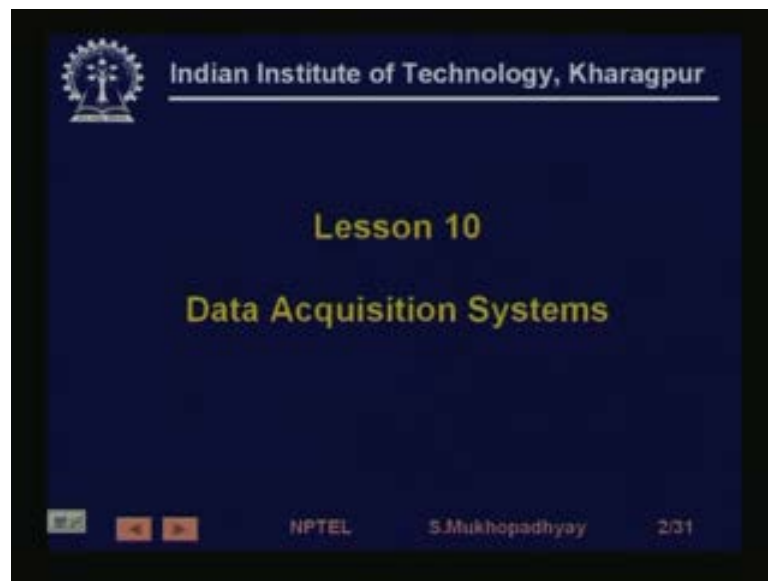


Industrial Automation and Control
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Lecture - 10
Data Acquisition Systems

Welcome to lesson 10 of the NPTEL course on industrial automation and control.

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So, today we are going to talk about data acquisition systems, and before we describe the instructional objectives let me say few words about why they are so important you know. So, far in the course we have first of all in the first 2 lessons we have seen that we have seen the industrial automation pyramid right.

So, we saw that one of the, one of the major features or characterizing features of advanced automation is that there is a lot of data flow up and down that is data actually gets into the into the computer, and it is all about computers. Because there are computers at every layer of the automation pyramid of various types, which do a lot of real time computing right and they do control optimization etcetera. And that is how the benefits of industrial automation are actually realized.

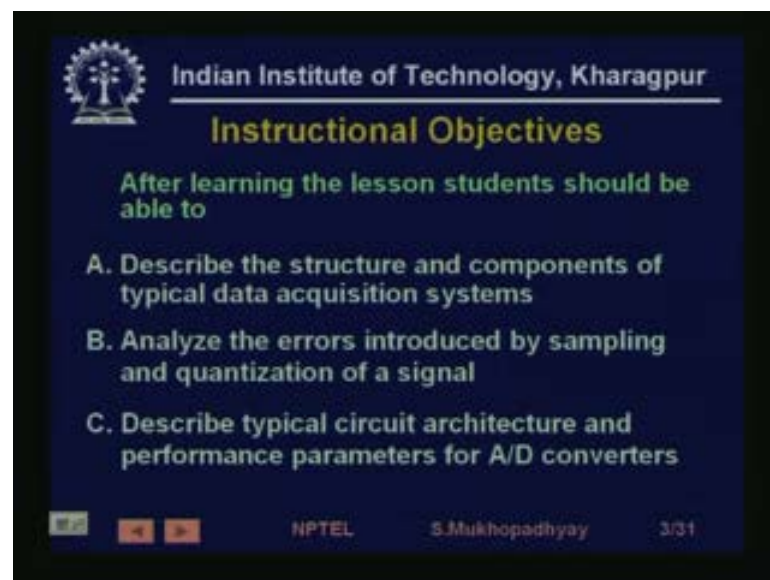
So, there is a lot of data flow from one level to the other right. And now all these data, which is actually basically acquired from the plant floor where the, where the, where the

actually machines are from the plants from the, from the, from the industrial equipment they are, they actually get into the these computers systems through the sensors.

So, we have studied sensors that how these process quantities are sensed and today we are going to end our sensor module by looking at the data acquisition systems, which will interface to the sensor on one side and to the computer on the other side. So, through the systems the data will the analog data usually analog there can be some digital data also.

So, the analog data will come through the sensors get converted from their physical forms in to some electrical forms, and then through the data acquisition systems will get in to digital form into the computer. Then, they are going to flow of then they are they are going to be utilized by the various algorithms residing at this computers, and they are going to get communicated to other computers after various processing and get utilized at the various levels of automation. So, what we are going to study today is how the data which is, which is coming from the sensors gets into the computers or how digital data is going to be acquired right. So, that is the subject of the lesson today.

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Instructional Objectives

After learning the lesson students should be able to

- Describe the structure and components of typical data acquisition systems
- Analyze the errors introduced by sampling and quantization of a signal
- Describe typical circuit architecture and performance parameters for A/D converters

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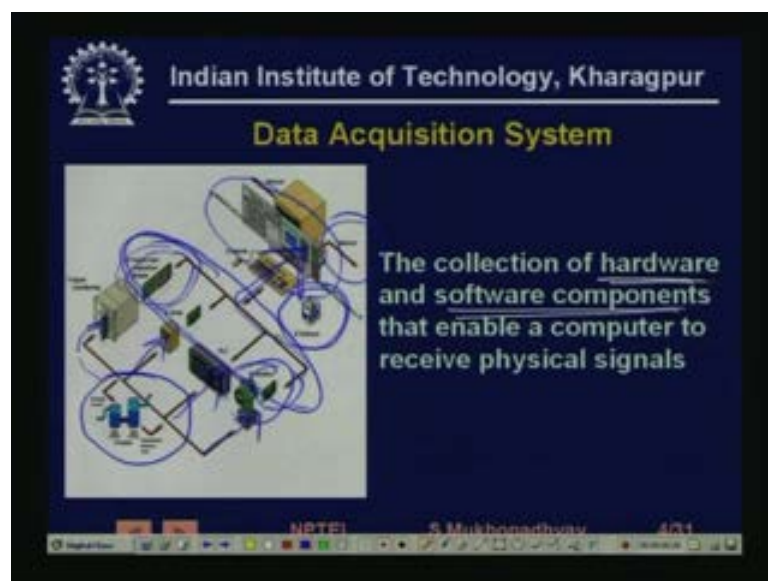
So, moving on we have as instructional objectives to get familiar with the structure and components of typical data acquisition systems, and to understand the basic mechanism of the process of sampling. You know by which data sampling and quantization because when we have digital data we do not have all the points on the, on the continuous

process. But we have points values of the signals, which are at close intervals or at intervals of the sampling time, and it is not only it is, it is not a because we are going to manipulate in the computer.

The computer although it has a large number of bits and usually quantization may or may not be an assume, but nevertheless there is a quantization issue as well, and that whether it is important or not that depends on the computer. So, it is, if it is an 8 bit computer then it could be important if it is a 32 bit floating point computer, it may not be important

In any case we will take a look at the basic concept of sampling and quantization and finally, we will see some typical circuit architecture. They are actually physically how is it that the analog electrical signals get converted to digital signals, which are interfaced with the computer hardware. So, we will we will look at that. So, these are the basic objectives.

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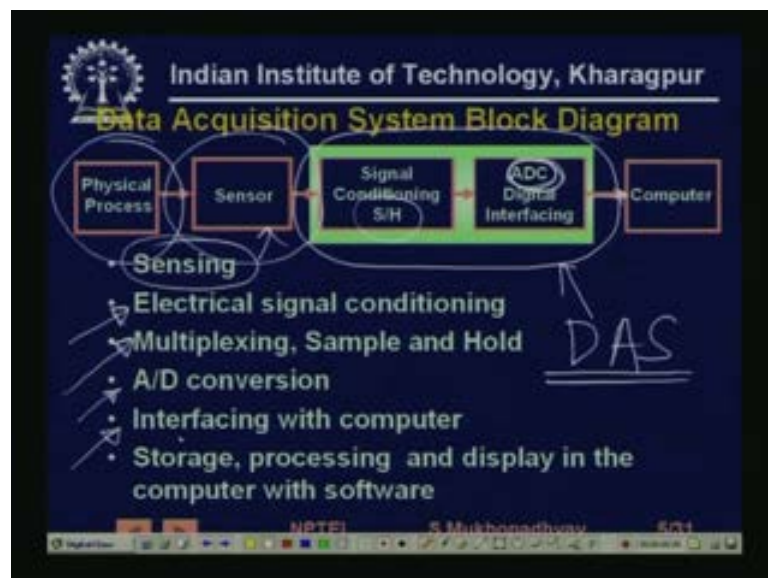
So, coming on to a data acquisition system, what this figure we want to define it first. So, we define it as follows: that it is a right. So, it is a collection of hardware and software components. Let me, let me choose. So, it is a collection of hardware and software components that enable a computer to receive physical signals. So, you see this is what this picture says that this is the process may be I have to change my pen again.

So, this is the process then and there are various you know hardware. For example, data is may first enter through signal conditioning modules it may, it may be serial data also sometimes it may go to, go to, got to P L C or it may go through. You know, these are some actuators. For example this look likes a, this looks like a valve some something like a motorized valve. And finally, from all these equipment there are through data acquisition processes it actually gets into a it gets into a computer.

So, once it gets into the computer through this sort of you know electrical boards, and there is some software residing in the computer. So, this software does 2 things it. Firstly, helps to helps these cards to transfer the data into the computer or rather interfaces and. Secondly, it may help in you know the actual usage of the data that is in terms of display, in terms of decision making trending alarm generation, what have you?

So, some of it may be utilized at that computer itself, where it is being acquired and some of it may actually be transmitted to other parts of the system through computer ne2rk. So, at that point it becomes pure computer communication. So, we are primarily going to look at this part of the system where from various equipment on the plant the data gets into the computer, right.

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So, let us first look at a block diagram, and see the major functionality terms, I am sorry, I do not know what is happening here. So, this is in this block diagram, we have this physical process change of pen again, this is the physical process, and then this is the

sensor up to this we actually understand. Now, from the sensor, now we as we have what we have learned that the sensor itself we have some signal conditioning, but at the same time there may be further signal conditioning required or in some case if the sensor signal conditioning. If you know it is if it is not possible to put signal conditioning electronics at the sensor.

Sometimes the sense the signal conditioning electronics, I think like amplification can be put as part of the data acquisition card itself. So, you have such signal conditioning here, which is typically analog signal conditioning and then you have a sampling, and hold circuit, which is, which is put. We will, we will see what it does and then finally, it gets into digital domain through a, through a circuit, which is typically referred to as the analog to digital converter or the ADC. This is a very well used to term and then.

So, at the output of the analog to digital converter you have bits. So, you have a number of bits, which represent the value of the analog signal at a particular sampling instant, and then that those lines or those bits have to be transfer transferred to the computer. So, you need an interfacing mechanism by which the computer can accept that digital data.

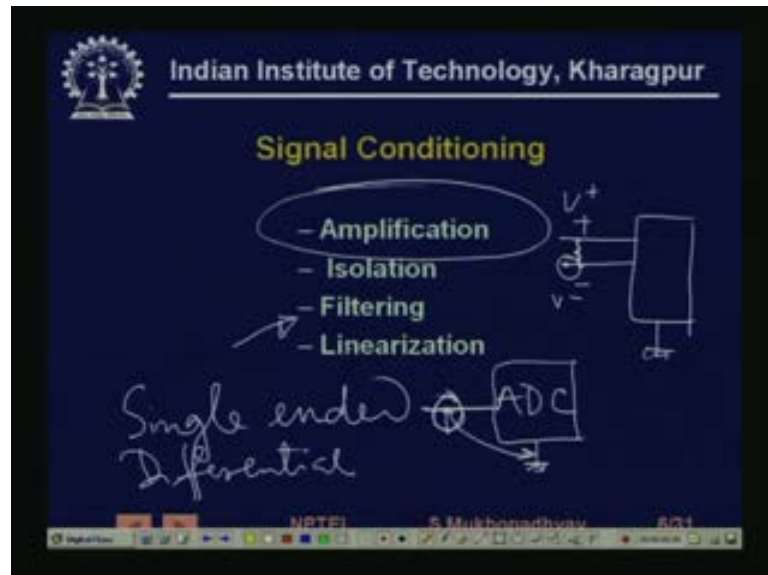
So, these are the typical you know this is the part, which is called the data acquisition system. This is the data acquisition system, which we will sometimes refer to as a D A S. So, this overall process, what does it? The sensing part is comes from the sensor, which is typically not a not part of the signal conditioning, and then there is electrical signal conditioning. Then, there is then there could be multiplexing sample and hold multiplexing means that you know time division multiplexing that is looking, if you have a number of analog sensors very quickly you scan the channel.

So, first see this one convert this to digital, then see that one convert this to digital and so on. So, that is called multiplexing and then sampled and hold. So, sampling and holding the signal for that small interval of time, when it is being converted. Then, A D conversion the process of converting it to digital and interfacing with computer, and then finally, this is also not strictly a part of the data acquisition board, but the, but sometimes it is a part of the data acquisitions.

It is considered to be a part of the data acquisition system because generally data acquisition vendors will not only give you this sort of components. They will also supply you with the software, which works with these data, and can you know display it can is

can store it can trend it analyze it. So, various functionality software are also provided for ease of use. So, that is what you do in typical data acquisition system.

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So, what you do, we do in signal conditioning. So, what you do in signal conditioning is you could do amplification. This is very important because as we will see in detail the important because every any A D converter has, what is called is dynamic range. And it is important that the analog signal that you are sort of presenting at the input port of the A D converter is utilizes the dynamic range of the A D converter.

Otherwise, you are going to have approximation errors larger approximation errors than are necessary. So, it is important to amplifier the signal to increase resolution isolation is typically required because the field signals can be sometimes be at high. For example, you know these analog channels generally come either as single ended or as differential.

So, when you have single ended it means that this value that is going to be the value of the analog voltage is actually with respect to the ground electrical ground of the A D converter. So, the A D converter is also an electrical circuit that has a ground. So, when you are applying it in a single ended mode this voltage will be measured by the A D converter with respect to its own ground and then convert it that value.

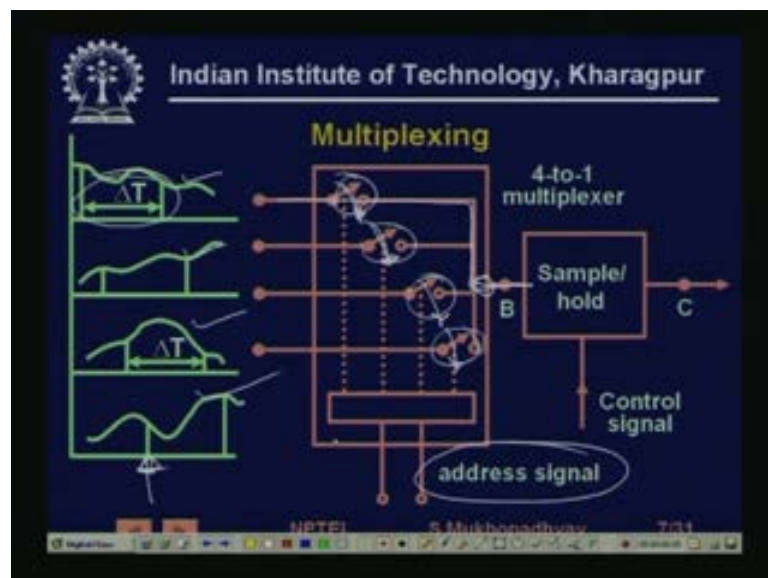
On the other hand when you, when you give it a differential input then what happens is that there are 2 inputs provided to the A D converter. So, there is a plus and there is a

minus terminal, and the difference in these 2 signals are provided, right. So, this the potential of this can be quite different from the ground. So, now the voltage difference V plus and V minus actually the signal value that you will get will be proportional to V plus minus V minus.

Now, this V plus and V minus can be at pretty high voltages, if they are coming from the let us say motor winding suppose, you want to measure the motor winding temperature. So, it is to, it is it is not. In fact, if you connect such high voltages to the A D converter, electrically it might it will it might get damage. So, therefore, what you do is you put an isolation circuit such that the input side is actually galvanic ally isolated with the output side.

You have various mechanisms of isolation like optical like you know capacitor based or transformer based inductive coupling etcetera. Then you have filtering is required for noise removal as we will see it is also required for phenomenon called anti aliasing. So, and we could do some linearization in the signal conditioner itself or you know sometimes you can do the one of, one of, one of the benefits of digital data acquisition is that you can do that linearization much probably much more easily in software.

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Now you have. So, this is, this is how you condition the each analog channel before you present it for A D conversion, right. Now, it turns out, if you see a, if you see any standard data acquisition system that they typically will specify, that they can take 8

analog channels simultaneously apparently simultaneously. Now, how do you get 8 analog channels simultaneously. So, typically it is the conceptually the scheme is something like this. So, it is, it is down through a process called multiplexing.

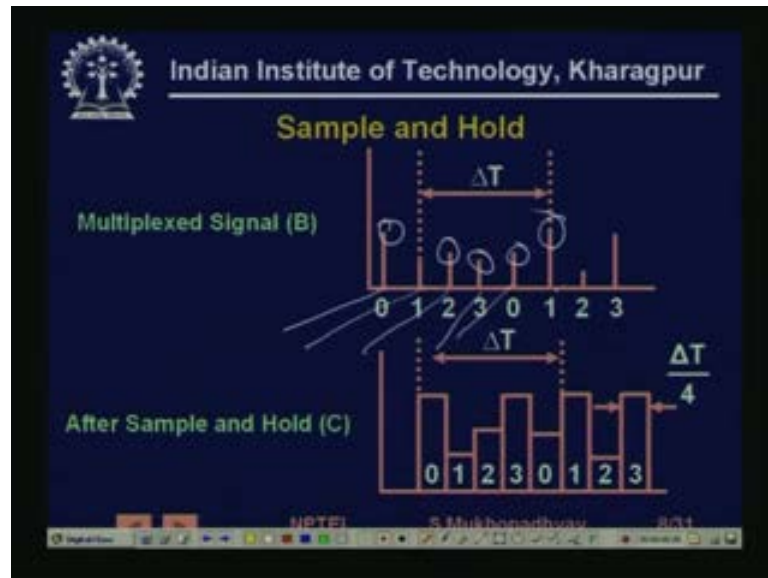
So, typically you have been now let us say, these 4 channels of analog signals are being presented at these 4 inputs right. So, what you do is if you these are you know, electronic switches, which can be put on or off depending on this address signals. So, may be this address is 0 0. This is 0 1, this is 1 0, and this is 1 1, right. So, what happens is that if you close this switch?

Then, this signal gets connected actually this is a, this is not connected here. This connected, and goes to the sample, and hold and goes to the A D conversion. On the other hand if this signal is connected this will go. So, actually, so if you connect these 4 switches in quick succession then over a time interval let us say the overall time interval is ΔT . Now, within this ΔT , if you divide ΔT by 4, and then apply these switches on at these one after the other within that overall time ΔT . Then, every ΔT interval you will get 4 values of these signals.

So, now if you, if you, if you, if you, if you are willing to ignore this slight difference between the timings that is if the, if the, if the, if the, if the timing is close enough compared to the rate of variation of this signals. Then, you can kind of, kind of implicitly assume that there are all signals, which are, which these are the 4 channels and after sampling this 4 channels you would get in to the computer 4 values.

So, you can for you future purpose you can assume that those where the values of the signals all the 4, which existed at sometime at the beginning of ΔT at time ΔT without differentiating between this you know ΔT by 4 and all that. So, this is called multiplexing.

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Then, we have sample and hold. So, why we have hold is that while the A D converter is converting the signal, the signal it is sometimes necessary. That the signal is maintained at the input, but for, but that maintaining need not be done by the switch, because you see the switches take finite time for getting switch on and off.

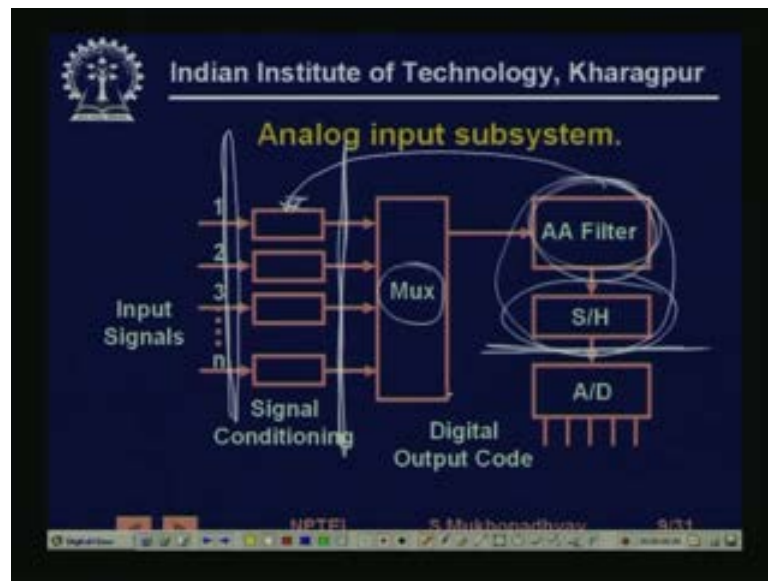
So, you just. So, you put another additional circuit such that you put on the switch. So, let this voltage be sensed by that circuit, and then this circuit is this circuit is called the sample and hold circuit, the circuit is such that it will hold that voltage value. Now, you can open the switch again, but this value will be held. So, the A D converter will actually see this held value even if this switch has been opened. So, this opening will take sometimes and it will be ready to close the next switch after let us say another delta T by 4 right.

So, this is called the sample and hold procedure where a particular time instant value is told, and held for a small interval within a circuit, right. This is called sample and hold. So, while at the switching point you may, you may close the switch for very small points of time. So, you get these values at the of the various see these are the samples. So, while this could be for channel number 1, channel number 2, channel number 3, channel number 4 and then again channel number 1 to 4. So, you have these 4 channels, but at the output of the hold circuit you will find that this value 0 is being held up till the value is

sensed to again the hold circuit is instructed that. Now, you release that held value and now acquire a new value.

So, by that time channel, the next channel switch has been put on and the value has stabilized there. So, now, the hold circuit senses a new value, and then again holds it for the next interval. So, it holds it for the next interval then again it senses the new value, and then again holds it. So, this is what happens by a sample and hold, right.

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That is because we need to hold it because otherwise sometimes the A D conversion can get in to error, if the signal, if while the converter is converting the signal vanishes from the input terminal, then the A D converter can get into problems. So, now we need see basically between the in between the signal conditioned input and the A D converter input. There is this block, which actually does two things: first it will multiplex several channels and second it has to do sample and hold. Now, this it can do in 2 ways and which one you will choose depends on, depends on how fast your signals are varying and how fast is your sampling frequency.

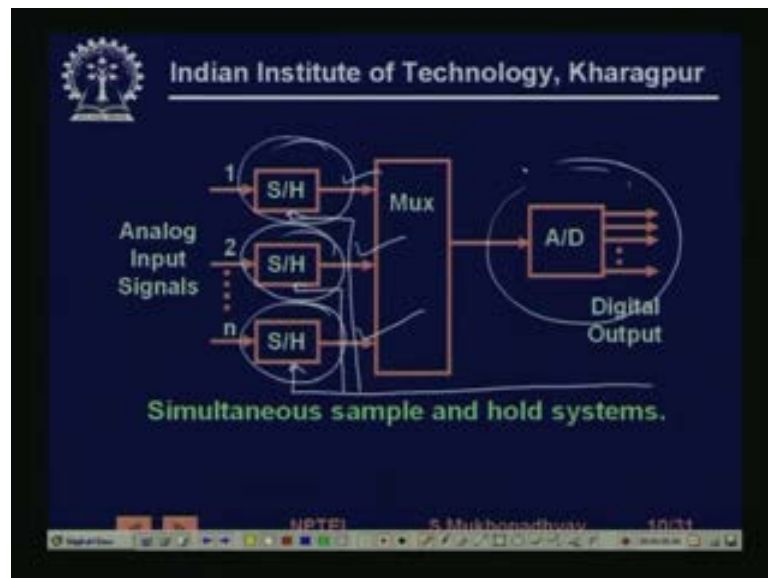
So, this is a case where you see that. So, we first present this architecture this is, this is the analog input subsystem. So, here is the A D converter. So, you are seeing between these input signals, then signal conditioning, then this is the A D converter input. So, here in this case what you are doing is that you have put a multiplexer on each channel and then you put, you know anti aliasing filter this can be also, this can also be a part of

this filter, if you want because the anti aliasing filter may be different for different channels and then you put a.

So, if that is not required to have different anti aliasing filter and in the transient response of the anti aliasing filter is fast enough, then you can actually have you can actually put one anti aliasing filter, and then one sample and hold amplifier. So, you see that this channels you put you are saving cost because you are putting only one anti aliasing filter and one sample hold circuit, and for all the let us say 4 or 8 channels. So, the assumption is that.

Now, what happens is the remember that. So, the difference is that now when you get these 4 values at the end of ΔT remember that these 4 values are actually sampled ΔT by 4 times later. So, they are actually not samples at the same theoretically at the same accurate instant of time. Now, if that makes a difference if that does not make a difference to you. Then, having one filter and one sample, and hold is, and that is cheaper right.

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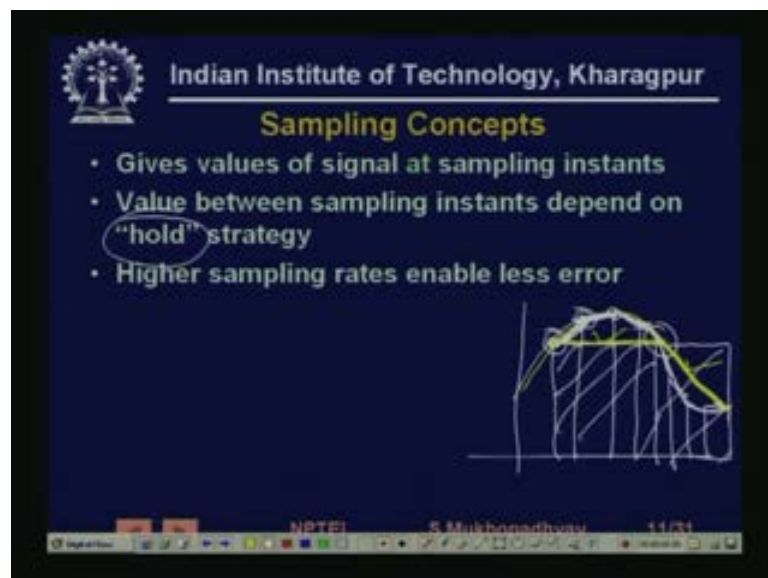
So, you go for this architecture, but if it does make a difference to you then have to look at the next architecture, which is called the simultaneous sample and hold circuit, where after signal conditioning. You have this sample and holds at for all channels individual sample and holds, and a what I have not shown, but what exists is that there is a control signal, which will go to all the sample and holds. So, that each sample and hold will

simultaneously sample all these channels that is possible now because you have because you have put separate sample and hold circuits. So, they will is. So, you, so they will now these, so they will hold the values.

So, now remember that the values, which are being held here correspond to samples at the same instant of time. So, they are going to be read into the A D little bit later, but they are time synchronous in the sense that the those 4 values represent values of process variables at the same instant of time.

So, the sampled in simultaneously and held it simultaneously, and then read it serially because you are having one A D converter. If, you had different A D converters, then probably you could have gone for simultaneously A D conversion also, but that is generally hardly necessary because the A D converters are quite fast and because we are talking about processes physical processes. So, there dynamics would be complete would be quite slow, and the A D converter speed is more than accurate. So, you would and the A D converter is very expensive. So, you do not want to have more than one A D converter rather than have more than one sample and hold channels right for simultaneously. So, that is simultaneously simultaneous sample hold.

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Now, before we look at the A D converter, let us take a look at some sampling concepts. You know, first thing that we must remember that the sampling value, which you get from the A D converter is at the sampling instant. So, it is just at the sampling instant and

ideally speaking. So, you have you have a signal. So, you have a signal analog and you are getting sampling, if here and here, and here I am sort of exaggerating, actually you do not sample it.

So, farther away you sample it much closer, but to just drive home the point, what I am saying is that you have got the value of the signal here, and the value of the signal here. And ideally speaking you do not know what the values of signals are over. Let us say, here or here, or here you do not know, because you have not got those values, right. So, what you, what you do is you make an assumption, right.

So, the hold, the whole why I say the hold strategy is that the this hold is not the sample and hold because the sample and hold will simply hold the signal, but when you are using the signal for if for example, suppose you want to plot this signal on a on a graph. So, are you going to plot it as if this signal is held up to this point, and then held up to this point, and then come comes down. So, are you going to plot it like this when you are going to plot or are you going to plot it like this, then let me use a different colors.

The alternate way of plotting it would be to plot it. So, between this and this it is a straight line like this, but between this and this you would say that I would interpolate I will do a linear interpolation. So, when I will plot it, I will plot this yellow line right. So, what I am saying is that, if you want to plot it as an, as an, as a continuous signal, then you may choose appropriate interpolation strategies for to construct a continuous signal, which will be an approximate version of the old signal.

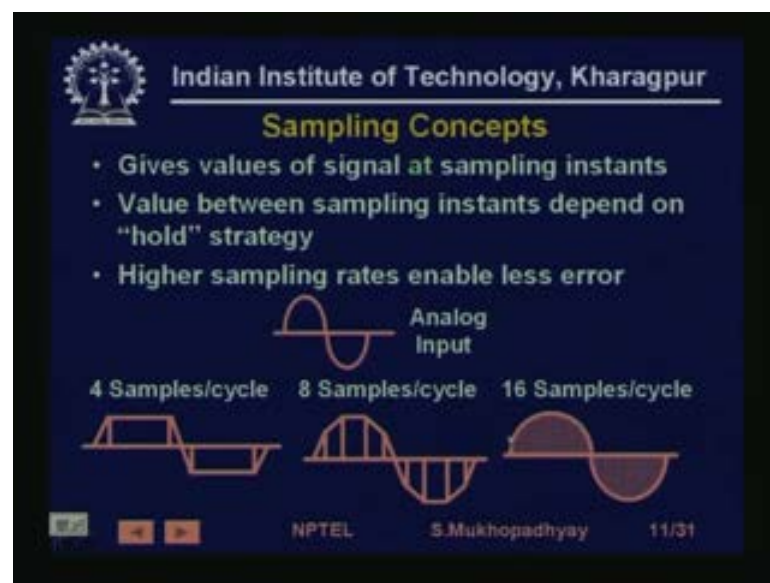
Now, the question so obviously, you can, you can understand that how accurate this digital approximation is with analog one depends on what. So, it depends on 2 things. Firstly, it depends on how close like for example, if you had, if you had taken rather than doing it this far if you had divided it in to let us say these intervals, and if you had got this value, and this value, and this value then your approximation would have been like this right.

So, you would have gone from here to here, from here to here, and from here to here, and then here to here. So, you see that you are able to do and then you would have got it here and, here and, here. So, you would have followed the analog signal much more close. So, in general closer sampling will give you better accuracy, but it is also a lot of, but it is also more work. So, more work faster is faster sampling and faster data processing.

So, you have to determine what is the appropriate level of error that you can tolerate, and what is the maximum amount of work that you can do within a given time right. So, it turns out that there are certain fundamental principles to be obeyed because if you do not do that if you, if you. So, ideally speaking you would like to sample you would like to sample at the lowest possible rate and trying to keep the error low.

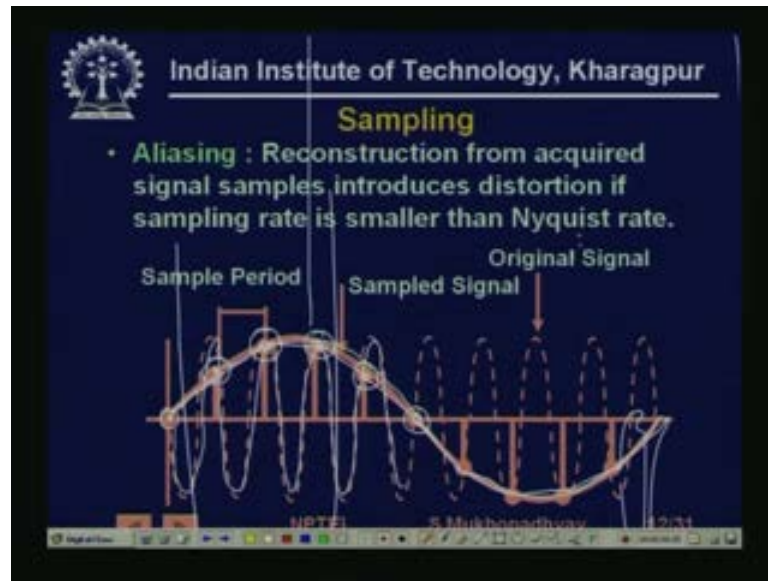
Especially, you would generally you would like to keep the high frequency error low you may have some high frequency errors, but generally you want that the low frequency errors. That is the let us say the average values over certain time intervals etcetera should be pretty accurate right. So, the generally the low frequency component of the, of the signal is actually of more use for the purposes that we are discussing and. So, we do not want low frequency errors right.

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So, that is what I am saying that if you have an analog input, and you have 4 samples per cycle, and you if you have 8 samples per cycle, and if you have 16 samples per cycle. You see gradually you are getting a better and better representation of the analog input

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Now, there is a you know benchmark rule, which everybody talks about is called the. So, called the Nyquist rate of sampling. Actually, this is this concept is explained in this diagram and imagine that you actually truly have this sine wave, which you are sampling this is the real analog wave right this is the real analog wave and being not aware of the Nyquist sampling theorem. You have sampled it at these points right

So, what happens is that now you would like to reconstruct the signal right you now in the computer you have got these values. So, when you reconstruct the signal, you will suppose you do the linear reconstruction. So, you will get this wave. So, you see that what you actually reconstructed is a much lower frequency sine wave, and this high frequency sine wave is completely lost. So, you made a major error here, it is not.

So, it turns out that theoretically speaking you cannot reconstruct a signal unless you sample it at twice the rate of the largest frequency content. That is largest frequency signal, that is present. So, suppose you have a signal, which is 5 hertz and another signal, which is 100 hertz. Then, theoretically speaking you cannot reconstruct the signal even with an infinite number of samples unless you sample it at least at 200 hertz right, but since we are not concerned with theoretical reconstruction, we have to actually reconstruct it. So, therefore, a practical rate would be 5 to 10 times of the maximum samples, maximum sampling, maximum frequency signal present.

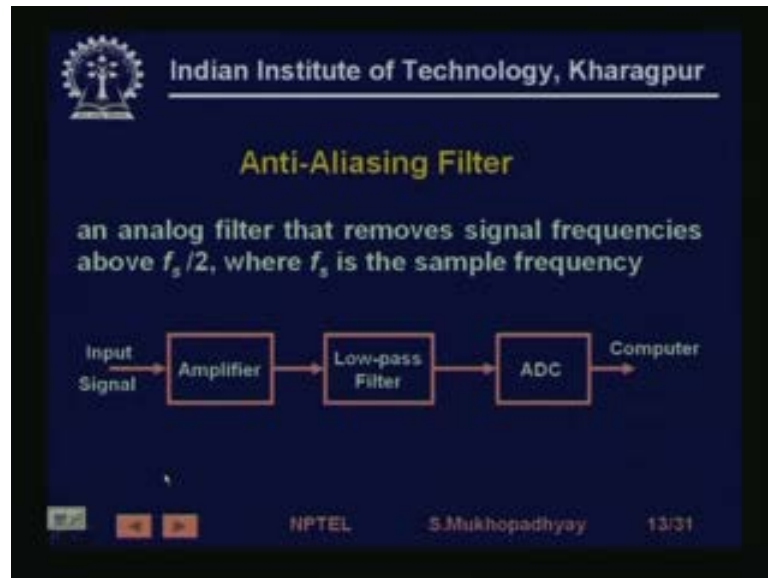
So, if you have 100 hertz present, you typically like to sample even 1000 hertz or minimum 500 hertz right. So, this says that. So, basically if you do not do that what there is something happens called aliasing means that one frequency signal will appear from the samples to be of completely a different frequency. It will appear as a different lower frequency signal.

So, you are going to get a lot of low frequency error, which is, which is bad generally. So, exactly that is, what is happening. So, your original signal was. So, your I do not know what is happening here. There is some problem with this. So, yeah may be problem. I understand what is happening, I understand what is happening, but I do not understand why how it came to be.

So, that is what is. So, that is what happens one frequency appear an another frequency and that is called aliasing. So, therefore, what do you do what you do is. So, now, this leads to a concept. So, what you do is you actually need to restrict see you have a certain sampling rate, which is fixed you cannot change that you do not know what frequency components are actually present in the analog signal.

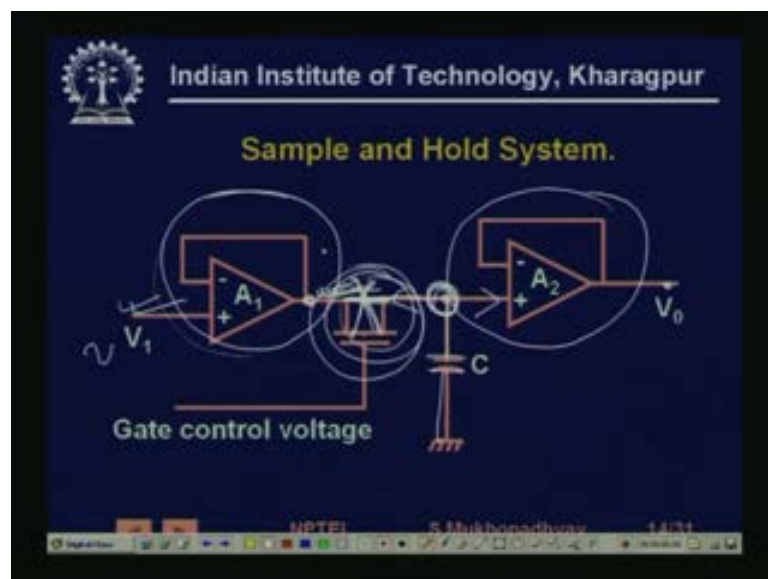
So, to make sure that you do not get low frequency errors what you do is you actually put a filter, which whose cutoff frequency ensures that you have no frequencies beyond. Let us say, one tenth of the sampling frequency. So, the sampling frequency is fixed, but the anti aliasing you actually put a filter, which is called the anti-aliasing filter, which will ensure that whatever the signal content is only those frequencies, which are less than one tenth of the sampling frequency are going to go through and appear at the input of the A D converter. The others are going to get blocked. So, that they cannot create any low frequency error, right.

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Having understood the basic idea of sampling we show. So, we say that an anti-aliasing filter is an analog filter that removes signal frequencies above $f_s/2$, where f_s is the sampling frequency. Actually, this $f_s/2$ is a theoretical rate it will not remove above $f_s/2$, but would perhaps in a practical case remove frequency above $f_s/5$ or even $f_s/10$. So, it is a low pass filter incidentally. So, this is... So, you put that filter typically between the signal conditioner and the ADC or in a multi channel case you put it either at the channels or you put it after the multiplexer.

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So, here is the sample and hold circuit very simple one. So, for example, So, you can see that you have 2 amplifiers this is this is nothing, but a buffer. So, this is nothing, but a buffer. So, whenever. So, this is the switch you know this is an electronic switch. So, whenever you turn the switch on this is, the this is, the this is the.

So, when you whenever you turn this switch on what happens is that this 2 point you can say that they are connected. So, this is the buffer unity gain buffer. So, whatever voltage you apply here, they will apply here and they will charge it up this capacitance very quickly this switch resistance is small. So, it will fast charge up and this capacitor voltage will as long as this switch is on this capacitor voltage will follow this voltage right. So, that is the sampling phase.

So, as long this switch is on this voltage is tracking this voltage changing along with that the movement you turn this switch off. So, now this switch is off and this capacitor voltage on this side, this is high impedance on this side also is high impedance. So, this charge in the capacitor cannot escape right.

So, this voltage the last voltage with the capacitor had is held and this is another buffer. So, this output will now be held at this voltage. Next time you put it on again this capacitor voltage is going to change according to V_1 . So, you first switch it on that is the, that is the sample command this capacitor voltage starts tracking this voltage. Then, you switch it off, and this voltage is held, which is transferred to the output right. So, this is a typical sample and hold circuit.

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A/D Converter: Throughput

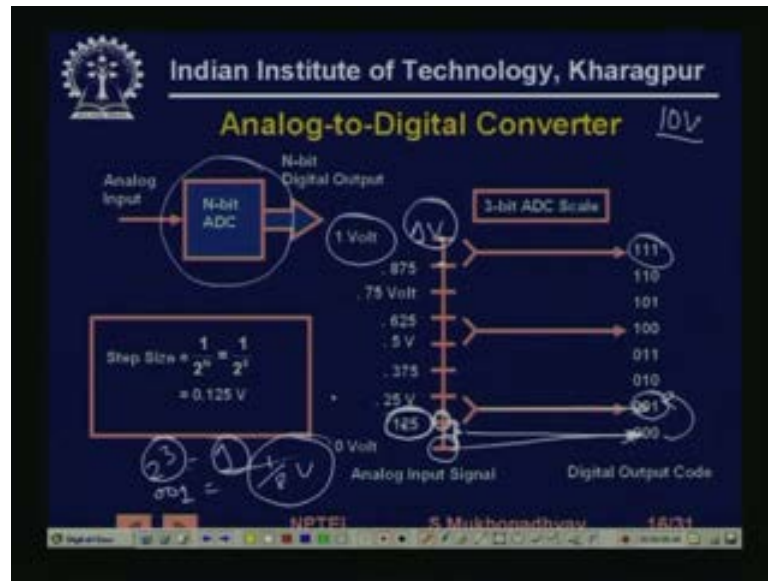
- Number of analog input channels
- Input channels differential or single ended
- Effective rate of each individual channel is inversely proportional to the number of channels sampled.

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So, we have to understand that there are number of input analog channels and the input channels can be differential or single ended as I said and I explained the meaning. Now, the this multiplexing says that the if you are the that is the maximum. Suppose, the A D converter can convert, let us say theoretically speaking hundred thousand samples per second. If, you are if you are having 8 channels then effectively each channel maximum can be sampled at hundred thousand samples divided by 8.

So, that is the through put and so the number of, so actually the often times it happens that the sampling maximum sampling rate specifications of the A D converter are given, and the number of channels are given. So, one has to be able to infer, what is the maximum sampling frequency per channel? So, that is the final sampling frequency per channel that is going to be effective on the each channel. So, that is why this 2 good question comes.

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So, next let us let, let us come to the question of quantization. So, we have now seen that an analog signal after sample and hold present itself at the analog input pin of the A D converter right. So, now, it gets converted to an N bit digital number right. So, if for example, here is a case that suppose, you have this is a 3 bit A D C right some are hypothetical, but 3 bit A D C, where it will. So, which means, so with 3 bits you can represent 8 numbers 2 to the power 3 right from 0 1 2 3 4 up to 2 to 3 power minus 1 that is 7. So, 0 to 7.

So, you see that the number 0 corresponds to 0 volts, and the first the one corresponds to 0.125 volt its. So, suppose the every A D C has what is called an analog reference voltage, which decides its dynamic range that I was talking about. So, hypothetically if the dynamic range is one volt generally it is 10 volts, 10 volts is a very typical figure. Suppose, it is one volt then the, then each. So, 2 to the power 3 is equal to 1. So, 1 2 to the power 3 digital is equal to one volt. So, one digital or the number 0 1 represents 1 by 8 volts right.

So, this signal will be come from 0 to from 0 0 0 to 0 0 1, when this signal reaches 0.125 volts or 01 by 8 volts. So, actually the any signal between 0 and 0.125 will get quantized to this 0, this is called quantization right. Similarly, 0.125 to 0.25 all these will get map to 0 0 0 1 and so on. And finally, 0.875 to 1 volts will get map to 1 1 1.

So, this is quantization and obviously, the higher the number of bits the smaller is that is the quantization interval, and the better is the resolution. But then we can always make an A D converter of 30 64 bits why not because these are related to you have to make them, we it is not just the number of bits. Finally, the they must represent that that mapping that one represents 0.125, this must be accurate. So, when you increase the number of bits you can understand that if you have let us say a 16 bit A D converter. Then, you have one by sixty-five thousand resolution right

So, the circuit should be able to resolve between 1 by 65000 will be you know something like not even mini volts right. It is, it is of the order of micro volts. So, the circuit should be. So, accurate that and the environment should be. So, less noisy that you will able to separate between those if you do not do that then you are A D converter just because of noise it will even if you present a constant signal. It is the, it is digital bits will start oscillating, and the last few bits will anywhere be useless because of noise. So, that is why it is very difficult to construct A D converters of number of bits more than, you know 18 may be and generally for this kind of things 12 to 16 are used 12 14 or 16.

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A/D Converter: Resolution and Range

Analog Signal 2-Bit Resolution 3-Bit Resolution

Range :

- Minimum and maximum voltage levels that the A/D converter can quantize
- May be selectable to improve resolution for a given number of bits

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So, we have. So, you see A D converter resolution and range. So, now final the signal this the accuracy is affected by the range the range is. So, finally, when you have a number, how do you convert that number? Suppose, you have a number 0 0 1, and it is a 3 bit A D converter. So, the number is actually 0 0 1 divided by 2 to the power 3. So, 1 0

$0.1 \text{ by } 2 \text{ to the power } 3$, which is, so it is one by 8 into the reference voltage in this case suppose it is one volt. So, its 0.25 volts

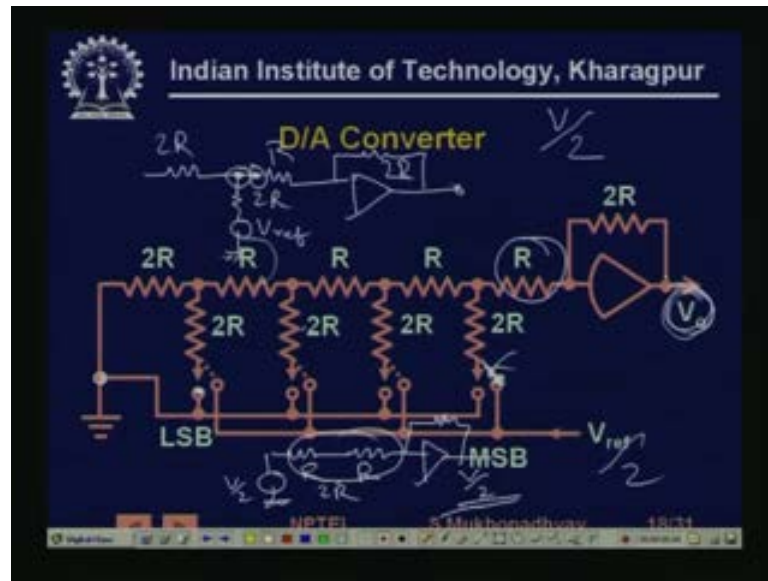
So, that is how so obviously you can understand that as the number of bits will go up the resolution will increase and the minimum. On the other hand, if you have a suppose you have a 10 volts range and you have a one volt signal, now the point is that you can actually amplify this one volt signal.

So, if you use a let us, say lets we can you write here also that suppose you have a converter, which is a dynamic range of 10 volts, and you are giving it a signal whose peak to peak value is 1 volt right. So, then peak to peak value is 1 volt means the lowest number and it is a 3 bit converter. So, then the lowest value is 1.25. So, this 0 to 1 very minus 1 to or let us say, let us say 0 to... Let us talk about unipolar signal. So, not negative.

So, suppose 0 to 1, if you have a signal then always you will get the value 0, it will the variation will not be caught at all. On the other hand, if you had amplified it this 1 volt to 10 volt, and then taken care of this amplification factor in your software. Then, this signal this 1 volt signal variation would have been captured up to 3 bit resolution.

So, this shows that always it is you must have the amplification enough amplification that is why people sometimes out programmable gain amplifiers, whose gains can be changed again under software control, but although they are expensive. So, you need to always amplify a signal. So, that the A D converter dynamic range that is the range between the reference voltages up to the reference voltage is acutely effectively utilized by the input. Then, you get the best resolution for a given number of bits, this is important to remember during data acquisition.

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So, now we are going to take a look at some A D converters, but then before we have a, we take a look at a D A converter because an A D converter circuit that is the most one of the common things like the successive approximation converter uses a D A converter. So, we just take a quick look at what is the digital to analog converter that is if we put a set of digital bits how can we get an analog voltage, which will be proportional to that number.

So, this is called an R to R ladder network for obvious reasons you can see that 2 R. So, there are some R and there are some 2 R resistances they are having double ratio. That is why this is called an R to R ladder network ladder because of this form structure. So, you see that you can it is rather easy to show using simple network theory that you see these are the switches, which are actually controlled by this bits.

So, now suppose, let us take the simplest case that if I switch only this M S B. So, it will get connected here and all the others are connected to this ground. You see this connected to ground. So, then what is the voltage that will appear here? That is very simple because you have 2 R and 2 R in parallel, now this is connected to ground. So, therefore, this resistance is R, now this that R and this R is in series.

So, it is 2 R again 2 R and 2 R in parallel again R, again R, and again R in series again 2 R and so on. So, finally, what you get finally, what you get is finally, you get this network. So, you have a 2 R. You have an 2 R, and then you have a voltage source. So,

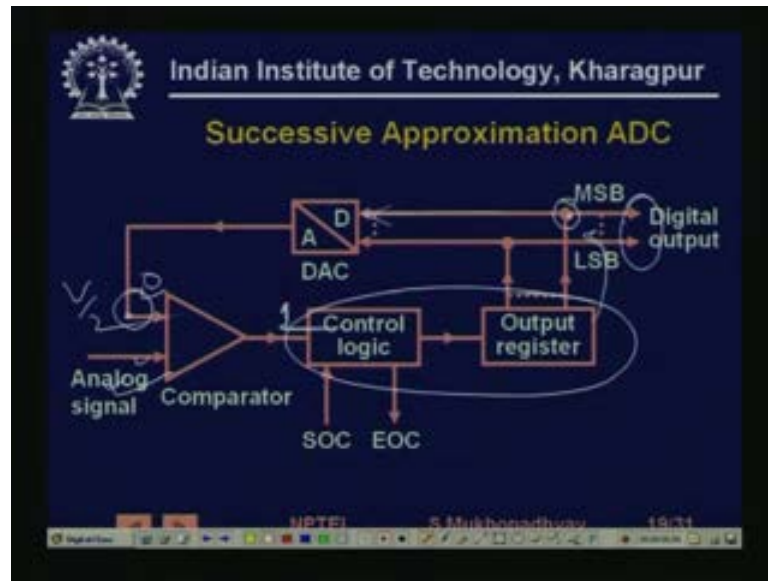
this is your V , this is your V_{ref} and then you have an R , and then you have an amplifier, and here you have a $2R$,

So, $2R$, if you do a Thevenin's circuit for this, what will be, what will be this voltage, this voltage is? So, the open circuit voltage is going to be $V/2$. So, it is going to be $V/2$ here and what is going to be the Thevenin impedance, it is going to be $2R$ parallel $2R$, which is R . So, this circuit can be further reduced in to a network of this form, it can be further reduced as $V/2$ in series with a resistance R , which is a Thevenin network. Then, another R , which is this one, and then the amplifier, which is $2R$.

So, this R and this R will make $2R$, and this $2R$ and $2R$ will give you a gain of 1. So, you get $V/2$ right and so if you have a total range of the A/D converters then the M/S/B should have a total range of N bit converter. Then, the M/S/B should have a weightage of $V/2$, and you can see that if you only switch on this M/S/B you get a $V_{ref}/2$ signals here.

In this way you can show that if you, if you keep all the others grounded and you put on the next bit on you would have got $V/4$, similarly $V/8$ and so on. So, now if you depending on your digital numbers of if you, if you, if you switch on some of them and do not switch on some of them. Then, you get a corresponding because this is a linear circuit. So, you are going to get superposition principle. So, the final signal is going to be V_{ref} properly weighted by the binary number weighting. So, you will get an analog voltage, which is going to be proportional to the digital number that is why it is called a digital to analog converter.

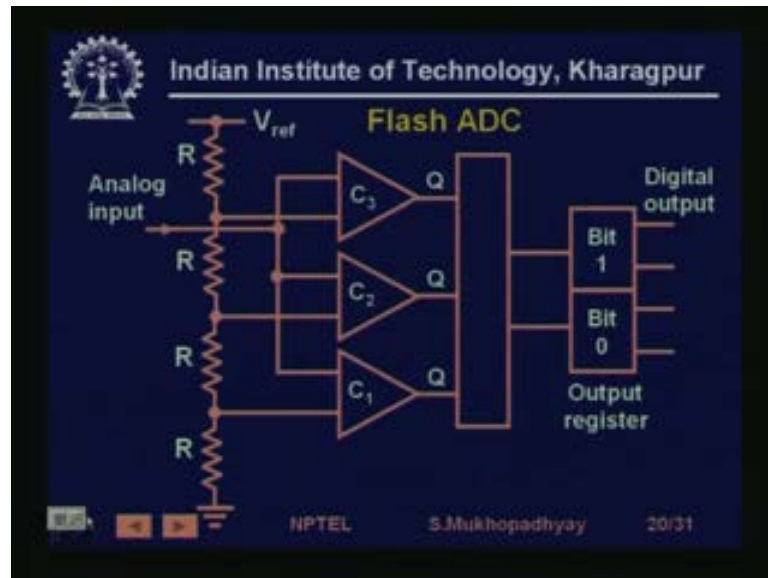
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So, in our so for example, in this A D converter this is which is called, which is called successive approximation converter principle is very simple you put a signal. The signal first compares with a suppose this is initially this is 0, initially this is 0 right. So, this just compares whether this should be this is higher or this higher. So, if this is higher it puts it one, now when it puts its one it sets this is something logic. So, this will put the maximum bit high that will no. So, maximum bit means V by 2. So, that will comes through D A converter and we will apply a V by 2 here.

Now, the question is whether this analog signal is greater than V by 2 or less than V by 2 if it is greater than V by 2, then it still one then the next bit will be put. So, in this way first M S B is put then the next bit put till this signal crosses this signal, and then at that level it is stopped. So, that is how you just compare it with a, with a number of digital, number of digital number till it exceeds and at that point you stop. So, you get. So, when you stop you get a set of digital bits, which is closest to this analog signal right.

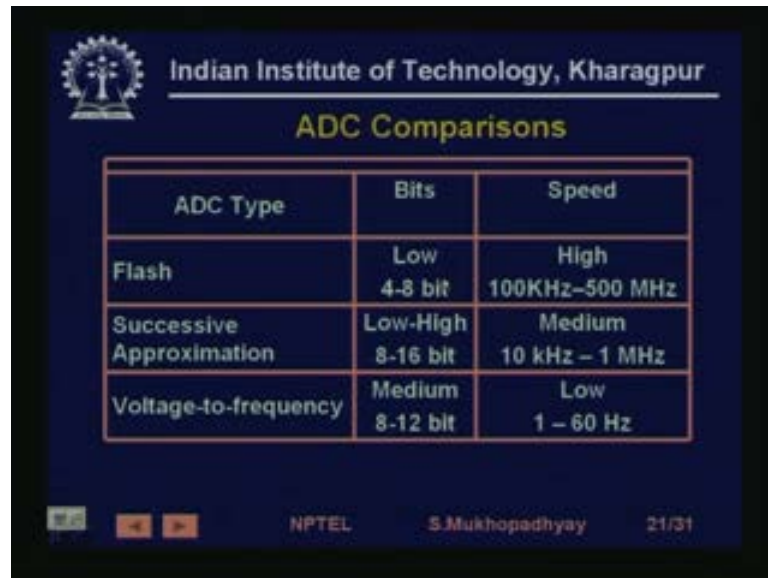
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So, this is how you convert this is, this is how a typical successive approximation converter is and this is one way. There are many various other converters. For example a very fast converter is called flash A D C, we are not going to discuss that because we do not have enough time, but this is this is converter where you know there you do it bit by bit. So, you need at least worst case you can need N times you can go through the cycle of setting and the setting bits. So, you need more time.

On the other hand in this converter, the whole conversion is just in one block cycle. So, it is much faster, but then it requires a huge resistance network, and they have to very precise otherwise, you are going to get error. So, this is a converter which is therefore, this converter has some problems and although it is very fast. So, we are not, we are not going to look at that too much.

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ADC Comparisons

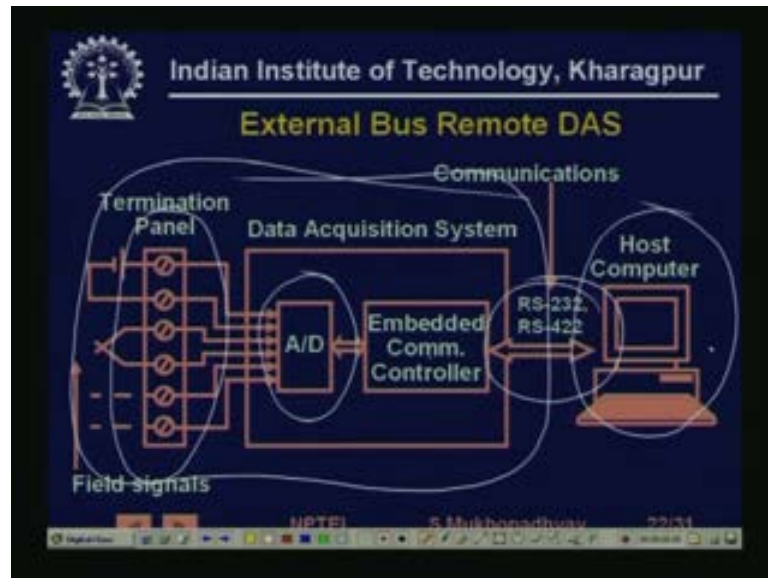
ADC Type	Bits	Speed
Flash	Low 4-8 bit	High 100KHz-500 MHz
Successive Approximation	Low-High 8-16 bit	Medium 10 kHz – 1 MHz
Voltage-to-frequency	Medium 8-12 bit	Low 1 – 60 Hz

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If, you compare there are various kinds of converters and we are for example, there are there are other converters too we are talking about flash. We should saw successive approximation. There are some converter, which are called voltage to frequency or integrating converters, which are very slow, but very accurate. So, you can have typical idea about the number of bits flash is 4 to 8 bits typically less number of bits, but very fast conversion. So, you can go up to you know 500 megahertz conversions.

Then, successive approximation is widely used 8 to 16 bits possible generally and medium sampling rates for most processes successive approximation is good enough, and then you have integrating converters, which are quite slow, but very accurate because they do an input averaging over long time. And therefore, they take have good response to noise.

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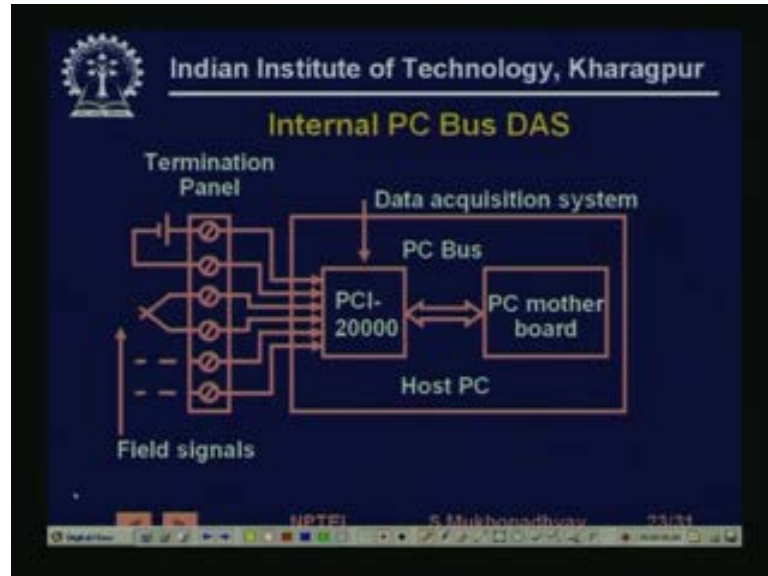
So, having seen that. So, now we have seen A D conversion. Now, we come to a system level discussion the typically they are 2 kinds of data acquisition system that you find. One are called external bus or remote. So, there what happens is that the computer is in a separate place generally, and I do not know what is happening? So, you see here the these are all the signals analog signals are getting terminated. Let us assume that it is signal condition, you are not if it is not done then an easy condition then the A D conversion, we have seen, what is A D conversion.

Then, now what happens is that the, is that this thing this physical the system boxes are separate are situated at a separate place. They have, they have their own power supplies, they are situated possibly much closer to the field and then here. So, after A D conversion. So, and they have their own processors. So, the value is firstly coming to that embedded communication controller processor, and then using some very standard communication protocol as we have you now R S 2 3 2 4 2 2 4 8 5 or typically 4 8 8. There are number of protocols by which to computer communication, it is coming to what is known as the host computer, where this data is going to be used.

So, this is a small box, which is going to be close to the process, and then one wire is going to come generally serial communication coming to the computer. So, this is an external bus remote data acquisition system. The advantages are that they can, they can

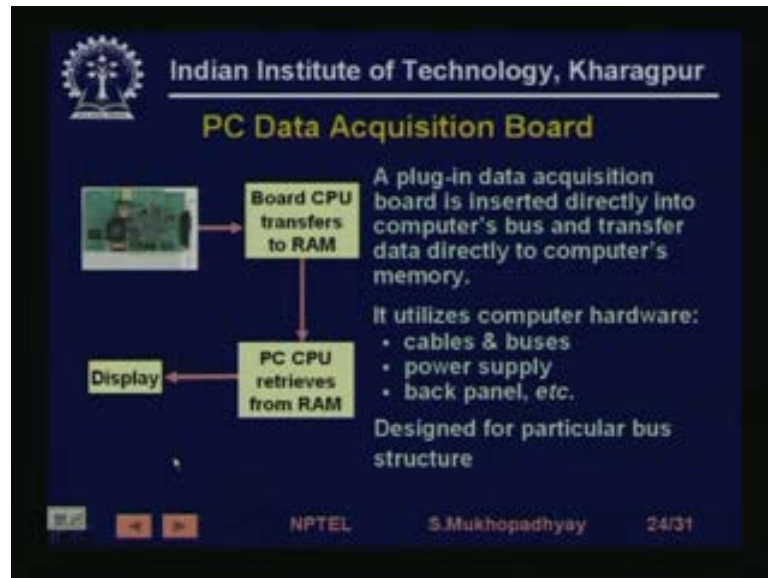
connect to any host computer it is. So, close to the field right, disadvantages is that generally data acquisition rates are limited because of this serial communication...

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So, you can you get slightly slower rates of communication, but the it, but it is that is good enough for you it is second kinds of things are internal P C bus data acquisition systems, where the data acquisition system sits right inside the P C box right. So, and then they will communicate with the P C through the, through the P C I bars, which is much higher speed and which is parallel in the field right. So, the data acquisition rates can go much faster, but it, but this means that the P C has to go close to the field otherwise you are going to have long cablings to the P C right. So, similarly these are generally designed for specific machines like the, like the, like window P C.

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So, this is, this is a typical picture of a P C data acquisition board. So, what happens is that the board itself has a C P U, which transfers the data because it sits on the P C bus. So, it will transfer the data to the P C memory, and then the C P U can actually take it from the memory, and then do a do a display right. So, they are, they are, they are generalized designed for the this you know kind of windows kind of P C.

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Typical Specification

Power consumption	+5 V supply 800 mA typ. /1 A max +12 V supply 2 mA typ. /5 A max	
Number of Channels	(DIP Switch)	8 single-ended/differential
Resolution		12-bit
Accuracy		0.01% of reading +/-bits
A/D Type		Successive approximation
Full scale range gain	+/- 5,10 VDC	0 to 10 VDC
Coding	Offset/True binary	Bipolar/Unipolar
Gain Tempco		+/- 50 ppm / C max
Zero Drift		+/- 10 ppm / C max
Acquisition time	4 μs to 0.1 % Typ. full scale step input	

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And they are very common and they are rather cheap and can especially can be used in a very much in a laboratory scenario and widely used. So, these are the basic, these are the

basic 2 types of data acquisition systems, if you take a, if you take a typical specification of a data acquisition system they will, they will mention things like you know power consumption see 5 volt supply has lot of current requirement because of the digital circuits. Then, a 12 volt supply for analog the number of channels. This is, this is very important typically you have about 8 analog channels either single ended, you can have 8 single ended 8 differential or a 16 single ended channels.

Typically number of digital channels are much more 64 like that you have to take only one signal each. Resolution says, what how many bit converter is being used. So, it is a 12 bit converter accuracy. The A D converter type then you know this full scale, then this is the, this is the, this is the dynamic range. So, it is saying that the, that the dynamic range is 0 to 10 volt D C.

Then, the A D converter codes can be available in various formats whether it is a 2 s compliment true binary offset binary various kinds of digital codes are there. The this these 2 specification gain and 0 drift specifications are for the signal conditioning block, which is there in the this thing. So, that if they will have an amplifier, and then amplifier gain variation and 0 drift can be specified and finally, the acquisition time, which says that the acquisition time is of the order of 4 micro seconds. You know, so these are, so these are typical specifications of a data acquisition system, which you need to look at when you select one.

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Data Acquisition Software

- Does not require programming.
- Enables developers to design the custom instrument best suited to their application.
- Examples: TestPoint, SnapMaster, LabView, DADISP, DASLAB, etc.

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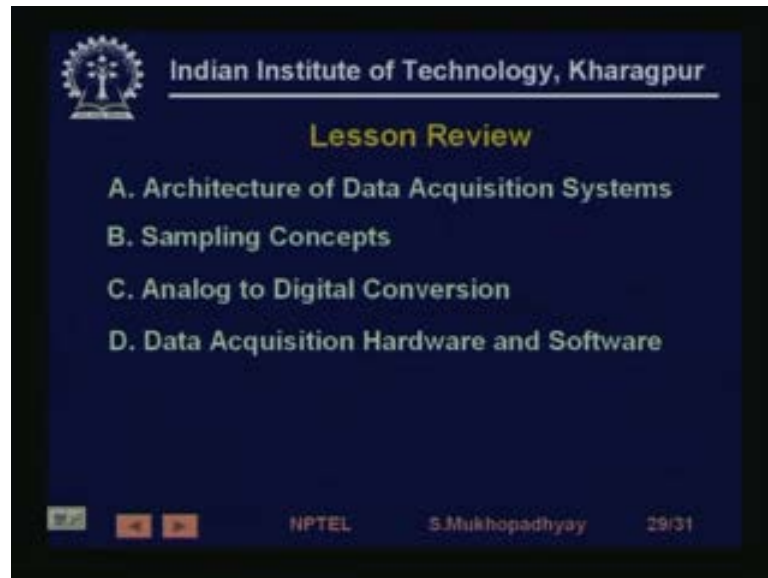
So, similarly, finally is a data acquisition software. I do not want to talk much about it only thing is that sometimes you can either go for. You know, this data acquisition software will only come up with some certain drivers. And then you can write your own C or basic programs to use the data acquisition systems, but that requires lot of programming although it can give you a lot of I mean flexibility in the sense that you can do anything you want with the data. Then, the raw data is made available, but this requires lot of programming, and often in a, in a industrial situation it is not preferred. So, people use data acquisition software packages.

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So, in programmable software you the advantage is flexibility, and the disadvantage is complexity and a very steep learning curve, while if you have data acquisition software. Then, it does not require programming generally it requires you know graphical programming in the sense that in a using very common sense, and domain related quantities. You can actually figure configure the system and get data and then display it. So, enormous developers to design custom instrument best situated to their application. There are various examples for example, a common one in which I am familiar is lab view, but there are few others also.

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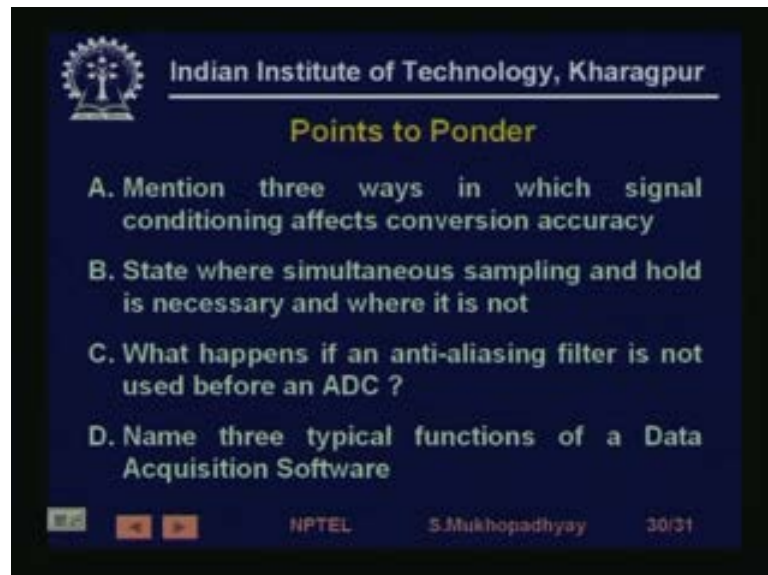
Lesson Review

- A. Architecture of Data Acquisition Systems
- B. Sampling Concepts
- C. Analog to Digital Conversion
- D. Data Acquisition Hardware and Software

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So, we have. So, what we have done during this course this lesson we have seen the architecture of data acquisition systems. We have seen the sampling concepts; we have seen the sub details of analog to digital conversion. Finally, we have taken a look at some data acquisition hardware and software features.

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Points to Ponder

- A. Mention three ways in which signal conditioning affects conversion accuracy
- B. State where simultaneous sampling and hold is necessary and where it is not
- C. What happens if an anti-aliasing filter is not used before an ADC ?
- D. Name three typical functions of a Data Acquisition Software

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So, points to ponder is mention 3 ways in which signal conditioning effects conversion accuracy. So, I have all ready talked about dynamic range you can think of some others state where simultaneously sampling and hold is necessary, and where it is not. So, as I

said this is related to the to the frequency content of the signals, and the speed of conversion of you're a D converter and the number of channels. Of course, what happens if an anti aliasing filter is not used before an A D C.

So, you have to get through, if the A D C is successive approximation or if the A D C is flash. So, choose the A D C and then go through it, and see that suppose in the middle. Suppose, the signal just goes to 0, then what will happen sometimes it may give you wrong results sometimes it may not name 3 typical functions of a data acquisition software. So, one of them could be displaying you can figure out the other 2 by looking at some of the, some of the software, which are advertised on the internet. So, that is all for today.

Thank you very much.

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Good afternoon and welcome to lesson eleven of the course on industrial automation and control. So far we have learnt about sensors, but before learning about actuators, I thought that it will be it will be more useful to learn about automatic controls mainly, because of the fact that several actuators are actually closed loop control systems themselves.

Thank you.