Switching Circuits and Logic Design
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Lecture – 49
Analog-to-Digital Converter (Part I)

So, in this lecture we start our discussion on the design of Analog-to-Digital Converter. So, earlier we had seen how we can design a digital to analog converter, now we are looking at the reverse. Given an analog voltage which is continuous valued how to convert it into an equivalent digital world.

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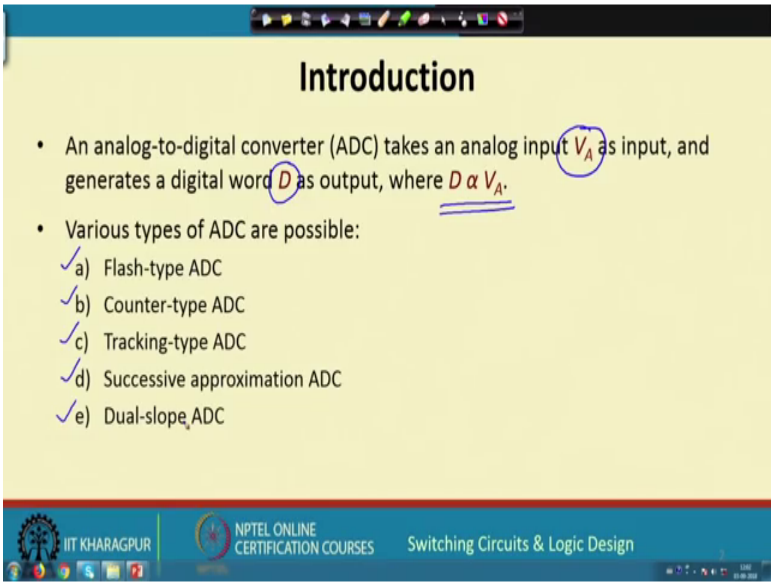

So, this is what is mean by A to D conversion let us see. So, an analog to digital converter said it takes an analog voltage V as input; that means a continuous valued voltage as input. Well, it can be the temperature of an environment, you are sensing the temperature and using a transducer you are converting into a voltage that can be V A for example. And it will generate digital word D as output such that there is a proportional relationship between D and V A.

But, for D A converter we only looked at a couple of designs, but here will see for an A D converter different designs are possible we shall be looking at 5 different designs; flash type, counter type, tracking type, successive approximation and dual-slope type, right, but before going into the designs there is some basic issues regarding analog to digital conversion that needs to be thought about; you see when you talk about a DA converter you are saying that well you are applying a digital input you will be getting an analog output now in digital input may be coming from some register or somewhere or some output of some gates. So, during the conversion the digital input can be held constant, but now think of an AD converter think of the example I talked about suppose we are measuring the temperature or means any kind of an analog signal from the environment maybe the temperature of an oven I am measuring.

Now, it is possible that because of some issue the temperature of the oven is changing or fluctuating very fast. So, it may so happen that while the conversion is going on your input is changing. So, you would expect that while conversion is going on input should be fixed then only the conversion takes place in a correct way, but the input continues to change. Then, there can be some error in conversion, ok. This is one issue for AD conversion you need to look at.

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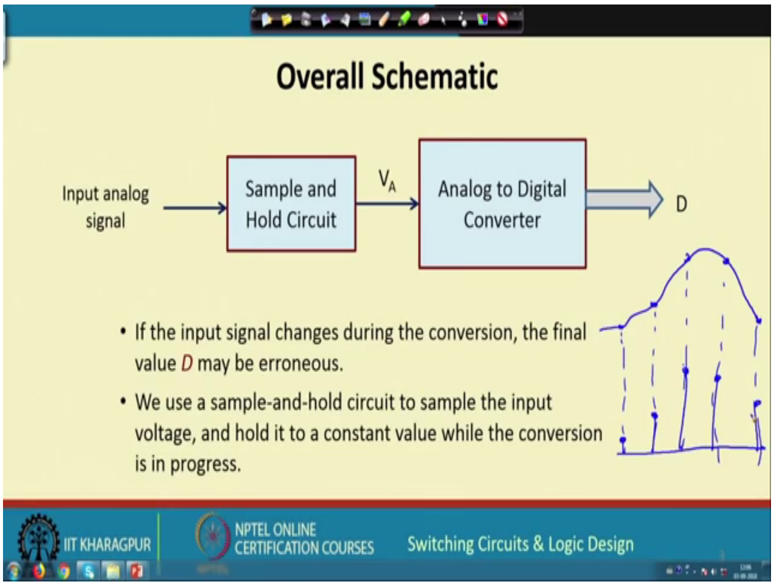

So, let us talk about the overall schematic here. So, here we have the analog to digital converter, but we need another lock before that sample and hold circuit to address the problem that I mentioned just now. Sample and hold circuit means you sample the input voltage what is the voltage value at this point in time suppose it is 5.3 volt that is your sampled value, and hold it somehow keep that 5.3 volt fixed or constant while the conversion is going on. So, that whatever digital output D is generated that will be proportional to this 5.3 volts, ok. If the input voltage changes then the D you really do not know what the value of D will be, fine. So, just what I have said that if the input signal changes during the process of conversion, then the final digital output value may be erroneous we use this kind of a sample and hold circuit to sample the input voltage and hold it to a constant value while the conversion is in progress.

You see the idea is like this suppose, my input signal let us say it is varying like this and let us say I am sampling the values periodically like this here and here. So, in the scale I will be getting one sample here corresponding this voltage may be another sample little higher corresponding this voltage third sample here may be somewhere here, fourth sample here may be somewhere here and fifth sample here may be somewhere here. So, with respect to D values I will be getting values like this. So, I will be not getting the original waveform, but I will be getting discretized values of the waveform these five values, these are called the sampled values.

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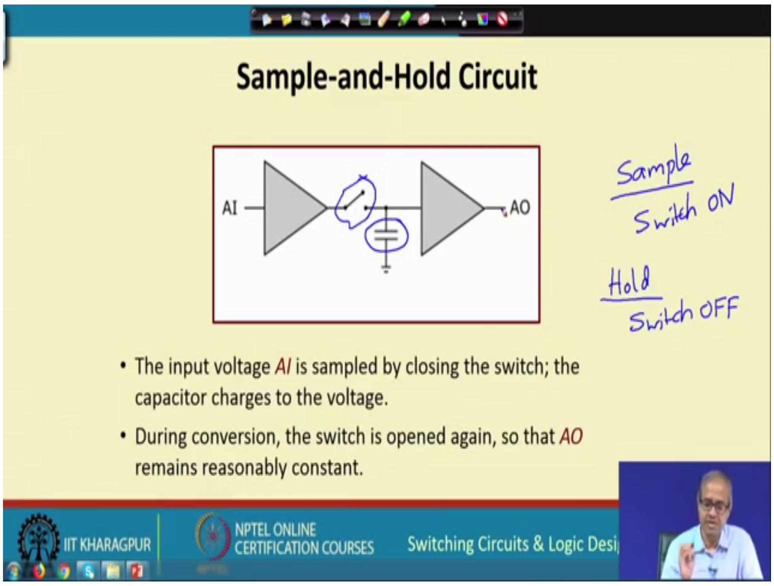

So, the sample and hold circuit I am not going into the detailed design of it. This is the little beyond this scope of this course, but conceptually let me tell you how it works. So, inside there is a capacitor there are some buffers these triangle symbols are two buffers and there is some kind of an electronic switch which can be turned on and off. So, while you are sampling the sample phase your switch is on switch is on means whatever is your analog input AI this voltage your capacitor will get charged to that voltage. So, your voltage is retained in the capacitor and during hold phase your switch is turned off. Switch-off means whatever voltage or charge was there on the capacitor it will remain the charge cannot leak through or discharge through any path.

So, the input of the buffer will remain at that voltage and this analog output will be will contain the sampled value. So, sampling is carried out by closing the switch and charging the capacitor and hold is carried out by opening the switch and using this buffer to transfer the voltage on the capacitor to the output, right. This is how sample and hold works.

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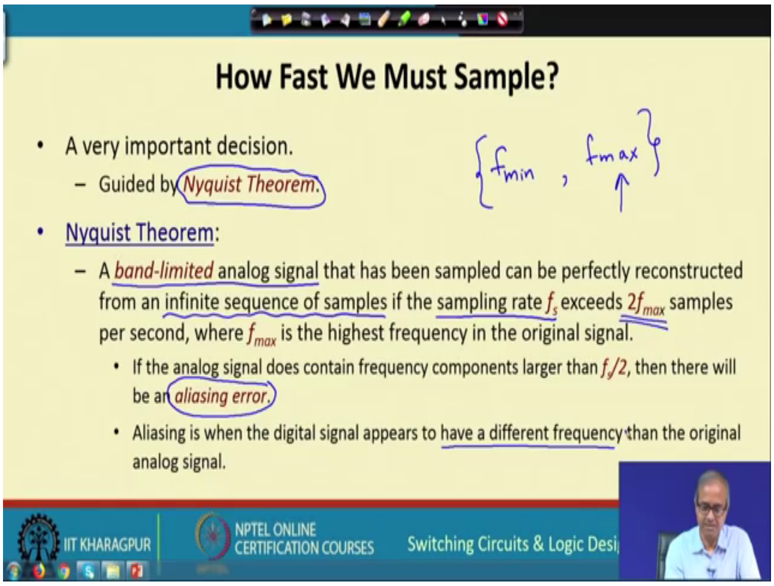

Now, there is some interesting observations and theorems. Now, the natural question to ask is how fast we should sample the signals. So, I have a signal I know some characteristic of the signals I know the frequency of the signal suppose it is an audio signal I am sampling some speech voice. So, I know the frequencies typically audio signals if we sample up to 5 or 6 kilo hertz we get good quality of reproduction, ok. So, let us see.

The question how fast we must sample this is answered by a theorem referred to as Nyquist theorem. Nyquist theorem goes like this it says a band limited analog signal; band limited means there is a signal you see in a signal there can be many frequency components there may be some minimum frequency, maximum frequency, a signal is a composition of all the frequencies; maybe I have a minimum frequency f min I have a maximum frequency f max. This range of frequency this is called a band band-limited means all frequencies are lying within this range.

Band-limited analog signal which we are sampling can be perfectly reconstructed if we have a sufficiently large number of samples if this is important if the sampling rate f s exceeds twice f max the maximum frequency in this range, where f max is the highest frequency in the original signal; that means, if we know what is the maximum frequency component in your original signal you must sample at a rate which is greater than twice of that Nyquist theorem specifies this condition, ok. So, if I know my maximum frequency is 6 kilo hertz I must sample at a frequency which is greater than 12 kilo hertz, ok.

So, if you do not do it then there will be something called as aliasing error, I should show some examples what is aliasing. Aliasing means because of this error the sampled value whatever you are sampling it will appear to have a different frequency than the original signal that you are sampling.

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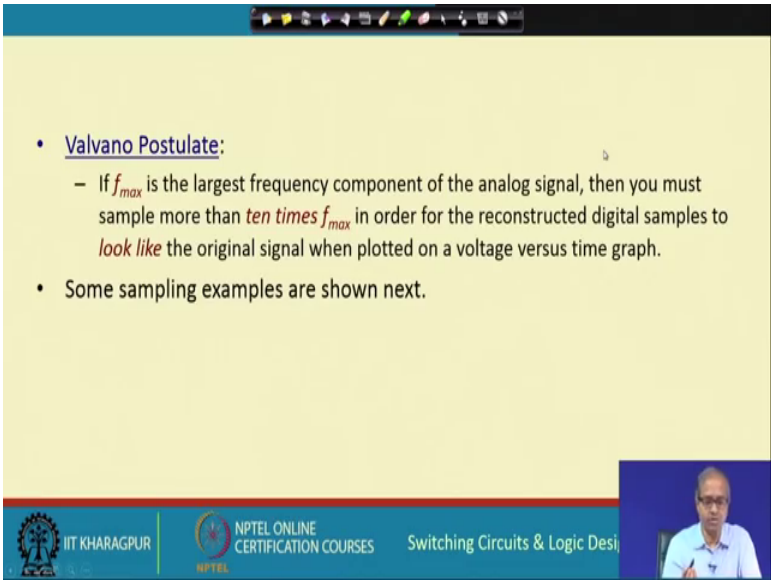

And, there is another kind of theorem or you can say postulate this was by someone called Valvano. Valvano’s postulate says this is just a role of the thumb it says again that if we know the maximum frequency in a signal f max then if you sample the signal at least 10 times then this sampled value will approximately look like the original. That means, the shape of the signal you can quite accurately guess from the sampled value.

The previous Nyquist theorem says that you can reconstruct this signal Valvano's postulate says just by plotting the sample values you can have a very nice idea regarding the original shape of the signal, ok; so some examples I am showing next some sampling examples.

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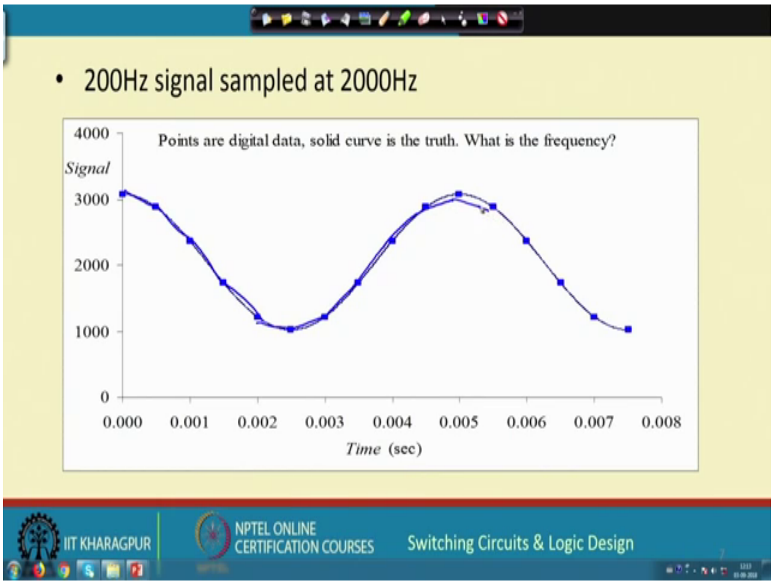

This is with respect to Valvano's postulate at 200 hertz signal let us say: a single frequency component which is sampled at 10 times the rate. So, here I am showing a sign wave 200 hertz this blue dots are the samples sample period which is you are sampling at 2000 hertz.

So, if you do not have the original waveform; if you only have the blue dots just by joining the blue dots by straight lines you will have a very fair idea how your original signal look like, just join them by straight lines. You will have a very fair idea about the original waveform this is what Valvano's postulate talks about.

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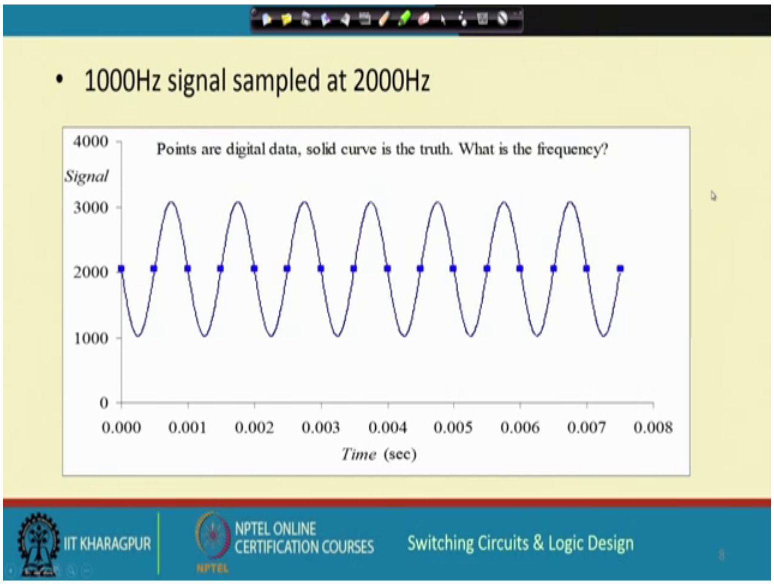

Let us take another example where we have a 1000 hertz signal for the sake of example I am taking a single frequency, because in real signals we have a combination of several frequency like an audio signal let us say. It is not an single frequency I am generating when I am speaking it is a combination of many frequencies that I amplitudes are different there all combined together. Normally when you do a Fourier transform we can find out the frequency components that are present in a signal you need not go into the detail.

But, you see here we are sampling this 1000 hertz signal at exactly at double the frequency not greater than double let say exactly at double what will happen. Well, it may so happen this is an extreme case that we are sampling at the dot points which are exactly in the middle which means you see; let see the signal values are ranging from 1000 to 3000; minimum is 1000, maximum is 3000, but if you sample at the blue points that double the frequency you will see all the sample values are 2000. So, you will have an illusion or means you will having a feeling that your signal is constant always at 2000. So, that it is a sine wave you are losing that information totally but this is happening, because you are not sampling at a frequency which is greater than twice f max, it is exactly equal to f max twice f max you are sampling right, but it should be greater than that, ok.

This is one nice example.

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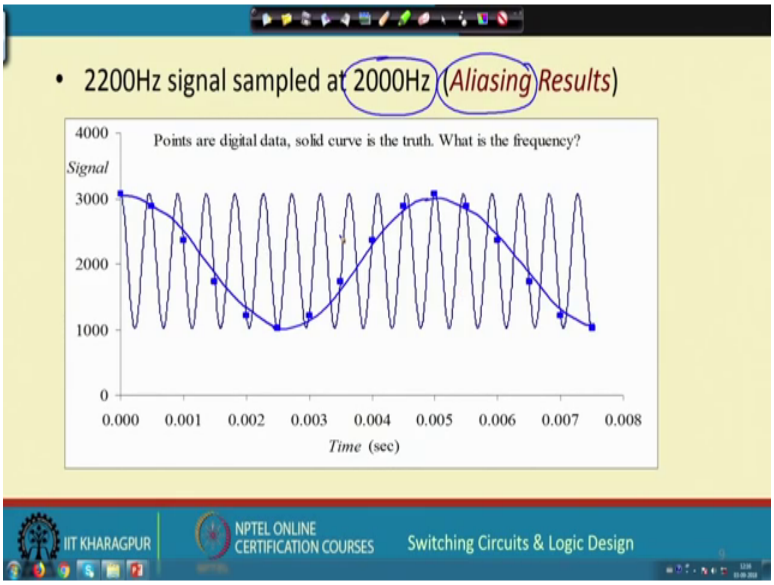

Now, let us say we are sampling at a frequency which is much less. This is again an example worked out at 2200 hertz signal which you are supposed to sample at a minimum greater than 4400, but you are let us say you are sampling at only 2000 hertz. So, the dots of the sampling points are shown. So, if you just join the dots you will see that it also looks like a sine wave, but the frequency of the sine wave is entirely different with respect to your original waveform.

So, your original waveform was at 2200 hertz, but here your this waveform is having a frequency which is much smaller may be about hundred hertz or so, right. This is called aliasing that because of improper sampling frequency you may be getting some sample values, where the frequency you are getting or you are guessing may be entirely different with respect to the original waveform.

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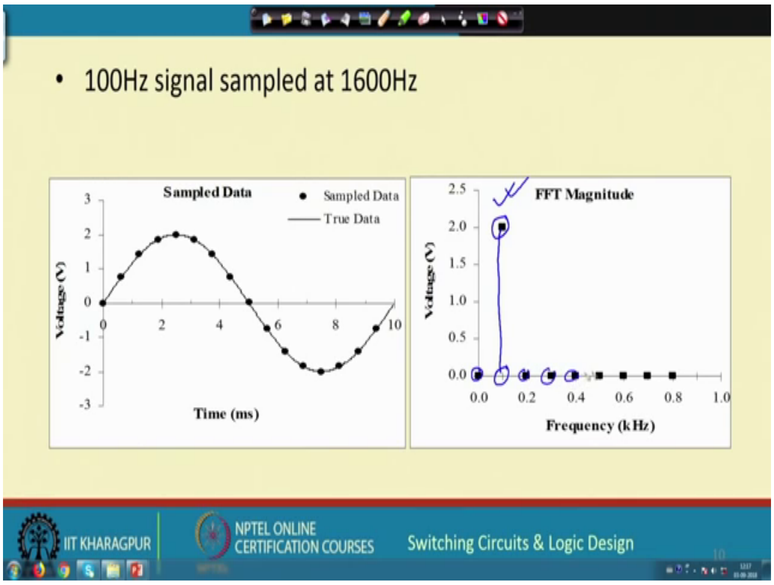

So, 100 hertz sampled at a sufficiently high frequency more than 10 times 16 times. So, 100 hertz signal sampled at sufficiently high frequency, while on the right side I am showing the Fourier transform results. So, if you take these sampled values and you can use something called a fast Fourier transform you will be getting the frequency components of all the frequencies. So, here you see that for this 100 hertz you have a maximum amplitude value coming. So, you are getting the correct result here and for all the other frequencies amplitudes are showing as 0. So, this no component, there is only one frequency component.

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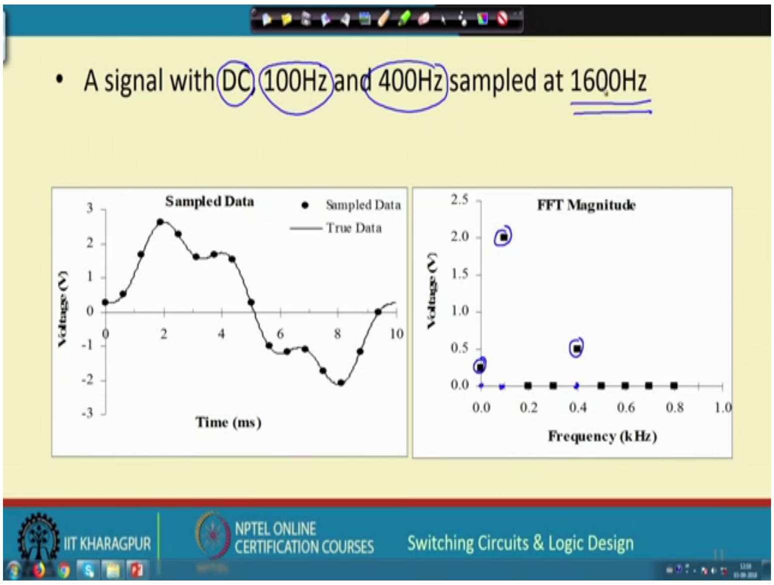

Let us take another this is a composite signal which consist of three frequency component frequency of zero which is a DC component; that means, average value is not 0 that is called DC 100 hertz and 400 hertz and you are sampling again at a sufficiently high frequency. So, again if you do a Fourier transform you see at DC, 0 frequency you are getting a frequency component, at 100 hertz you are getting a frequency component, at 400 hertz you are getting a frequency component and all others are 0. So, you are getting correctly you are retrieving the frequency components, ok. So, if you sample at a sufficiently high frequency.

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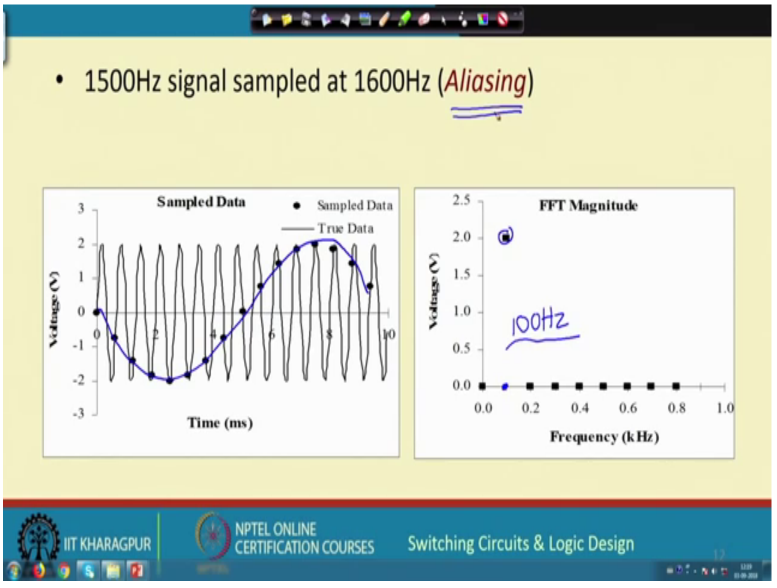

Let us take an example here while you are not sampling at a sufficiently high frequency 15 hertz signal sampled at 1600 hertz 1500 hertz. So, you see this sample value is again if you just join them they appear to have a much lower frequency. And this will be very apparent if you compute the first Fourier transform of the sampled values you will see that not 1500 hertz, 1500 hertz is much towards a right you are getting a very high amplitude at 100 hertz which is wrong, right. In your original waveform there is no 100 hertz component, it is a single waveform of 1500 hertz. So, here aliasing is taking place.

So, you should be very careful when you are choosing the sampling frequency when you are designing the input circuit to an AD converter. When you are getting your input ready by sampling you should be very much aware of the frequencies of the signals ok, so that you know; what is the sampling rate you should use so that whatever you are sampling and converting that is very much commensurate and corresponds to your original waveform right.

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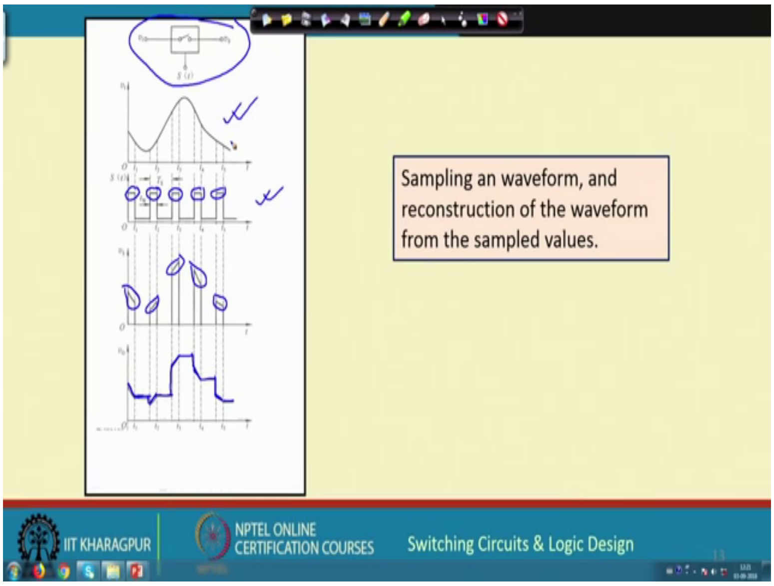

So, you look at a complete picture here. Suppose, I am showing a schematic of a sample and hold circuit like this and input voltage is coming and the switch where turning on and off let say and your input voltage is like this is your input voltage. And, here I am showing a sampling a clock kind of a thing which tells you where I am sampling I am sampling here I am sampling again here I am sampling here and here, right. So, with respect to the input waveform I was sampling here; that means, at this point then again after a gap I am sampling here; that means, here again here then here then here.

Now, in between that I am not sampling anything. So, I am assuming that the signal is not changing. So, if I reconstruct the signal form the sampled values it will look like this sampled value then here, I am assuming signal is not changing then again sampled value here again signal is not changing. I am assuming next the signal sample value is here again not changing again sample value is here. So, there is a drop again not changing again sample value is here not changing. So, you see your original waveform is like this your reconstructed waveform will be something like this right

So, more frequent is your sampling your reconstructed signal will be looking more like your original waveform, right. This is what you should remember.

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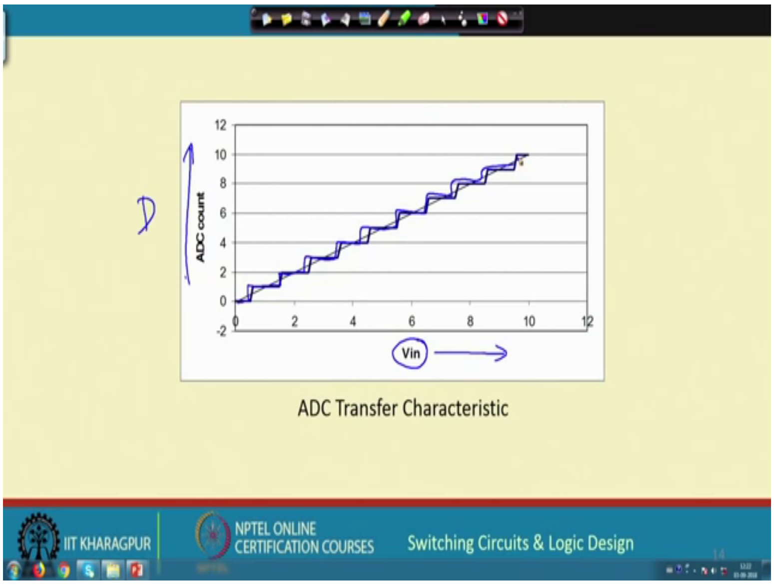

And, now that you are trying to design an analog to digital converter your ADC transfer characteristic will be looking something like this; means along the x axis you can plot your input voltage which is analog which is continuous. Let say from 0 to 12 volts and in the output side you will be potting the digital value let say from 0 these are the digital value. So, even if you are applying your inputs continuously the output value can only be discrete they will be some discrete levels. So, they will be some jumps 0 to 1 2 3 4 again this kind of a staircase kind of a waveform will be there, right.

So, this we shall be discussing in detail later when you talk about the design of AD converter circuits. Now, the design of AD converter circuits will be starting to discuss from the next lecture onwards. So, in this lecture we come to the end where we talked about a very important facet to analog to digital conversion that of sampling the input signal and holding it to a stable value while the conversion is in progress.

And, also you should be aware of the frequency components of the input signal in order to take a decision that how fast to sample. You cannot do this blindly, if you do this blindly then such aliasing and other problems I talked about that can occur which may mislead the sample values that you are converting. Because, ultimately this sampled values will be failing to your digital circuit or to some other sub system where some computation will be carried out. Now, if the data you are feeding are not consistent then the computation may also be wrong.

So, I talked about several different types of AD converter from the next lecture, we shall be discussing the designs of those AD conversions techniques.

Thank you.