Op-Amp Practical Applications:
Design, Simulation and Implementation
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Lecture - 40
Introduction to Data-Acquisition

Hi welcome to this module and in this module we will learn Analog to digital converters and digital to analog converters. So, when we talk about analog to digital converters what does it mean and how it can use, how we can use it? Similarly if you talk about DAC is short form is ADC short form is DAC, how we can use ADC and DAC while designing your signal conditioning circuits. So, that is a idea to help you out what kind of ADC is are there what kind of DAC are there out there we will talk about that. There are a few parameters that you need to understand when you are designing or using ADC or DAC.

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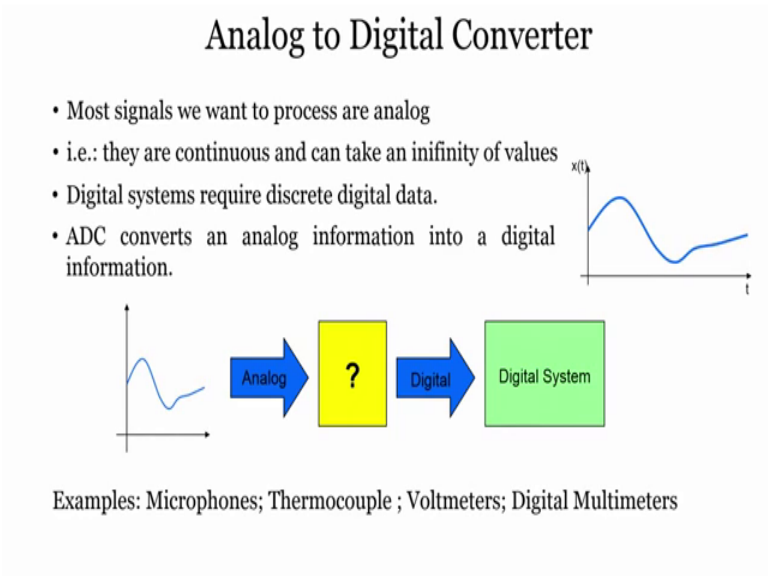

So, let us see if you see the first slide what you get is analog to digital converter most signals we want to process are analog in nature and there continuous and they can take infinite number of values. For example, if you talk about this particular signal how many values on this signal there are infinite values right. You can have like this and you go on designing anything like this infinite values you cannot really calculate so and for and there also continuous.

But if you when you talk about digital systems it requires discrete digital data, so the analog to digital converter converts your analog information into digital information right. So, analog and the digital, this is analog signal digital system and this guy is nothing but your ADC. So, examples are microphones, thermocouple, voltmeter digital meters.

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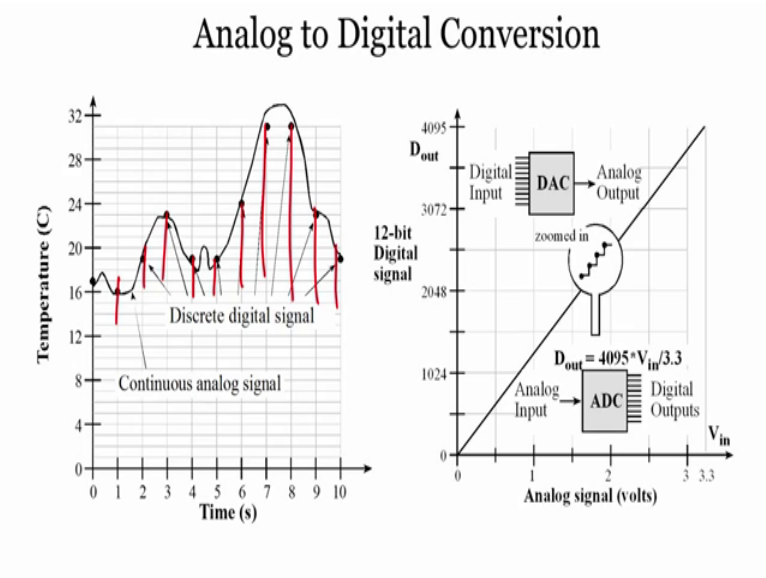

So, there are lot of examples where ADC can be utilized, you can see very clearly that if there is a continuous signal which is write over here right and you want to create a discrete signal from this, then there can be a discrete points right and if you watch those points then it is the same information. If you just see here, analog input digital outputs if you zoom in this looks linear right, but when you actually zoom in what do you find is there are a lot of steps in between. Same way digital input and it gives you analog output it will called DAC, this is analog signal was digital signal for twelve bit digital signal you can see the plot.

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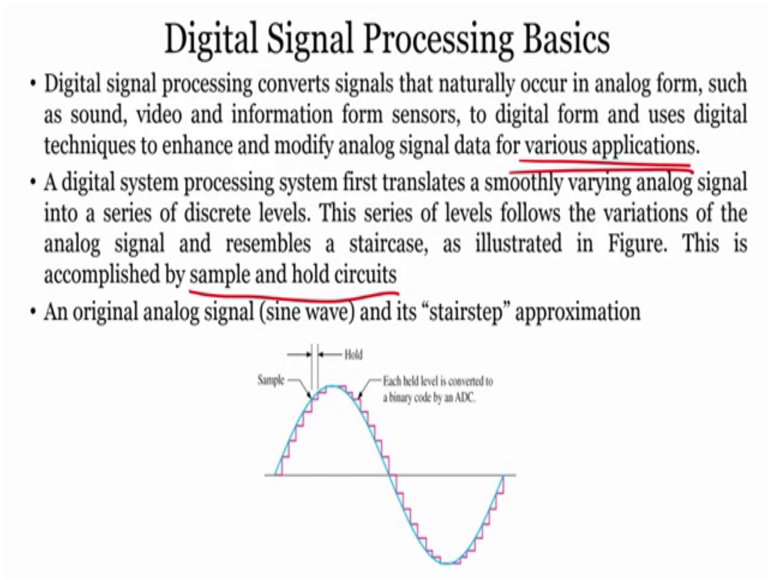

Now, when you talk about digital signal processing basics, then digital signal processing converse signals that naturally occur in analog form such as sound, video information from sensors to digital form. So, from analog to digital right so analog forms can be sound video information and lot of other things which are from the sensors and it has to convert to a digital form and use this digital techniques to enhance modify analog signal data for various applications right. So, a digital system processing system or digital system processing first translates smoothly varying analog signal into series of discrete levels. You can see here a smoothly varying analog signal into series of discrete values right.

So, the series of discrete values or the series of discrete levels, this series of levels follows the variation of analog signals and resembles a staircase, you can see here it looks like a staircase. Now to do to use this we can use a sample and hold circuit which you will see at some point in ADC and this is accomplished by using sample and hold circuits which is also written over here and original analog signal which is a sine wave and it is stair step approximation is shown.

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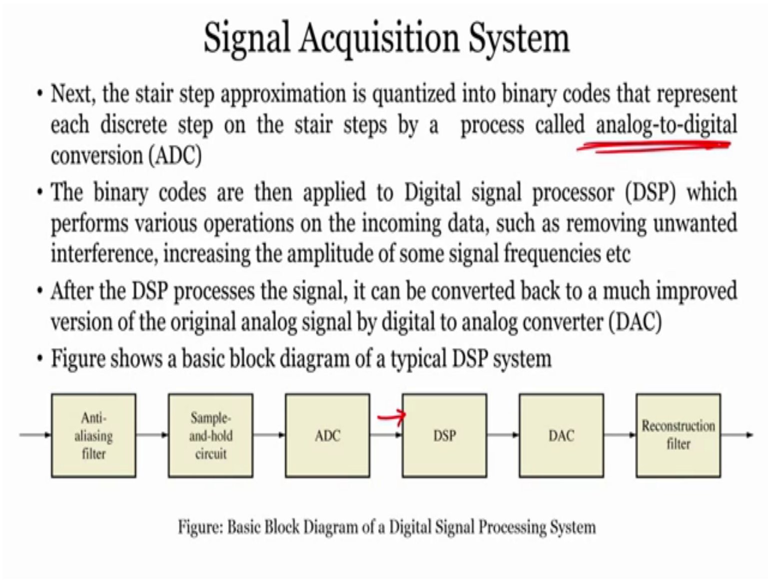

When you talk about signal acquisition system, so what does that mean? So, next the stair step approximation is quantized into binary codes that represent each discrete step. So, when we are talking about converting your analog signal to digital format or digital domain and you can see this stair step approximation. Then you have to further use that stair step approximation and it has to be quantized or quantisized into binary codes that represent each discrete step on the stair steps by processor called a to d converter right.

So, there where your ADC comes into picture, so if you see here there is anti aliasing filter sample and hold circuit and then there is something called ADC, then there is a digital signal processing it converts the DAC and the final reconstruction filter. So, these are the blocks were start from anti aliasing filter to sample and hold ADC, DSP, DAC and reconstruction filter.

So, this is how the signal processing block works and this is a very basic idea of how the block diagram of DSP system looks like. So, the binary codes are then applied to these a signal processor which is write over here right and which performs various operation on the incoming data such as removing unwanted interference increasing the amplitude of some signal frequencies etcetera. Once it is processing DSP it can be converted back to it is because, we finally we require the information that was sent right. So, after DSP processing the signal we can we converted back to much improved version of original analog signal by DAC and the figure shows the block diagram.

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Signal Acquisition

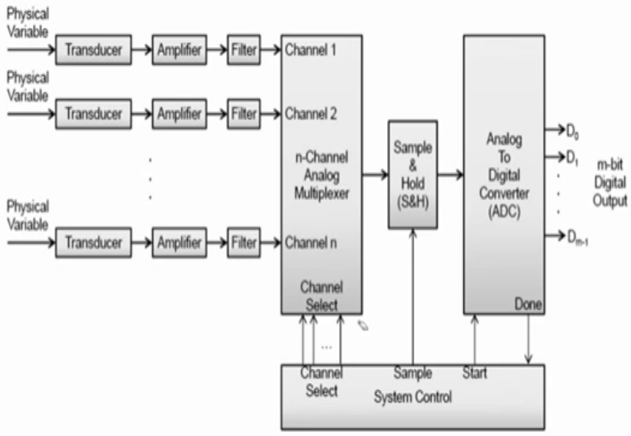

Figure: Block Diagram of a Signal Acquisition System

So, this is a block diagram signal acquisition system and here you can see the starting variable is physical variable followed by a transducer, followed by amplifier, filter and then there is an N channel Analog multiplexer followed by sample internal circuit to ADC and then the channel select and start and stop and sample and hold these signals are given through the block shown here which is your system control right. So, this is how the signal acquisition system block diagram looks like.

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Conversion Process

3 steps:

- Sampling
- Quantification
- Coding

These operations are all performed in a same element:
the A to D Converter

And now when you talk about conversion process then there are 3 steps that you do remember, the first step is sampling second step is quantification and third step is coding these operations are all performed in a same element. So, which is your A to D converter, so 3 things happens one is called sampling one is called quantification and one is called coding. So, 3 things happen when you convert the signal.

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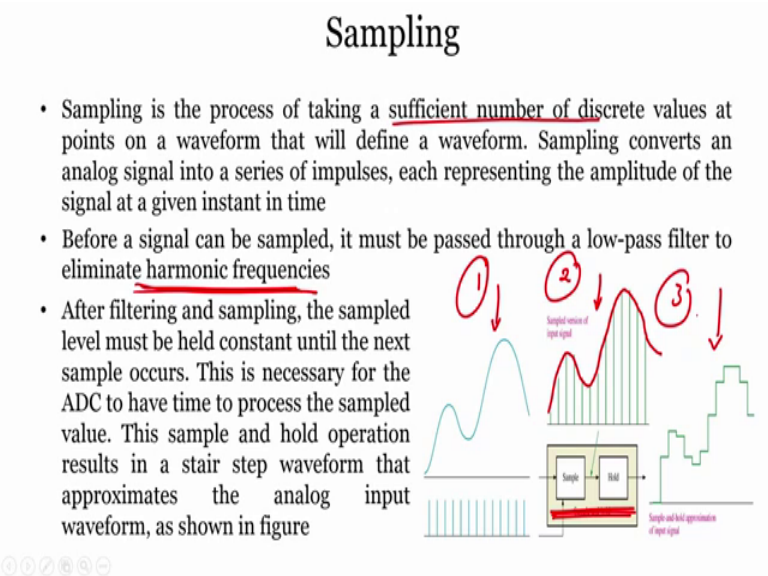

Now what is sampling let us see; so sampling is the process of taking a sufficient number of discrete values at a points on a waveform that will define a waveform. So, suppose I have this waveform write like this and I take sufficient points on this waveform let us say here, here, here, here, here and if I just point it out it looks like this know. So, is it sufficient yes it is sufficient because from this lines like this like you have seen here if I draw end in if I join all these lines, then you see what happens you have a similar signal right you see this one and this one that what is called sampling.

Sampling is the process of taking sufficient number of discrete values at a point on a waveform that will define a waveform, sampling converts an analog signal into series of impulses each represent representing the amplitude of the signal at a given instant in time. The first signal can be sampled it must pass through a low pass filter to eliminate any harmonic frequencies right we are to eliminate the harmonic frequencies.

So, what is the filter that we can use you can use low pass filter we have seen the advantage and the diagram of low pass filter, high pass filter, bend pass filter, band reject filter in the in the first lecture. As well as we have also seen in earlier course which was meant for understanding how these circuits can be designed and can be operated now we are using those circuits. Now after filtering and sampling the sample level must be held constant until the next sample occur, so once you filter and sample then you have to hold that thing right until the next sample. So, this is necessary for ADC to have time to process a sample value, this is done with the help of sample and hold circuit right.

So, this sample internal operation results in a stair step waveform that approximates the analog input waveform as shown in this particular figure. So, this is analog signal this one and this is sample version of input signal and this one is sample and hold approximation of input signal; this is how your sampling works.

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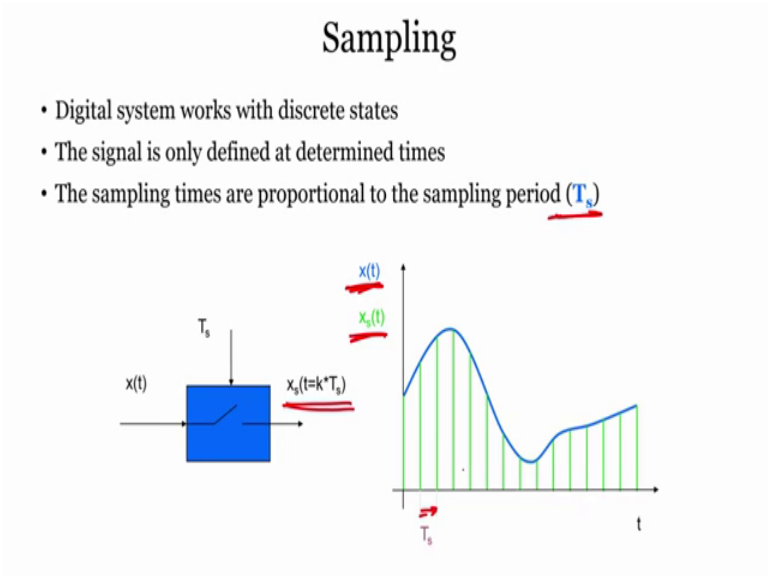

Now further if you see sampling the digital system works with discrete states, so the signal is only defined as determine times, the sampling times are proportional to the sampling period T s. So, if I apply x t if there is a T s then I have my value x equals to x s equals x s which is nothing but t equals to k times T s. So, this is the sampling time or sampling period this is my x s t this is my analog signal x t, so when I apply x t and you have a sampling rate or sampling period, then your value would be t equals to k times T s right. This is from this plot you can understand what this equation means.

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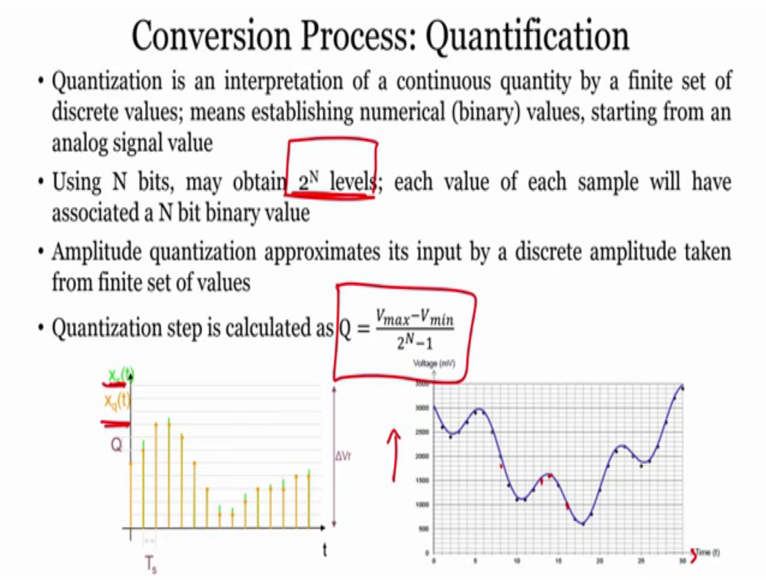

Now, once you do the sampling then let us see the quantification. So, not quantification, quantization is an interpretation of continuous quantity by a finite set of discrete values means establishing numerical or binary values starting from an analogue signal value right. So, using N bits we may obtain 2 raise to N level right each value of each sample will be have associated and bit binary values right. So, for N bit we are 2 raise to N levels, so amplitude quantization approximate it is input by discrete amplitude taken from finite set of values.

So, if you just want to calculate your quantization then you can use this formula where quantization is nothing but V max minus V min into 2 by ratio of V max minus V min and 2 raise to N minus 1. So, if you see here you have a plot here right, the x q t x s t and then you have here which is analog voltage or milli volts and you have time. Now you have this discrete values here that you have calculated and if you join this discrete values then you can form this particular signal right or you can break this signal into discrete values. There is another way of looking at it the point is that for N bits we will have to raise to N levels right and each sample will have associated N bit binary values that is very important to remember.

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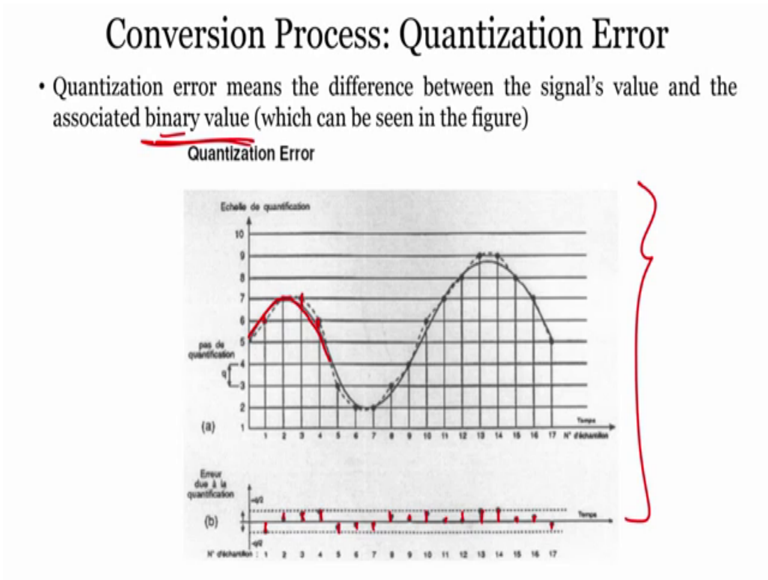

Now, if you see quantization error what does it means, it means the difference between signal value and the associated binary value right. So, you can very easily see here that what are the errors in this particular block right. The signal value which is right over here and associated binary value, associated binary value is you see there is a difference here, difference here, difference here and if I plot it is like this correct. So, this is a difference of the signal value and the binary value this is your error.

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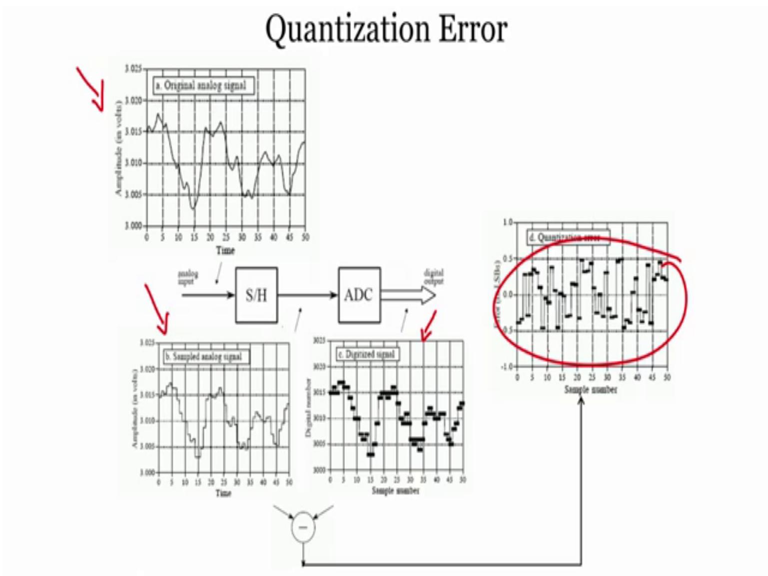

Now, you can here also see that original analog signal is this guy and then when you apply to sample and hold then sample analog signal is this one, when you go for digitized signal digital signal looks like this and if there is a quantization error then you can see here right. This is how the quantization error will occur and then you have when this is mixed and it is here then you have this quantization errors quantization error should be minimum.

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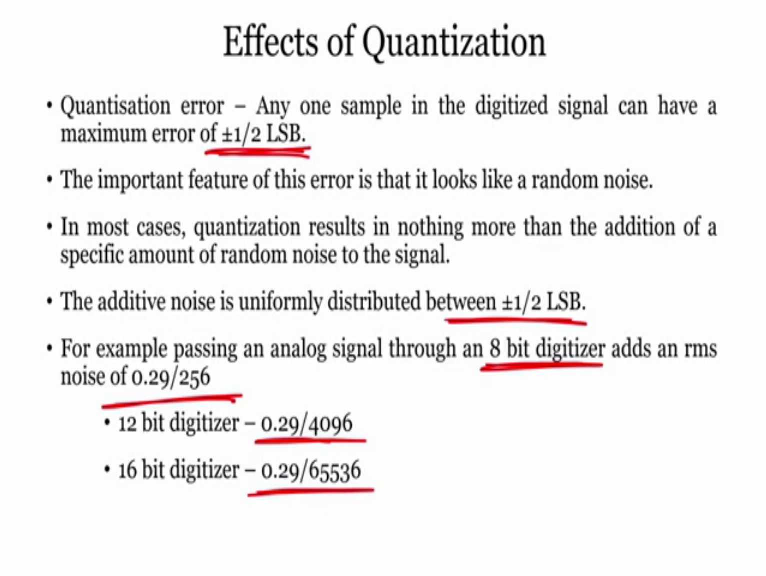

Now, effect of quantization so what will be the effect anyone sample in the digitized signal can have maximum error off, so that is another thing that you have to remember. When you design your ADC is that quantization error should always be plus minus half of least significant bit and it cannot be sorry, these quantization error can have a maximum value of plus minus half of least significant bit and the important feature of this error is that it looks like a random noise. So, if you really see it does not look like some kind of signal it really looks like a noise signal right, we just look at the analog noise then what you feel like this right.

So, that is another point the additive noise is uniformly distributed between plus minus half LSB, for example passing analog signal through 8 bit digitizer adds an RMS noise of 0.29 by 256. So, for 12 bit digitizer it is 0.29 by 4096, for 16 bit digitizer it will be 0.29 by 65536. So, this is very important to again I remember that quantization error should be minimum, in most of the cases it is nothing but more like addition of a significant amount a specific amount of noise and it is uniformly distributed between plus minus half LSB. And if you want to calculate passing an analog signal through 8 bit digitizers how much RMS noise is added then we can calculate like 8 is 2 raise to 8 to 2 raised to 8 is 256. If you are 12 bit digitizer is 2 raise to 12 is 4096, if you have 16 bit 2 to the power 16 will give you 65536, so 0.29 by 4096 0.29 by 65836 then comes another value which is called coding.

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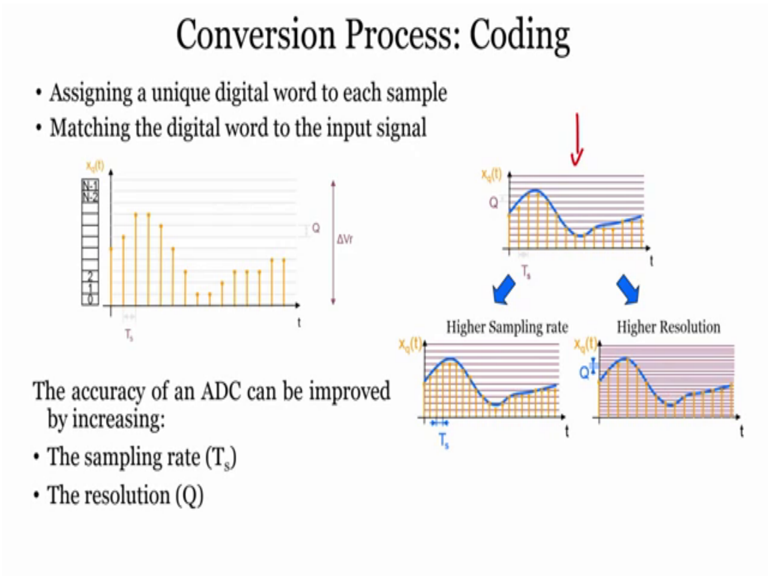

So, as assigning a unique digital word to each sample and matching the digital word to the input signal right. So, you understand here now that if this is the signal and if I have a higher sampling rate and if I have a higher resolution what is the difference that you have to have better sampling rate and better resolution. So, by increasing the sampling rate that how many times you sample, if this is the T s if I have a one smaller T s you see then I have a sampling rate which is higher; by higher sampling rate with for my analog signal in a better format right. So, I can reconstruct my signal at some stage in a way better format, same way goes for the higher resolution higher the resolution you can have a better accuracy of the entire system.

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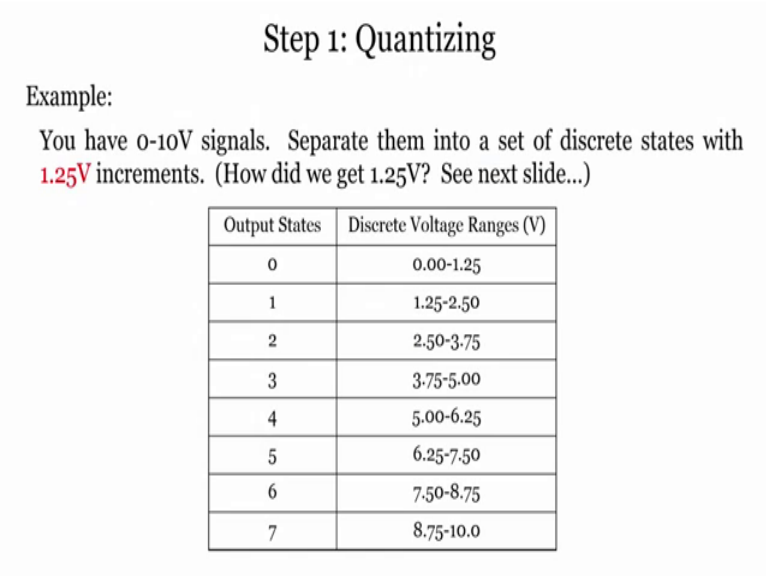

Output States	Discrete Voltage Ranges (V)
0	0.00-1.25
1	1.25-2.50
2	2.50-3.75
3	3.75-5.00
4	5.00-6.25
5	6.25-7.50
6	7.50-8.75
7	8.75-10.0

Now, if I give an example then if you have 0 to 10 volt signal, even if it is separate in set of discrete state which 1.25 volts increment how did we get 1.25 volts we will see in the next slide. But let us see here so you can write 0 to 7 right because, 0 to 10 volts is there let us say output states are 0 to 7 0.00 1.25; 1.25 to this, so when you have this kind of discrete voltages is you have output state which is from 0 to 7.

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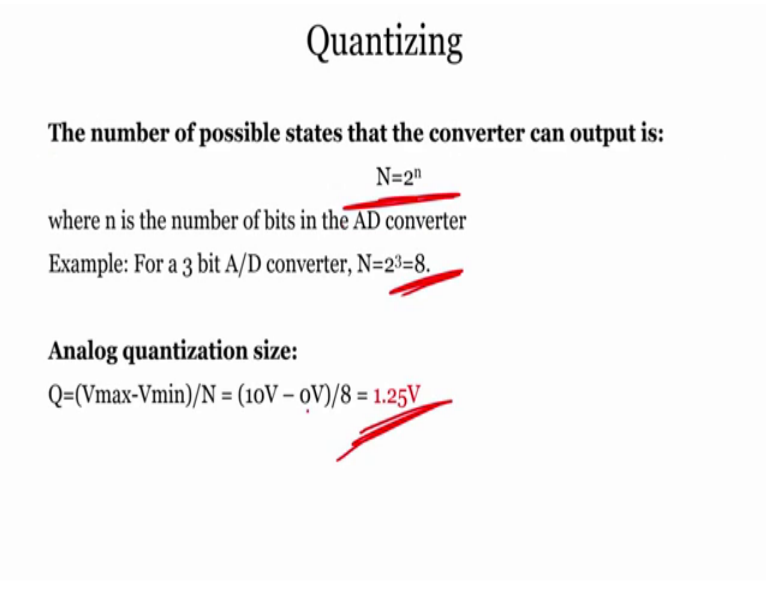

Now, the number of states that converter can output is N equals to 2 raise to N, now what is our thing we have said that if we use 3 bit ADC then it is 2 raise to 3 which is 8. So, Analog quantization size would be V max minus V min by N what is V max V max is our 10 volts right what is V min is here 0, so we max minus V min divided by N. So, 10 volts minus 0 volt what is our N, N is 2 raise to n so divided by 8, so we get 1.25 that is why we have discrete states of 1.25 volts each so this is a quantizing.

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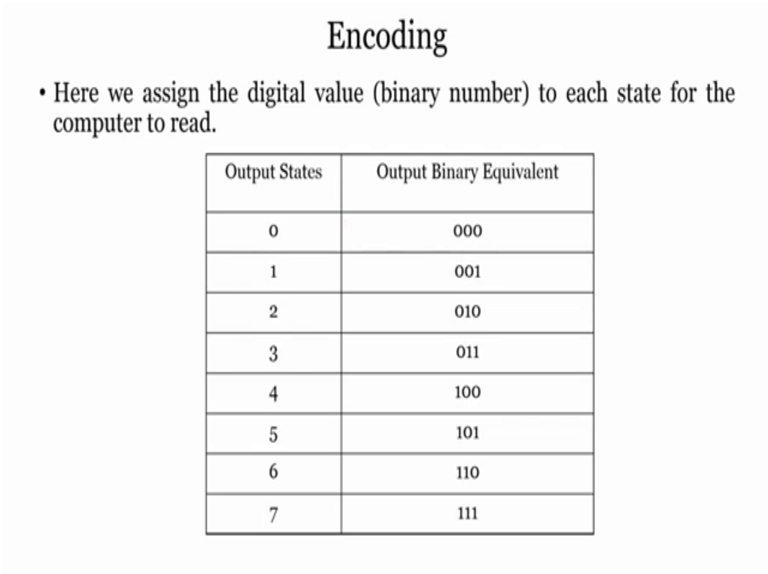

Encoding

- Here we assign the digital value (binary number) to each state for the computer to read.

Output States	Output Binary Equivalent
0	000
1	001
2	010
3	011
4	100
5	101
6	110
7	111

Now, if you talk about encoding then here we assign digital value to each state for the computer to read. So, if you have 0 to 7 because you have within 0 to 7 you can you can put your analog signal at an increment of 1.25 that is why you have 0 to 7 and if you want to put 0 to 7 in binary state then you can you can easily use it right because this is 3 digit we are using 3 digit a to d, so we have 0 is 000, 1 001 and so on. You have 7 equals to 111 you guys know it write how to do it is 2 raised to 0 2 raised to 1 2 raised to 2 there will be your 8 bit, 3 digit or 2 raised to 8 is 256.

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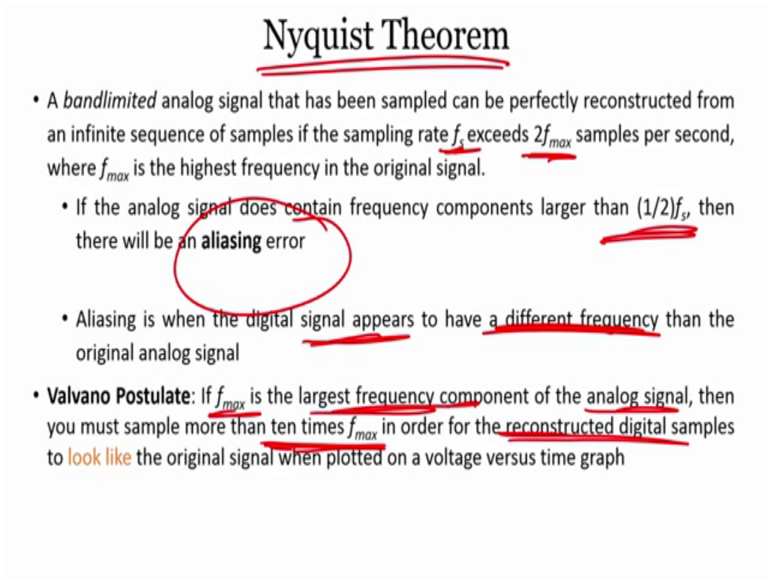

So, Nyquist Theorem, now this is again very important there band limited analog signal that has been sampled can be perfectly reconstructed from an infinite sequence of samples in the samples rate f s exceeds 2 f x samples per second, so where f s is our highest frequency of the original signal. So, what it says that for an analog signal that has been sampled, if you want to perfectly reconstruct then you should have a sampling rate which exceeds 2 times f max, where f max is your highest frequency in the original signal. So, if the analog signal does not contain frequency larger than 1 by 2 f s then there will be an aliasing error, we will see what is arising error. So, aliasing is when digital signal appears to have different frequencies than the original analog signal there is a aliasing.

So, when your digital signal appears to have a different frequency, you will see what is aliasing in the next slide and then there is a Valvano postulate which says that if your f max is largest frequency component of the analog signal, then you must sample more than 10 times f max. Why? In order to reconstruct digital samples to look like the original signals when plotted out up voltage versus time graph. So, if you have your sampling which is 10 times, then your f max then you can reconstruct your digital samples exactly or to look alike your original analog signal.

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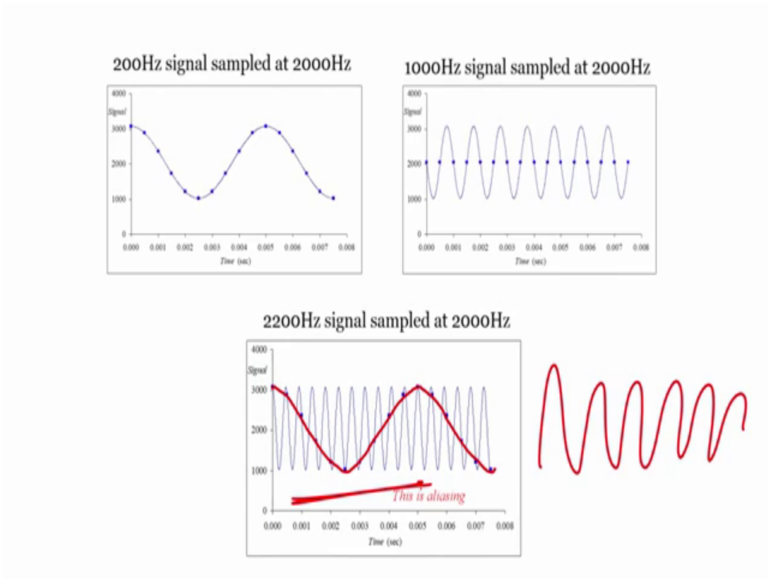

Now, if you see here you have 2000 hertz signal amplified at 200 hertz signal amplified at 2000 hertz you can see here. Now 1000 hertz signal amplified at 2000 hertz you can see this waveform and 200 and 2200 hertz amplified to 2000 hertz you can see this waveform right. So, here you can see very clearly that there is an aliasing error because if I do this right, then you see if I am reconstructing this signal this is what I am getting. But what is my original signal my origin signal is this is not is not this right is not this, so this is aliasing error that is why you should always have f s greater than 2 times your f max right.

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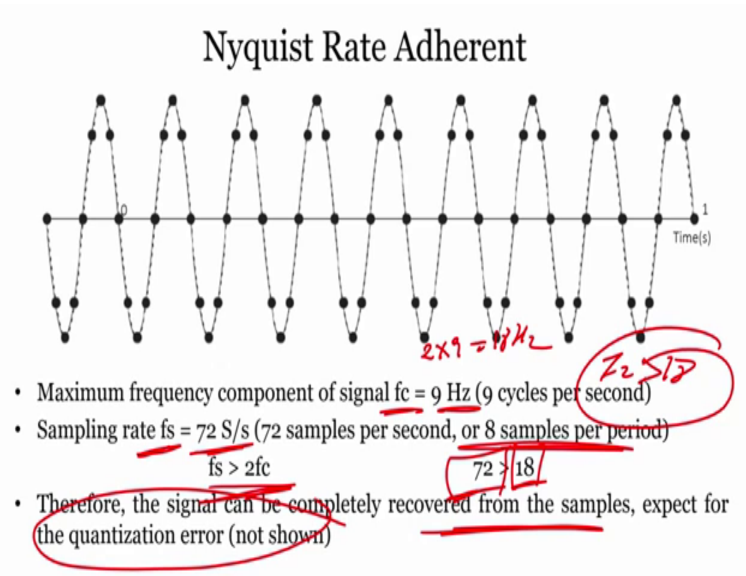

So, now suppose you talk about here maximum frequency component for signal fc is 9 hertz that is 9 cycles per second and sampling rate would be fs equals to 72 samples per second or 8 samples per period. In this case fs should be 2 times fc so 72 is greater than 18, is it correct? No, therefore but fs is what 72 samples per second is greater than 2 times fc. So, 72 is greater than 2 times fc or not is greater than 2 times 9 2 times 9 is 18 hertz 72 is greater than 18, so we have we have we have stick to the Nyquist rate and that is why the signal can be completely recovered from the sample except quantization error which is not shown here.

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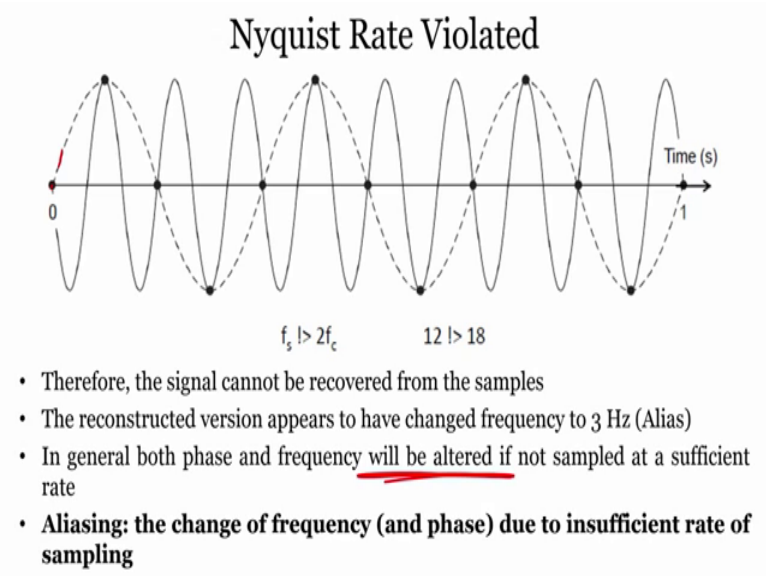

Now, if I do not follow the Nyquist rate and if I violate the Nyquist rate which is write over here, then the signal cannot be recovered from the samples you can see very clearly right. You can see how it is violating and because of that the signal that I am reconstructing is very different from actual signal which is right over here correct. So, if you do not adhere to the Nyquist rate then you will not get the reconstructed proper signal. In general both phase and frequency will be altered if not sample at a sufficient rate and the aliasing the change of frequency and phase due to insufficient rate of sampling is called your aliasing.

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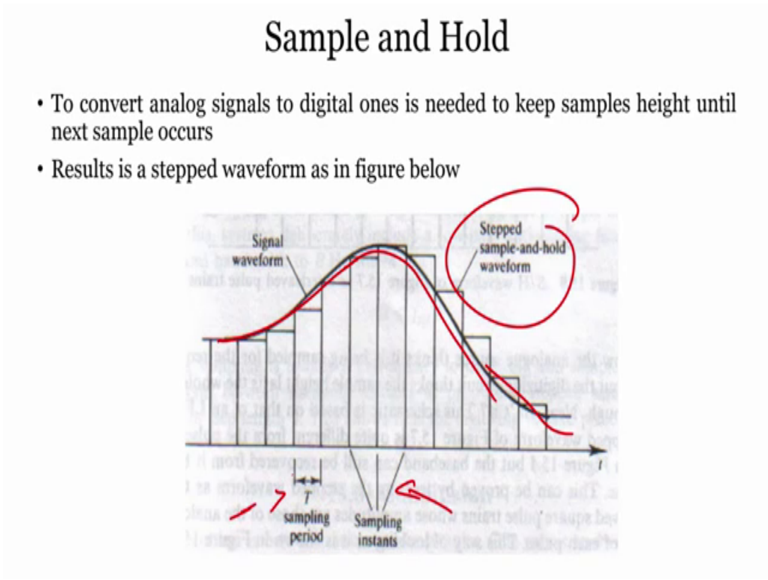

Now comes to your sample and hold, so to convert your Analog signal to digital ones we require the sample height until the next sample occurs right. So, this will one is needed to keep in a samples height until the next sample occurs, result is a step waveform which is shown here, right sample step waveform. These our original signal right these our original signal and this is a sampling period T this is sampling instance which are shown in the plot and if you see here what is the need of sample and hold circuit.

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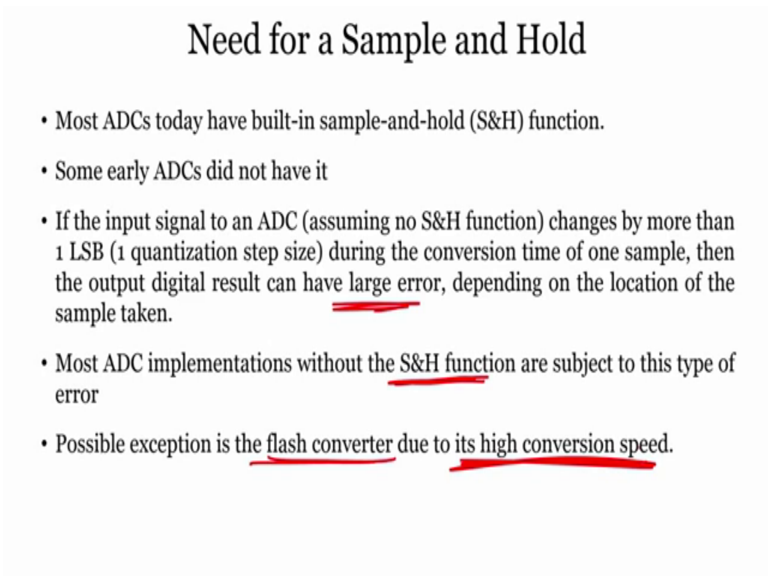

The need of a sample and hold circuit in most of the ADC is today they have built in sample and hold circuits some already say ADC is did not have that and if the input signal to ADC changes by more than 1 LSB. Then during the conversion time of one sample then output digital result can have larger error right and depending on location of the sample taken. Similarly most ADC implementation without sample and hold functions are subject to this type of error and possible exception is the flash converter due to it is high conversion rate, so accept less converter other ADC is will suffer if you do not use sample and hold circuit.

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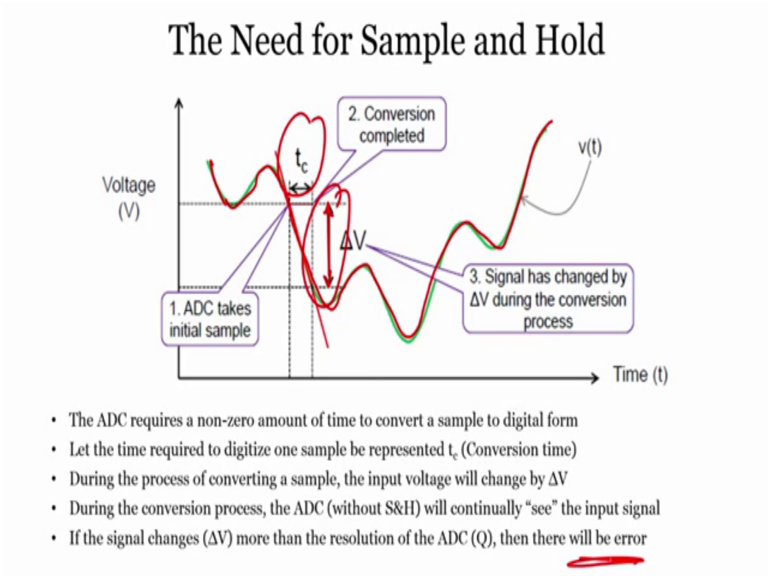

The ADC requires a nonzero amount of time to convert a sample to a digital time or form let the time required digitize one sample bill is represented by tc which is write over here right and during the process of converting a sample the input voltage will change by delta V which is write here right. So, this is the signal actually and we are trying to see how we can use the sample and hold during the converter process the ADC without sh will continuously see the input right.

So, the one that is not using s signal and sample and hold circuit with just see the input signal continuously, but if the signal changes to delta V right then more than resolution of the ADC and then there will be error. Because if you do not use sample and hold then what you see is this signal which is going here right, but if our error is more than delta V right or the signal change is more than delta V then it is difficult for ADC to follow right so there will be an error, so there is a importance of sample and hold circuit.

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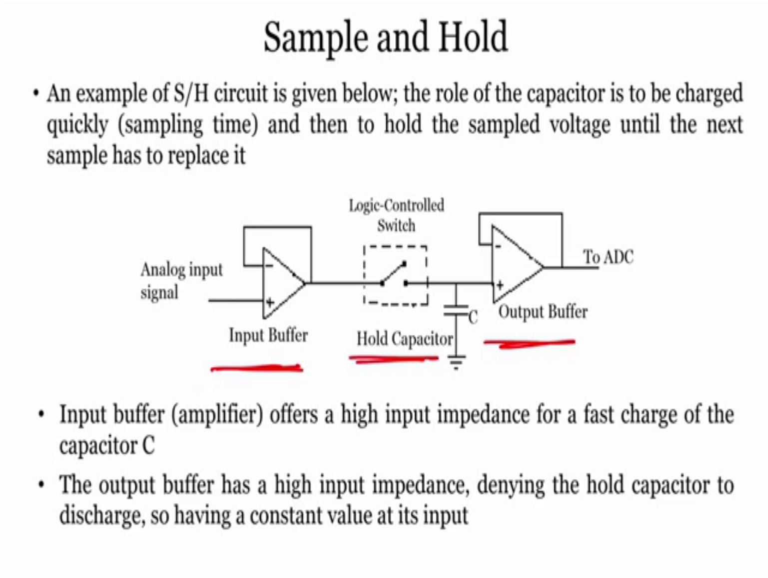

Now, an example of sample and circuit is given below and the role of the capacitor is to be charged quickly during the sampling time and to hold the sample voltage until next sample has to replace it there is a role of the hold capacitor. So, when you sample it analog signal then you have to hold the capacitor and you do resample it. So, input buffer amplifier offers a high input impedance which is write over here for a fast charge of the capacitor and the output buffer which is write over here has a high input impedance denying the hold capacitor to discharge. So, having a constant value at the input, that is the role of the 2 buffers input and output buffer across the capacitor and this is how your parameters for ADC will work.

So, to understand how the ADC works you have to go through these parameters understand how the coding works, how the sampling works, how the quantization effect works, how the sample and hold circuit works and what is an Nyquist theorem, what is an Nyquist rate why fs would bigger than 2 times f max? If you do not adhere to that what kind of signals are changing this is what is given write over here right. In the next module we will see how the digital to analog converter works all right, till then you read this and I will see you in the next module take care.