**Mechatronics**
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**Lecture – 18**
**Analog to Digital Converters**

I welcome you all to this NPTEL online certification course on Mechatronics. Today we are going to talk about a very important component Analog to Digital Converters. Many of the sensors which we are using in a Mechatronics system give analog output. And the microprocessor or microcontroller which is going to process these signals requires digital input. So, is the need for analog to digital converter. So, in this lecture; I am going to talk about this I hope you will enjoy that.

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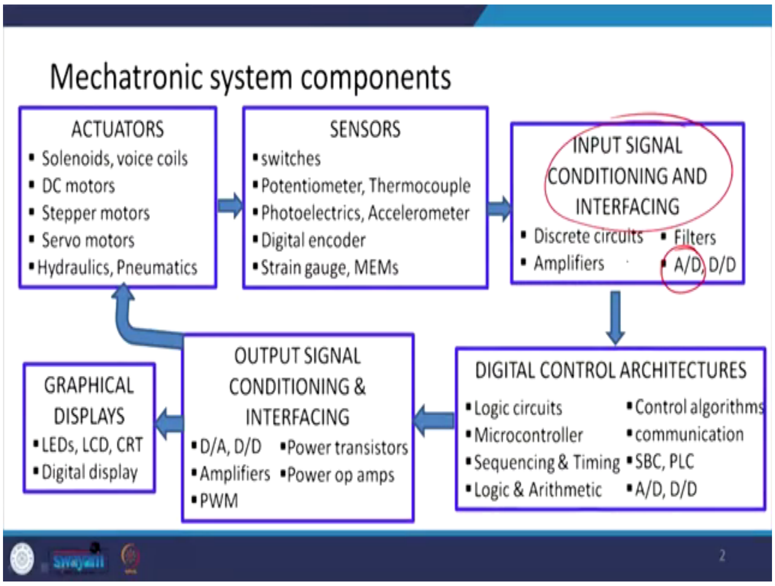


So, here we are at this phase input signal conditioning and interfacing and here I am going to talk about this analog to digital converter.

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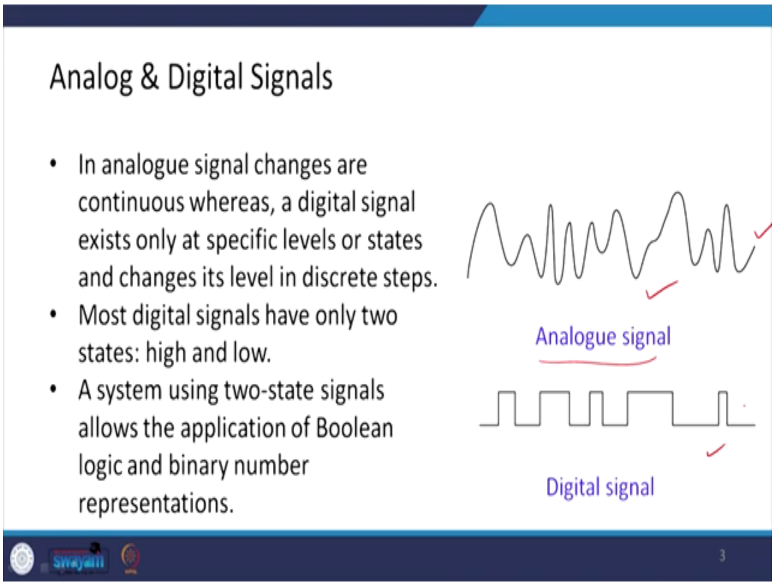


So, see how does the analog signal look like? So, the analog signals are continuous signals and this is how they look like; whereas, the digital signals are they look like this they are either on or off.

So, most of these digital signals have got only 2 states as I said on and off or high and low. A system using 2 states signal allows the application of very powerful logic a very powerful tool called Boolean logic and binary number representation. So I will be talking about these also in my upcoming lectures.

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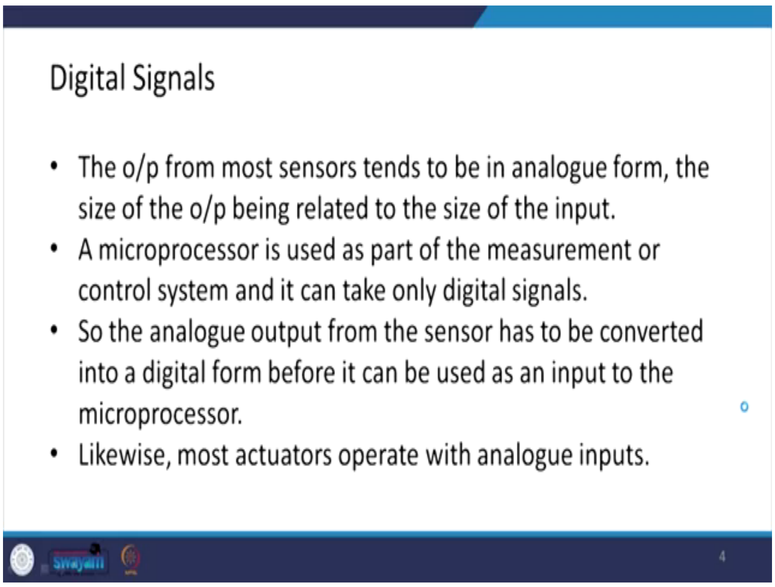

So, let us talk a little about the digital signal, as I said the output from more sensors tends to be analog form, and the size of the output is related to the size of input one way or the other. And I just said a microprocessor is used as a part of measurement or control system and it can take only the digital signal.

So, this sensor signal needs to convert into the digital form before it can be used or it can be processed by the microprocessor. Likewise, most actuators operate with analog input, so for that, we require the conversion from digital to analog and that I am going to talk about in my next lecture.

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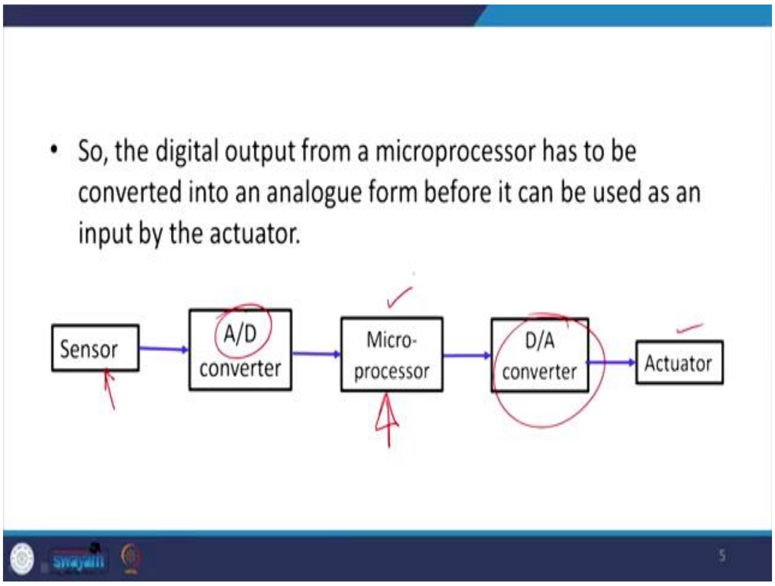


The analog to digital; digital to analog conversion; so, the digital output from microprocessor has to be converted to analog form before it can be used by the actuator as I said and I am going to talk about this component that is digital to analog converter in my next lecture.

So, the signal can be used by the actuator, and here as you see we have the microprocessor and I need the signal for microprocessors to be a digital signal. So, since most of the sensors are analog sensors, I need one analog to digital converter over here, so that the signal can be processed by the microprocessor.

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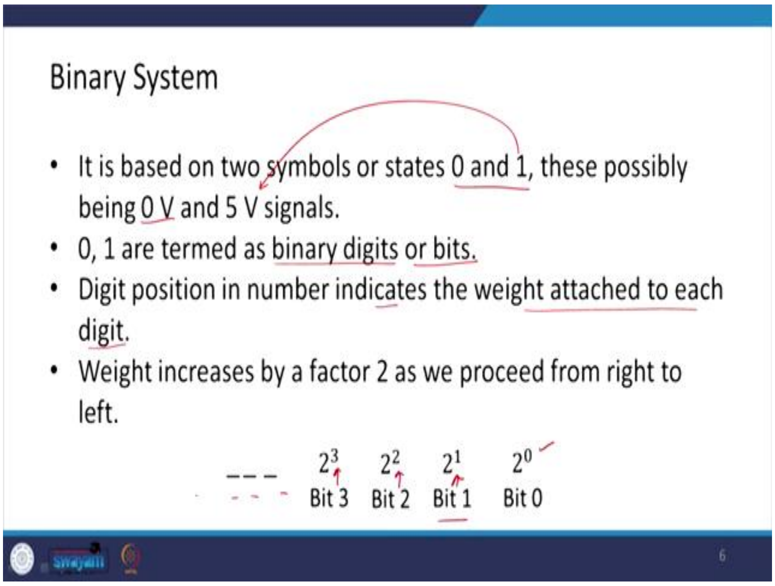


As I said the beautiful thing with the binary with the digital signal is that we can use the binary or Boolean logic in signal processing. So, before we proceed, let us have a little about the binary system so, that we have some understanding about it before we proceed further. The binary system is based on 2 symbols or states that are what we call 0 and 1 and these possibly 0 may correspond to a 0 volt and 1 may correspond to the 5 volts and these 0 and 1 are termed as binary digits or bits. Digit position in a number indicates the weight attached to each digit.

In a binary number, the digit position in number is indicated by how much weight is attached to that, and weight is increased by a factor of 2 as we proceed from right to left. So, this is like this a bit 0 is represented by 2 to the power 0, bit 1 is a 2 to power 1, bit 2 is 2 to power 2 and bit 3 is 2 to power 3, and so on.

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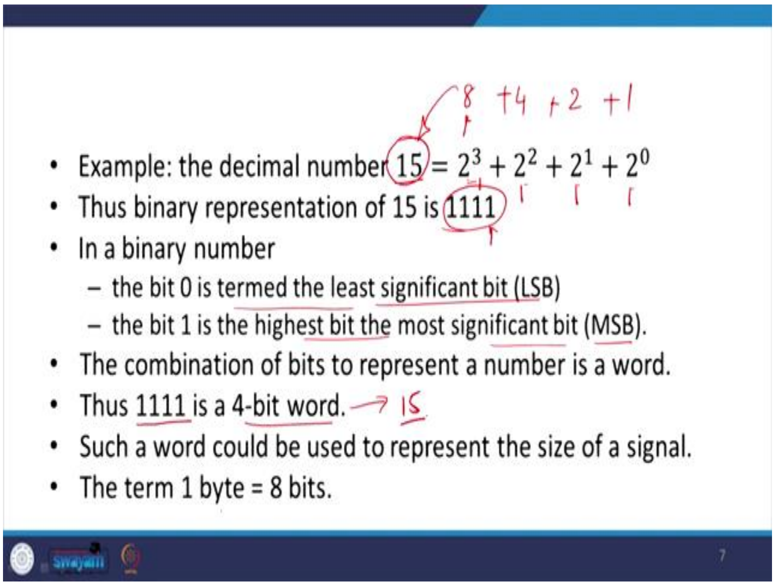


So, let us take an example number 15. This number 15 can be represented by or rather I can write 15 as 2 to power 3 which is nothing but 8, 2 to power 2 which is 4, and 2 to power 1 which is 2, and 2 to power 0 which is 1 so this gives me the 15.

Now, here all these are present. So, we represent the presence of all these by number 1. So, you have 1 here, 1 here, 1 here, so this is the binary representation of 15, that is 1 1 1 1, and the combination of bits represents the volts. So, for 1 1 1 1 is a 4-bit word and this 4-bit word represents a number 15 in the decimal system which as we have seen. So, such a word could be used to represent the size of a signal, and to have a little bigger unit 1 byte is taken equal to 8 bits.

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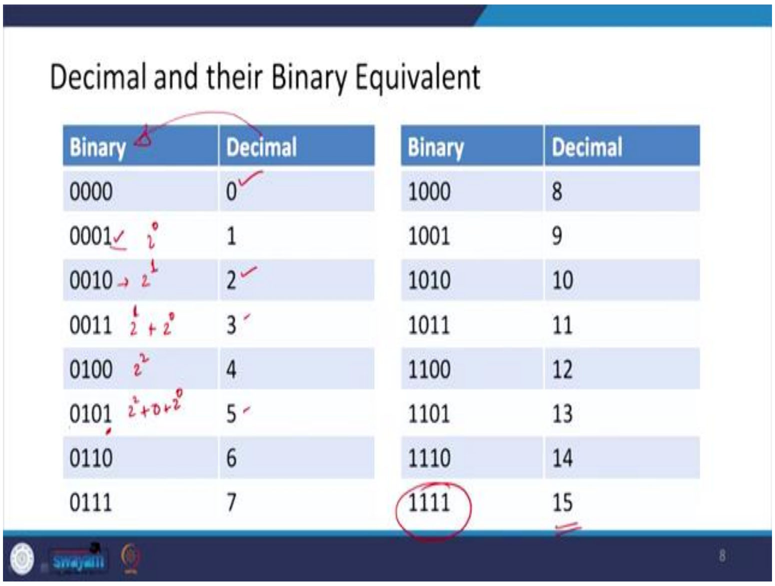


So, this table indicates the decimal and their binary equivalents. So, as I have just talked about 1 1 1 is equal to 15 and so in binary, if I am writing 0 0 0 here that is my all are absent. So, this is 0 0 0 1 is a is equal to 1, so this is what? 2 to the power 0.

And this is what is equal to 2 to the power 1 that is present. So, this is 2 and this is what is 2 to the power 0 plus 2 to the power 1. So, both these are present so this is equal to 3, and here this is what 2 to the power 2, so this is 4, and this is 2 to the power 2 plus 0 plus this is present. So, this is 2 to power 0, so you have 5. So, likewise so likewise we can have established that is we can have a binary representation of a decimal number in a 4 bit presentation like this.

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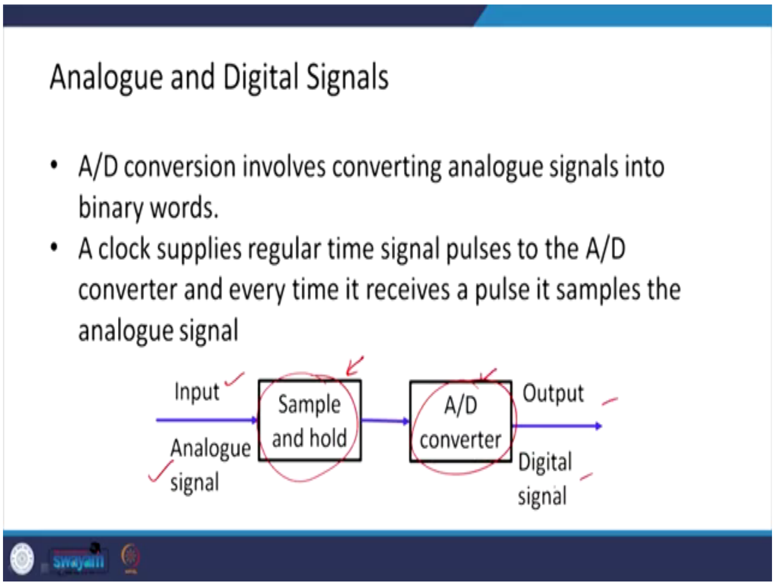


Now, let us look further at Analog and Digital signals. So, analog to digital conversion involves converting an analog signal into a digital signal, and digital signal we know that is nothing in the form of binary words. So, a clock supplies regular time signal pulses to the analog to digital converter, and every time it receives a pulse it samples the analog signal.

So, this analog to digital converter needs and a sample and hold circuit for the sampling purpose. So, you have the input side you have an analog signal that is a sample and hold circuit, for sampling and holding of it and then this is given to an analog to digital converter and you get the output as the digital signal.

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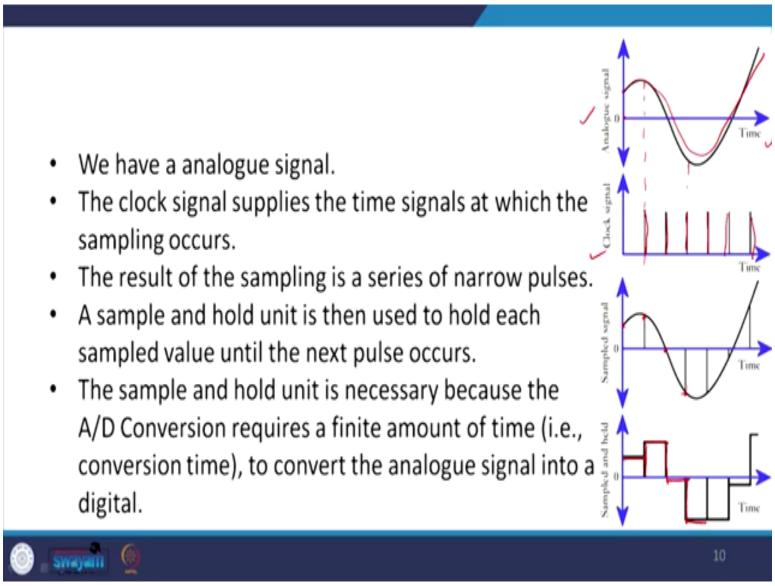


So, this is how it is done. So, suppose this is my analog signal as you can see over here. So, I have plotted analog signal versus time and this is my clock signal. So, I am putting my clock at this time. So, at this time if I am clicking my clock, so this value is being picked up over here and at 0 this value is being picked up and here this value is being picked up, likewise, I am picking up the values and at this time I am putting up a clock. So, this value is being picked up like this.

Now, you see so this is the sampled signal and if I talk about the sample and hold. So, this signal is held till the next ticking comes. So, this signal is held up to here and then at this time interval another value will be held and it will be this value. So, this value is held up to this duration, and then here it comes to 0, so the 0 will be held and it will be coming over here; the same 0 is held up to here and until unless this is held.

So, this way this signal is being sampled and holds the analog signal. So, as a result of the sampling, we get this one, and of this sample and hold unit is necessary because analog to digital conversion requires a finite amount of time that is conversion time to convert the analog signal into a digital signal.

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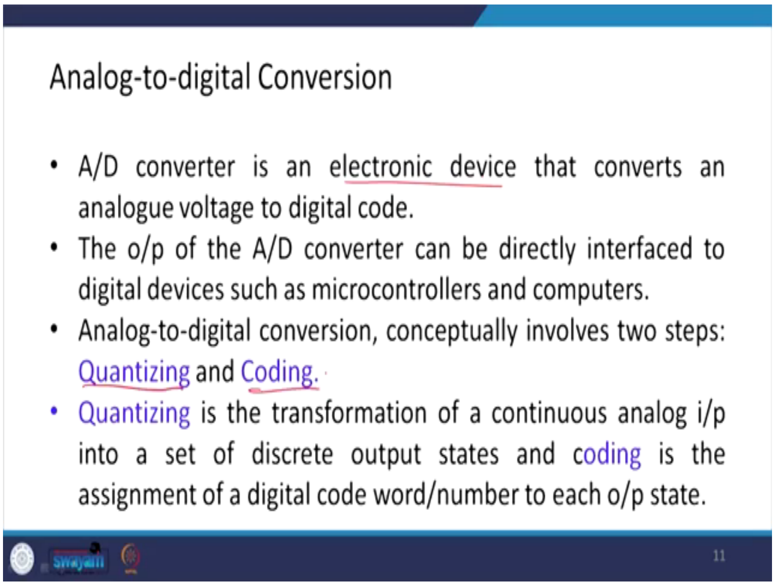


So, looking at analog to digital conversion; as I said analog to digital converter is an electronic device that converts the analog voltage into a digital code, the output of the A to D converter can be directly interfaced to the digital devices such as microcontroller and computers.

And analog to digital conversion conceptually involves 2 steps, what we call quantizing and coding. So, that is first you have to quantize the signal and then you have to give a code to that particular quantized value. So, the quantizing and the coding involved in the conversion of the analog signal into a digital signal. So, what is quantizing? Quantizing is the transformation of a continuous analog input into a set of discrete output states and coding is the assignment of a digital code for that particular quantized value.

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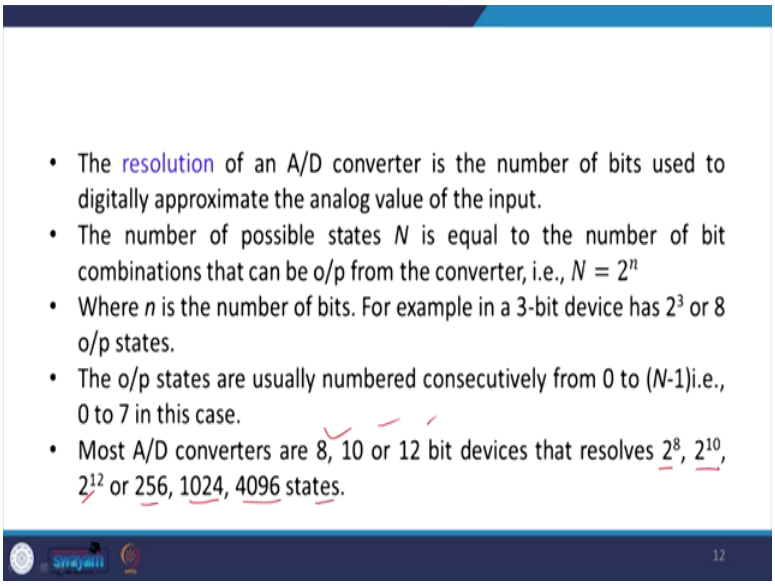


And the resolution of an analog to digital converter is the number of bits used to digitally approximate the analog value of the input. So, the analog value of input you are being represented by how many bits tells you the resolution of the analog to digital converter. So, the number of possible states is equal to the number of bits combinations that can be output from the converter.

So, if there is n number of bits then your number of states is going to be equal to 2 to the power nor the in case of a 3-bit device it is going to be 2 to the power 3 or that is 8 output states. And the output states are usually numbered consequently from 0 to N minus 1 and that is in this case 0 to 7 and in total, we are going to have 8 states. So, most analog to digital converters uses either 8, 10, or 12-bit devices that resolve 2 to the power 8, 2 to power 10, 2 to power 12 or 256, 1024, or 4096 states.

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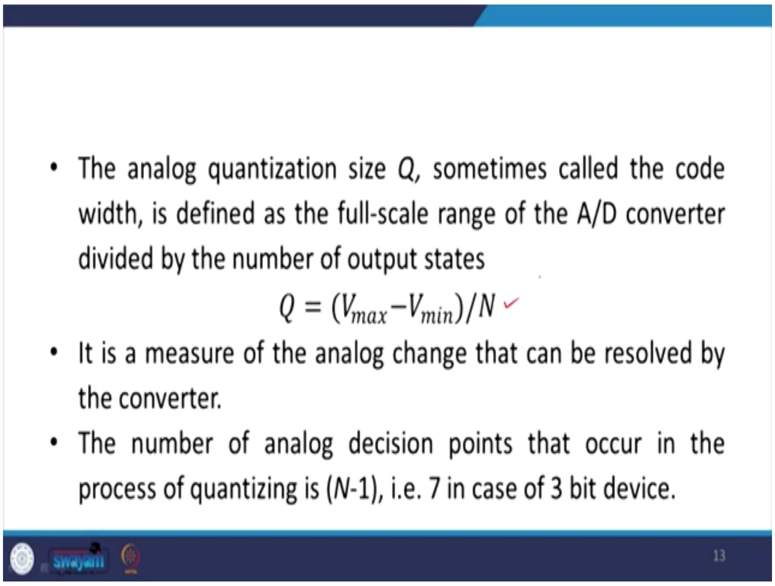


Then the analog quantization size is sometimes called the code width and this is defined as the full-scale range of the A D converter divided by the number of output states. So, it is,

$$Q = (V_{max} - V_{min})/N$$

where N is the number of the output states. It is a measure of the analog change that can be resolved by the converter. So, the number of analog decision points that occur in the process of quantization is (N-1) that is 7 in the case of a 3-bit device.

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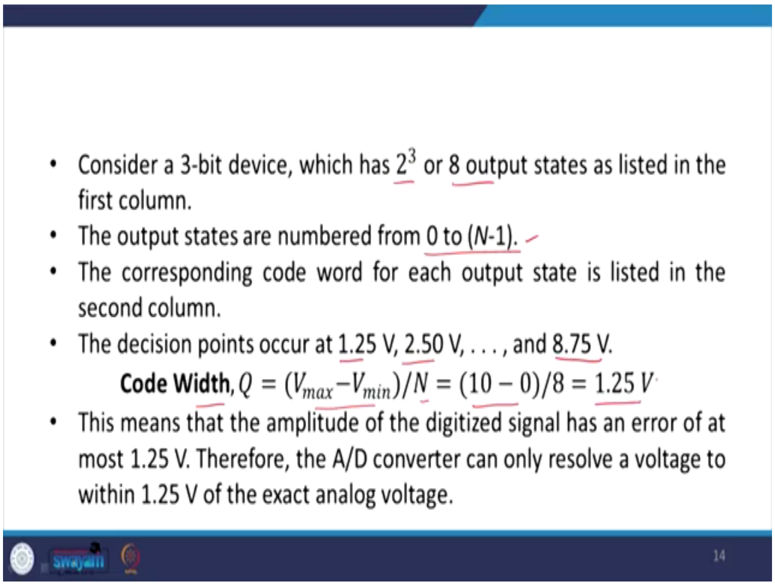

So, consider letting us take a 3-bit device that has got 2 to power 3 or 8 output states as listed I am going to illustrate that. So, the output states are can be numbered from 0 to (N-1), and the corresponding code word for each output state is listed as I am going to show in the next slide.

And these are going to be the decision points and so the code width will be$(V_{max} - V_{min})/N$ that is (10 -0)/8, so this is around 1.25 volts. And this means that the amplitude of the digitized signal has an error of at the most 1.25 volts. So, the A D converter can only resolve a voltage to within 1.25 volt of the next analog voltage.

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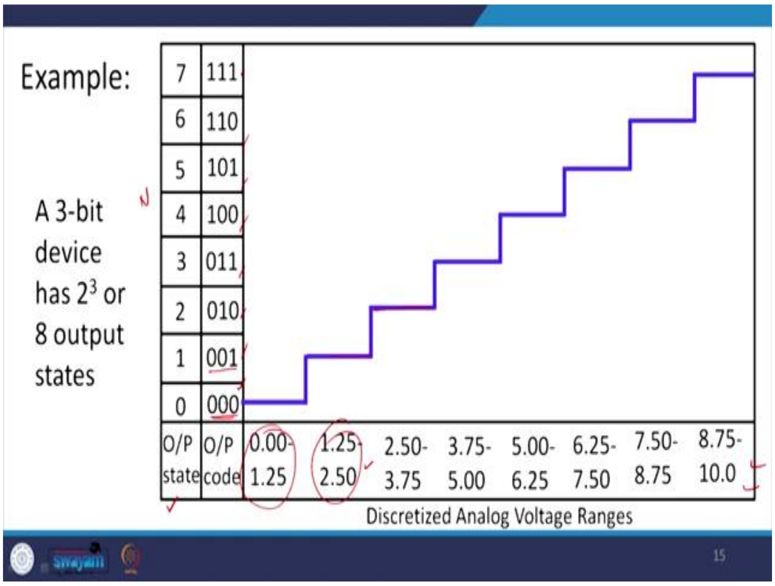


So, here this 10-volt range 0 to 10 divided by 8 gives us 1.25. So, this is 0 to 1.25, 1.25 to 2.5 so, these are going to have and output states are going to be here, this is your capital N which I was talking to you 0 1 2 3 4 5 6 7 8.

And these are the numbers means 0 to 1.25 volts I am representing by a binary number 0 0 0 and 1.25 to 2.5 volt I am representing by binary number 0 0 1 or 2.5 to 3.75 volt I am representing by binary number the 0 1 0. So, this is how this signal analog signal is represented by the digital signal. So, particular voltage range values are given a particular binary representation after the quantization.

So, the question arises that if we are sampling your analog signal, what should be our frequency of sampling, and that criteria have been given by Shannon. So, Shannon sampling theorem helps us or rather gives a guideline that how frequently we should sample our analog signal when we are going to represent it digitally.

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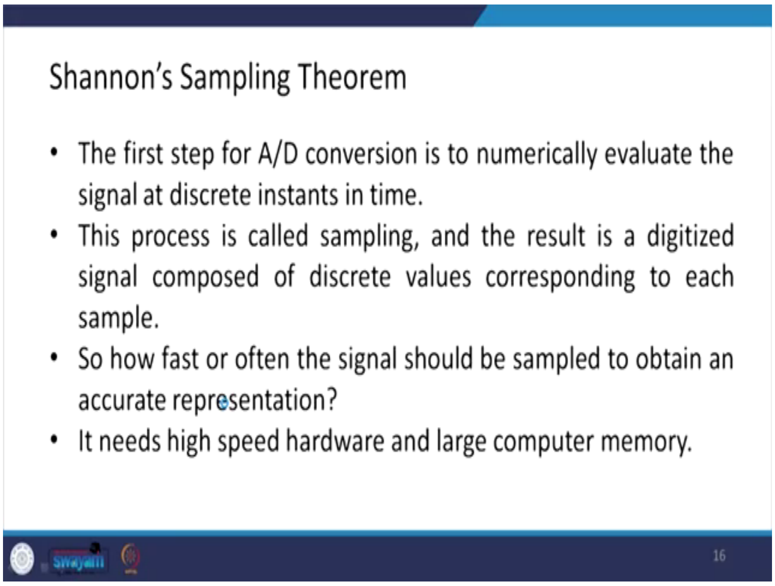


So, the first step for analog to digital conversion is to numerically evaluate the signal at the discrete instant in time and this process is what is called sampling. And the result is a digitized signal composed of the discrete value corresponding to each sample.

Now, how fast or how often the signal should be sampled to obtain an accurate representation, is a very important factor because if it needs high-speed hardware and large computer memory naturally if you are going to do the sampling very often.

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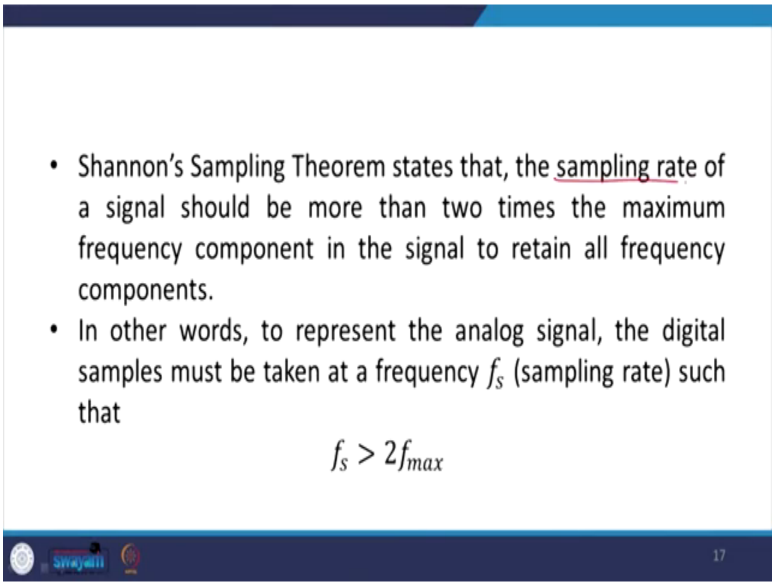


So, the Shannon sampling theorem states that the sampling rate of the signal should be more than 2 times the maximum frequency component in the signal. So, whatever maximum frequency component is there in that signal you have to sample it more than 2 times of that, in order to retain all frequency components. So,

$$f_s > 2f_{max}.$$

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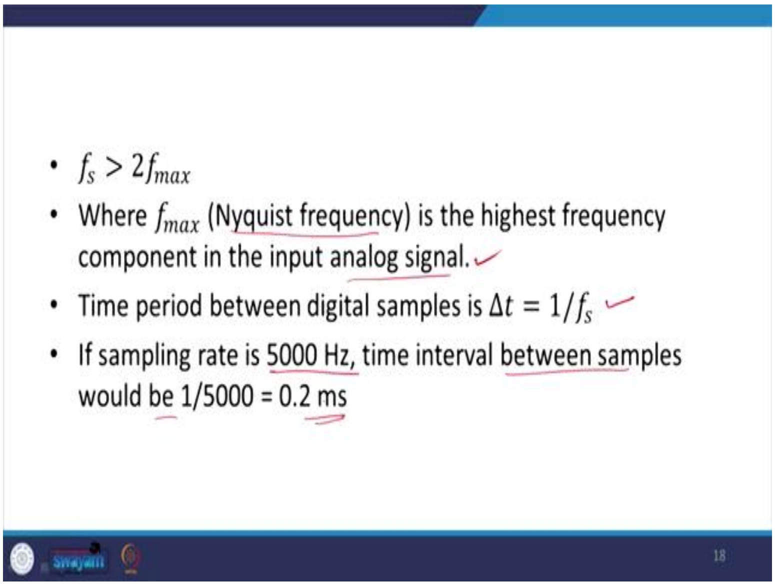

So, where $f_{max}$, or what we are also called Nyquist frequency is the highest frequency component in the input analog signal and the time period between. So, for the digital sample, we could just evaluate by putting it 1/ $f_s$.

So, if the sampling rate is 5000 hertz. What does this mean? This means that the time interval between samples would be 1/ 5000 which is 0.2 milliseconds so you have to do this sampling of your analog signal at the interval of 0.2 milliseconds.

Then let us look at a phenomenon called aliasing.

If a signal is sampled at less than 2 times its maximum frequency component, then aliasing will occur. So, 12 equally spaced samples are taken over 10 cycles of the original signal, therefore the sampling frequency is around $1.2f_o$, where $f_o$ is the frequency of the original sine wave.

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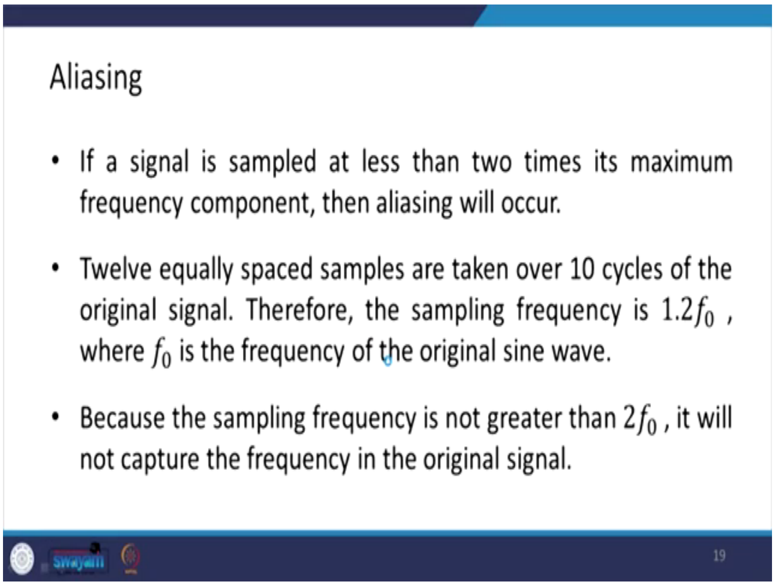


And because the sampling frequency is not greater than $2f_o$, it will not capture the frequency in the original signal.

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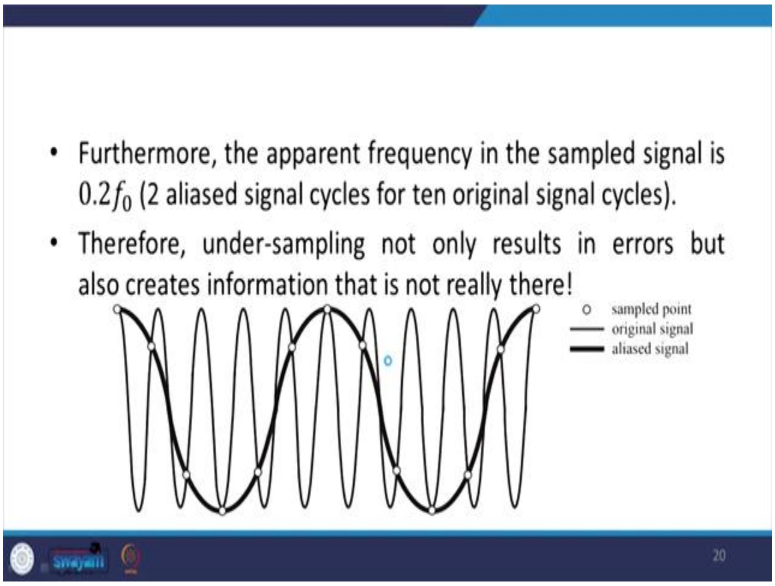


So, the apparent frequency in the sampled signal is $0.2f_o$ that is going to be there or the 2 aliased signal cycles for 10 original signal cycles it is there. And because of this under-sampling not only result in an error but also create information that is not really there. So, here you can see in this figure you have the sample points over here and you have the original signal and you have the aliased signal. So, this signal which you have got by sampling is the alias signal.

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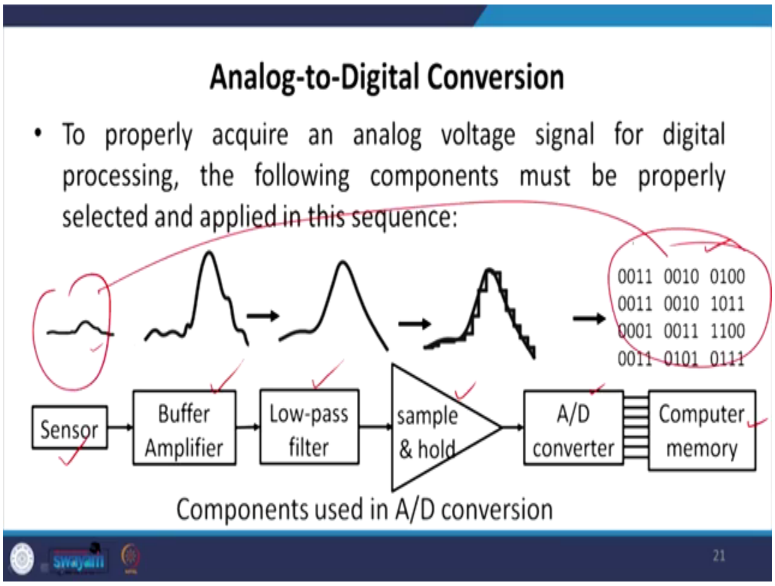

Then to properly do the analog to digital conversion, that is to properly acquire an analog voltage signal for digital processing the following components must be properly selected and applied in this sequence. So for example, the sensor is there which gives an analog signal you need a buffer amplifier in order to amplify that signal then you need a low pass filter and then you need a sample and hold the device and once you have the sample and hold device then you need an A to D converter for that and then, of course, this can go to the computer memory like this. And this way you can convert this your analog signal into a digital signal like this. So, the buffer amplifier isolates the output from the input and provides a signal in the range close to but not exceeding the full input voltage range of the A to D converter.

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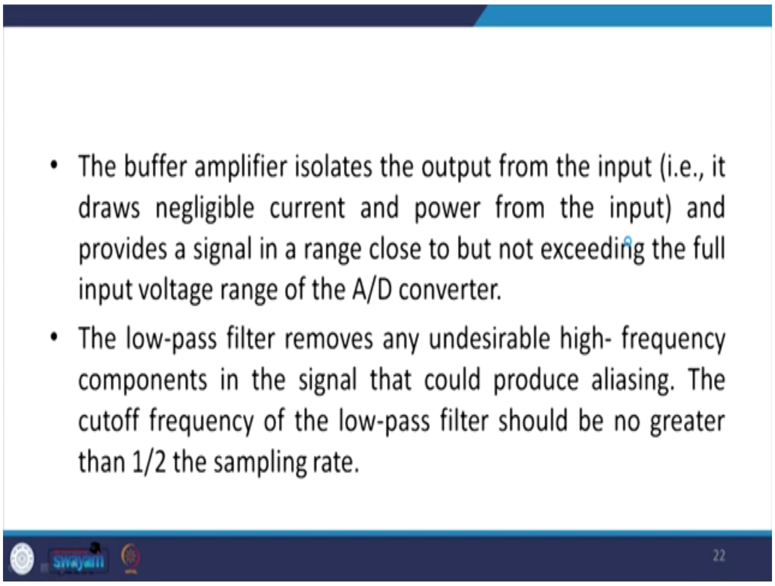


The low pass filter removes any undesirable high-frequency components in the signal that could produce aliasing. Because if you have a high-frequency component and if your sampling frequency is less than 2 times that high-frequency component, you are going to get the aliasing. So, the cutoff frequency of the low pass filters should be no greater than half the sampling rate. The sample and hold amplifier maintains a fixed input value.

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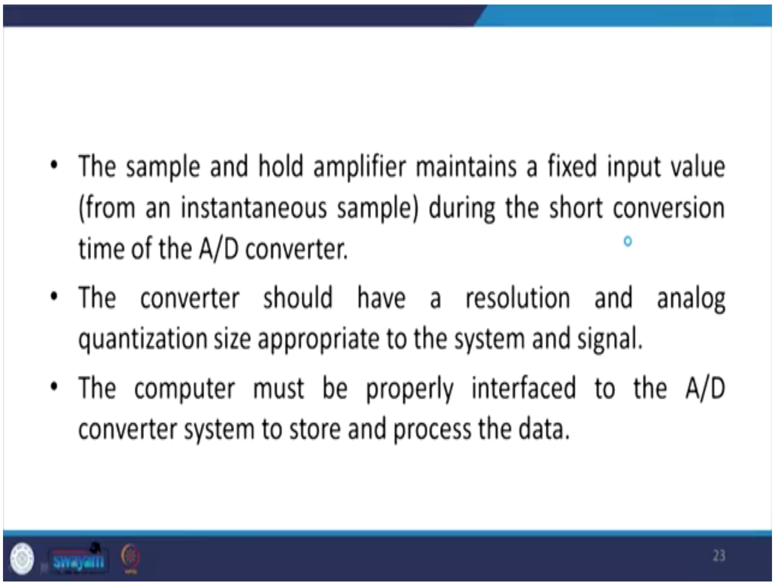


During the short conversion time of the analog to digital converter and the converter should have a resolution and analog quantization size approximate to the system and signal. And the computer must be properly interfaced with the A to D converter system to store and process the data.

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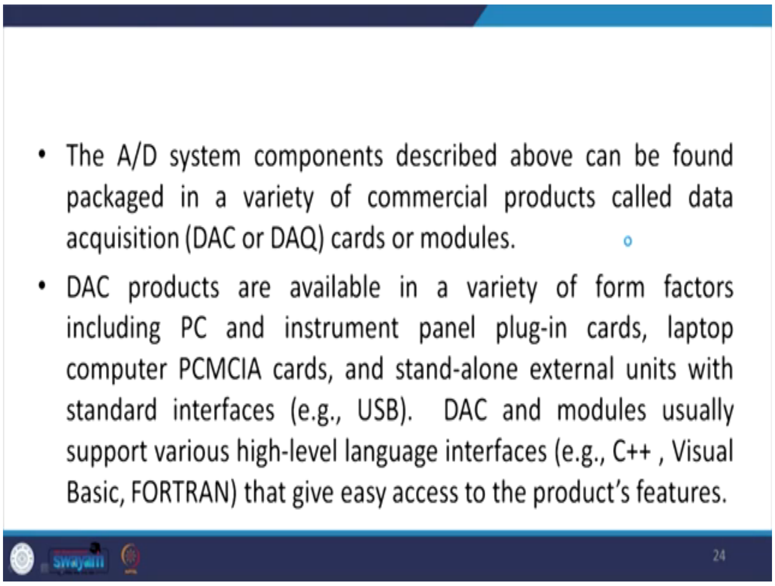


The A to D system components described can be found packaged in a variety of commercial products called data acquisition, DAC, or DAQ cards or modules. And these DAC products are available in a variety of form factors including PC and instrument panel plug-in cards, laptop computers, PCMCIA cards, and stand-alone external units with standard interfaces that are USB alright. A DAC and modules usually support various high-level language interfaces also that is either C plus or visual basic, FORTRAN that gives easy access to the product features.

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So, these are the commercially available product from the national instrument lab view software national instrument are there, and various forms of data acquisition products from the national instruments as you can see over here.

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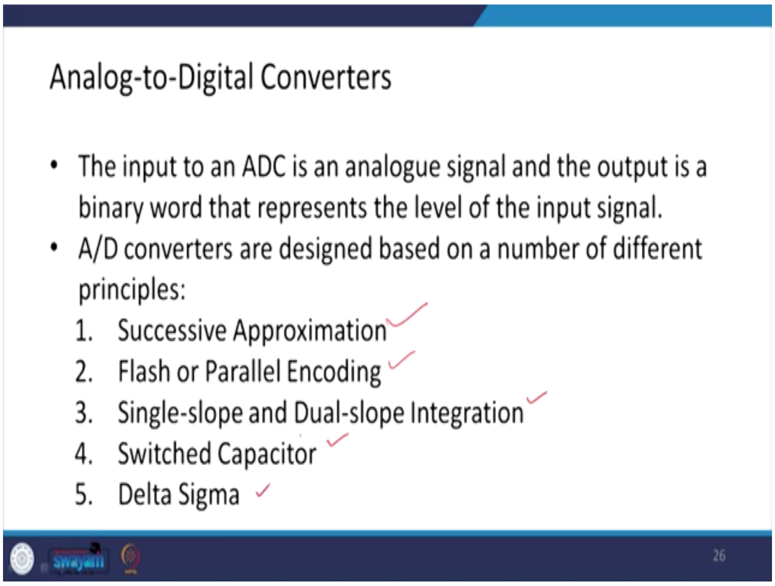

Now, analog to digital conversion, what are the methods and how this is done so, let us try to understand. The input to ADC is an analog signal and the output is the binary word and that represents the level of the input signal, we have already seen by quantizing and coding. These A to D converter converters are designed based on different principles; such as successive approximation, flash or parallel encoding, single slope and dual slope integration, switched capacitor, and sigma delta-sigma. So, I am going to talk about the first method that is the Successive Approximation.

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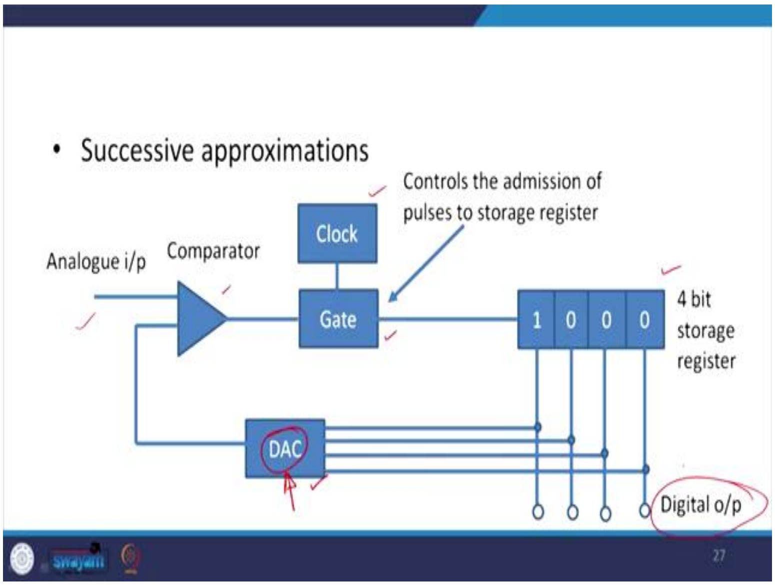


In this case of a successive approximation what is done is that you have an analog input being supplied to a comparator, there is a clock over here and there is a gate and this gate which I will be talking about in my coming lectures. So, this controls the admission of pulses to the storage register, and here is your 4-bit storage register. There is a digital to analog converter this is being used and again as I said; I will be talking about digital to analog converter in my next lecture. So, a digital to analog converter is being used in this method of successive approximation, and from here we get the digital output.

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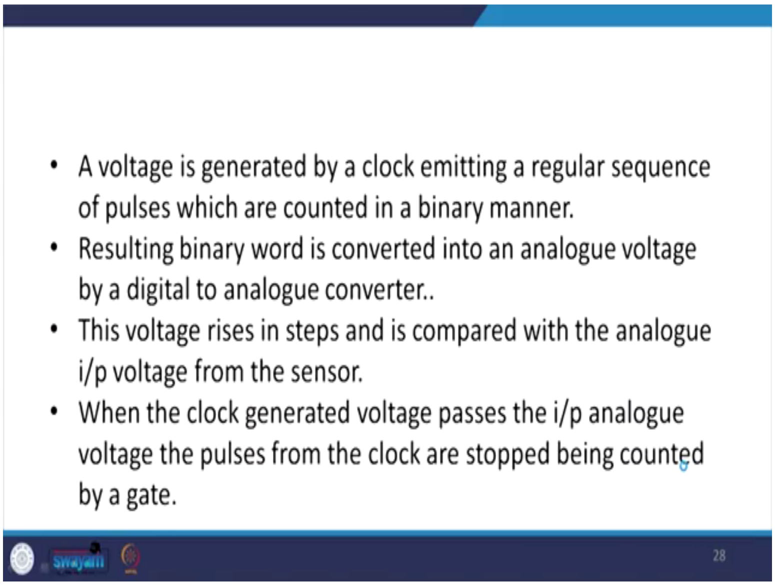


So, how the process takes place is illustrated over here, how this conversion from analog to digital takes place. A voltage is generated by a clock emitting a regular sequence of pulses which are counted in a binary manner and the resulting binary word is converted into an analog voltage by a digital to analog converter. This voltage rises in steps and is compared with the analog input voltage from the sensor and when the clock generated voltage passes the input analog voltage the pulses from clocks are stopped being counted by a gate. And the output of the counter at that time is the digital representation of the voltage.

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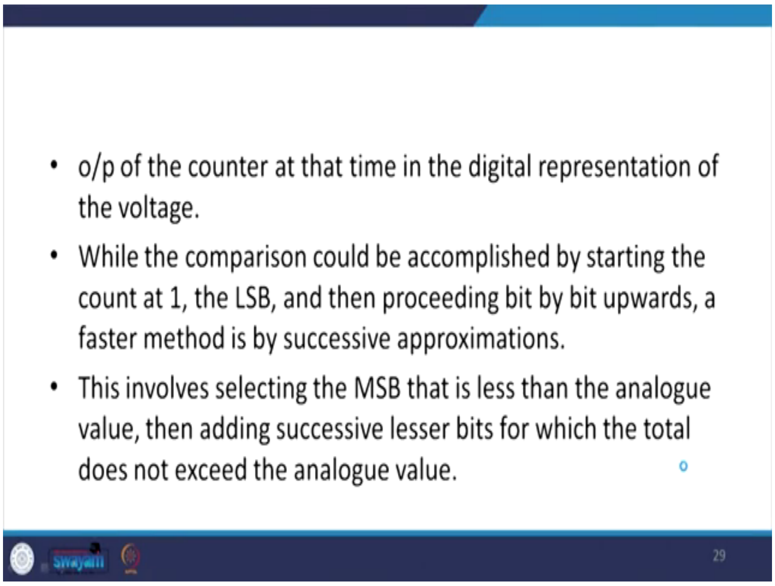

While the comparison could be accomplished by starting the count at 1 the least significant bit and the processing bit by bit upwards a faster method as I said is by this method that is the success approximation. This involves selecting the most significant bit that is less than the analog value and then adding successive lesser bits for which the total does not exceed the analog value.

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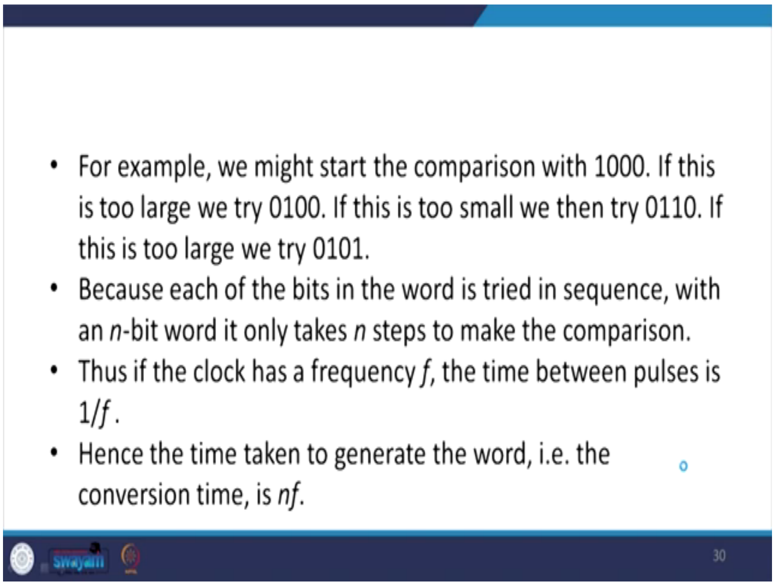


For example, we might start the comparison with 1 0 0 0 this byte and that is with these 4 bits. If this is too large we try with then 0 100 and if this is too small, then we try with 0 11 0 and if this is too large we again try with a different number. Because each of the bits in the word is tried in sequence with an n bit word it only takes n step to make the comparison. So, in this successive approximation, you have an analog signal and you sample that analog signal and give a representation to a digital representation to that analog signal, sampled analog signal and you check and that digital representation is converted into an analog for comparison purposes with the actual analog.

And when the comparison is there and the deviation is not found, then whatever the digital value has been there that digital value is taken as the digital representation of the analog signal. So, this is how this successive approximation works I will again explain you.

So, you have a clock you sample it and after sampling gives a digital representation to that. And this digital representation is also available in the form of output, but whether this output has to be taken or not taken or this is correct or not to check that we take this digital signal and do use digital to analog converter and compare it with the analog input. And I have already explained in my previous lecture about the comparator. When this converted analog signal and the actual analog signal are compared. Then this particular value of the bit or digital signal is selected or taken as the analog that is taken as the digital representation of this analog signal. So, this way you have the digital value of that. So, if the clock has frequency $f$ then the time between pulses is naturally going to be $1/f$. Hence the time is taken to generate the word that is the conversion time is going to be $n f$, where $n$ is the step to make the comparison.

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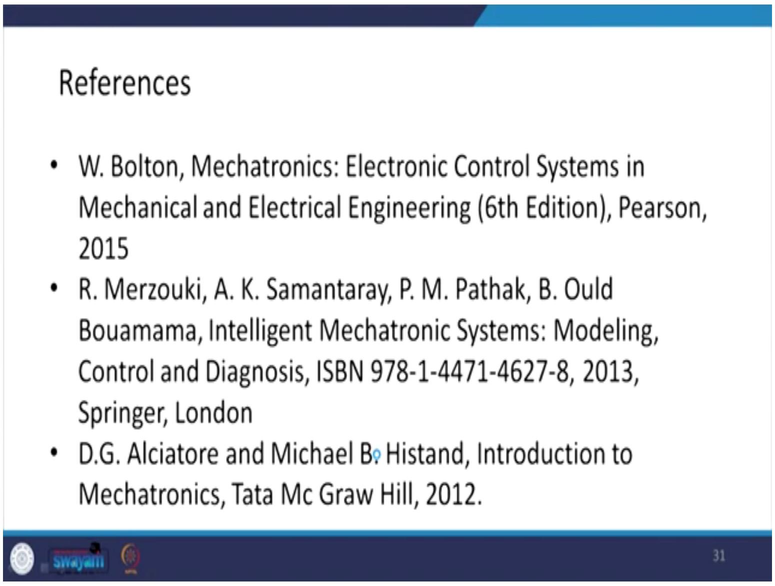


So, these are the further references which you would like to go through for further reading.

Thank you very much.