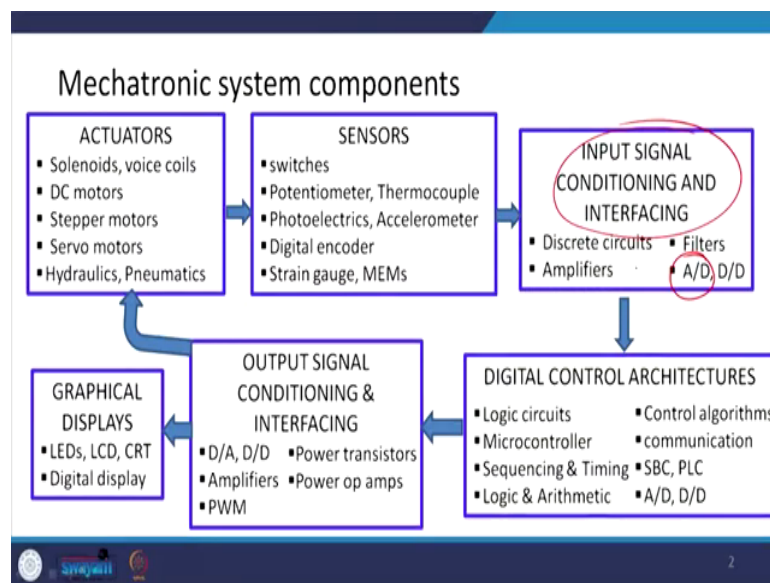


**Mechatronics**  
**Prof. Pushparaj Mani Pathak**  
**Department of Mechanical and Industrial Engineering**  
**Indian Institute of Technology, Roorkee**

**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.

(Refer Slide Time: 01:41)

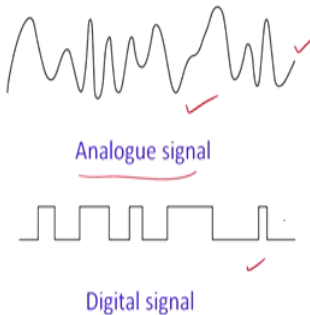


So, here we are at this phase input signal conditioning and interfacing and here I am going to talk about this analog to digital converter.

(Refer Slide Time: 01:46)

### Analog & Digital Signals

- In analogue signal changes are continuous whereas, a digital signal exists only at specific levels or states and changes its level in discrete steps.
- Most digital signals have only two states: high and low.
- A system using two-state signals allows the application of Boolean logic and binary number representations.



Analogue signal

Digital signal

3

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.

(Refer Slide Time: 02:55)

### Digital Signals

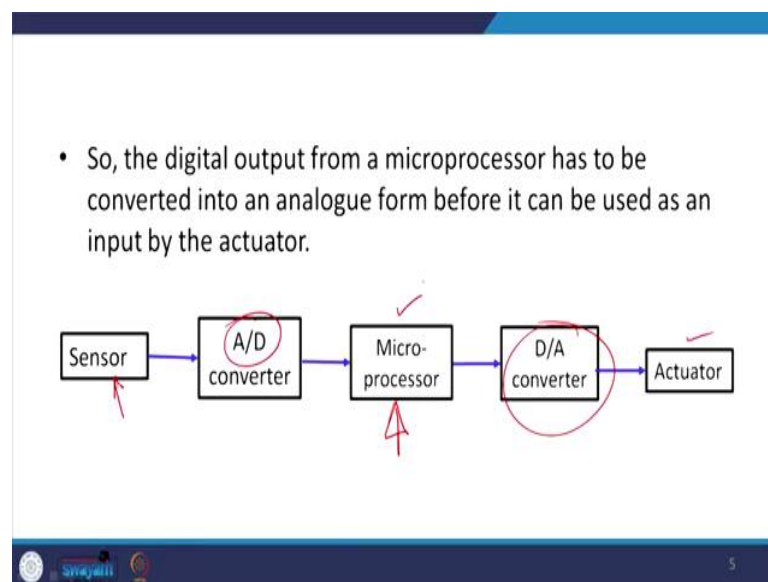
- The o/p from most sensors tends to be in analogue form, the size of the o/p being related to the size of the input.
- A microprocessor is used as part of the measurement or control system and it can take only digital signals.
- So the analogue output from the sensor has to be converted into a digital form before it can be used as an input to the microprocessor.
- Likewise, most actuators operate with analogue inputs.

4

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.

(Refer Slide Time: 03:40)



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.

(Refer Slide Time: 04:36)

**Binary System**

- It is based on two symbols or states 0 and 1, these possibly being 0 V and 5 V signals.
- 0, 1 are termed as binary digits or bits.
- Digit position in number indicates the weight attached to each digit.
- Weight increases by a factor 2 as we proceed from right to left.

Diagram illustrating bit weights:

---	$2^3$	$2^2$	$2^1$	$2^0$
---	Bit 3	Bit 2	Bit 1	Bit 0

The diagram shows four bit positions labeled Bit 3, Bit 2, Bit 1, and Bit 0 from left to right. Above each bit label is its corresponding weight:  $2^3$ ,  $2^2$ ,  $2^1$ , and  $2^0$ . Red arrows point from the bit labels to their respective weights. A red curved line connects the top of the  $2^3$  weight to the top of the  $2^0$  weight, indicating the progression from right to left.

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.

(Refer Slide Time: 06:16)

The slide contains a list of bullet points and a diagram. The diagram shows the decimal number 15 in a circle, with arrows pointing to the powers of 2: 8, 4, 2, and 1. Below this, the equation  $15 = 2^3 + 2^2 + 2^1 + 2^0$  is written. The binary representation 1111 is shown with arrows pointing to each bit, corresponding to the powers of 2 above. The text explains that bit 0 is the least significant bit (LSB) and bit 1 is the highest bit the most significant bit (MSB). It also states that a combination of bits to represent a number is a word, and that 1111 is a 4-bit word representing the number 15. Finally, it notes that 1 byte = 8 bits.

- Example: the decimal number 15 =  $2^3 + 2^2 + 2^1 + 2^0$
- Thus binary representation of 15 is 1111
- In a binary number
  - the bit 0 is termed the least significant bit (LSB)
  - the bit 1 is the highest bit the most significant bit (MSB).
- The combination of bits to represent a number is a word.
- Thus 1111 is a 4-bit word. → 15
- Such a word could be used to represent the size of a signal.
- The term 1 byte = 8 bits.

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.

(Refer Slide Time: 07:50)

Decimal and their Binary Equivalent

Binary	Decimal	Binary	Decimal
0000	0	1000	8
0001	1	1001	9
0010	2	1010	10
0011	3	1011	11
0100	4	1100	12
0101	5	1101	13
0110	6	1110	14
0111	7	1111	15

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.

(Refer Slide Time: 09:23)

### Analogue and Digital Signals

- A/D conversion involves converting analogue signals into binary words.
- A clock supplies regular time signal pulses to the A/D converter and every time it receives a pulse it samples the analogue signal

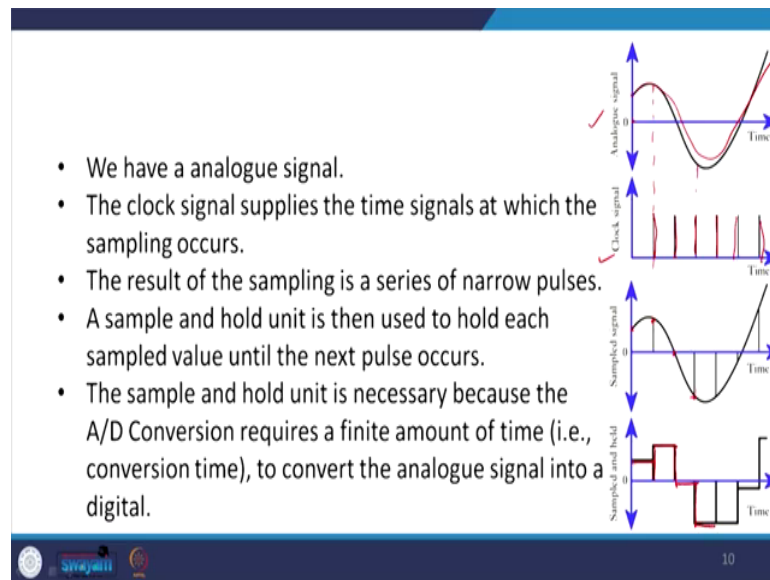
```
graph LR; Input[Input Analogue signal] --> SH[Sample and hold]; SH --> ADC[A/D converter]; ADC --> Output[Output Digital signal];
```

The diagram shows a block diagram of an A/D conversion process. It consists of two main blocks: 'Sample and hold' and 'A/D converter'. An 'Analogue signal' enters from the left into the 'Sample and hold' block. The output of the 'Sample and hold' block goes into the 'A/D converter' block. The output of the 'A/D converter' block is a 'Digital signal'. There are red checkmarks and arrows indicating the flow of the signal.

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.

(Refer Slide Time: 10:31)



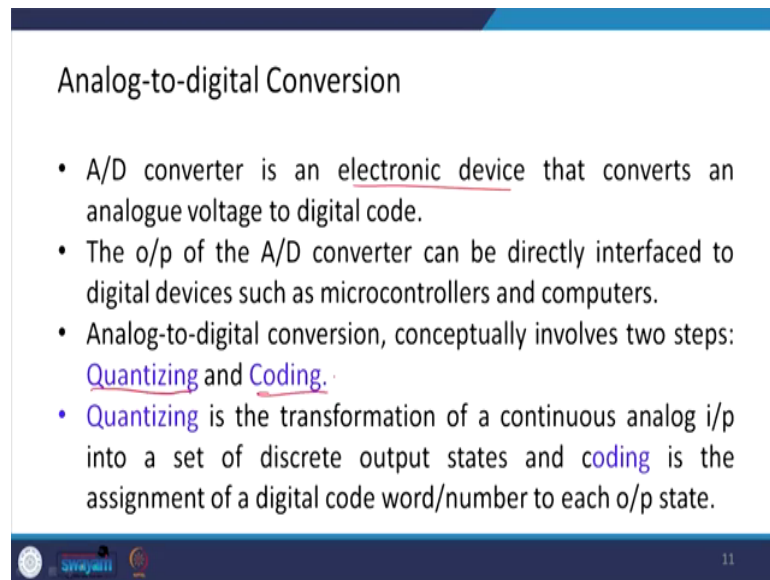
So, this is how it is done. So, suppose this is my analog signal as you can see over here. So, I have plotted analog signal versus time and this is my clock signal. So, I am putting my clock at this time. So, at this time if I am clicking my clock, so this value is being picked up over here and at 0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



(Refer Slide Time: 12:51)



The slide is titled "Analog-to-digital Conversion" and contains the following text:

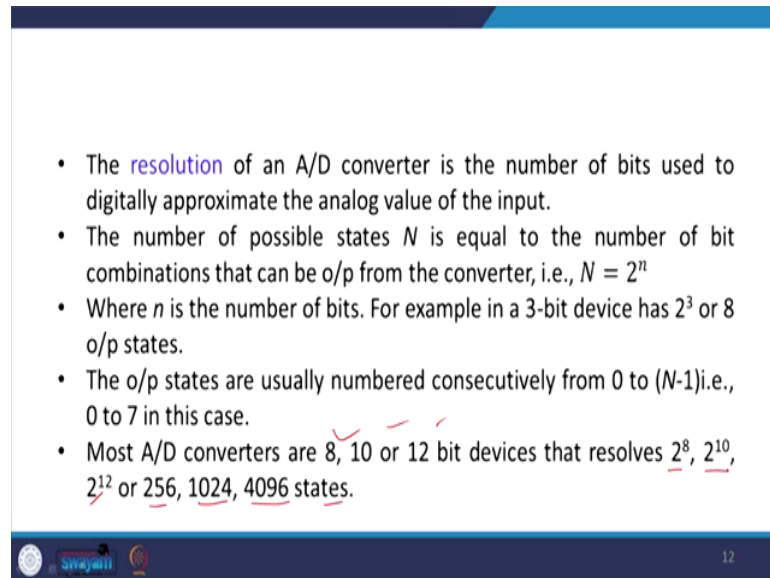
- A/D converter is an electronic device that converts an analogue voltage to digital code.
- The o/p of the A/D converter can be directly interfaced to digital devices such as microcontrollers and computers.
- Analog-to-digital conversion, conceptually involves two steps: Quantizing and Coding.
- Quantizing is the transformation of a continuous analog i/p into a set of discrete output states and coding is the assignment of a digital code word/number to each o/p state.

At the bottom of the slide, there are logos for "Sri Jayati" and "11".

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.

(Refer Slide Time: 14:00)

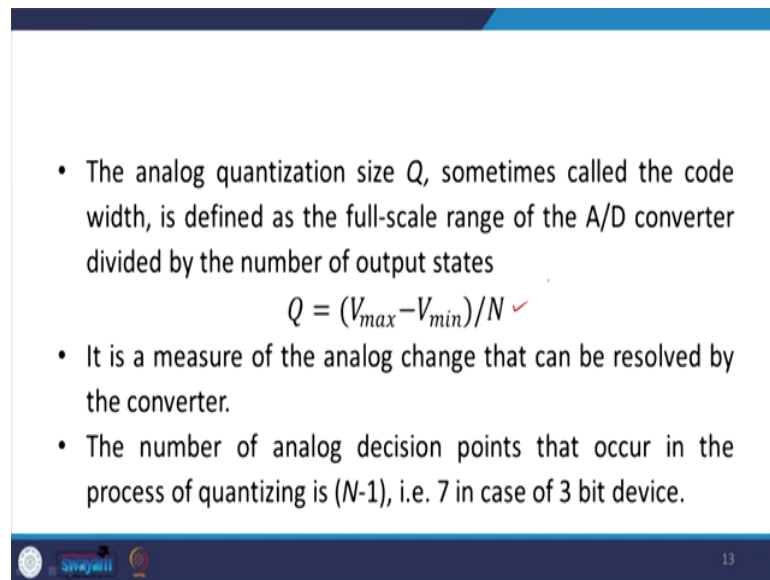


- The **resolution** of an A/D converter is the number of bits used to digitally approximate the analog value of the input.
- The number of possible states  $N$  is equal to the number of bit combinations that can be o/p from the converter, i.e.,  $N = 2^n$
- Where  $n$  is the number of bits. For example in a 3-bit device has  $2^3$  or 8 o/p states.
- The o/p states are usually numbered consecutively from 0 to  $(N-1)$  i.e., 0 to 7 in this case.
- Most A/D converters are 8, 10 or 12 bit devices that resolves  $2^8$ ,  $2^{10}$ ,  $2^{12}$  or 256, 1024, 4096 states.

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  $n$  or in the 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 use 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.

(Refer Slide Time: 15:39)



• The analog quantization size  $Q$ , sometimes called the code width, is defined as the full-scale range of the A/D converter divided by the number of output states

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

• It is a measure of the analog change that can be resolved by the converter.

• The number of analog decision points that occur in the process of quantizing is  $(N-1)$ , i.e. 7 in case of 3 bit device.

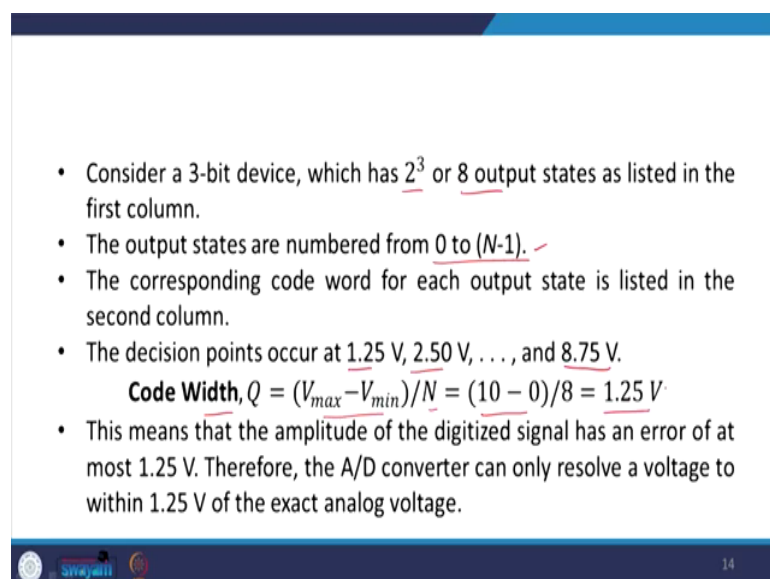
13

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.

(Refer Slide Time: 16:24)



• Consider a 3-bit device, which has  $2^3$  or 8 output states as listed in the first column.

• The output states are numbered from 0 to  $(N-1)$ .

• The corresponding code word for each output state is listed in the second column.

• The decision points occur at 1.25 V, 2.50 V, . . . , and 8.75 V.

**Code Width,  $Q = (V_{max} - V_{min}) / N = (10 - 0) / 8 = 1.25 V$**

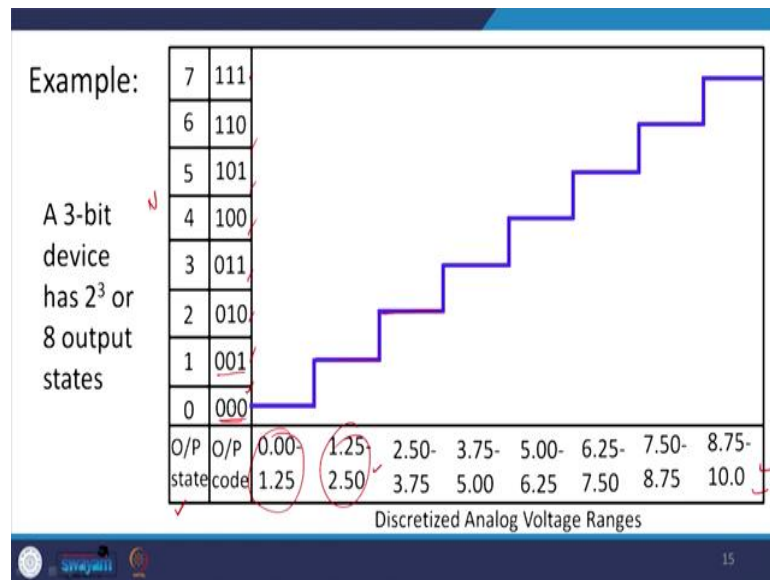
• This means that the amplitude of the digitized signal has an error of at most 1.25 V. Therefore, the A/D converter can only resolve a voltage to within 1.25 V of the exact analog voltage.

14

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.

(Refer Slide Time: 17:36)



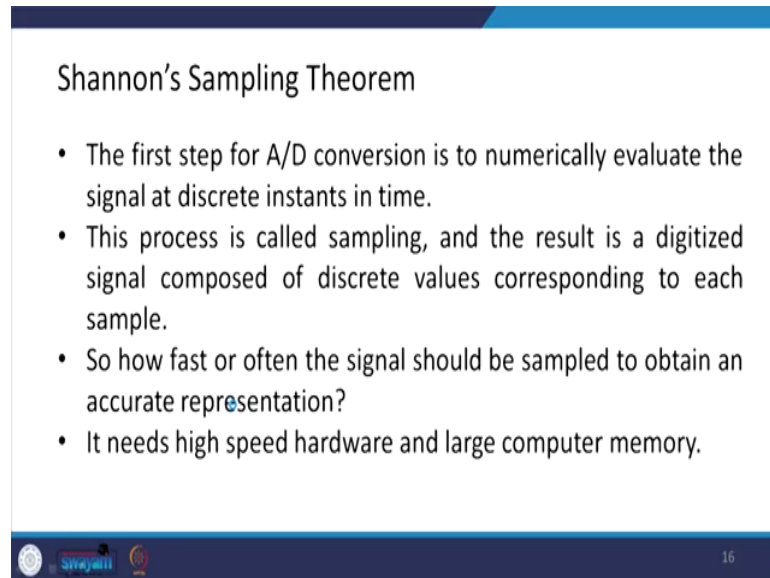
So, here this 10-volt range 0 to 10 divided by 8 gives us 1.25. So, this is 0 to 1.25, 1.25 to 2.5 so, these are going to have and output states are going to be here, this is your capital N which I was talking to you 0 1 2 3 4 5 6 7 8.

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.

(Refer Slide Time: 19:40)



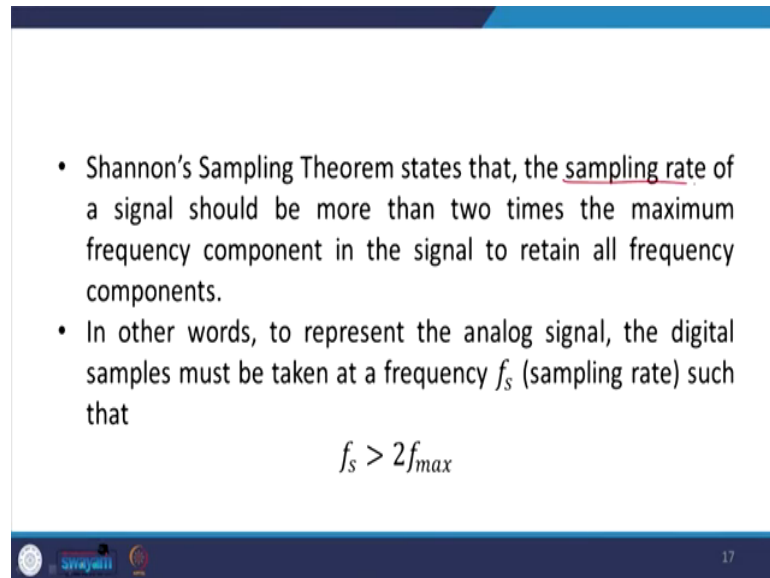
Shannon's Sampling Theorem

- The first step for A/D conversion is to numerically evaluate the signal at discrete instants in time.
- This process is called sampling, and the result is a digitized signal composed of discrete values corresponding to each sample.
- So how fast or often the signal should be sampled to obtain an accurate representation?
- It needs high speed hardware and large computer memory.

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.

(Refer Slide Time: 20:31)



• Shannon's Sampling Theorem states that, the sampling rate of a signal should be more than two times the maximum frequency component in the signal to retain all frequency components.

• In other words, to represent the analog signal, the digital samples must be taken at a frequency  $f_s$  (sampling rate) such that

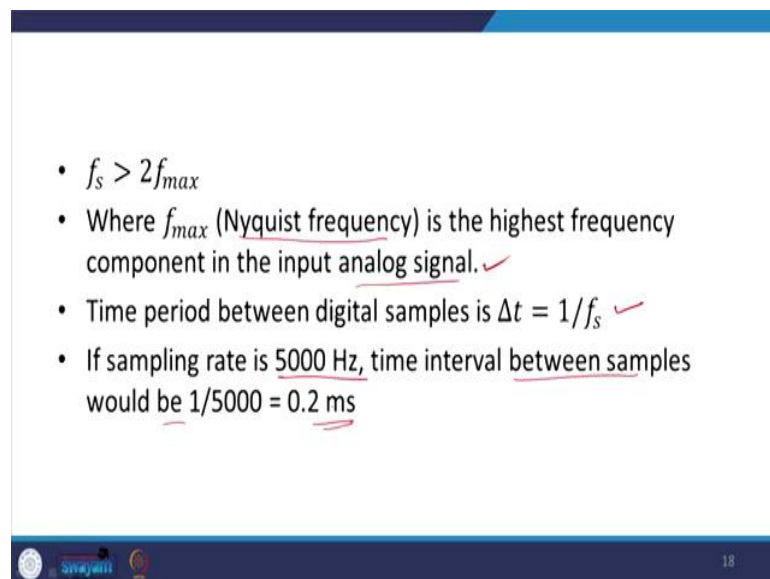
$$f_s > 2f_{max}$$

17

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,

$$f_s > 2f_{max}$$

(Refer Slide Time: 21:08)



•  $f_s > 2f_{max}$

• Where  $f_{max}$  (Nyquist frequency) is the highest frequency component in the input analog signal. ✓

• Time period between digital samples is  $\Delta t = 1/f_s$ . ✓

• If sampling rate is 5000 Hz, time interval between samples would be  $1/5000 = 0.2$  ms

18

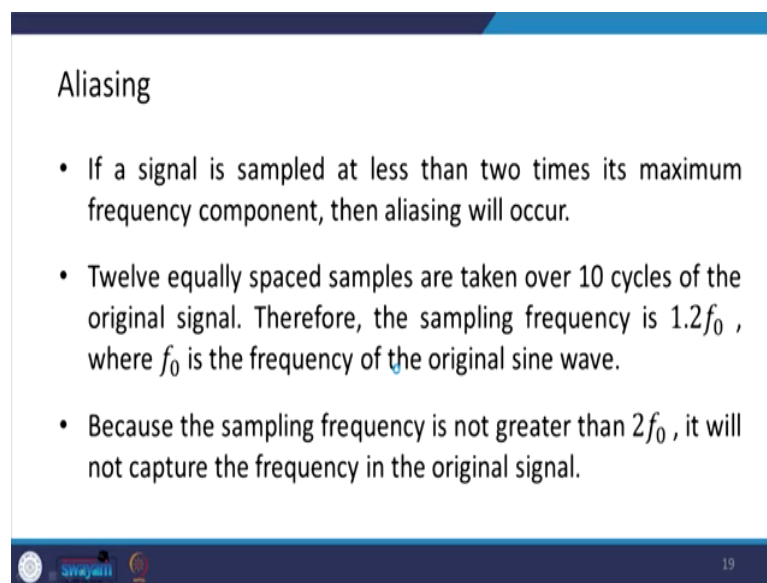
So, where  $f_{max}$ , or what we are also called Nyquist frequency is the highest frequency component in the input analog signal and the time period between. So, for the digital sample, we could just evaluate by putting it  $1/f_s$ .

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.

(Refer Slide Time: 22:33)



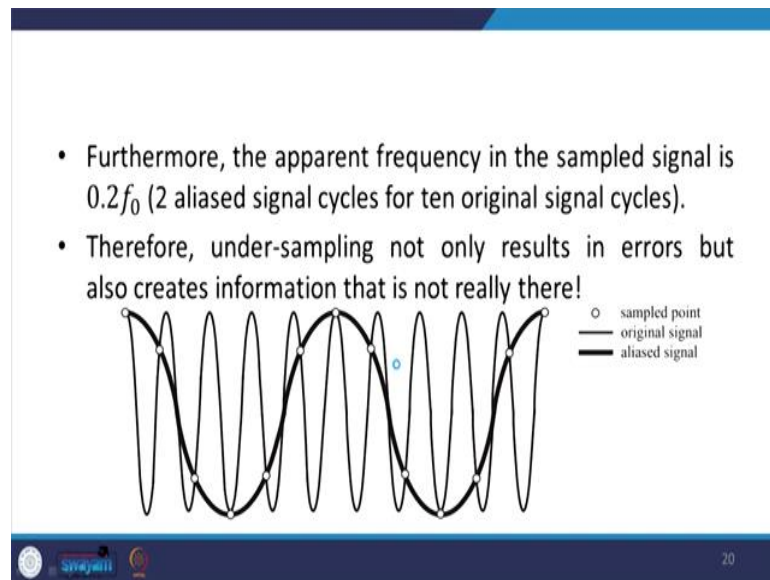
The slide is titled "Aliasing" and contains three bullet points. The first bullet point states that if a signal is sampled at less than two times its maximum frequency component, aliasing will occur. The second bullet point provides an example: twelve equally spaced samples are taken over 10 cycles of the original signal, resulting in a sampling frequency of  $1.2f_o$ , where  $f_o$  is the frequency of the original sine wave. The third bullet point concludes that because the sampling frequency is not greater than  $2f_o$ , the original signal's frequency cannot be captured.

Aliasing

- If a signal is sampled at less than two times its maximum frequency component, then aliasing will occur.
- Twelve equally spaced samples are taken over 10 cycles of the original signal. Therefore, the sampling frequency is  $1.2f_o$ , where  $f_o$  is the frequency of the original sine wave.
- Because the sampling frequency is not greater than  $2f_o$ , it will not capture the frequency in the original signal.

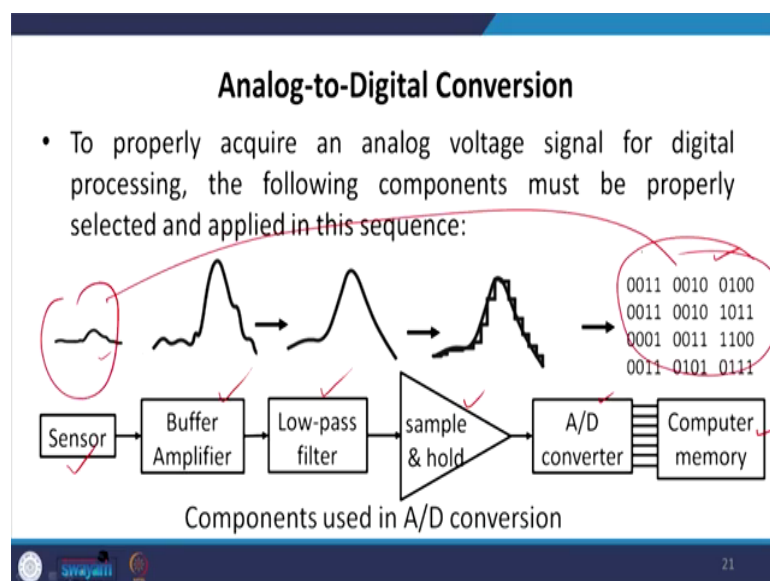
And because the sampling frequency is not greater than  $2f_o$ , it will not capture the frequency in the original signal.

(Refer Slide Time: 22:47)



So, the apparent frequency in the sampled signal is  $0.2f_0$  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.

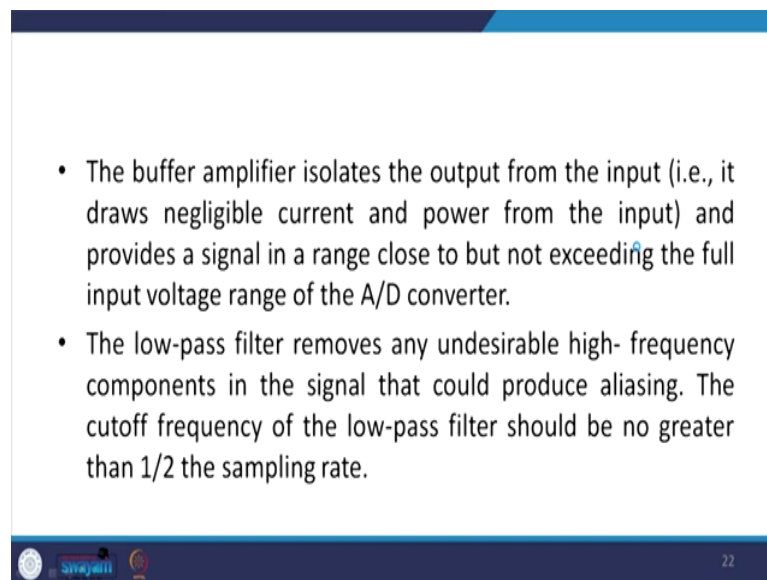
(Refer Slide Time: 23:45)





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.

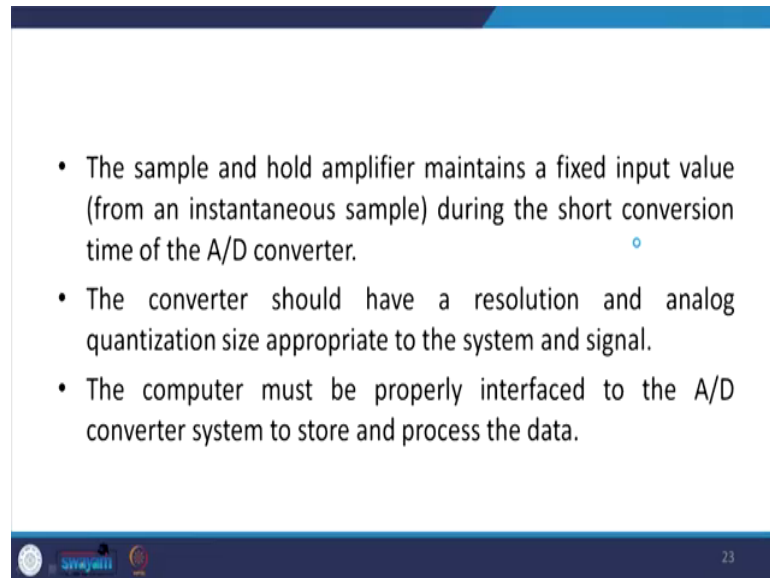
(Refer Slide Time: 25:04)



- The buffer amplifier isolates the output from the input (i.e., it draws negligible current and power from the input) and provides a signal in a range close to but not exceeding the full input voltage range of the A/D converter.
- The low-pass filter removes any undesirable high-frequency components in the signal that could produce aliasing. The cutoff frequency of the low-pass filter should be no greater than  $1/2$  the sampling rate.

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.

(Refer Slide Time: 25:48)



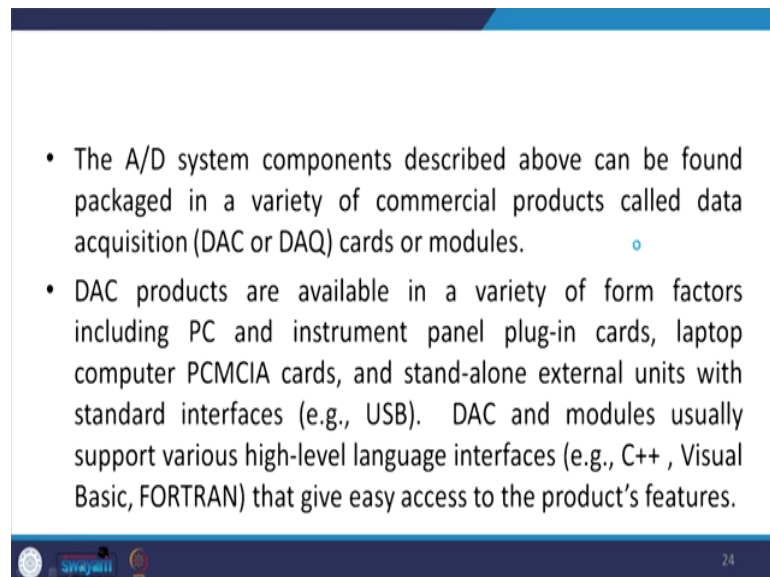
Slide 23 contains three bullet points:

- The sample and hold amplifier maintains a fixed input value (from an instantaneous sample) during the short conversion time of the A/D converter.
- The converter should have a resolution and analog quantization size appropriate to the system and signal.
- The computer must be properly interfaced to the A/D converter system to store and process the data.

The slide footer includes the Swayam logo and the number 23.

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.

(Refer Slide Time: 26:10)



Slide 24 contains two bullet points:

- The A/D system components described above can be found packaged in a variety of commercial products called data acquisition (DAC or DAQ) cards or modules.
- DAC products are available in a variety of form factors including PC and instrument panel plug-in cards, laptop computer PCMCIA cards, and stand-alone external units with standard interfaces (e.g., USB). DAC and modules usually support various high-level language interfaces (e.g., C++ , Visual Basic, FORTRAN) that give easy access to the product's features.

The slide footer includes the Swayam logo and the number 24.

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

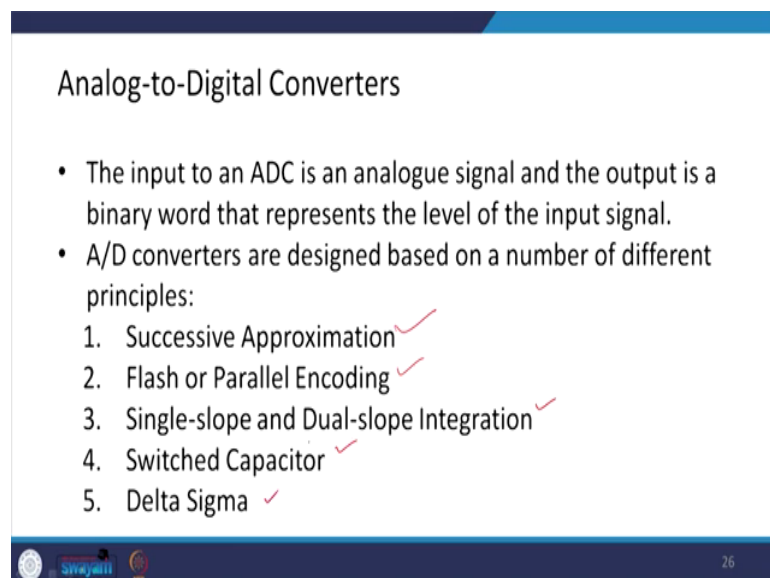
plug-in cards, laptop computers, PCMCIA cards, and stand-alone external units with standard interfaces that are USB alright. A DAC and modules usually support various high-level language interfaces also that is either C plus or visual basic, FORTRAN that gives easy access to the product features.

(Refer Slide Time: 27:04)



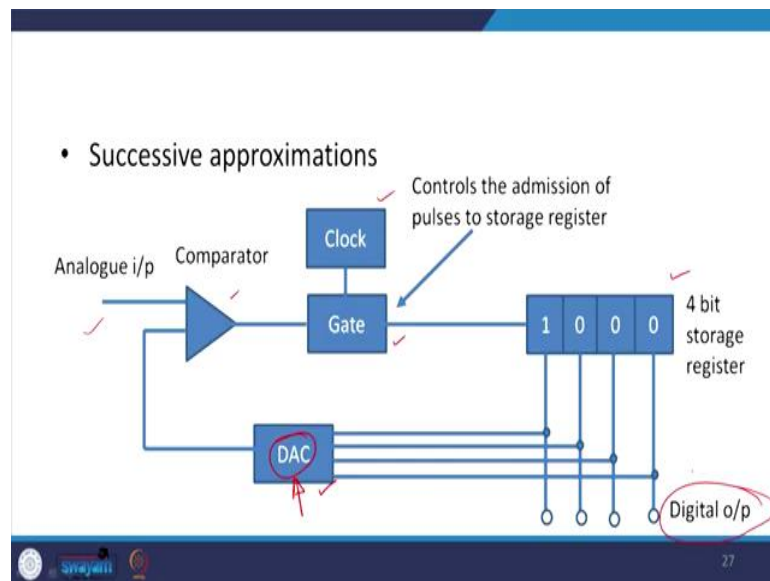
So, these are the commercially available product from the national instrument lab view software national instrument are there, and various forms of data acquisition products from the national instruments as you can see over here.

(Refer Slide Time: 27:25)



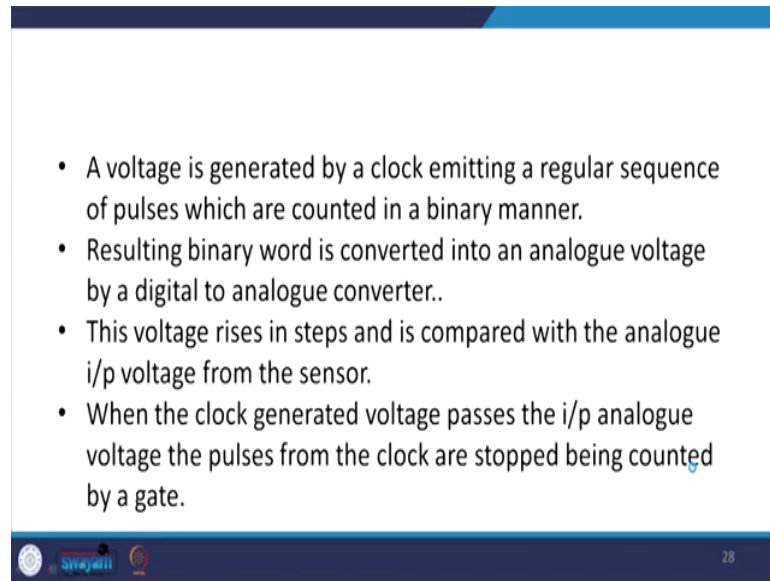
Now, analog to digital conversion, what are the methods and how this is done so, let us try to understand. The input to ADC is an analog signal and the output is the binary word and that represents the level of the input signal, we have already seen by quantizing and coding. These A to D converter converters 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.

(Refer Slide Time: 28:26)



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.

(Refer Slide Time: 29:24)

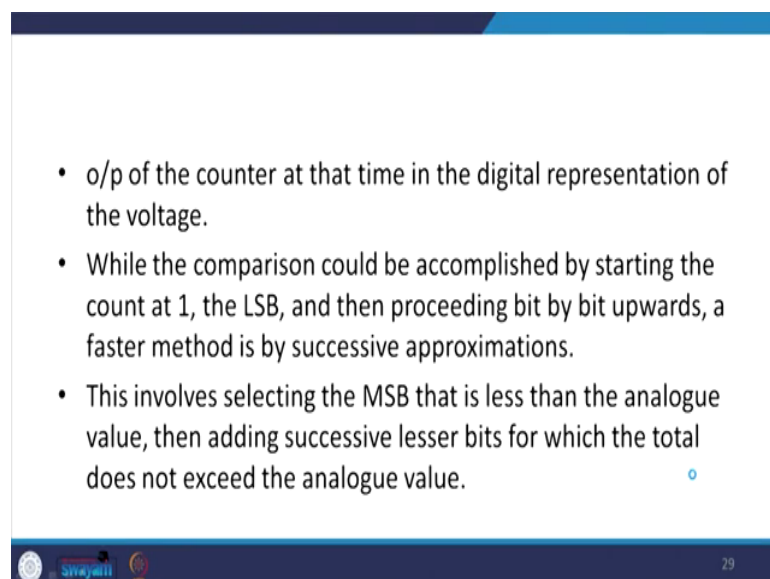


- A voltage is generated by a clock emitting a regular sequence of pulses which are counted in a binary manner.
- Resulting binary word is converted into an analogue voltage by a digital to analogue converter..
- This voltage rises in steps and is compared with the analogue i/p voltage from the sensor.
- When the clock generated voltage passes the i/p analogue voltage the pulses from the clock are stopped being counted by a gate.

28

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.

(Refer Slide Time: 30:42)

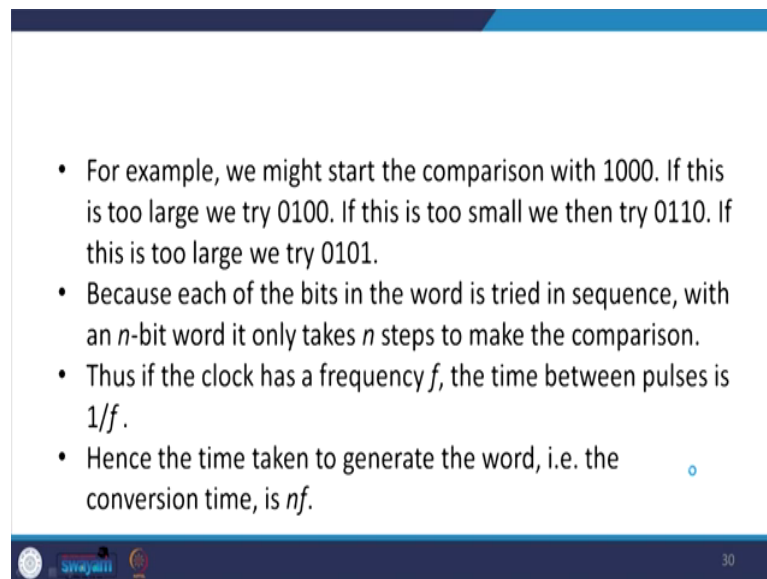


- o/p of the counter at that time in the digital representation of the voltage.
- While the comparison could be accomplished by starting the count at 1, the LSB, and then proceeding bit by bit upwards, a faster method is by successive approximations.
- This involves selecting the MSB that is less than the analogue value, then adding successive lesser bits for which the total does not exceed the analogue value.

29

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.

(Refer Slide Time: 31:13)



- For example, we might start the comparison with 1000. If this is too large we try 0100. If this is too small we then try 0110. If this is too large we try 0101.
- Because each of the bits in the word is tried in sequence, with an  $n$ -bit word it only takes  $n$  steps to make the comparison.
- Thus if the clock has a frequency  $f$ , the time between pulses is  $1/f$ .
- Hence the time taken to generate the word, i.e. the conversion time, is  $nf$ .

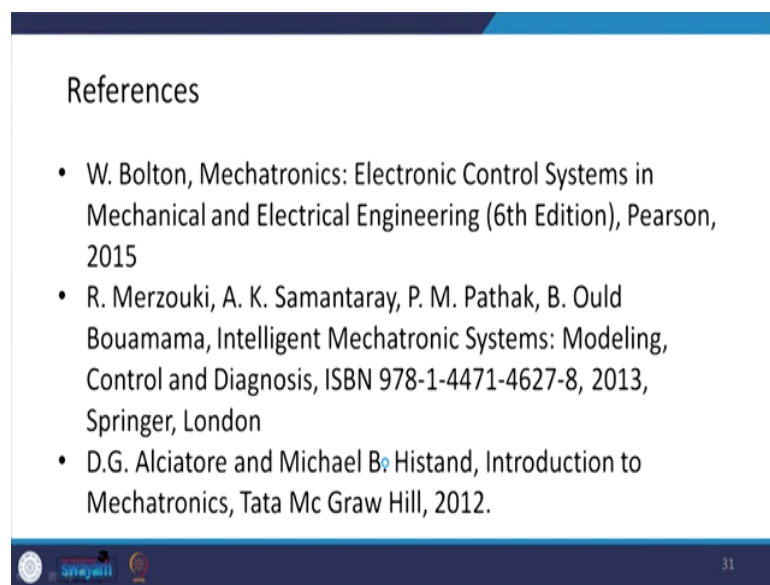
For example, we might start the comparison with 1 0 0 0 this byte and that is with these 4 bits. If this is too large we try with then 0 100 and if this is too small, then we try with 0 11 0 and if this is too large we again try with a different number. Because each of the bits in the word is tried in 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.

(Refer Slide Time: 34:51)



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

Thank you very much.