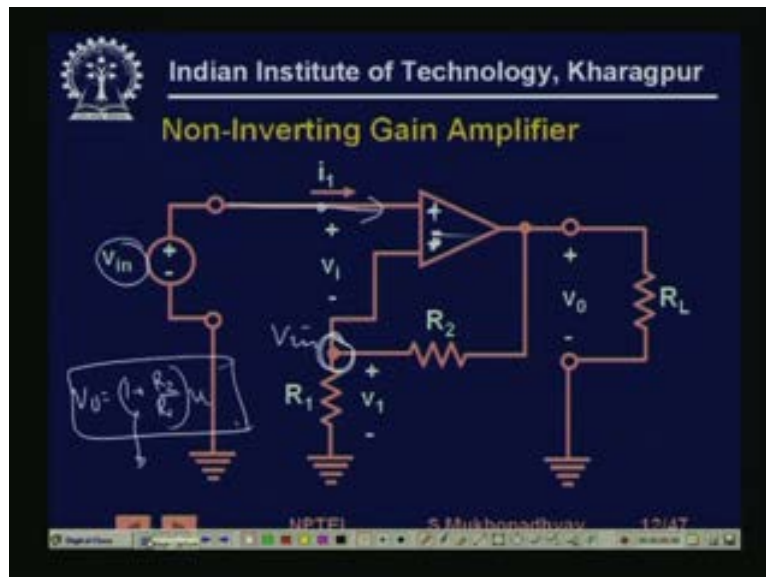


So, the first configuration that we looking at is an inverting gain amplifier where, you see this is the operational amplifier element. Along with that, it is now connected to these external elements R_1 R_2 , this is the load resistance and one of the terminal this is the called non-inverting input. Since, the input actual signal input is actual coming to the inverting terminal and because the voltage, if you apply a positive voltage here, you will get a negative voltage here. That is why this amplifier is called as inverting amplifier or inverting gain amplifier. So, as we know that by virtual ground principle since the basic principle is simple, since the open loop gain is very-very high and if this voltage has to be finite, then this potential difference must be 0.

So, this is called the virtual ground principle since this in ground therefore, this also at ground very close to ground and that it is maintained very close to ground by the feedback. So, what is the current flowing in to this? The current flowing in to this is V_m by R_1 . Since, the amplifier presents an extremely high input impedance therefore, no current is flowing into this. So, therefore, this same current is flowing into this, this is at ground. So, therefore, this potential is minus I in to R_2 is equal to what is I ? I is V by R_1 so it is minus R_2 by R_1 into V .

So, this minus R_2 by R_1 is the gain, so this is called the inverting gain amplifier. It is very simple requires only two resistive elements. Only drawback is that the input impedance seen by the source is this resistance. So, V by R_1 is the current flowing into it. So, if this resistance is low, then this overall amplifier has a low input impedance.

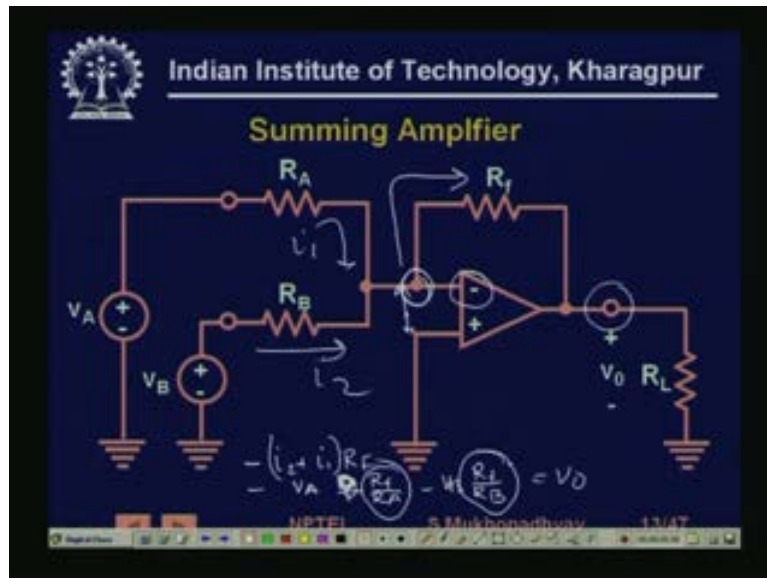
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So, now we go for another the non inverting gain amplifier configuration. So, in the non-inverting gain amplifier configurations see that, this is actually wrong. So, this should be plus and this should be minus. In fact, you can ponder, this a very interesting question that what if this was minus and this was plus, then you would have a case of positive feedback and the circuit would not behaved like a linear amplifier at all.

So, anyway we are not discussing that case, but this is plus this is minus. So, now, you see that here what happens is that. So, this potential now no current is flowing into it so this amplifier 1 of the plus points is that this amplifier has very high input impedance equal to that of the basic of an and since by virtual ground. Therefore, this potential is V_{in} , so therefore this potential is also V_{in} . So, you can easily calculate just by solving the DC network that V_0 will be given by $1 + R_2$ by R_1 into V_i . So, the gain of the inverting amplifier is $1 + R_2$ by R_1 . Note that there is no minus sign therefore, it is a rather than it is non-inverting amplifier.

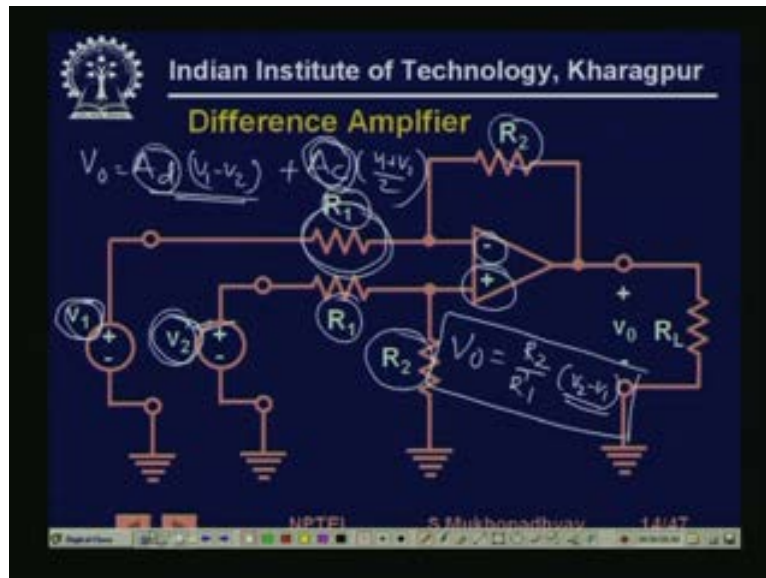
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Then, in many cases instrumentation we need to add up signals. For example, you can take sensor suppose, you want to compute the average value. Actually have put two sensors, which are both measuring temperature and you want to measure the average temperature. So, then what you have to do is, you have to multiply each one of them by again say 1 by n and then add all of them. So, that is require a summing amplifier.

So, this is a particular summing amplifier, in this case it is a summing and inverting amplifier because you see that we have basically taken the inverting amplifier configuration. Only thing is that now we are connecting two sources V_A and V_B and two currents are being. This is still at 0 potential because of virtual ground, so now we are pushing two currents. So, this is I_1 and this is I_2 and those two currents are flowing through this. So therefore, this potential is going to be at minus of I_2 plus I_1 into R_F . So, we will get it is equal to minus V_A into R_F by R_A minus V_B into R_F by R_B . So, this is one of the weights, this is the other weight. So, you have this is equal to V_0 . So, you have two voltages input voltages and weighted them by two gains. So, it calculates a weighted sum of two signals.

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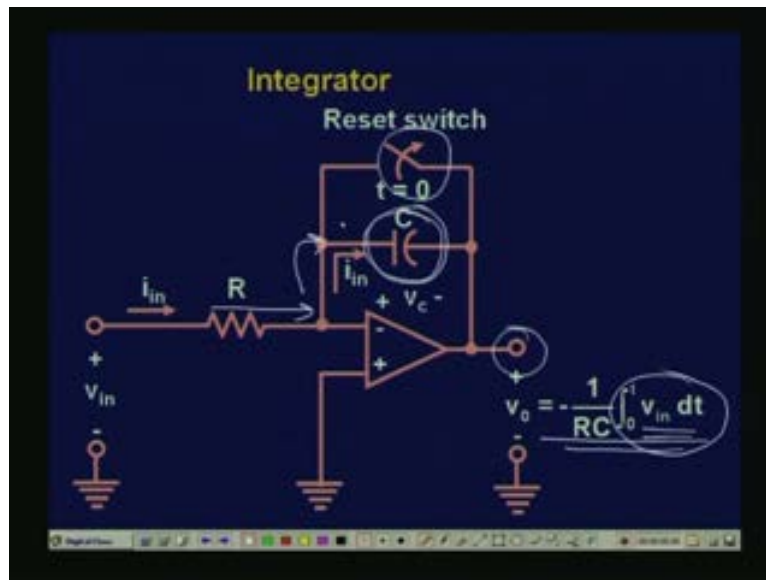
Similarly, we can have a difference amplifier. Now, see this configuration is slightly different in the sense that one of the sources are connected to the non-inverting input through a resistance network and the other is connected to the inverting input. So, it is a combination of inverting, non-inverting and also note that the resistance values are made equal. So, what happens, what is the result? Result is that you get that V output is equal to R_2 by R_1 into V_2 minus V_1 .

So, you purely calculate the difference very interestingly you see that, it does not depend on the value of V_1 or the value of V_2 , but depends only on that difference. So, they can be 10 volts and 9 volts or they can be 2 volts and 1 volt and still the output is going to be the same. There are many-many cases where this amplifier is used because in many cases what happens is that, we want to have an amplifier where we only want to amplify or take as a signal as the output signal, which is the difference of two potentials and not the individual potentials themselves. So, that is how what you do by the reference amplifier.

Only disadvantage of this amplifier is that again, the input impedance seen by this amplifier are low. So, we need to improve on the input impedance and there is also a particular factor for difference amplifier, which is called the CMRR or the common mode reduction ratio. That is all though I am saying that it is non dependent on the levels of the individual voltage is, but only one the difference actually what happens is that, the output becomes actually is proportional to some difference gain into V_1 minus V_2 . Then some common mode gain, this

is not completely dependent on this into $V_1 + V_2$ by 2. So, the whole exercise in circuit design is to make this gain small and this gain large, then it will behave like an ideal difference amplifier.

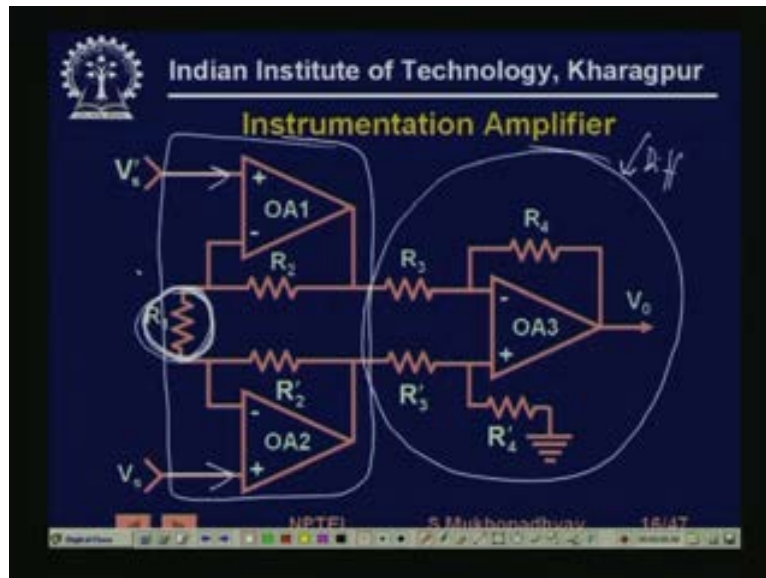
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So, circuits are available with very high CMRR values. Now, we come to a circuit called the integrator. So, if we want to integrate what we do is, we put a capacitor here. So, now this current flowing will now still flow through this switch is open when it is working. So, therefore, the output will be $\frac{1}{RC}$ into integral 0 to t $V_{in} dt$. So, the output voltage in the output voltage we have produced a voltage, which is proportional to the integral of V_{in} .

This is also very much necessary in number of instrumentation applications. For example, we may like to integrate flow to get level or to compute total sum of flows and so many things. So, integrate velocity to get position etcetera. This resets which is added such that the initial condition on the capacitor can be nullified and we made 0 before the circuit starts operating.

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Now, we have what is known as the instrumentation amplifier. So, the instrumentation name comes directly into play because this signals can accurately the signal. Firstly, it offers a very high input impedance. Secondly, the input impedance is also symmetric. In the previous case it will not symmetric on, if you see from the inverting input you will see one input impedance. If you see from the non-inverting input, you will see another input impedance, but in this case the input impedance is symmetric.

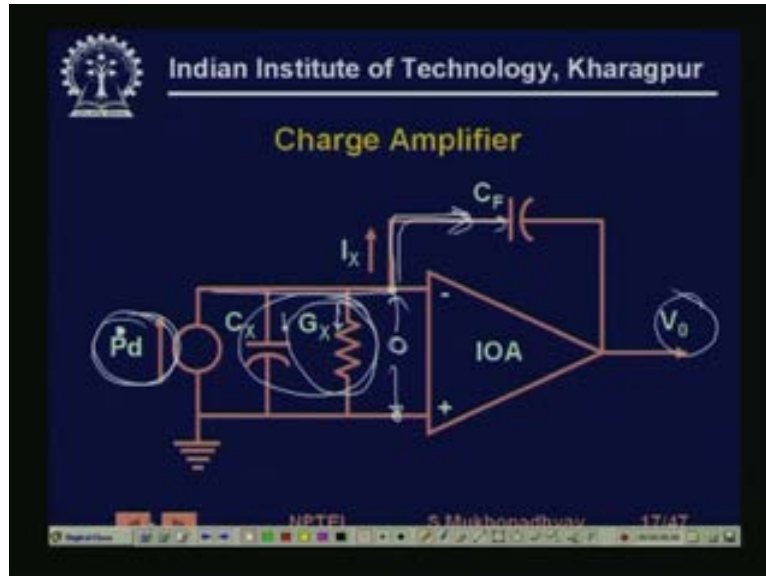
If you examine the circuit, you can find out that this is nothing, but a difference amplifier stage, this is difference amplifier. Along with that an input stage is added so basic job the input stage is to do three things. Number 1 is increasing input impedance, so you see each one of the inputs are feeding in to the non-inverting input. So, this see a very high input impedance. So, there is no question of loading.

Second thing is that the input impedances seen are equal they are not different. Third thing is that normally in this resistance if you see commercial instrumentation amplifiers, you will find that the gain of the input stage can be programmed. Generally, the gain of the differential amplifier stage is not disturbed. It is because the differential amplifier is typically designed for having very high CMRR and that stage is generally not disturbed

If you want to program the gain, you program by programming these resistor either by connecting jumpers or by connecting external elements. So, this is the instrumentation

amplifier and now, it will have the gain of this will be you know the product of these; obviously, so you can learn more about it.

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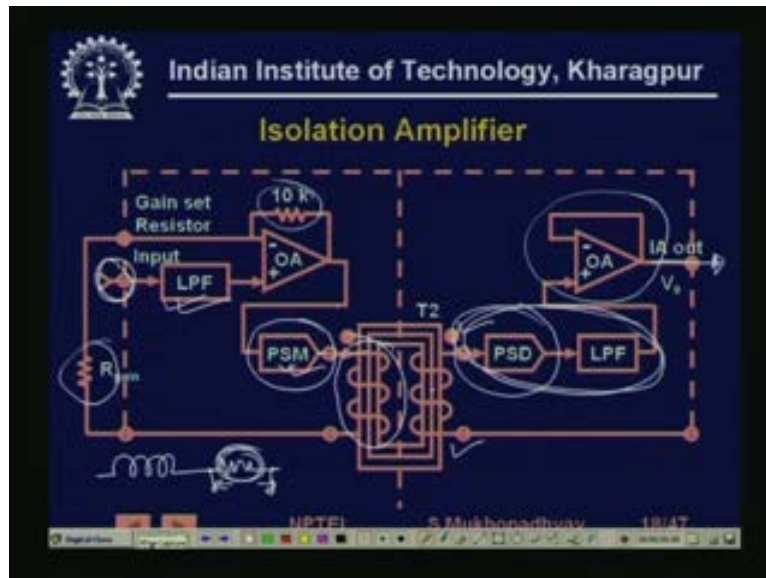


Next we have a charge amplifier. So, you see that there are sudden cases where, we need to produce a voltage which is proportional to the charge. For example, as we have seen in the piezoelectric crystal, that the piezoelectric crystal produces charge which is proportional to the force applied or the displacement of the piezoelectric crystal surfaces. So, in this case you can see that the input signal is actually assumed to be a pressure so we have put P dot.

So, if you consider this P dot as a P produces charge. So, P dot or the derivative of P will produce current derivative of charge. So, therefore this is modeled as a current source. Now, we see that this current will actually flow in to this path, why not this path? Because of the fact that these resistances because this is an amplifier so this is maintained by virtual ground so this resistant potential difference is nearly 0.

So, therefore, with the impedances here the current flow into this impedances is actually is going to be negligible. So, actually this is the low impedance path and therefore, the current will flow into this capacitor and will get integrated. So, when it will get integrated this is pressure dot. So, the output voltage is going to be proportional to pressure. So, this is very-highly used in piezoelectric crystals also because of the fact that even if these resistances vary it is not going to affect the result. So, you can connect a cable and the cable capacitance does not affect the sensitivity of the device as we have seen in the case of piezoelectric.

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Next, we talk about an isolation amplifier. You know there are certain cases where the input sensors may be at some very high voltage. For example, suppose you want to measure a motor current. Now, we typically what we do is we take the winding of the motor and in series with the winding. Sometimes we just connect a resistance and we tap this voltage just to sense the current through the motor winding.

So, now although this potential difference is actually proportional to the current, but these can be these points can be at very high potential compared to ground at hundreds of volts. So, therefore, it is important that you do not directly connect these, such signals into electronic circuits it may even damage the circuit.

So, for such thing isolation amplifier are used. So, there are various ways of isolation there are optical ways of isolation, there are transformer coupling of isolation. So, in the isolation amplifier from the input side and the output side, there is no as such no galvanic electrical connection. So, the signal is transmitted from one side to the other by some other means, in this case it is the mutual inductance or flux linkage. So, what happens is that the input we applied here. See this part is actually floating, this part has is not connected to the ground of the circuit at all. So, therefore there is no problem applying high voltage here.

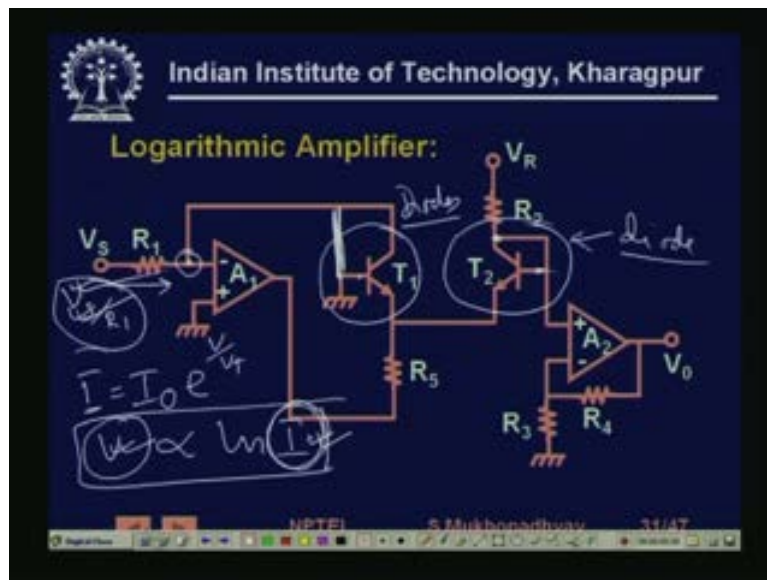
So, the input is amplified with these resistances and then applied to this. This is the input actually applied with a phase sensitive modulator because this is the transformer coupling, so therefore it has to be AC. So, therefore basically a sine wave is modulated by the input. So,

the amplitude sine wave goes up and down as the input increases and decreases. So, that sine wave is actually applied to the primary of the transformer.

So, we shall know that in the secondary we will get the same wave only we will get it by a some trans ratios. So, in the secondary we get the same wave, but see that there is no electrical connection between the primary and the secondary. So, therefore, this the secondary side grounds and reference potentials are completely different from the prime primary side ground.

So, now this signal is taken and passes though a phase sensitive detector we will see what it is, which will strip it of that sine wave which was added here. So, this PSD and low pass filter will finally, extract this signal, which we applied and then it will be passed through this is nothing but a buffer. So, it will be applied to the buffer and then will get the voltage out. So, you see that we have this is called an isolation amplifier because the input side and the output side are said to be completely ohmically or galvanically isolated. So, this is another kinds of amplifier which is often used.

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Then in some cases for example, we need to do various kinds of computation. For example, we may be interested in sensing a voltage and a current signal and then measuring power in which case we need to measure, we need to multiply. There may be cases for example, in the case of flow measurement as we have seen that the flow is actually proportional to root over

of ΔP and what we sense from a meter is essentially ΔP . So, we need to compute root over of ΔP .

So, for such signals as we know that a logarithmic amplifier is of great use because if you want to have a multiplier. Then if you take the log then in the log domain it becomes an addition. So, you can actually first compute the log of signal A then compute log of signal B then add them. So, you get $\log A$ plus $\log B$, which is equal to $\log AB$ and then you take pass it through an anti-log amplifier which will give you AB . So, you have multiplied using two log amplifier, one normal summing amplifier another anti-log amplifier.

So, this is the principle which is used and the logarithmic and the anti-log amplifiers are basic building blocks of that and in the logarithmic amplifier we will not go into a detail derivation of the circuit, but the basic idea is that look at this transistor, the collector in the base are shorted. So, therefore, this is behaving like a diode.

See this is a virtual ground this is also at ground, so effectively this collector and base are also short. So, these are two diodes actually and we know that in diodes, what is the voltage current relationship? So, the voltage current relationship is I is equal to $\sum I$ naught in to $\sum e$ to the power V by \sum constant say V by V_T . So, actually the voltage is proportional to log of the current. So, if you can generate a voltage in the output which current is actually proportional. So, in this case see what is the input current, the input current is V_s by R . So, the input current is actually proportional to the input voltage. So, if you can use this amplifier, the output voltage is going to be proportional to the log of current or it will be going to be the proportional to the log of the voltage. So, this is the basic principle and in which we use this log amplifiers.

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Assuming $i_{B1} \ll i_{C1}$ and $i_{B2} \ll i_{C2}$

We get

$$V_0 = -V_T \left(\frac{R_3 + R_4}{R_3} \right) \left[\ln \frac{V_0 R_2}{R_1 V_R} \right]$$

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For example, for this circuit there are certain assumptions which you can make which are actually valid and then you will get that the output is actually proportional to this constant and this is say it is proportional to some constant time log of the input voltage. So, this is the logarithmic amplifier and exactly opposing.

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Antilogarithmic Amplifier:

The diagram shows a circuit with an input voltage source V_S connected through resistor R_4 to the base of a BJT. Resistor R_3 is connected from the base to ground. The emitter is connected to the base of another BJT through resistor R_E . The collector of the second BJT is connected to the base of a third BJT through resistor R_1 . Resistor R_2 is connected from the collector of the first BJT to ground. The output V_0 is taken from the collector of the third BJT.

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Using the same principle one can also construct an anti-log amplifier. Actually this circuit is as it is drawn is erroneous, this line will not be there, this line should be deleted. So, you see again we have a diode here and we have again because of this virtual ground, we have a

diode here. So, in this case you can understand that voltage here at this point is actually proportional to V_s . It is actually R_3 by R_4 plus R_3 in to V_s .

So, this is connected in negative feedback. So, at this potential also you have you have the same potential so the current that is flowing that it cannot flow in to this. So, therefore it will flow entirely through this diode and the output voltage will be such that the current that is flowing will be also entirely through this diode. So, using these principles you can find out that the opposite effect can be achieved, if you write the equations.

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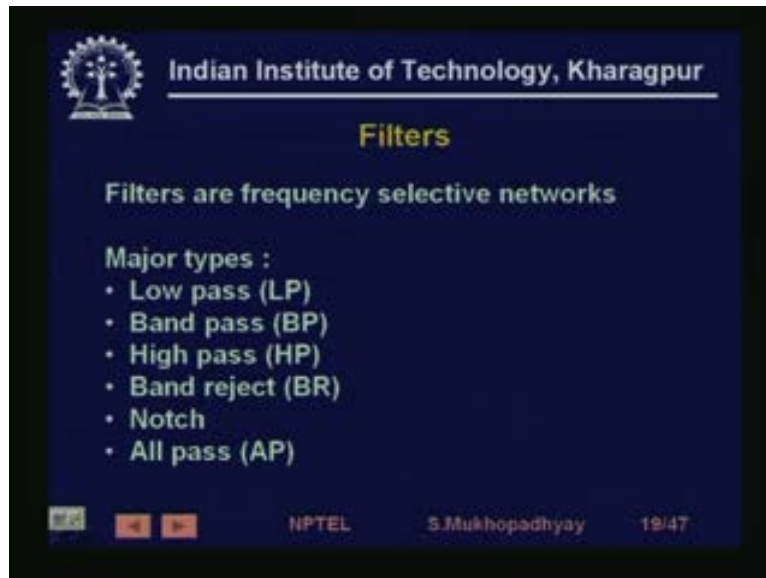
Here

$$V_o = \frac{R_1 V_R}{R_2} \exp \left\{ \frac{-V_s}{V_T \left(\frac{R_3}{R_3 + R_4} \right)} \right\}$$

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So, the output voltage in this case becomes exponential of V_s , so it is an anti-log operation.

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Now we come to the other important block, which is called filter. So, we have various kinds of filters for example, the filters are basically frequency selective network. So, as we all know that any arbitrary waveform can be always thought of as a sum of sinusoidal waveforms of various frequencies. The job of the filters is that they will allow some of the frequencies or they will amplify some of the frequencies and they will not allow some of these frequency.

So, they are frequency selective. Some frequency for example, it often turns out in instrumentation we are interested in accurately measuring signals. See in many cases the process variable that we want to measure is actually in a slowly varying quantity, so it has only low frequency components. On the other hand while we are acquiring the signal may be it is going through a cable, may be somewhere some you know some high frequency noises like from the power lines are etcetera getting into the signal.

So, we would like that we can separate out the signal of our interest and this undesirable signals like this power signals. So, if you want to do that for example, we can say that our signal is within 1 hertz actual signal power signal is 50 hertz. So, can I have a network which will have nearly 0 gain for at 50 hertz and quite good gain at 1 hertz's. Then if I pass my noisy signal through it, then the power frequency signal will be killed and the real signal will be passed. So, this is the basic purpose of filters

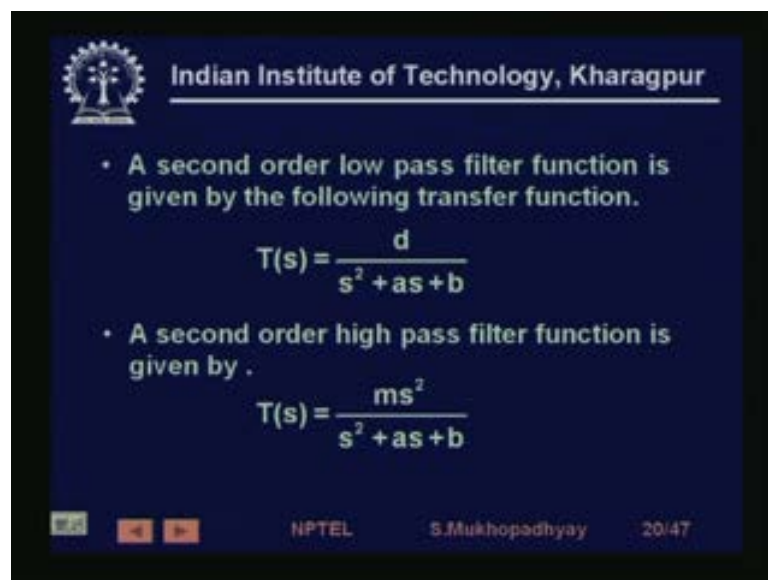
So, they are actually frequency selective networks and there are various types of filters. So, for example, we have low pass filters in which we pass low frequency and stop high

frequency. We have band pass filters where we only pass within a certain band from some frequency f_{min} and f_{max} . The lower than f_{min} are also rejected, higher than f_{max} are also rejected, such filters are called band pass.

Similarly, we have high pass in which we reject below a certain level. For example, suppose we know that we do not have any DC signal in our actual case. Like for example, typically when we measure vibration, we cannot have a DC signal vibration. Vibrations are always you know kind of high frequency. So, we can eliminate, but some DC's may come due to other kinds of biases. So, you can eliminate them using high pass.

We can also have some band reject for example, I was saying that we can reject 50 hertz. So, we can create a filter which will just reject a very low gain around 50 hertz and good gain around both sides of it, low and high, such filters are called band reject. Notch is nothing but a narrow band reject and all pass. Sometimes we have all pass signals where we are not interested in changing the gain, but we are interested in changing the phase. So, these are we mention typically because they are the major types of filters, but typically the low pass, band pass, high pass signals are mainly used.

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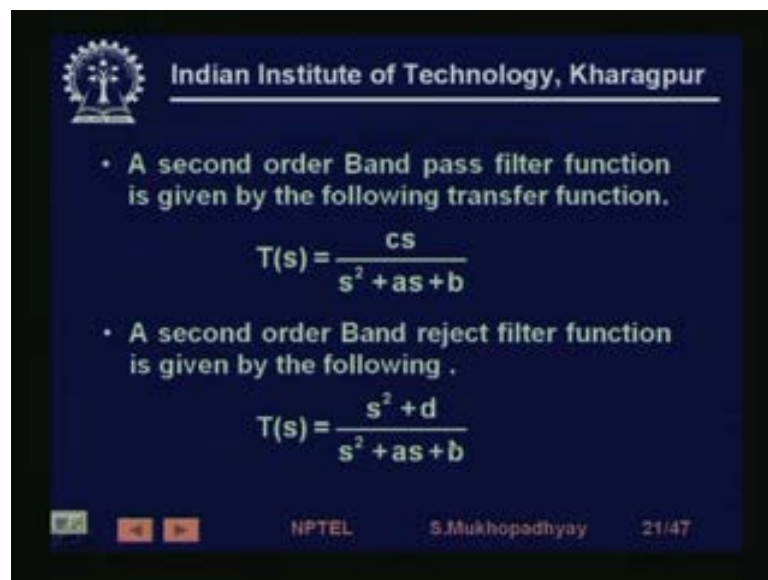
The slide features the IIT Kharagpur logo and name at the top. It contains two bullet points with their respective transfer functions. The first bullet point describes a second-order low pass filter with the transfer function $T(s) = \frac{d}{s^2 + as + b}$. The second bullet point describes a second-order high pass filter with the transfer function $T(s) = \frac{ms^2}{s^2 + as + b}$. At the bottom, there are navigation icons, the text 'NPTEL', the name 'S. Mukhopadhyay', and the slide number '20/47'.

Low pass being the most common. So, if you see the domain of transfer function, then if you see second order transfer function then this is the typical low pass transfer function. You can find out where it is low pass because if s tends to infinity, then the denominator becomes very high and the value of $T(s)$, which is the gain of the filter becomes nearly 0. While as s

tends to 0 the gain of the filter become d by b, which is a constant. So, at low frequency it has a constant gain, but at higher frequencies the gain falls and approach a 0.

For high pass on the other hand, there is an s square term on the top. So, when s tends to infinity, it becomes m, m is the high frequency gain which is constant. As as s tends to 0 because the denominator tends to b, but the numerator tends to 0. So, the gain goes to zero, this is high pass.

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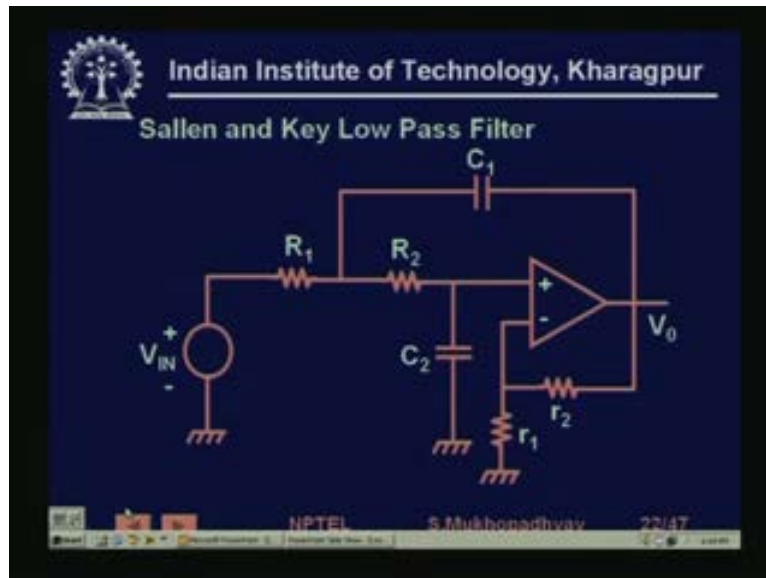


The slide features the IIT Kharagpur logo and name at the top. It contains two bullet points with their respective transfer functions. The first bullet point describes a second-order band pass filter with the transfer function $T(s) = \frac{cs}{s^2 + as + b}$. The second bullet point describes a second-order band reject filter with the transfer function $T(s) = \frac{s^2 + d}{s^2 + as + b}$. At the bottom, there are navigation icons and the text 'NPTEL S.Mukhopadhyay 21/47'.

Similarly, you have a band pass in which you can see that if s tends to 0, then also this gain tends to 0. If s tends to infinity, then it becomes you know s and this s square term become dominating. So, this becomes cs by s square so again it goes to 0. So, on both sides for very low values it is 0, for very high value it is 0, in between it has a gain therefore, it is band pass. In this case this is band reject because if you have s close to d, then you can see that this s square term if you put normally we put s is equal to j omega. So, then you have a term d minus omega square.

So, as omega approaches root over d the gain of the filter will fall. As omega tends to infinity this gain is one and as omega tends to 0 this gain is d by b. So, on both sides you have good gain suddenly around d, the gain will fall to 0 this is a band rejected filter.

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There are various ways in which filters are realized. You can have purely passive network, RLC networks for realizing filters, but most modern filters are realized using RC elements. And because that is what is typically used in electronics and using amplifier because you simultaneously want to have a certain amount of gain in the filter. These are very well known architecture call the Salle and key low pass filter. So, it turns out that this is the low pass architecture. In fact, you can have a Salle and Key low low pass high pass filter. In fact, if you change the resistances and the capacitances, we inter change them, then the filter transfer function goes from low pass to high pass.

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From circuit we find that

$$\frac{V_o}{V_{IN}}(s) = \frac{k/R_1 R_2 C_1 C_2}{s^2 + s \left[\frac{1}{R_1 C_1} + \frac{1}{R_2 C_1} + \frac{(1-k)}{R_2 C_2} \right] + \frac{1}{R_1 R_2 C_1 C_2}}$$

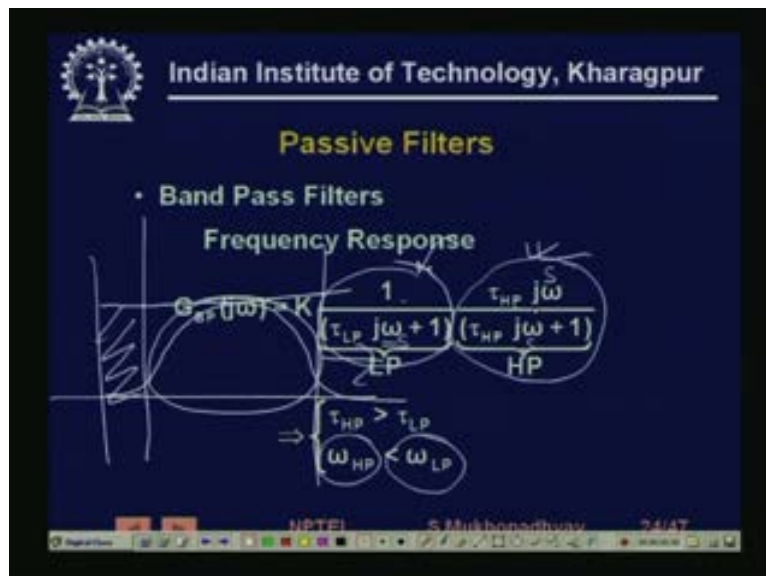
Or, $\omega_p = \sqrt{\frac{1}{R_1 R_2 C_1 C_2}}$

So, in the Sallen and Key filter the transfer function is given like this where, this K is actually the gain of the last hop block, which is $1 + \frac{R_2}{R_1}$ because it is a non-inverting block if you have noticed. So, obviously from the transfer function, we realize that it is a low pass transfer function, the top is constant and the bottom you have an second order polynomial in s.

So, you can see that for second order transform whenever we have a filter we are concerned with what is the pass band and what is the stop band. So, we want to know that which are the bands in which we can assume that the signal will be passed with good gain constant good gain and which are the bands in which the signals will be stopped. So, while the stop is not abrupt. In fact, you can make it more and more abrupt by designing more and more complex filters of higher and higher order, but generally the gain will fall continuously. So, at some point we say that we call it a bandwidth.

So, we say that if the low frequency gain falls by a quantity called three db or when there low frequency gain becomes $\frac{1}{\sqrt{2}}$, then we call that three db bandwidth. So, that is the term which you know roughly signifies that up to what frequency the gain will be more or less high. So, that is given by in this case $\frac{1}{\sqrt{2}} \frac{1}{R_1 R_2 C_1 C_2}$.

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Generally, you know what order filter you will design etcetera depends on the application itself. So, using low pass and high pass filters, you can very easily construct band pass filters. For example, see it is very simple that this is a low pass first order transfer function. You can

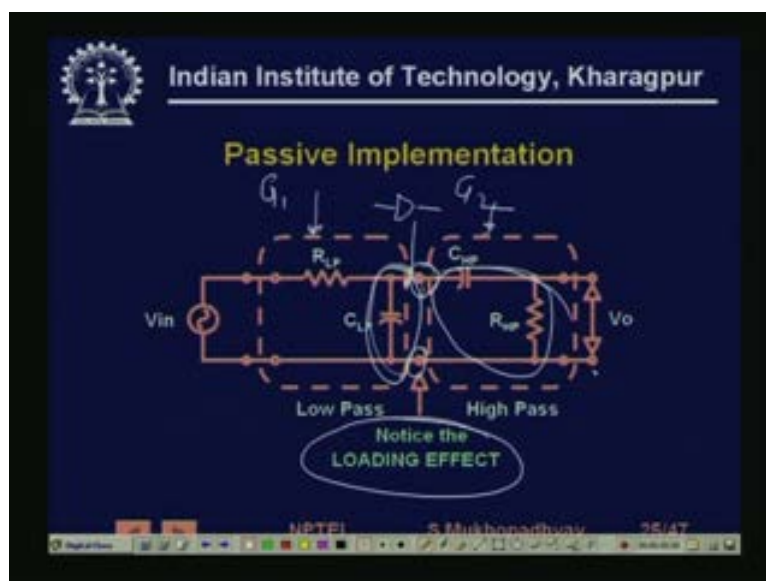
see that it is low pass you can treat take it as s and then as s tends to infinity, this gain will go to zero, so it does not pass high frequency.

On the other hand this is high pass because this is another s , this another s and this is another s . So, s goes to 0, this goes to 0, but as s goes to infinity this will go to 1 nearly. You can see that what we have done here is that we have cascaded a low pass and high pass and we are ensuring that the cut off frequency of high pass is actually lower than the cut off frequency of low pass.

So, if you see on the frequency axis then the cut off frequency of the low pass is like this and this is for the high pass and this is lower than the cut off frequency of the low pass. So, you see that when you multiply them around here, the gain is 0, why because the high pass gain is 0, there are in series. Around here the gain is almost 0 because low pass gain is 0 and in between there is a frequency band in which none of them are 0 and they are substantial and you get a gain.

So, you get a band pass filter by cascading a low pass and band pass filter and ensuring that the cut off frequencies of the high pass is lower than that of the low pass. So, this is a simple way of creating a band pass filter. Band pass filters are very often used in a you know vibration monitoring instruments, which are widely used in industries to check the health of rotating machinery like turbines motors large motors.

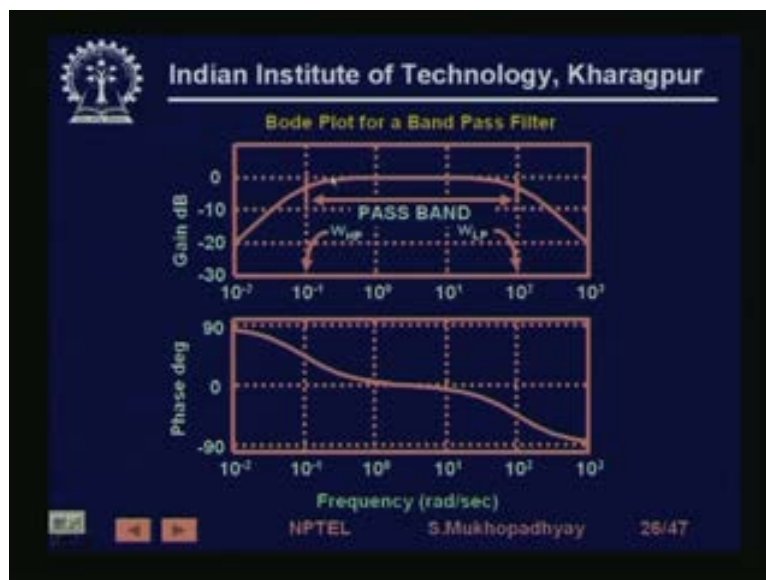
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So, this is the circuit implementation. You have a low pass and you have high pass and only one thing is to be remembered that there is a loading effect meaning that just because this has a low pass transfer function G_1 and this has a transfer function G_2 . This resulting transfer function is not going to be $G_1 G_2$ exactly because of what is known as loading.

So, if you want to prevent loading you can put a buffer here. Otherwise, it is going to be still it is going to be band pass, but it will be slightly different than $G_1 G_2$, because of electrical loading because this side is not open. When we derive this transfer function, we assume that this force are open, but this is not open. So, it will see the effective impedance of this branch will be this in parallel with this, so that that will happen.

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So, this is a typical band pass plot as we have shown. Low at the low frequencies, low at the high frequencies and in between high.

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Passive Filters

- Notch (Band Stop) Pass Filters

Frequency Response

$$G_{BP}(j\omega) = \frac{1}{(\tau_{LP} j\omega + 1)(\tau_{HP} j\omega + 1)}$$

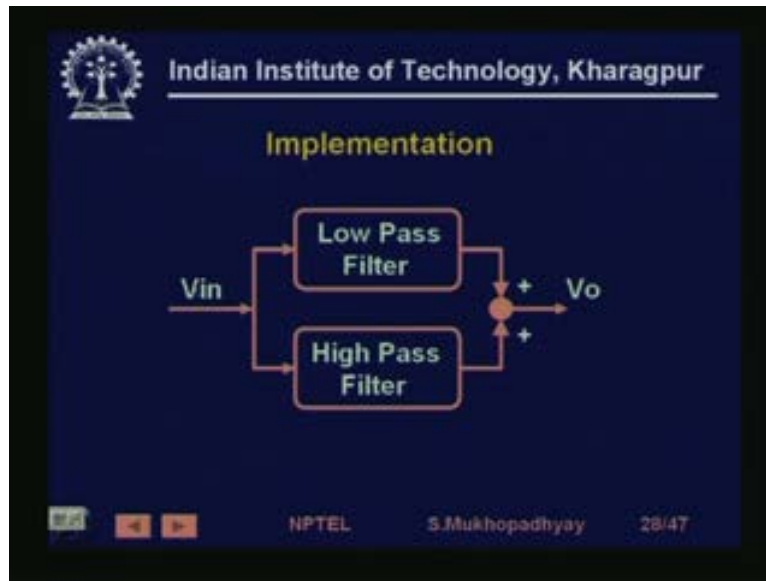
LP HP

$\tau_{HP} < \tau_{LP}$
 $\omega_{HP} > \omega_{LP}$

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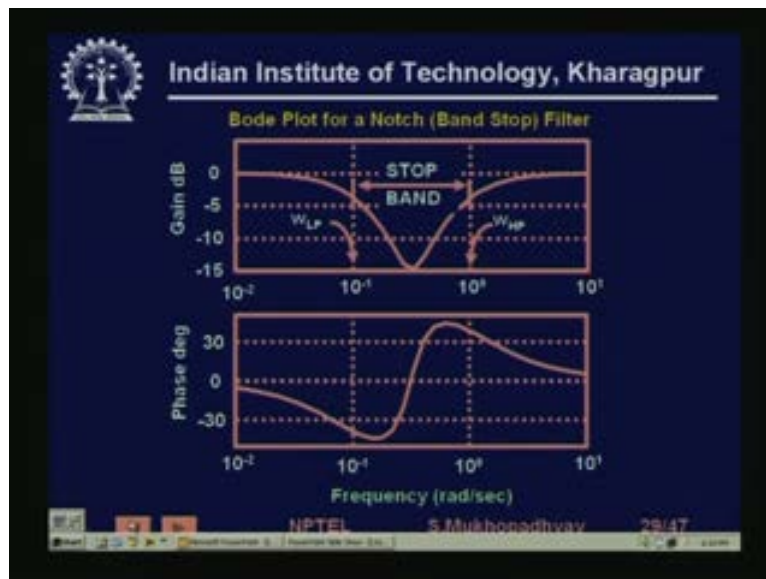
Similarly, you can have a band reject filter this is a band stop filter. So, if you have a band stop filter and now if you have the cut off frequencies where the omega high pass. So, this is low pass and this is high pass. So, now cut off frequency of high pass is actually higher than the cut off frequency of the low pass. So, now if you actually see if you connect these two filters in parallel then what will happen is that, at the low frequencies this low pass filter channel will pass. So, the signal will pass in the low frequencies. Now, in the high frequency the parallel the high frequency channel will pass the signal, but in between since both gains are zero. So, therefore non of the channels will pass and you have what is known as band stop filter.

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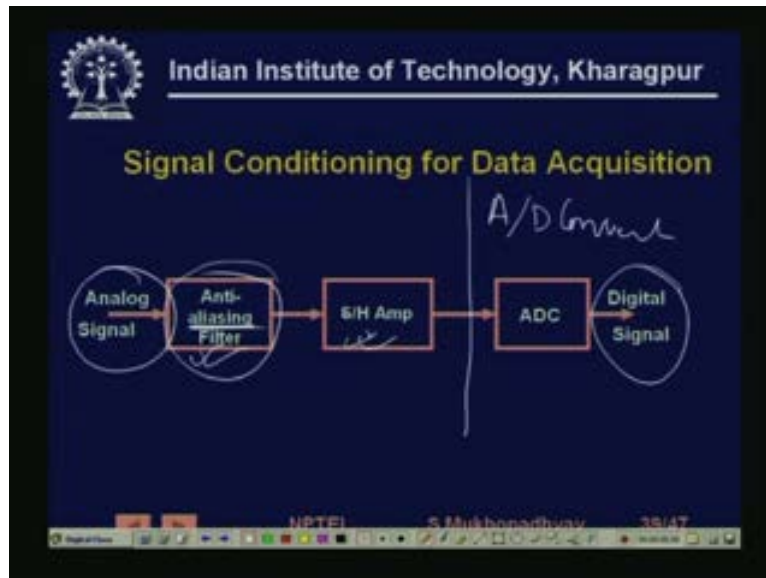
So, that is what is shown here, that low pass and high pass are shown in parallel.

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So, Bode plot is also shown. The Bode plot as you see you can see that falls down at some frequencies. So, reject this stop band.

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Now, we have seen the amplifiers and we have seen the filters. So, now we look at some two other types of signals, which are very common instrumentation. For example, it is very common that the sensor signals are all analog, but it is very common now a days that you would like to acquire those signals into a computer or some sort of a digital computing hardware and then process it further. May be send it by some communication means etcetera.

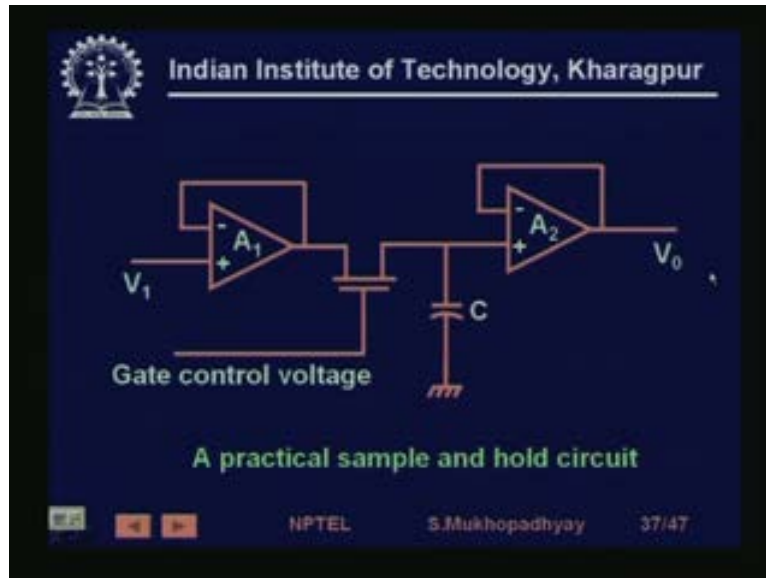
So, the first step that has to be achieved for that, is that generally what you do is you connect it through some sort of a data acquisition hardware which actually takes as input the analog signals and produces as output a set of digital signals, which are read by the computer on the bus and then stored on the memory etcetera, etcetera. So, there are large number of such devices you get such as acquisition cards, you get a data loggers. So, all these things are used in instrumentation widely. So, we are going to see how this thing basically takes place.

So, we have an analog signal here and we finally, want to produce it is digital equivalent right it turns out that; obviously, at the final stage we need an A to D converter, A D converter. So, but it turns out that you need to sample the signal from time to time and it also turns out that you need to put a filter before you put the sample and hold amplifier. This is related to the fact to an effect which is called aliasing.

So, therefore, you need actually put an anti-aliasing filter, which is simply a low pass filter and whose cut off frequency is related to the sampling frequency, which you using here. The sampler and hold amplifier basically takes certain values of signals and then holds it constant,

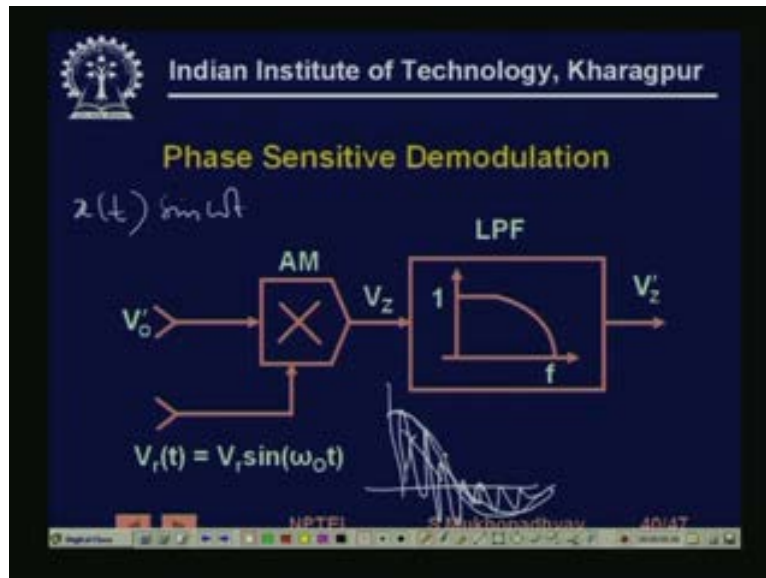
again takes a value of signal again holds it constant. So, it samples the signal and then holds it constant in between. So, you need to put these three typical blocks and anti-aliasing filter, then followed by a sample and hold amplifier and then followed by an ADC.

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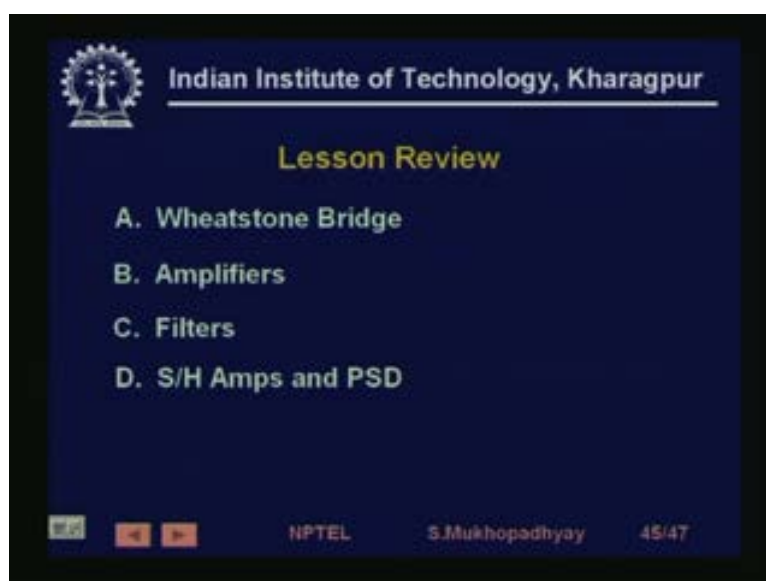
So, this is sample and hold amplifier. So, you can see that this is basically a MOS if you turn on this MOS, this is the buffer. So, this voltage will actually come and charge this capacitor and then if you switch off this MOS then there is no way that this capacitor can discharge. This side is also open, this side is high impedance, so this voltage on the capacitor will be held if you keep it discharge. So, this voltage will be held here. So, every time you want to sample you turn this on, this signal will be transferred here, then you turn it off it will be held. So, this is called a sample and hold amplifier.

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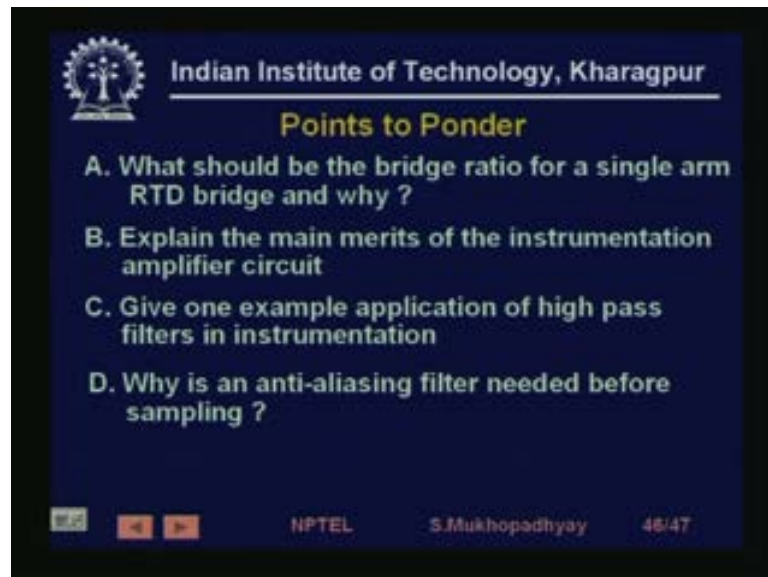
Lastly, we have a phase sensitive demodulation circuit where, you know we have a modulated wave and then we have to demodulate it. So, a modulated wave is typically looks like you have a signal $x(t) \sin \omega t$. So, you have a signal $x(t)$ and $x(t) \sin \omega t$ will look like this. So, you want to from there, you have to extract this $x(t)$ signal and you have to reject this $\sin \omega t$ signal, this is the problem of demodulation. So, what will happen? So, what you do is you actually multiply this signal. So, actually what will do is we will talk about this little bit in our next lecture and straight away go to the lesson review. So, this phase sensitive demodulation we will talk about it in the next lecture.

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So, in the lesson review we have talked about Wheatstone bridges, we have talked about amplifiers, we have talked about various kinds of filters and we have talked about sample and hold amplifiers in the lecture and the phase sensitive demodulation we will talk about it in the next lesson 10.

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Points to Ponder

- A. What should be the bridge ratio for a single arm RTD bridge and why ?
- B. Explain the main merits of the instrumentation amplifier circuit
- C. Give one example application of high pass filters in instrumentation
- D. Why is an anti-aliasing filter needed before sampling ?

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So, points to ponder you can think about what should be the bridge ratio for a single arm resistance temperature bridge. It should be low because of the fact it should be well you should decide that because the temperature is RTD bridge quite high in contrast with the sensor with the strain gauge bridge. Then explain the main merits of the instrumentation amplifier circuit, we have discussed this all ready. Give one example application of high pass filters in instrumentation you can think of something where there is no low pass signal, it is the noise. And why is anti-aliasing filter at all needed before sampling, for this you have to learn what is known as a Nyquist sampling theorem. So, that is the end of the lesson today. Bye.