**INDIAN INSTITUTE OF TECHNOLOGY**
**KHARAGPUR**

**NPTEL**
**ONLINE CERTIFICATION COURSE**

**On Industrial Automation and**
**Control**

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**Topic Lecture – 08**
**Data Acquisition Systems**
**(Contd.)**

Welcome to lesson 10 so that simultaneously, simultaneous sample-and-hold.

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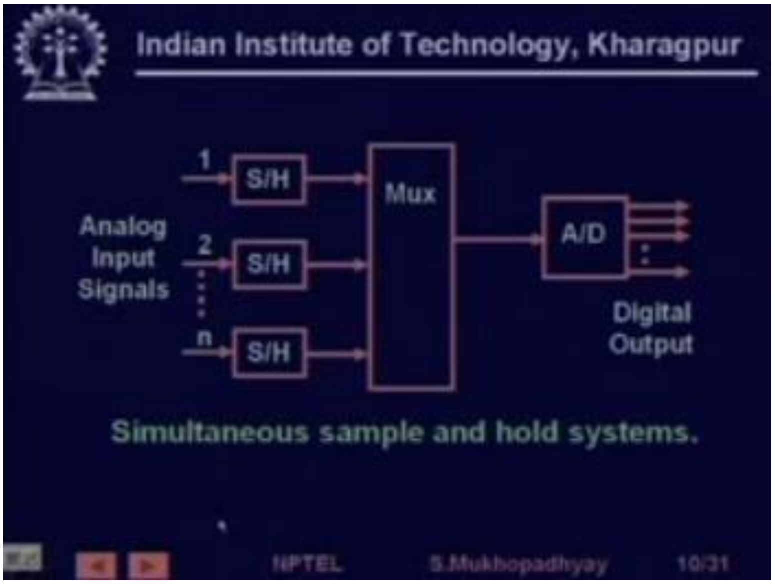

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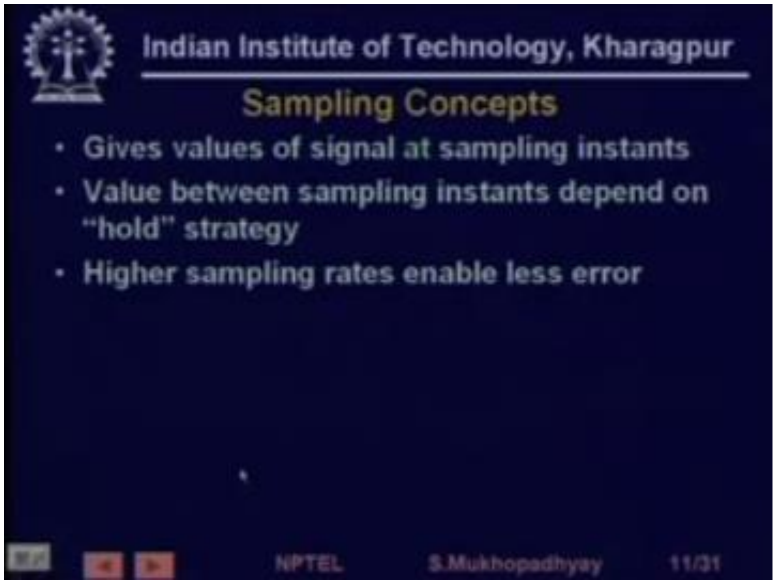


Now before we look at the A/D converter let us take a look at some sampling concepts. You know first thing that we must remember that the sampling value which you get from the A/D converter is at the sampling instant. So it is just at the sampling instant and ideally speaking so you have you have a signal.

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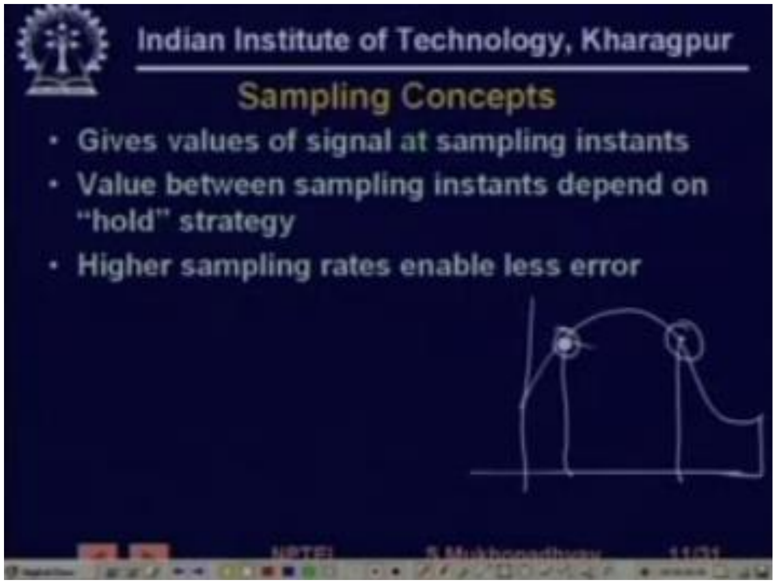


So you have a signal analog and you are getting this sampling it here and here and here I am sort of like the exaggerating actually you do not sample it so further away you something much closer but we just to drive home the point what I am saying is that you have got the value of the signal here and the value of the signal here and ideally speaking you do not know what the values of the signals are.

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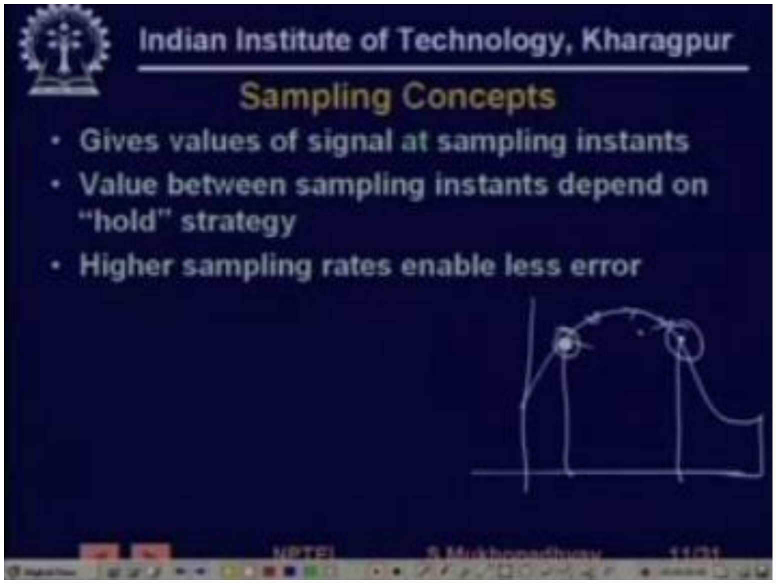


Over, over let us say here or here or here, you do not know that because you have not got those values, right. So what you, what you do is you, you make an assumption right, so, so the whole.

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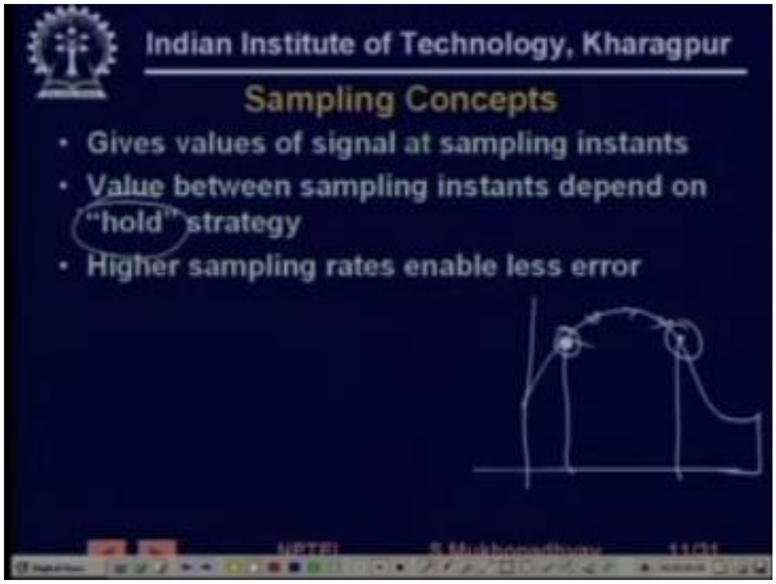


Why I say the whole strategy is that this, this whole dis not the sample-and-hold, because the sample-and-hold will simply hold the signal but when you are using the signal, so for example suppose you want to plot this signals on, on a graph so are you going to plot it.

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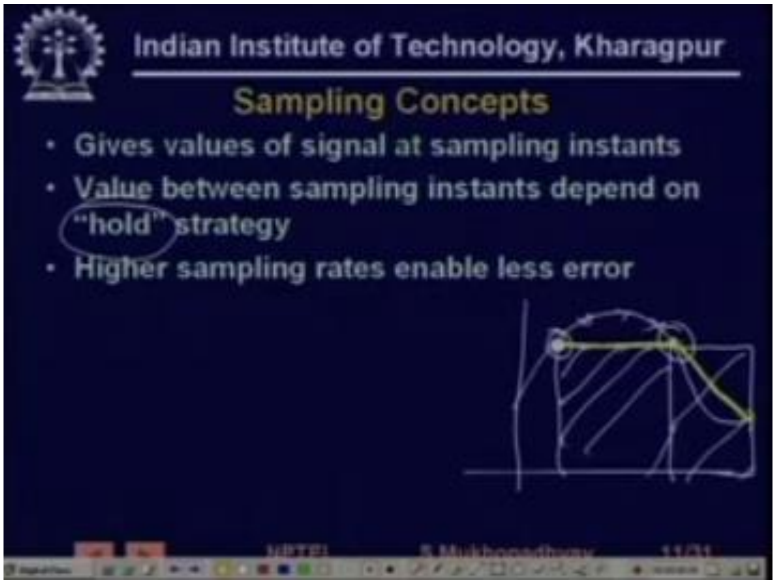


As if this signal is held up to this point and then held upto this point and then comes down so are you going to plot it like this when you are going to plot it or are you going to plot it like this then let me use a different color the alternate way of plotting it would be to plot it from between this and this it is a straight line like this, but between this and this you would say that I will 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 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 is so obviously, you can you can understand that how accurate this.

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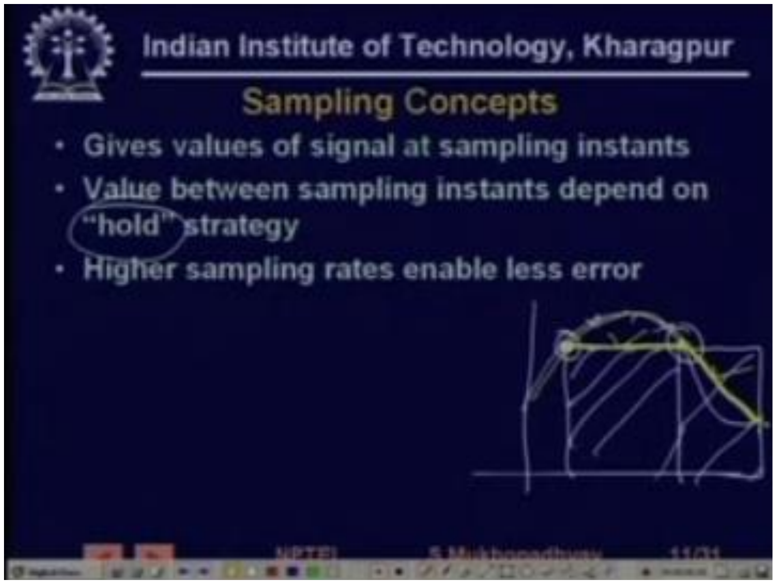


Digital approximation is with the analog one depends on what, so it depends on two things, firstly it depends on how close.

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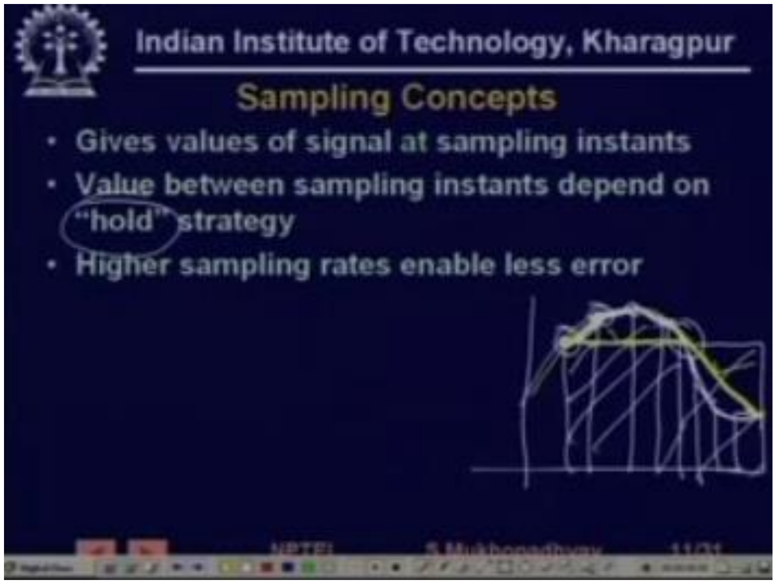


Like for example, if you had, if you had taken rather than doing it this far if you had divided it into let us say these intervals and if you have got this value and this value and this value then your approximation would have been like this, right so you would have gone from here to here from here to here and from here to here and then here to so you will 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, but it is also a lot of but is also more work so more work 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.

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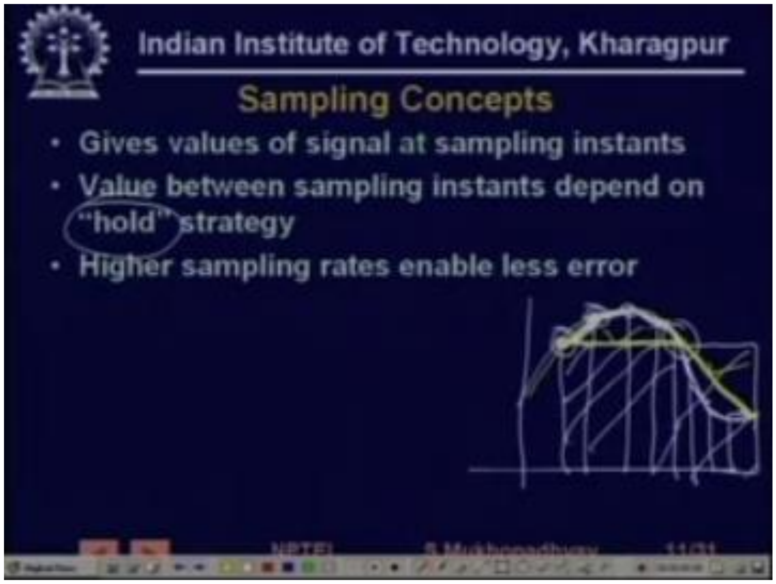


So it turns out that there are certain fundamental principles to be obeyed because if you do not do that if you, you so ideally speaking you would like to sample, you would like to sample at the lowest possible rate and trying to keep the error low and especially you generally you would like to keep the high frequency errors low you may have some high frequency errors but generally you want at the low frequency errors that is let us say the average values over certain time intervals extra, should be pretty accurate right. So the generally the low frequency component of the signal is actually of more use for.

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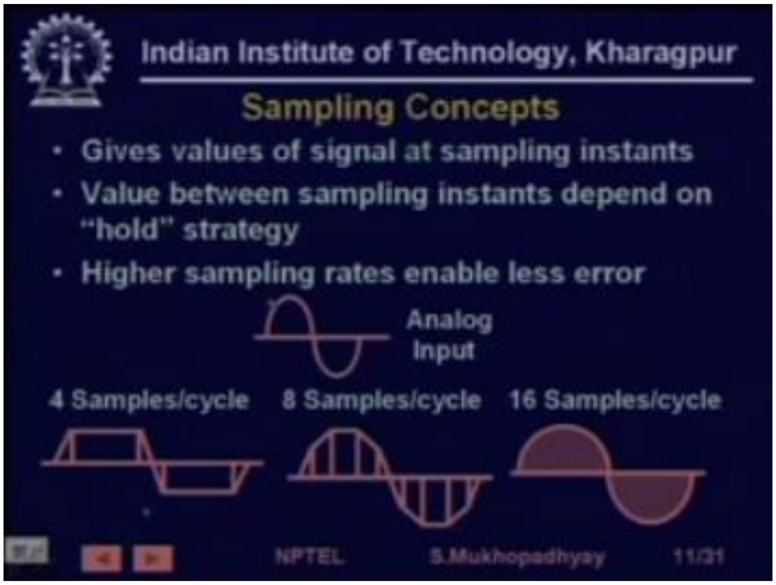


The purposes that we are discussing and so we do not want low frequency errors right. So that is what I am saying that if you have an analog input and if you have four samples per cycle and if you have eight samples per cycle and if you have 16 samples per cycle so you see gradually you are getting a better and better representation of the analog input.

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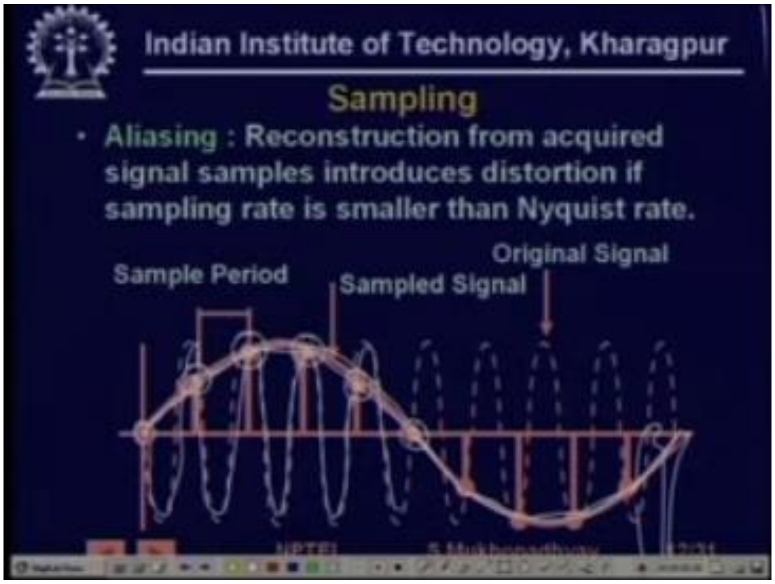


Now there is a theorem bench mark rule which everybody talks about is called the so called the Nyquist rate of sampling actually this is this concept is explained in this diagram that imagine that you actually truly have this side wave which you are sampling, this is the real analog wave right, this 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 the now you would like to reconstruct the signal right, now in the computer you have got these values so when you reconstruct the signal you will suppose you do a 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 five 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 reconstructed, 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 it at 1kilohertz or minimum 500 Hertz right.

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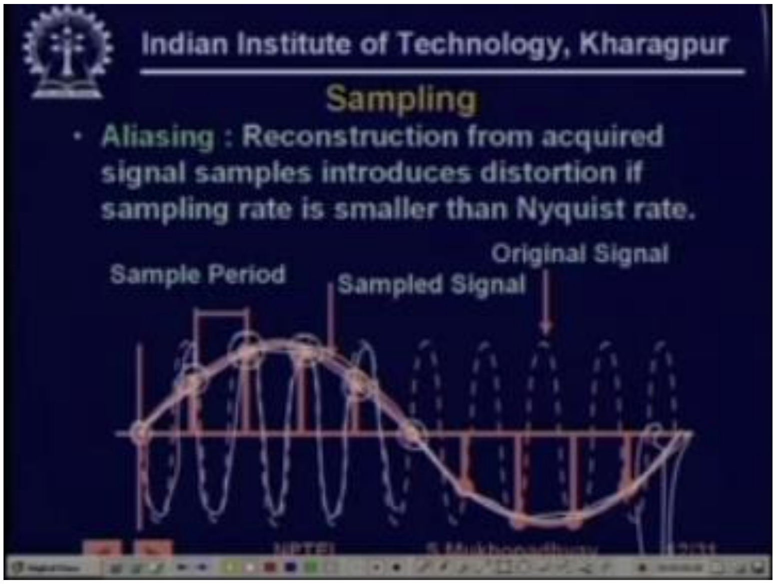


So this say is that, so basically if you do not do that what there is then something happens called aliasing, aliasing means that one frequency signal will appear from the samples to be of completely a different frequency, it will appear as a different lower frequency signal, so you are going to get a lot of low frequency error which s which is bad generally.

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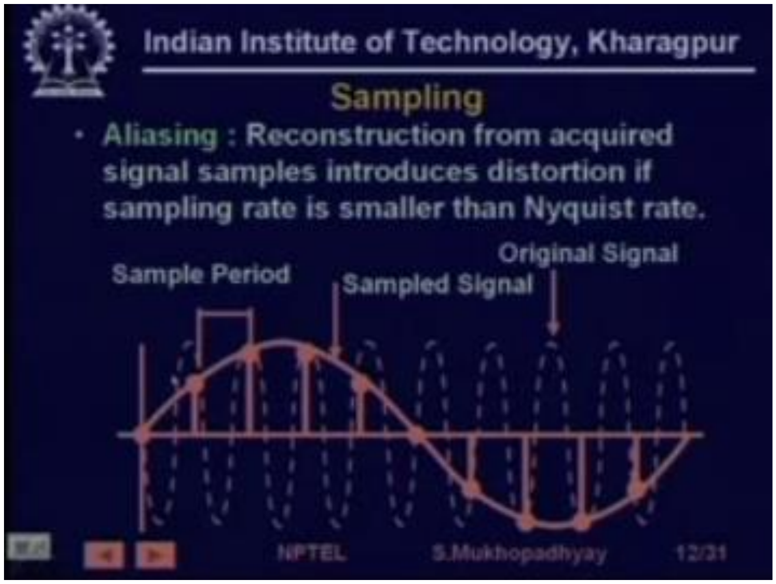


So exactly therefore it is happening, so your original signal was oops, so you are I do not know what is happening here there is some problem with this. Let me, so yeah, maybe now it will work now. Oh, oh I understand what is happening, I understand what is happening yeah, yeah but I do not understand why how it came to be, okay. So, so that is what is, so that is what happens one frequency appears on another frequency and that is called aliasing. So therefore what you do, what you do is so now this leads to.

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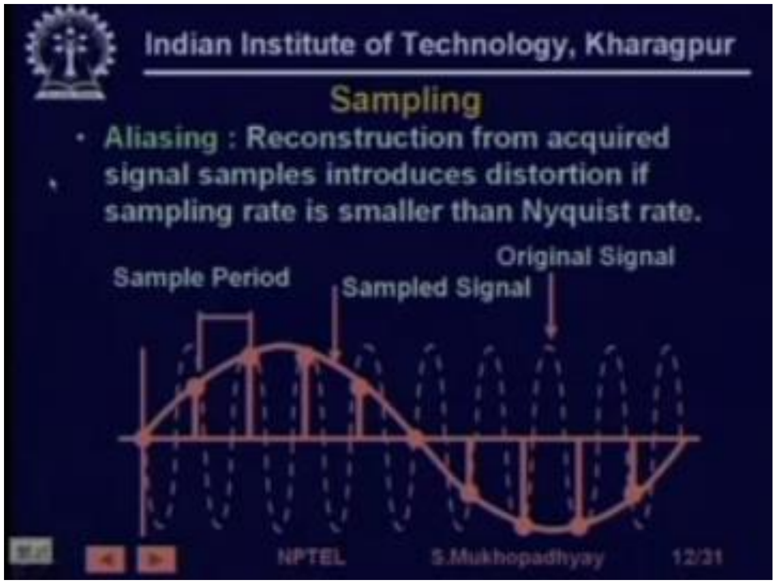


A concept so what you do is you actually need to restrict so you have a certain sampling rate which is fixed, you cannot change that you do not know what frequency components are actually present in the analog signal, so to make sure that you do not get low frequency errors what you do is you actually put a filter which whose cutoff frequency ensures that you have no frequencies beyond let us say one-tenth of the sampling frequency.

So sampling frequency is fixed but the anti aliasing 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, okay.

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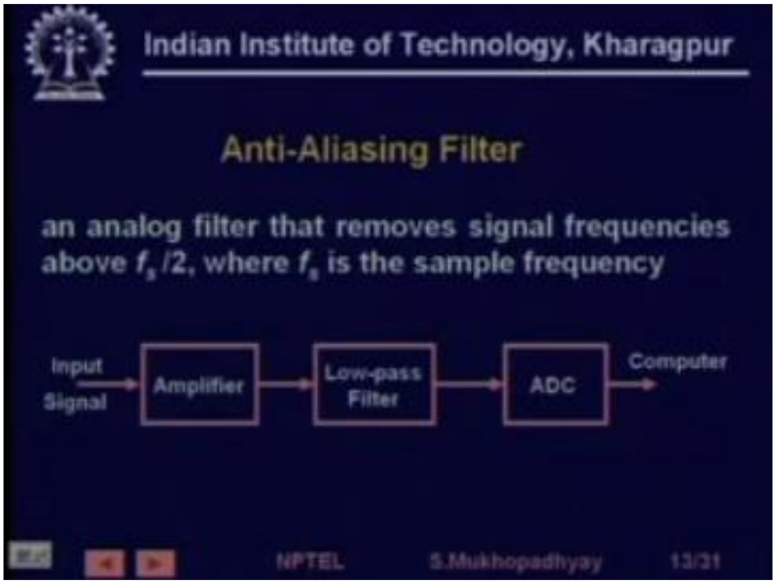


Having understood the basic idea of sampling we show the, so we say that an anti-aliasing filter is an analog filter that removes signal frequencies above $f_s/2$ where $f_s$ is the sample frequency, actually the $f_s/2$ is actually a theoretical rate it will not remove above $f_s/2$ but would perhaps in a practical case remove frequency of $f_s/5$ or even $f_s/10$. So it is a low-pass filter incidentally, so this is.

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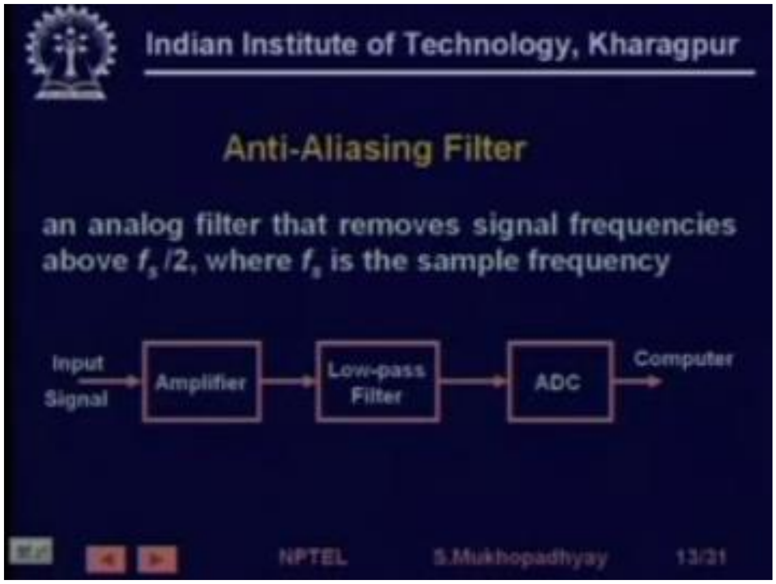


So you put that filter typically between the signal conditioner and the ADC or in a multi-channel case you put it either at the channels or you put it after the multiplexer.

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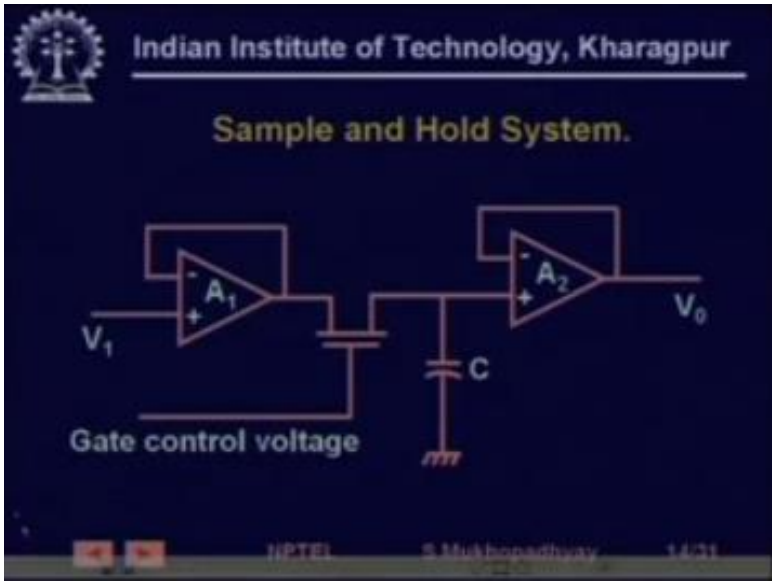


So here, is the sample and hold circuit very simple one.

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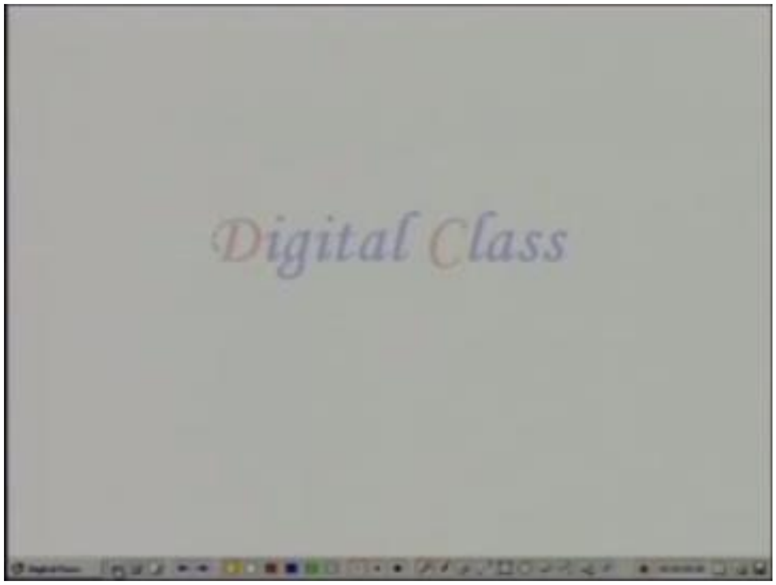

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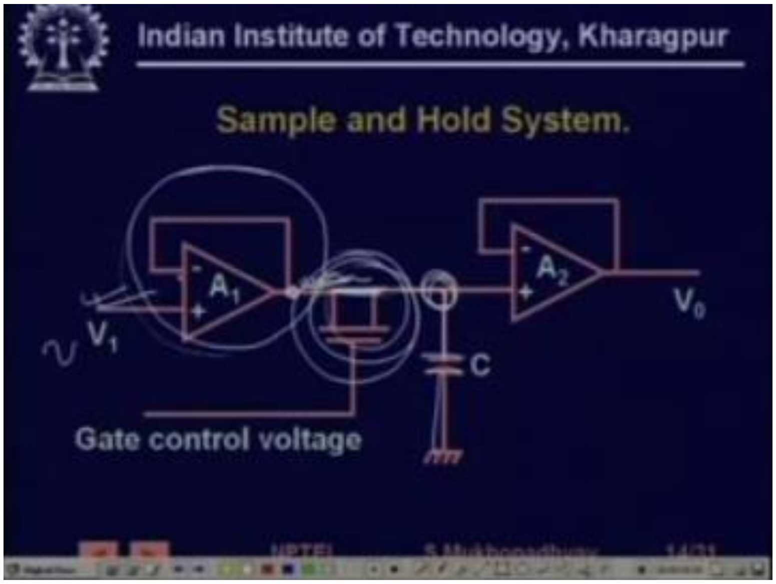


So for example, so you can see that you have two amplifiers this is nothing but a buffer okay, so this is nothing but a buffer so whenever, so this is a switch you know this is an electronic switch so whenever you turn the switch on this is a mass, so wherever you turn the switch on what happens is that these two points you can say that is they are connected, okay. So this is a 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 as long as the switch is on this voltage is tracking this voltage, changing along with that, the moment you turn this switch off.

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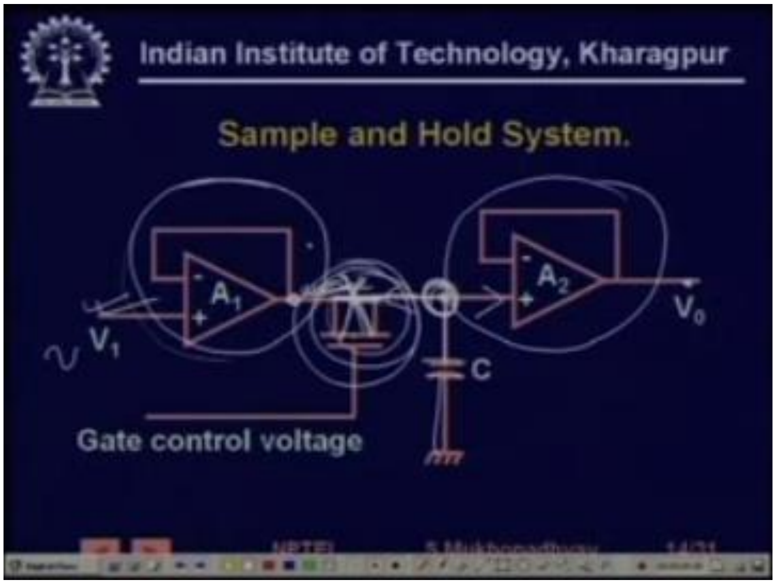


So now this switch is off and this capacitor voltage on this side sees high impedance on this side also sees high impedance so the charge in the capacitor cannot escape, right. So this voltage the last voltage with the capacitor had is held and this is another buffer, so this output will now be held 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 a, that is the sample command this capacitor voltage starts tracking this voltage there is 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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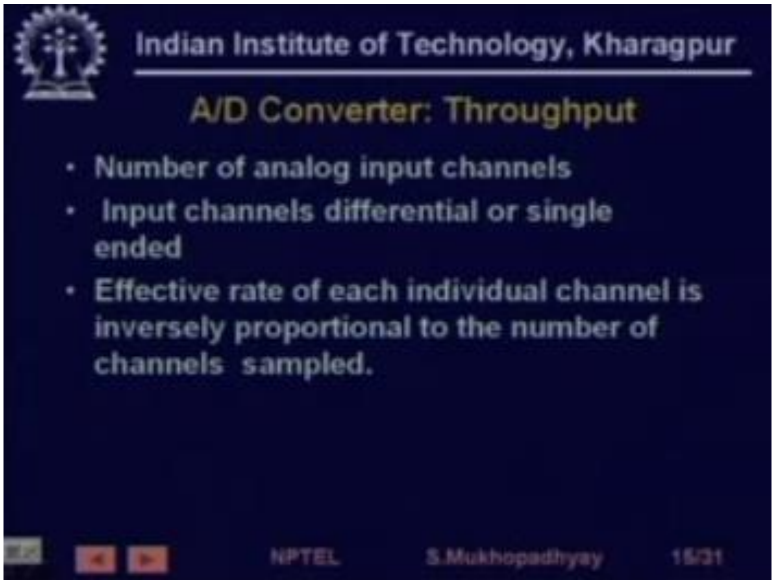


So we have to understand that when there are a number of input analog channels and the input channels can be differential or single-ended as I said and I explained the meaning now that this multiplexing says that the if you are the, that is the maximum suppose the A/D converter can convert let us say theoretically speaking 100,000samples per second. If you are having eight channels then effectively each channel maximum can be sampled at 100,000 samples divided by 8, so that is the throughput and so the number of.

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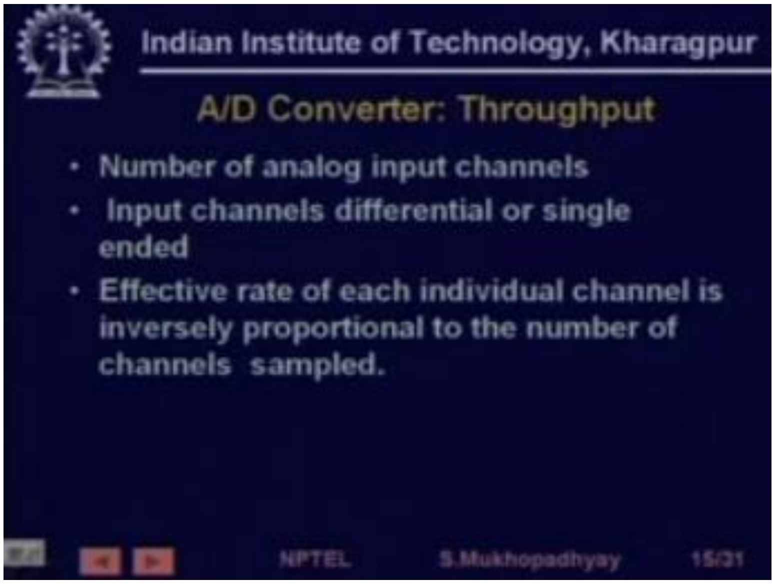


So actually the often times it happens that the sampling maximum sampling rate specifications of the A/D converter are given and the number of channels are given so one has to be able to infer what is the maximum sampling frequency per channel, so that is the final sampling frequency per channel that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

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So next 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.

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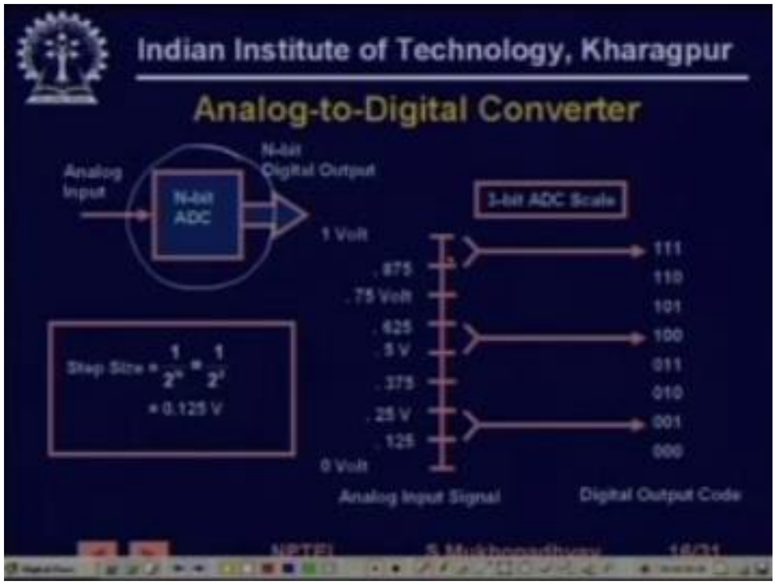


So now it gets converted to an N-bit digital number right, so for example here is the case that suppose you have this is a 3-bit ADC right, some are hypothetical but 3-bit ADC where so which means so with 3-bits you can represent eight numbers to the power 3 right from 0 1 2 3 4 up to $2^3$-1 that is 7 so 0 to 7, so you see that the number 0.

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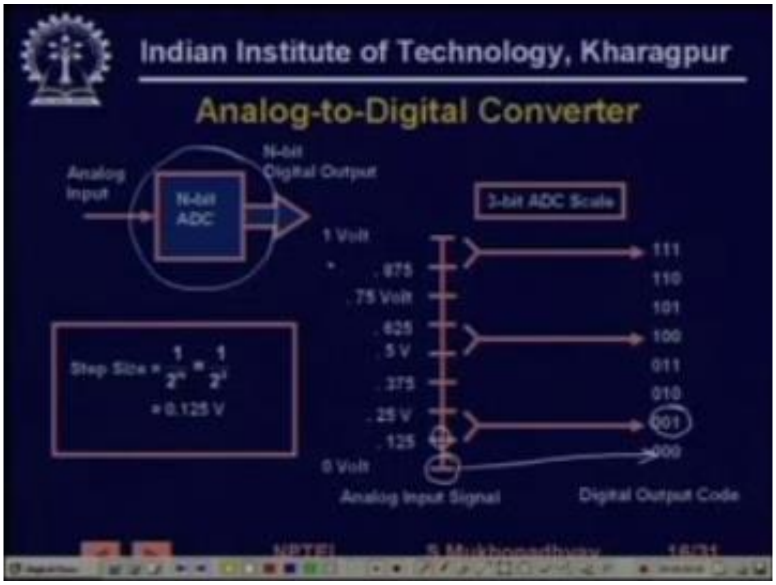


Corresponds to 0 volts and the first the one corresponds to 0.125 volt it so suppose the every ADC has what is called an analog reference voltage which decides it's dynamic range when I was talking about, so hypothetically.

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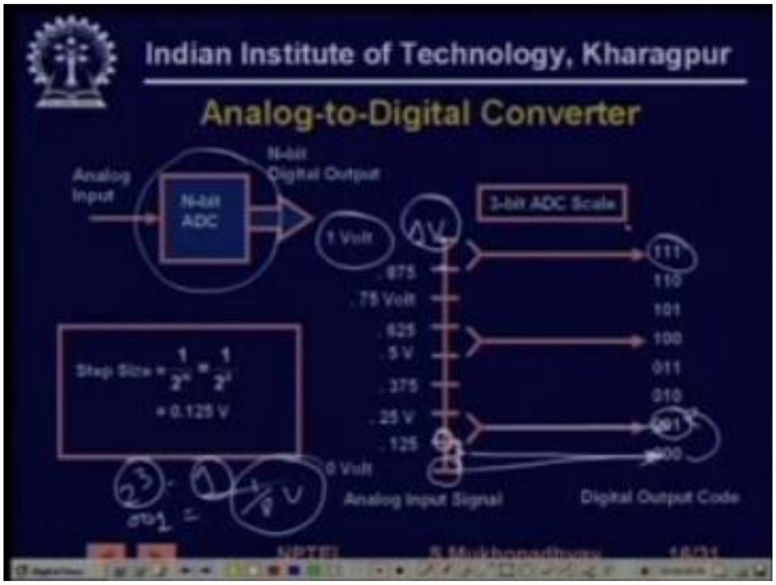


If the dynamic range is 1volt generally it is 10 volts, 10 volts which is a typical figure, suppose it is 1volt then the, then each so $2^3$=1 so 1 $2^3$ digital is equal to 1 volt so 1 digital or the number 001 represents 1/8 volts right. So this signal will become from 0 to from 000 to 001 when the signal reaches 0.125 volts or 1/ 8 volts, so actually the any signal between 0 and 0.125 we will get quantized to this 0, this is called quantization, right.

Similarly 0.125 to 0.25 all these will get mapped to 001 and so on, and finally point 0.875 to one volts we will get map to is 111, so this is quantization and obviously the higher the number of bits the smaller is the quantization interval and the better is a resolution. But then we can always make an.

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A/D converter of 32-bit, 64-bit why not, because these are related to, because you have to make the, 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 1/65,000 desire resolution, right. So the circuit should be able to resolve between 1/65,000will 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 be able to separate between those, if you do not do that then you A/D converter just because of noise it will this even if you present a constant signal it is the digital bits will start oscillating and the last few bits will anyway the 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 maybe.  And generally for this kind of things 12 to16 are used 12, 14 or 16.

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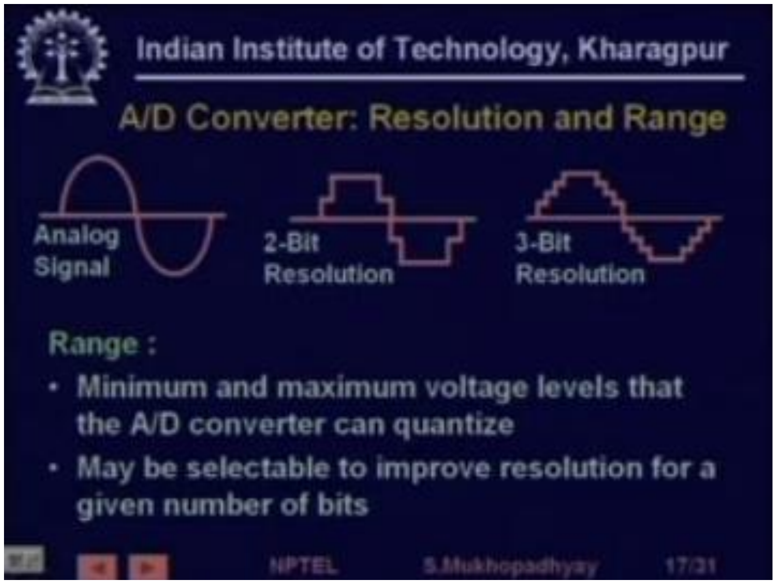


So we have so you A/D converter resolution and range, so now finally the signal this the accuracy is affected by the range the range is, so finally when you have a number how do you convert that number suppose you have a number 001 and it is a 3-bit A/D converter so the number is actually 001 divided by $2^3$ so 1, 001/ $2^3$ which is so it is 1/8 into the reference voltage in this case suppose 1 volt so it is 0.125 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.

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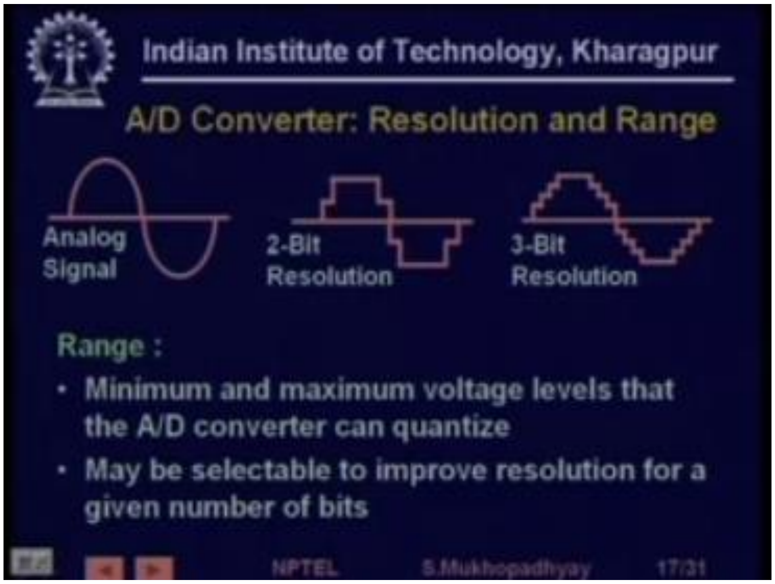


On the other hand if you have a, suppose you have a 10 volts range and you have a 1 volt signal. Now the point is that you could actually amplify this 1 volt signal.

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So if you use a, let us say we can you write here also that suppose you have a converter which is the 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 where -1 to or let us say 0 to, let us talk about unipolar signals 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 one volt signal variation would have been captured up to 3 bit resolution.

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So this shows that always it is you must have amplification that is why is people sometimes put programmable gain amplifiers. Amplifies those gains can be changed again under software controls, but although they are expensive. So you need to always amplify the signal so that the A/D converter dynamic range that is the range between the reference voltage up to the reference voltage is actually 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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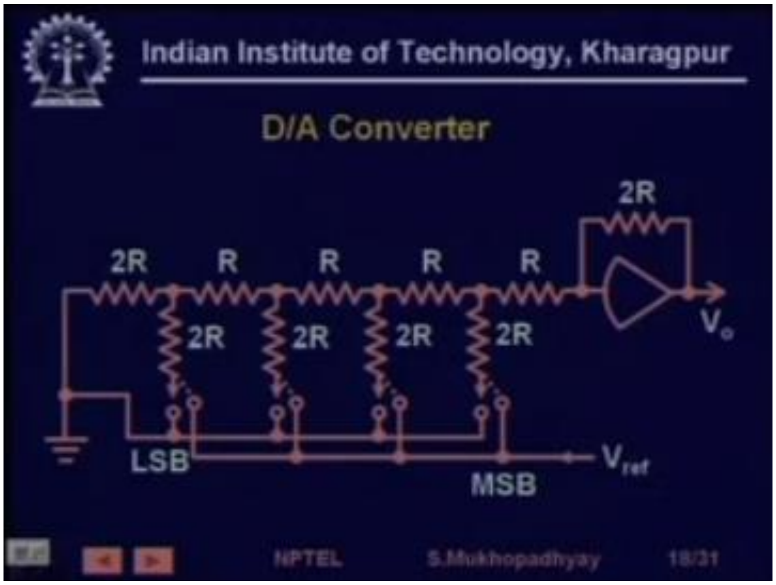


So now we are going to take a look at some A/D converters but then before that we have a, we take a look at a D/A converter because the A/D converter circuit this the most, one of the most common things like the successes convert approximation converter uses a D/A converter so we just take a quick look 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 2R ladder network.

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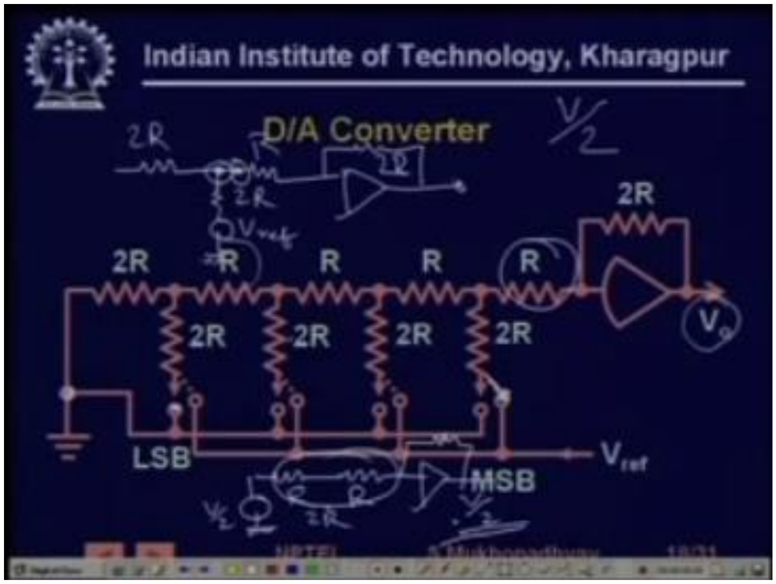


For obvious reasons you can see that 2R, 2R so there are some R and there 2R resistances there a double ratio, so that is why this is called an R 2R ladder network ladder because of this form structure. So you see that you can it is rather easy to show using simple network theory that you see these are the switches which are actually controlled by these bits, so now I suppose let us say the simplest case that if I switch only this MSB so it will get connected here and all the others are connected to this ground if this is connected to ground.

So then what is the voltage that will appear here, that is very simple because you have 2R and 2R in parallel now this connected to ground so therefore this residence is R, now this that R and this R is in series switch 2R again to 2R and 2R in parallel again R and R in series again 2R and so on. So finally what you get is a finally you get this network, so you have a 2R you have one 2R 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 R, 2R, 2R, 2R so if you do a therein circuit for this, this what will be this voltage, this voltage is so these are 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 therein impedance it is going to be 2R parallel 2R which is R, so this circuit can be further reduced into a network of this form, so it can be further reduced as V/2 in series with the resistance R which is the Therein 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 MSB should have a total range of n bits converter then the MSB should have a weight age of D/2 and you can see that.

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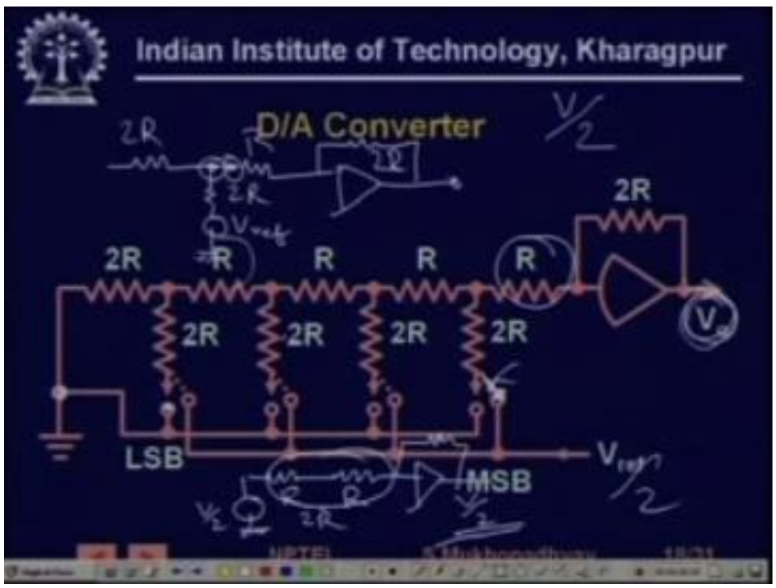


If you only switch on this MSB you get a $V_{ref}/2$ signal here, in this way you can show that if you if you kept all the others grounded and you put on the next bit on you would have got V/4 similarly V/8 and so on.

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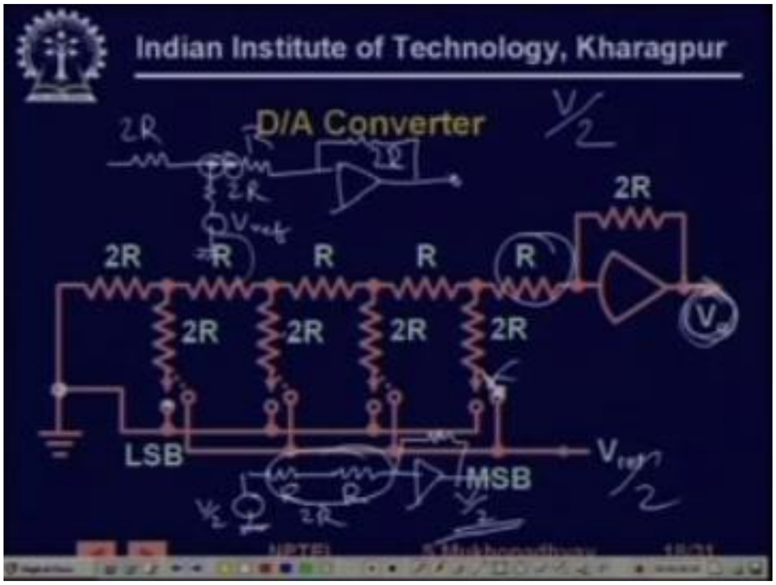


So now if you depending on your digital numbers if you switch on some of them and do not switch on some of them then you get a corresponding, because this is a linear circuit so you are going to get superposition principle so the final signal is going to be $V_{ref}$ properly weighted by the, with the binary number waiting.

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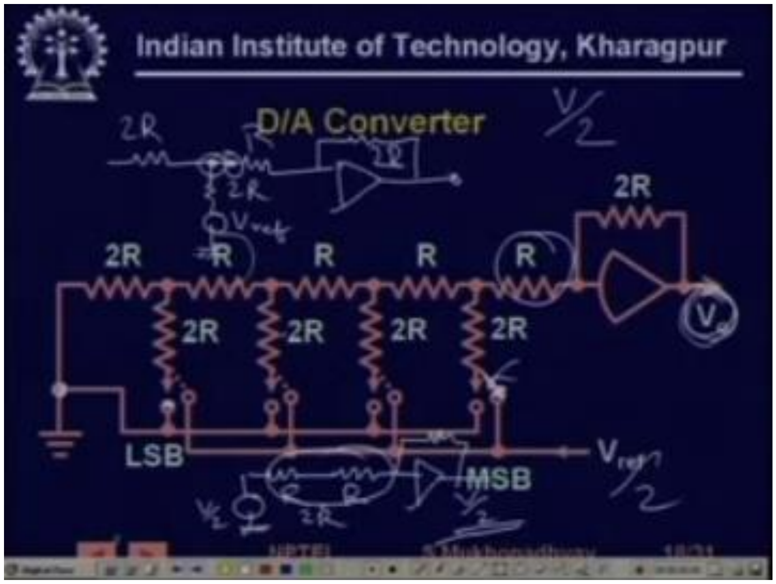


So you will get a analog voltage which is going to be proportional to the digital number that is why it is called a digital-to-analog converter so in our.

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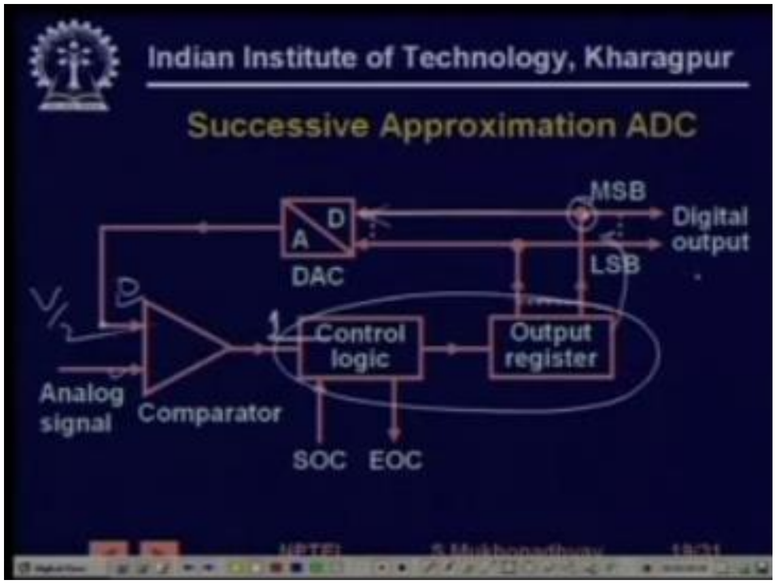


So for example in this A/D converter switch which is called successive approximation converter, principle is very simple you put a signal the signal first compares weather suppose this is initially this is 0, initially this 0 right, so this just compares whether this should be, this is higher or this is higher, so if this is higher it puts it 1, now when it puts it 1 it sets the this is some control logic, so this will put the maximum bit high that will know so maximum bit means V/2 so that will come to the D/A converter and will apply a V/2 here.

Now the question is whether this analog signal is greater than V/2 or less than V/2 if it is greater than V/2 then it is still 1 then the next bit will be put, so in this way first MSB is put then the next week till this signal crosses this signal and then at that level it is stopped. So that is how you just compare it with a number of digital, number of digital number still it exceeds and at that point you stop, so you get so when you stop you get a set of digital bits.

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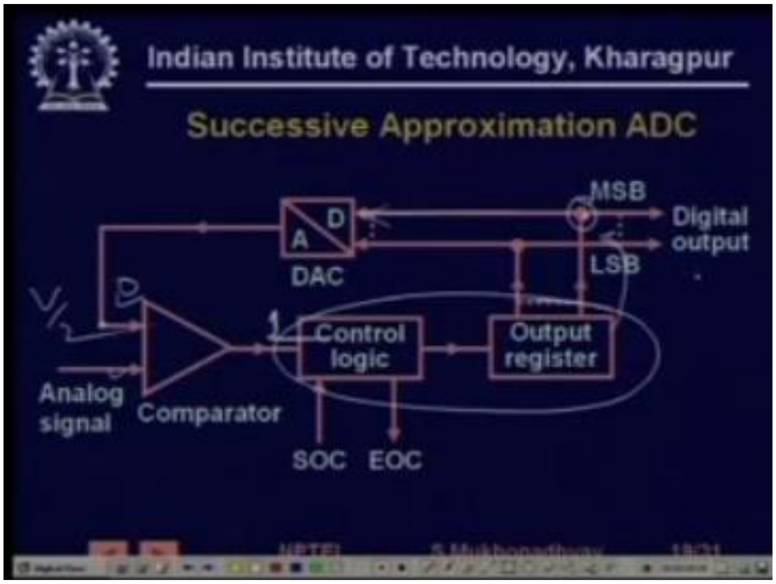


Which is closest to this analog signal right, so this is how you convert this is how a typical successive approximation converter is and.

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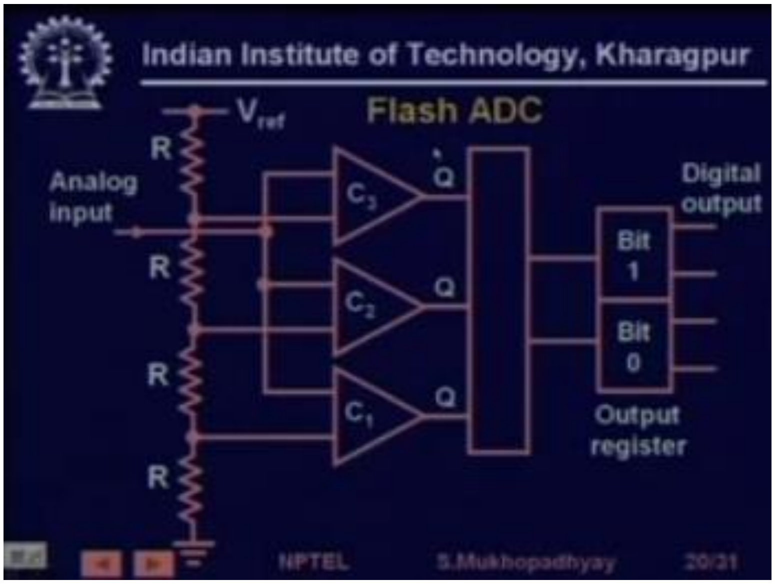


Just one way there are many various other converters for example a very fast converter is called flash ADC, we are not going to discuss that because you do not have enough time but this is a converter where you know there if you do it bit by bit so you need at least what case you can need n times you can go through this cycle of setting and resetting bits. So you need more time on the other hand in this converter the whole conversion is just in one clock cycle so it is much faster but then it requires a huge resistance network and they have to be very precise otherwise you are going to get errors so this is a converter which is therefore.

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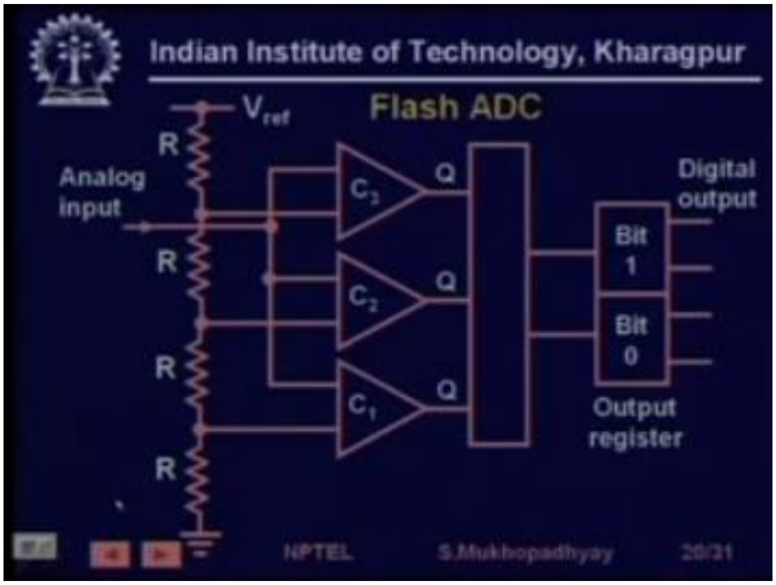


This converter has some problems and although it is very fast, so we are we are not going to look at that to do much.

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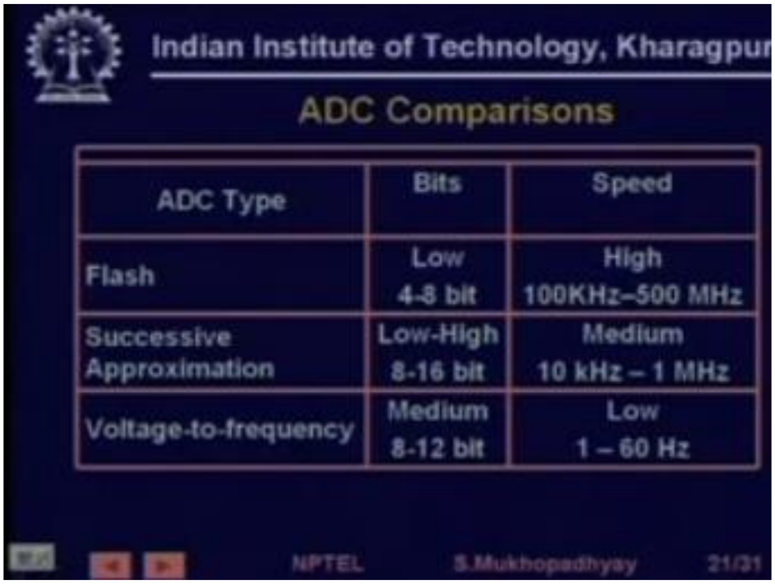


If you compare there are various kinds of converters and we are for example there are other converters to we are talking about flash we should saw successive approximation there are some converters which are called voltage to frequency or integrating converters which are very slow but very accurate, 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 we can go up to you know 500 megahertz kind of conversions.

Then successive approximation is very 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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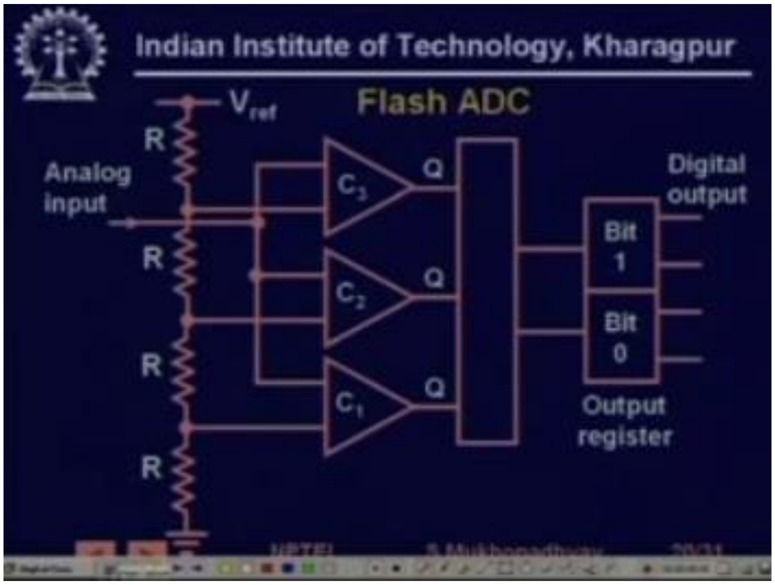


So having seen that, so now we have seen A/D conversion now we come to the system level discussion that typically there are two kinds of data acquisition system that you find one are called external bus or remote, so there what happens is that the computer is in a separate place generally.

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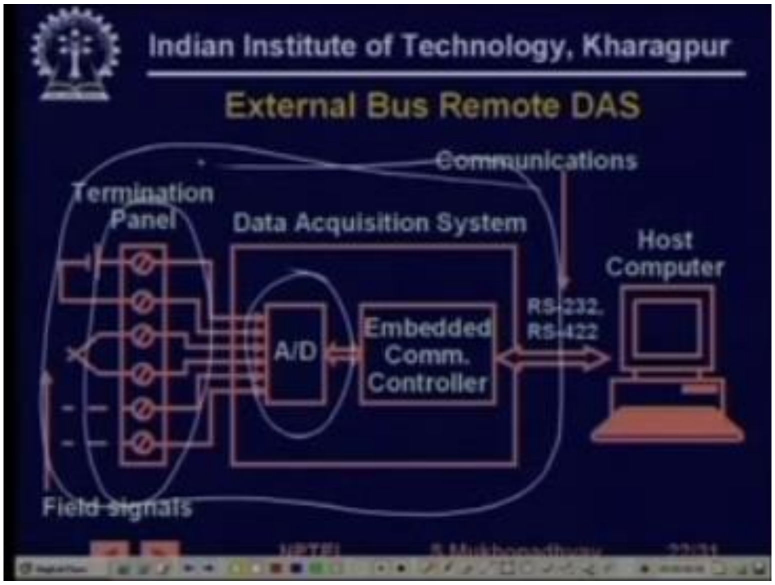


And oops I do not know what is happening yeah, so you see here these are all these signals analog signals are getting terminated. Let us assume that it is signal conditioned we are not if it is not that then it will be conditioned then the A/D conversion we have seen what is A/D conversion then now what happens is that is that this thing this physical these such systems boxes are separate or situated a separate place they have their own separate power supplies they are situated possibly much closer to the field.

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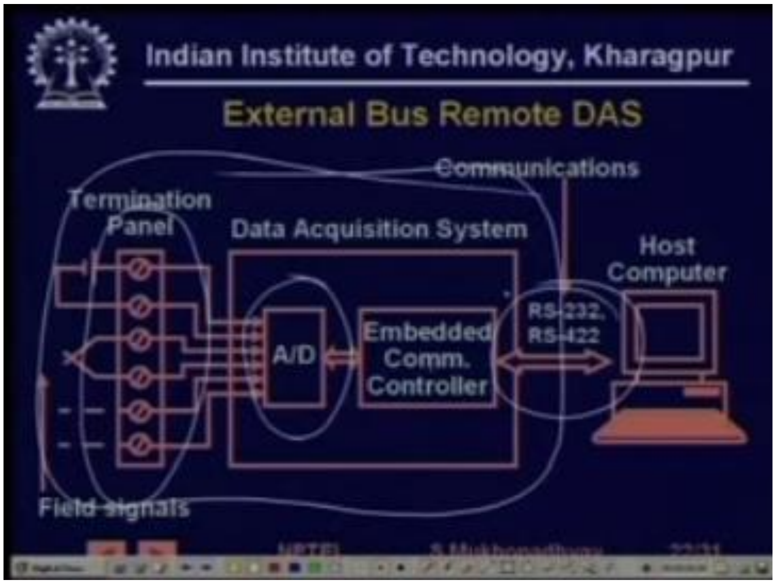


And then here this also after A/D converters and they have their own processors so the value is firstly coming to that embedded communication controller processor, and then using some very standard communication protocol either you know RS-232, 422, 485 or IPP 488 there are number of protocols by which through computer communication it is coming to what is known as the host computer.

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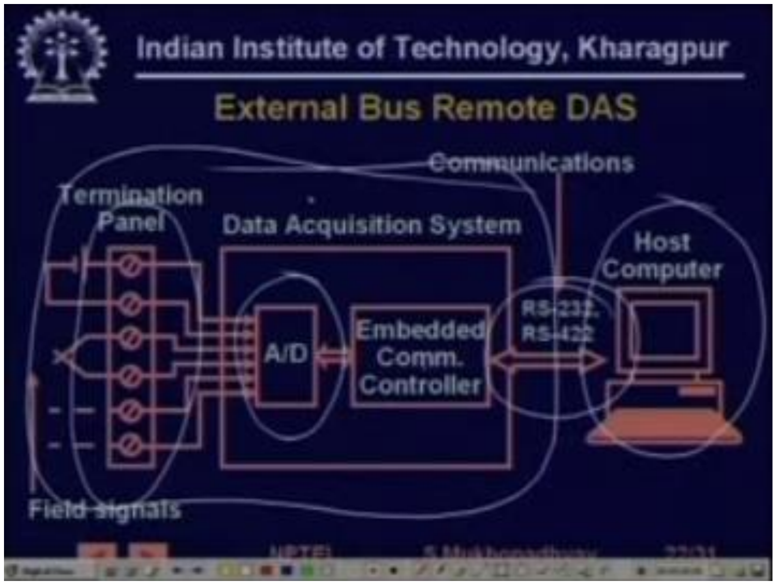


Where this data is going to be used, so this is a small box which is going to be close to the process and then one wire is going to come generally serial communication coming to the computer, so this is an external bus remote data acquisition system.

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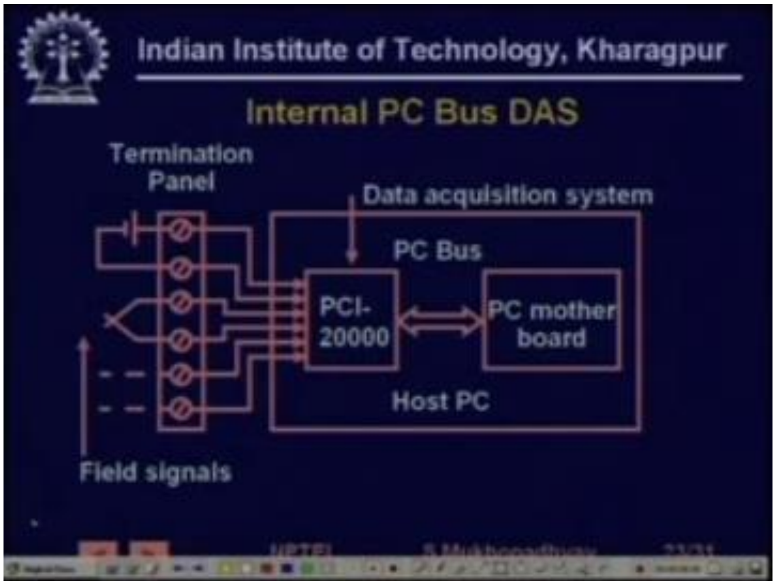


The advantages are that they can we can connect to any host computer it is so close to the field, right disadvantage is that generally the accusation rates are limited because of this serial communication so you can you get slightly slower rates of communication but if that is good enough for you it is, okay. Second kinds of things have internal PC bus data acquisition systems where the data acquisition system strikes inside the PC box right.

So and then they will communicate with the PC through the PCI bus which is much higher speed which is parallel in the facing, right. So the data acquisition rates can go much faster but because it but this means that the PC has to go close to the field otherwise you are going to have long cable links to the PC, right.

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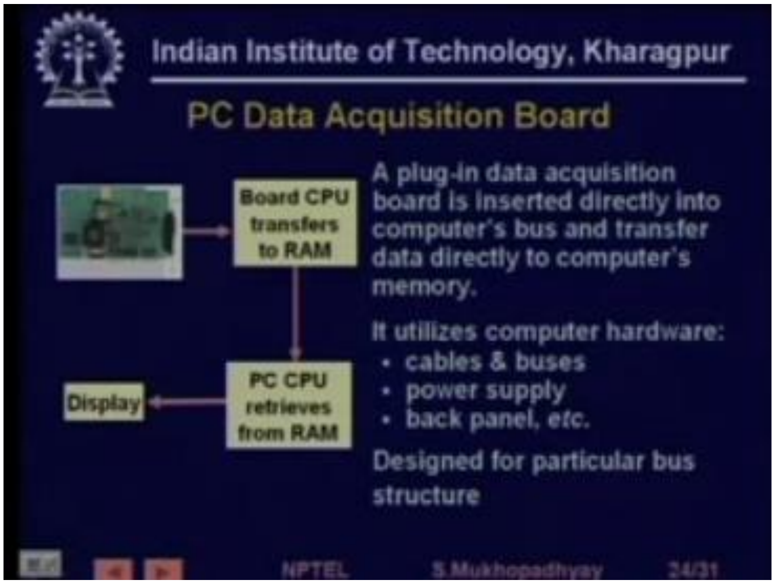


So similarly these are generally designed for specific machines like the like Windows PCs so this is a typical picture of a PC data acquisition board, so what happens is that the board itself has a CPU which transfers the data because it sits on the PC bars so it will transfer the data to the PC memory and then the PC CPU you can actually take it from the memory and then do a do a display, right.

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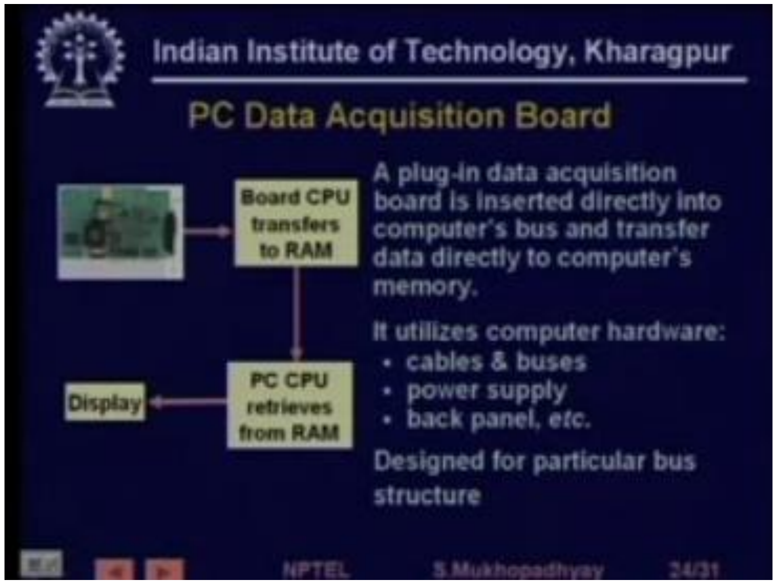


So there they are generally designed for this you know kind of Windows kind of PCs and they are very common and they are rather cheap.

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| | | |
|---|---|---|
| Power consumption | +5 V supply 800 mA typ. /1 A max<br>+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 | |

And especially can be used in a very much in a laboratory scenario and widely used. So these are the basic two types of data acquisition systems, if you take a typical specification of a data acquisition system they will mention things like you know power consumption see 5 volt supply has lot of current requirements because of the digital circuits then a 12 volt supply for analog the number of channels is very important typically you have about 8 analog channels either single-ended you can have 8 single-ended 8differential or 16 single-ended channels.

Typically number of digital channels are much more 64 like that because they take only one signal each, resolution says what.

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How many bit converter is being used, so it is a 12-bit converter accuracy the A/D converter type then you know this is this full scale this is a dynamic range so it is saying that the dynamic range is 0 to 10 volt DC then the A/D converter codes can be available in various formats whether it is 2s complement true binary offset binary various kinds of digital codes are there.

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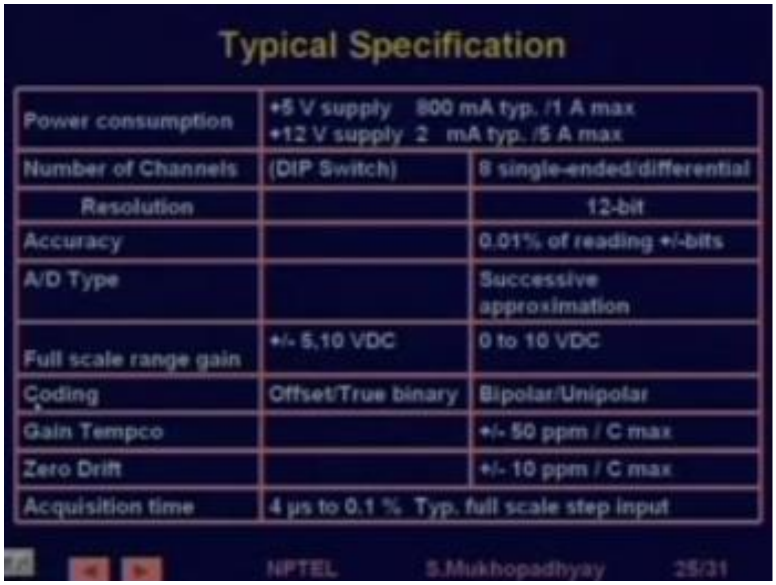


These thing this these two specification gain and zero drift specifications are for the signal conditioning block which is there in the thing so that if they will have an amplifier and then amplifier gain variation and zero drift can be specified and finally the acquisition time which says that the acquisition time is of the order of 4 micro seconds, you know so these are typical specifications of a data acquisition system which you need to look at when you select 1.

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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 cannot this 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 a lot of programming although it can give you a lot of I mean inflexibility in the sense that you can do anything you want with the data the raw data is made available.

But this requires lot of programming and often in an industrial situation is not preferred so you people use data acquisition software packages.

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So in programmable software you at the advantages flexibility and the disadvantages of complexity and a very steep learning curve.

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While if you have the data acquisition software then it does not require programming generally it requires you know graphical programming in the sense that using very common sense and domain related quantities you can actually configure the system and get data and then displayed it. So Airbus developers to design custom instruments best suited to their application.

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There are various examples, for example a common one in with which I am familiar is LabView but there are few others also.

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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 this sampling concepts, we have seen the sub details of another to digital conversion and finally we have taken a look at some data acquisition hardware and software features.

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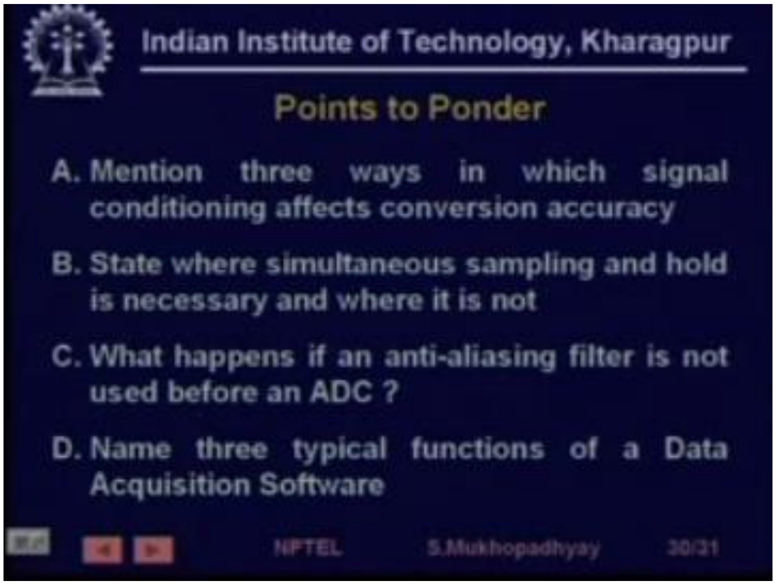


So points to ponder is mention three ways in which signal conditioning affects conversion accuracy, so I have already talked about dynamic range you can think of some others. State where simultaneous sampling and hold is necessary and where it is not, so as I said it is relate to get to the frequency content of the signals and the speed of conversion of your A/D converter and the number of channels of course.

What happens if an anti-aliasing filter is not used before an ADC so you have to go through if there is a successive approximation or if the ADC is flash so you choose the ADC and then go through it and see is that suppose in the middle, suppose that the signal just goes to 0 then what will happen. Sometimes it may give you wrong results sometimes in may not.

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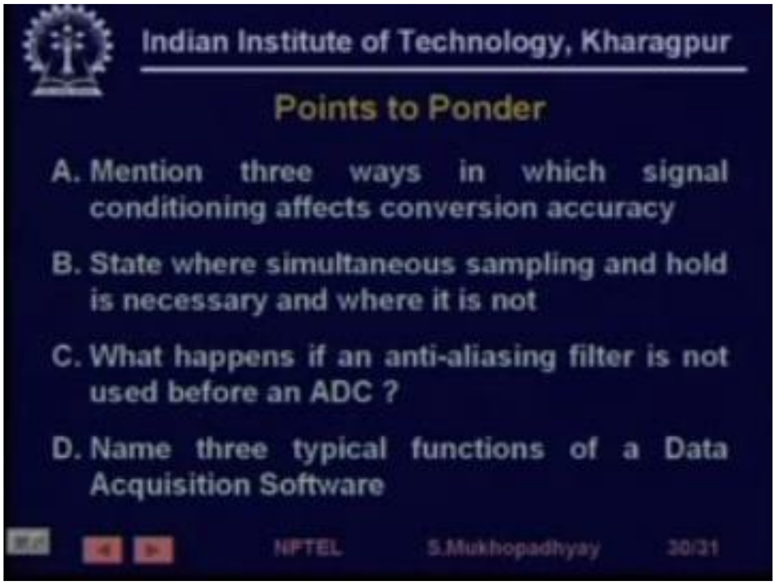


Name three typical functions of a data acquisition software, so one of them could be display you can figure out the other two by looking at some of the software which are advertised on the internet, so that is all for today thank you very much.