Data Communication
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Lecture No # 8
Transmission of Digital Signal - II

Hello viewers welcome to today's lecture on transmission of digital signal.

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This is the second lecture on this topic. In this lecture we shall cover the following topics.

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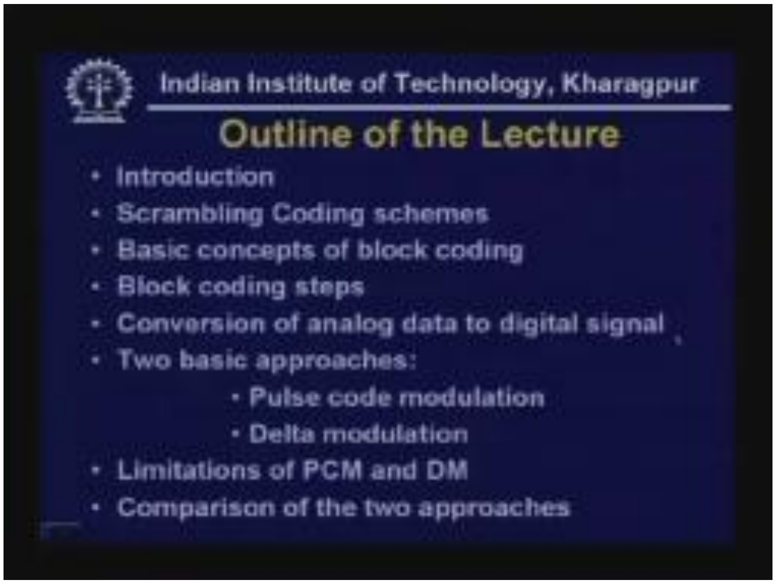

First I shall give a brief introduction this will be followed by scrambling coding schemes then we shall discuss basic concepts of block coding, block coding steps, conversion of analog data to digital signal which will be a different type then two basic approaches of analog data to digital signal that is Pulse Code Modulation and Delta Modulation then we shall discuss limitations of Pulse Code Modulation and Delta Modulation and compare these two approaches.

And on completion of this lecture the students will be able to explain scrambling coding schemes, explain the need for block coding, explain the operation of block coding, the students will be able to explain the coding techniques used for conversion of analog data to digital signal they will be able to distinguish between the coding techniques such as PCM and DM which are used for coding from analog data to digital signal that is conversion of analog data to digital signal and then they will be able to compare the advantages and limitations of Pulse Code Modulation and Delta Modulation.

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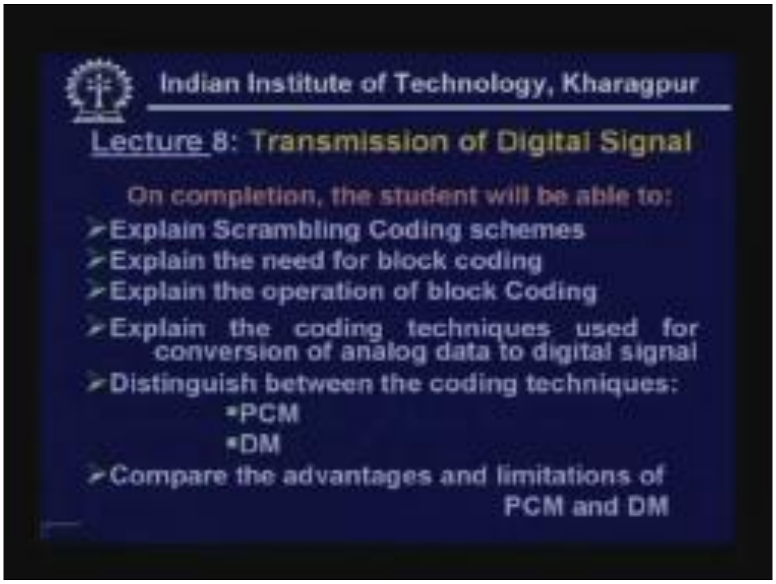

Here is the summary of what we discussed in the last lecture. In the last lecture we started our discussion of transmission of digital signal. Essentially we discussed various schemes for conversion of digital data to digital signal. So here is the summary. And as you know the conversion technique is known as line coding. They can be classified into three broad categories; the unipolar, polar and bipolar. Unipolar as you know is not really used but the polar scheme has the number of different types like NRZ, RZ, Manchester and differential Manchester. All of these coding techniques we have discussed in detail.

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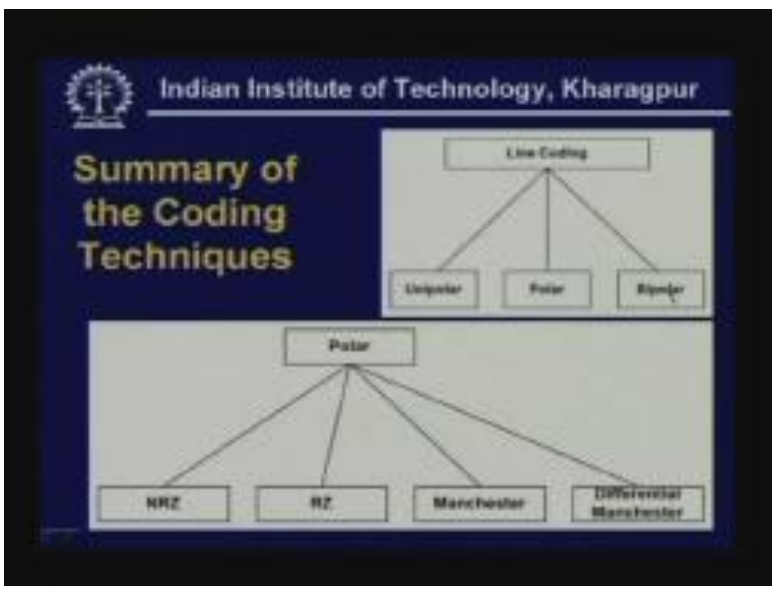

We have also discussed one particular technique of bipolar encoding that is your amplitude AMI Amplitude Mark Inversion.

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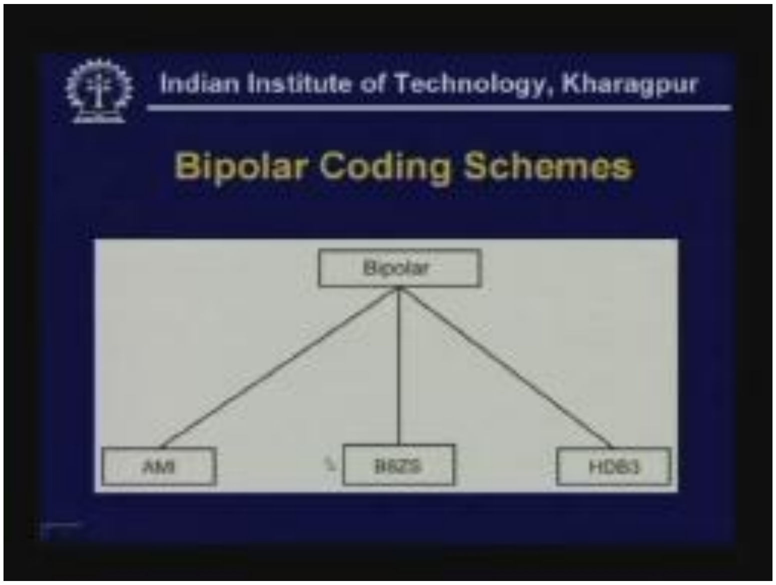

Let us pick up where do we left in the last lecture. I mentioned that although AMI is a good scheme it has some limitation because it does not provide you synchronization although it is very good from the view point of bandwidth it does not have a large bandwidth, the bandwidth is not high and that's why it is attractive from the bandwidth point of view but it does not provide good synchronization. So how this problem can be overcome retaining the advantage or good feature of bandwidth, that is what is being tried in B8ZS which stands for Bipolar with 8 Zero Substitution.

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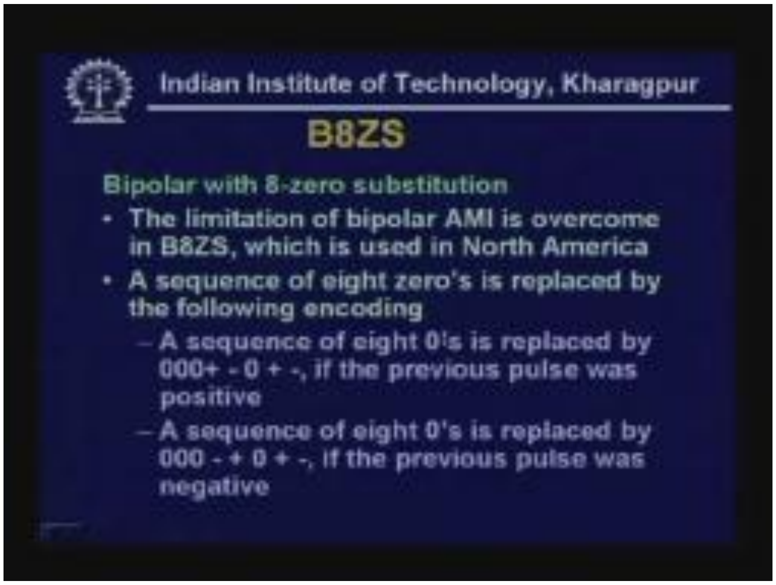

As I mentioned the limitation of bipolar AMI is overcome in B8ZS which is particularly used in North America. As we have seen the main limitation of amplitude mark inversion was that whenever you have a long sequence of zeros there is lack of synchronization. Synchronization fails because there is no signal transition whenever you have a long sequence of zeros. Today we shall discuss about how that can be overcome.

What is done in this particular case is whenever you have got 8 zeros that is being replaced by the following encoding. Whenever you have a sequence of 8 zeros it is replaced by 0 0 0 plus minus 0 plus minus if the previous pulse was positive. On the other hand if the previous pulse was negative then that sequence of 8 zeros is replaced by 0 0 0 minus plus plus minus 0 plus and minus. So you see normally it should be 8 zeros so in place of 8 zeros some of the zeros are being replaced by positive pulse, negative pulse and 0. Now this should be done at the transmitting end and also it should be done at the receiving end so only then it is it is possible to do communication. So it is some kind of protocol that is being followed by both sides for data communication.

So as you see here as long as the number of zeros does not exceed eight then there is no problem. When it is eight or more then 8 zeros or replaced by this code. Let me illustrate it with the help of an example.

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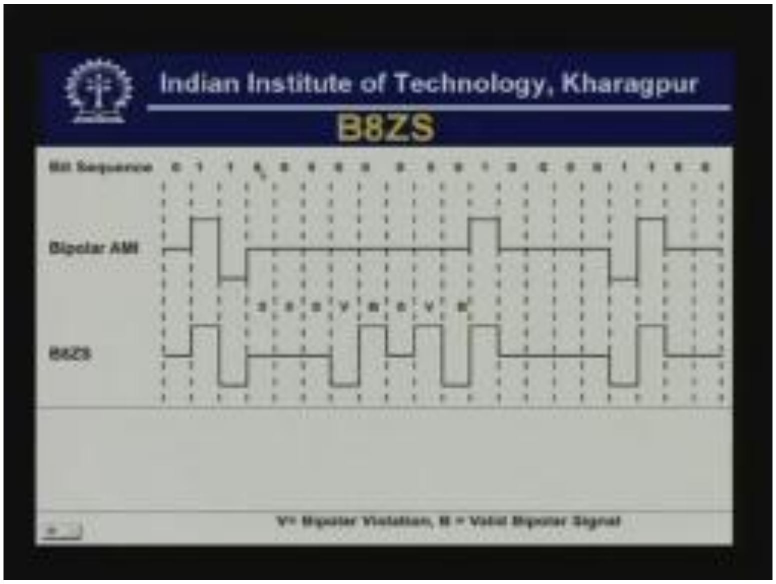

Suppose the bit sequence is 0 0 0 1 1 0 0 0 0 0 0 0 0 here you have got 8 zeros 1 2 3 4 5 6 7 8 and after that it has got one 0 0 0 0 there are 4 zeros after that there is a 1 1 0 0. So if we use bipolar AMI as we find there is a transition from 0 to 1 here and another one is there so again there is transition from plus 1 to minus 1 so as we know the bipolar AMI uses three levels 0 level, positive plus b and minus b positive and negative and for each occurrence of one as we know there is inversion of the pulse. That means for this one it is plus b and for the next one it is minus b and as you see here as long as there is no 1 here all are zeros so there is no transition. Therefore only whenever there is a 1 here at this point then again there is opposite transition. Here it was negative transition and here it is a positive transition.

Now these 8 zeros are replaced by 0 0 0 and here V stands for bipolar violation. Why we are introducing violation?

The reason for that is the other side should be able to understand that this is not a proper code. So it has been forcibly introduced for the purpose of synch synchronization so it will be replaced by all zeros. That means here as you can see these two will remain then after 0 0 0 there will be a violation then the next one is b then 0 positive then again violation and as you see here (Refer Slide time: 8:36) there are two violations and then opposite one is minus b. So in this case sensitivity is followed by a negative transition as we have seen in the previous case if the previous pulse was negative. then it will be 0 0 0 negative positive 0 positive negative that is precisely what is being done in this particular case. And as you can see here now we have enough transitions so that at the other end the receiver will be able to regenerate clock and synchronization will be achieved.

However, in this particular case as you can see there are four Zeros 0 0 0 0 these zeros however is not replaced. But as I said as long as the number of 0 does not exceed 8 no replacement is done. so this is how B8ZS code is sent and this is being sent over the line and at the other end this will be received and this will help you to achieve synchronization and then you will be able to get back your original signal like 0 1 1 0 0 0 whatever it was that can be retrieved and from these violations you will know that this has been introduced so that enough transitions are there. But these are not really the transitions related to that bipolar AMI code. This is how the B8ZS works.

Now another alternative code is the high density Bipolar-3 Zeros and in this case this particular alternative is used in Europe. The previous one was used in America but this is particularly used in Europe and Japan.

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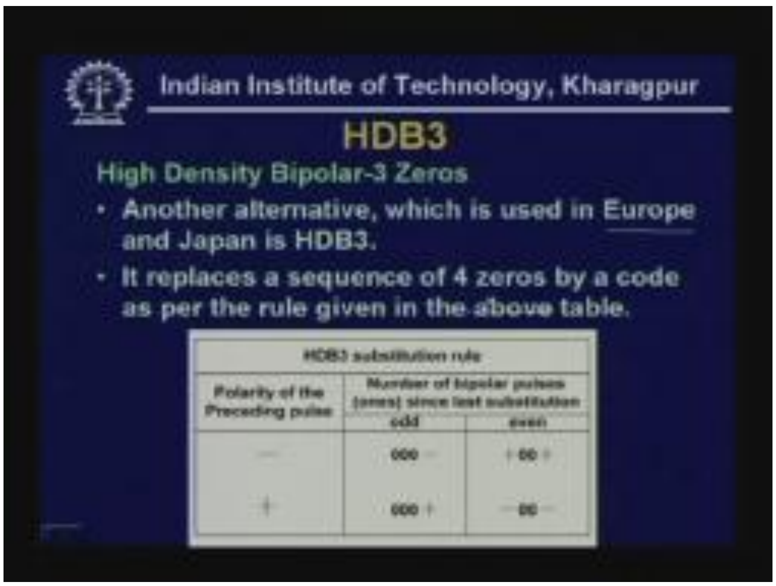

So it replaces a sequence of 4 zeros by a code as per the rules given in the table below so here as you can see it also depends on the polarity of the previous pulse. If it was negative then it is replaced by 0 0 0 and if the number of bipolar pulses since substitution is odd then it is replaced by 0 0 0 negative. On the other hand if the number of substitutions was even then it is replaced by positive 0 0 0 positive. You may be asking why this is being done. This is done so that the average value is 0 that is the purpose.

Similarly if the polarity of the preceding pulse was positive and if the number of bipolar pulses since last substitution was odd then it will be replaced by 0 0 0 positive and if it was even it will be replaced by negative 0 0 negative. Let me illustrate this with the help of an example. The same example is shown here.

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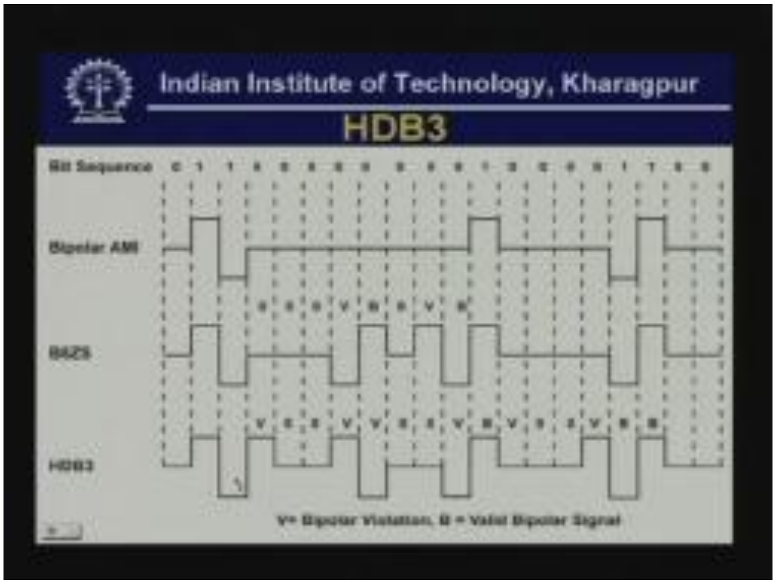

Here as you can see the number of substitutions was two and the previous pulse was negative. Since the number of substitutions that have already taken place is two that means even and the previous pulse is negative so it shows that it is even and number of pulse is negative so it is replaced by positive 0 0 positive. So you see that here it is positive 0 0 positive.

Now here again another 4 zeros are there (Refer Slide Time: 12:28). In this case since the previous pulse is positive here this will be done as per this role negative 0 0 negative so here you see negative 0 0 negative. This is how all the 8 zeros the first 4 zeros replaced by this code and next 4 zeros are replaced by this code and you see here two positive and two negative are there so average value will be 0.

Similarly, here again 0 0 0 0 there are 4 zeros. So far there are one two and three ones so number of ones has occurred odd and the previous pulse is positive so in this particular case the rule followed is odd and it was positive so it is 0 0 0 positive. Therefore here we get 0 0 0 0 positive. So you see that here also the violations are introduced. As we know as per bipolar AMI code the next pulse has to be positive but it is not done here. So forcibly violations are introduced so that they are identified as signals sent for synchronization so that the proper data is retrieved from the received signal and at the same time synchronization is achieved.

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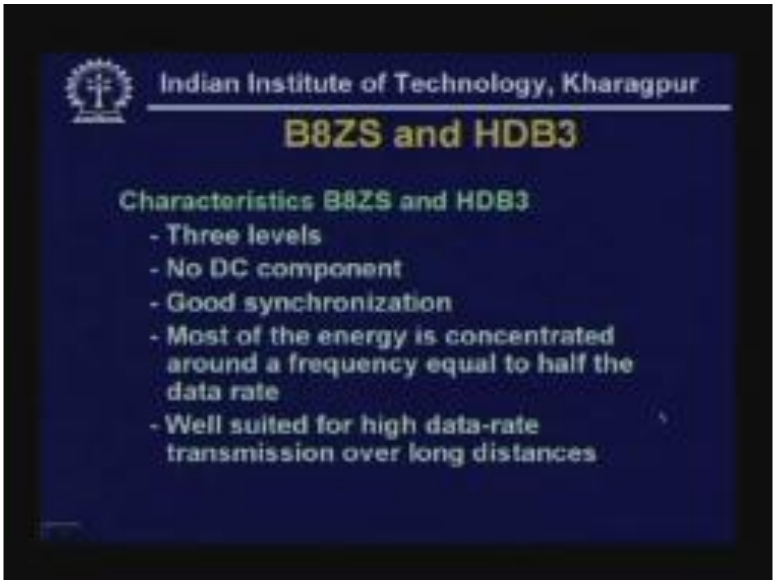

So here is the summary of the characteristics of B8ZS and HDB3. As we have seen it uses three levels positive, 0 and negative and there is no DC component which is a very good parameter or very good feature and it allows good synchronization for both the codes and the advantage is most of the energy is concentrated around a frequency which is equal to half the data rate. So the bandwidth is less and it is well suited for high data rate transmission over long distances.

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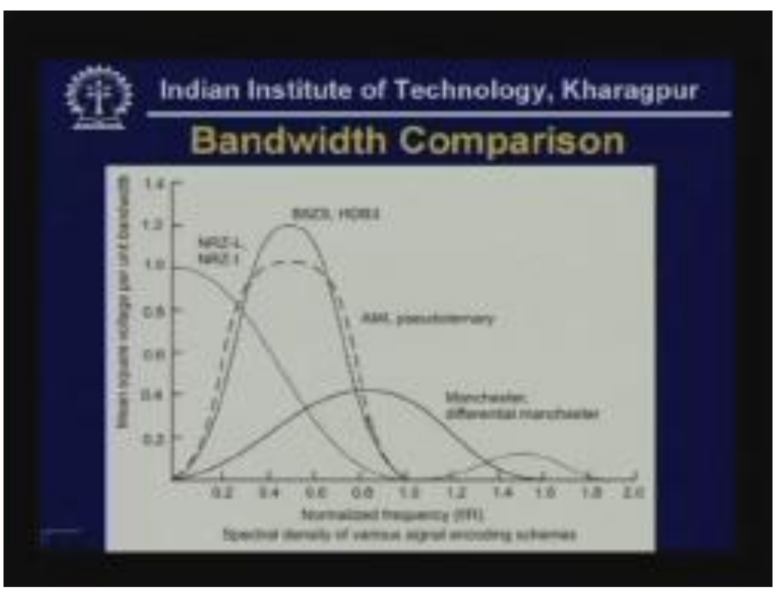

Let us look at the frequency spectrum. Here as you can see as compared to other course we have already discussed here is the B8ZS and HDB3. Here is a nice spectrum that we have got and most of the energy centered around the middle of the bandwidth.

So here one corresponds to the bandwidth so most of the energy is concentrated in the middle of the bandwidth which is definitely a very good feature and that way it will allow transmission through a medium with lesser distortion and attenuation because it will match perfectly with most of the bandpass nature of transmission media.

So we have completed all the codes. Now there is another type of coding known as block coding. This block coding was introduced to improve the performance of line coding. So far we have discussed various types of line coding which is used for conversion of digital data to digital signal for sending through the transmission media.

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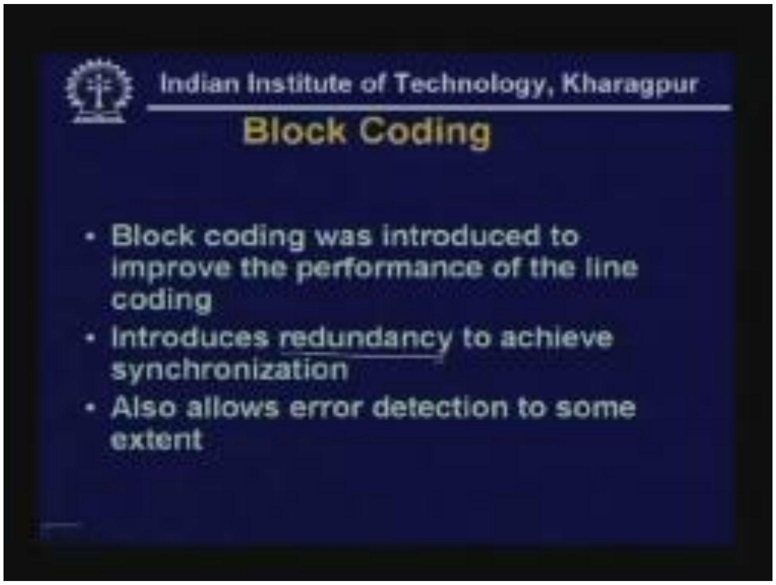

But before you do the line coding a co a technique known as block coding was introduced so that the performance of line coding can be improved. Performance in terms of what? Performance in terms of synchronization, performance in terms of bandwidth, performance in terms of error detection these are the three important parameters which are considered. As I mentioned the basic concept that is being used here is redundancy.

So it introduces redundancy to achieve synchronization, also because of the redundancy introduced here it allows error detection and apart from synchronization it also allows error detection to some extent.

Let us see the basic scheme used. Basic scheme has got three distinct steps division. For example here is your input data so the division is an mb division that means this sequence is divided in blocks of m bits.

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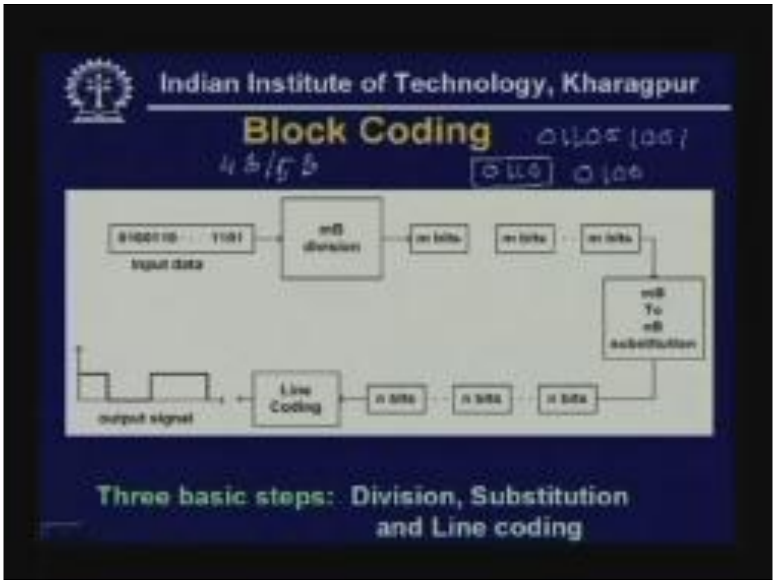

For example, if we are using 4b 5b encoding in that case this m bits will be 4. For example, if the data is 0 1 1 0 0 1 0 0 1 then it will be divided into 0 1 1 0 this is one block then 0 1 0 0 another block (Refer Slide Time: 17:22) so this is how it is divided so it is divided in terms of m bits then that m bit is applied to mb to nb substitution so each m bit is replaced by n bits. So in this case each of these four bits will be replaced by five bits for this particular example. Then ultimately these n bits are line coded and that output signal is transmitted through the transmission media. We have already learnt about line coding. This is the basic scheme.

Let us see the example of 4B/5B encoding. While doing the encoding here you see the encoding is done little cleverly.

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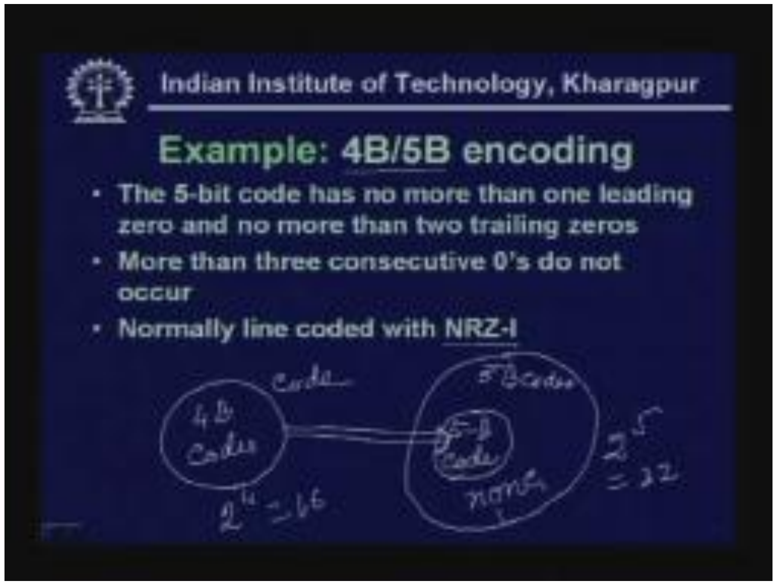

The five bit code is done in such a way that it has got no more than one leading 0 and no more than two trailing zeros. So it is ensured that transitions are there then more than three consecutive zeros do not occur. So these are the two things that are done and finally the line coding is done by using NRZ I code. so what you are essentially doing is suppose here it is the code we set as 4B codes and in 4B codes as you know there will be 2 to the power 4 symbols and obviously the 5B codes will be larger so here the 5B codes will have two to the power five that means here it is 16 and here it is 32. So what will be done is this four B code should be mapped to a subset so it is mapped to a subset such that these properties are satisfied. That means five bit code has no more than one leading 0 and no more than two trailing zeros and more than three consecutive zeros do not occur this is how it is being done.

You may be asking how error detection is done. So you see this is a code and this is also a code (Refer Slide Time: 19:35) but here it is a five bit code. Now suppose at the receiving end if the code is not within this code so it becomes a non code. When it becomes a non code then we know that some error has occurred during transmission from the transmitter side to the receiver’s side. That’s how the error detection is possible in this case.

Let us see the example of how the encoding is done. Here there were 4 zeros 4 zeros have been replaced by 1 1 1 1 0 this is five bit and this is four bit, 0 0 0 1 is replaced by 0 1 0 0 1. And you can see here that when long sequence of zeros like 4 zeros or 4 ones are there then there are zeros and ones that are introduced and as I mentioned it was necessary to have no more than one leading 0 and no more than two trailing 0 that is being satisfied here.

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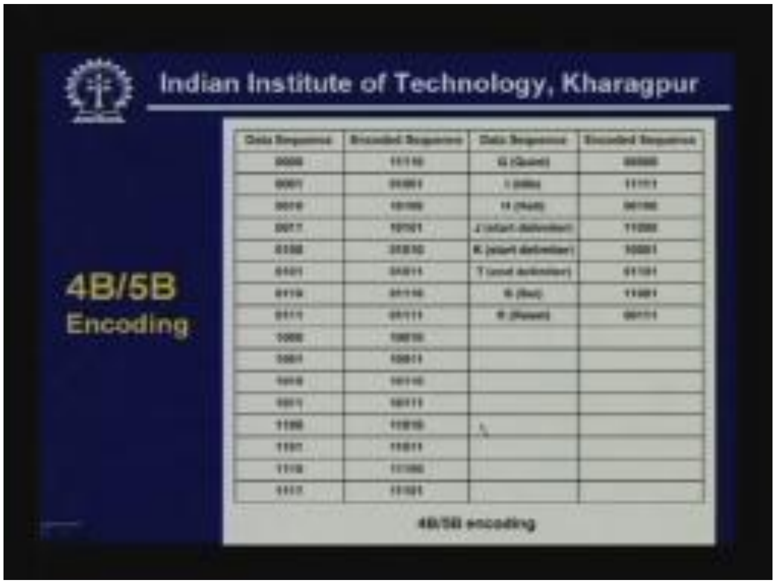

Here you find no more than one leading 0 and no more than two trailing zeros. Here there are two trailing zeros but you will not find three trailing zeros anywhere. so this is how the encoding is done. However you have got some other codes which are used for other purposes like quiet, idle, halt essentially control codes. But so far as the data codes are concerned it satisfies that synchronization property and it helps you to achieve error detection.

Now other block coding examples are 8B to 10B so here you have got 8-bit data codes data blocks which are substituted by 10-bit code. So each 8-bit data is encoded by 10-bit data. Obviously it allows more redundancy.

More redundancy means it provides more error detection capability but unfortunately it leads to increase in bandwidth. Whenever you have got more redundancy there is more error detection capability. So unfortunately this error detection capability is advantageous but there is increase in bandwidth. So this is acceptable and this is not acceptable (Refer Slide Time: 22:10), how do you overcome this problem? This problem can be overcome by using suitable line coding.

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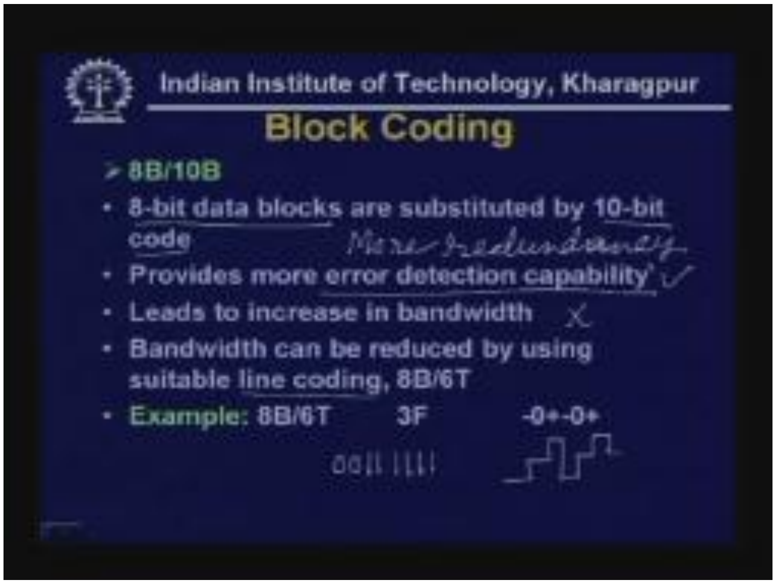

For example here let's assume that 6B to 8B encoding is done then the line coding can be done from 8B to 6T. So suppose after encoding the 8-bit data is 3F so 3F means this is 0 0 1 1 and F is 1 1 1 1 so now while doing the encoding you are not generating 8-bits but you are generating 6 values for transmission so negative, 0, positive, negative, 0 and then positive so you are generating six symbols during transmission. So obviously the bandwidth of this will be lesser compared to 8-bit signals. So here is the advantage of this line coding, it reduces the bandwidth but at the same time it provides you higher redundancy, higher error detection capability and also synchronization.

So here we now change gear from digital data to digital signal to analog data to digital signal. There are many situations where we have to send analog data but we want to send in digital form or we want to send in the form of digital signal. Because sending signal in the digital form has many advantages from the view point of signal to noise ratio and other things. For example analog data such as voice, video and music are to be converted into digital signal for communication through transmission media. One example is voice that is being sent through telephone line. So here we are trying to send analog data. But as you know in modern telephony it is not sent in analog form but it is converted into digital form so digital signal is transmitted. Similarly, video and voice, video and music can be sent. For that purpose there are two basic approaches.

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The two basic approaches are; first one is known as Pulse Code Modulation (PCM) and second one is known as Delta Modulation (DM).

So in this lecture today we shall discus about these two schemes; Pulse Code Modulation and Delta Modulation one after the other.

Let us first concentrate on Pulse Code Modulation. The Pulse Code Modulation has got three basic steps; sampling, quantization and line coding.

Suppose this is the analog signal so this analog signal has to be converted into digital form for sending through a transmission media. First step is known as quantization or first step is known as sampling.

The sampling process converts the analog data analog signal you can say as this is essentially analog signal into a form known as Pulse Amplitude Modulation. so PAM stands for Pulse Amplitude Modulation. How it is done? This analog signal is sampled at regular interval, this is the interval this is the interval at which it has to be sampled (Refer Slide Time: 26:32) obviously they will be equispaced. When you do the sampling you have to you have to follow the nyquist criteria.

Nyquist criteria says if the signal has a bandwidth B then the sample rate has to be 2B. Suppose you are sending voice signal the voice signal has the bandwidth of 4 KHz so your sample rate has to be 8 KHz so that is the rate here in this direction that is being sampled. now the sampling is done here, here, here, here at these points (Refer Slide Time: 27:20) so as you sample it now you are also holding that value till the next sample then you are again increasing then you are holding it to the next sample then you are going to the next sample again holding it till the next sample then you are going there and holding it to the next sample and then you are holding it to the next sample value and so on. And this sample value again you are holding it to the next sample value, here again you are holding it to the next sample value then again you are sampling it holding it to the next sample value and so on and so forth. In this way you convert it into a signal and this signal is known as PAM. that means this signal as we have obtained after sampling it and then holding it for the next duration with the help of a hardware known as sample and hold circuit you will be able to do this. Therefore the signal that is being generated at the output of the sample and hold circuit is known as PAM.

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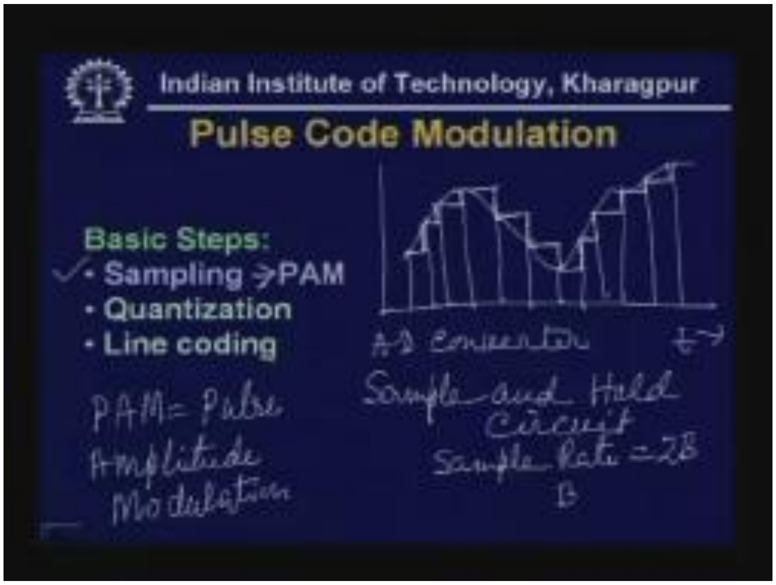

Now this PAM has got different values so you have to now do the quantization. Quantization is done with the help of a hardware known as analog to digital converter A-D converter.

AD converter will do the quantization because that analog to digital conversion is done and the analog to digital converter will have limited number of bits say 8-bit or 6-bit. If it is 8-bit then it will have 256 different quantization levels. Now the analog value is having all possible voltage levels. It is now converted into some discrete values depending on the number of bits used in the analog to digital converter and that leads to what is known as quantization error. After we have got the digital data it can be converted into digital signal using suitable line coding which we have already discussed.

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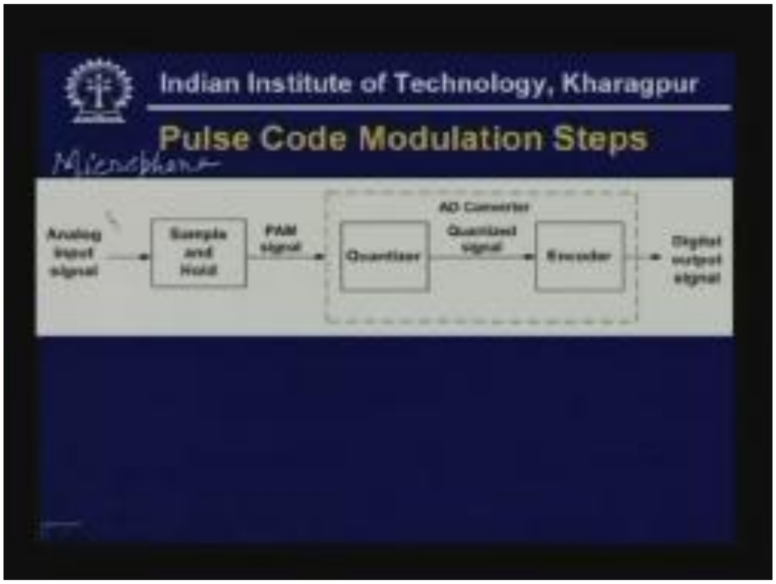

This is the process analog input signal which is obtained with the help of a microphone and if it is voice then using sample and hold you are converting into PAM signal then AD converter has got two parts quantizer and encoder then it generates the digital output as it is shown here.

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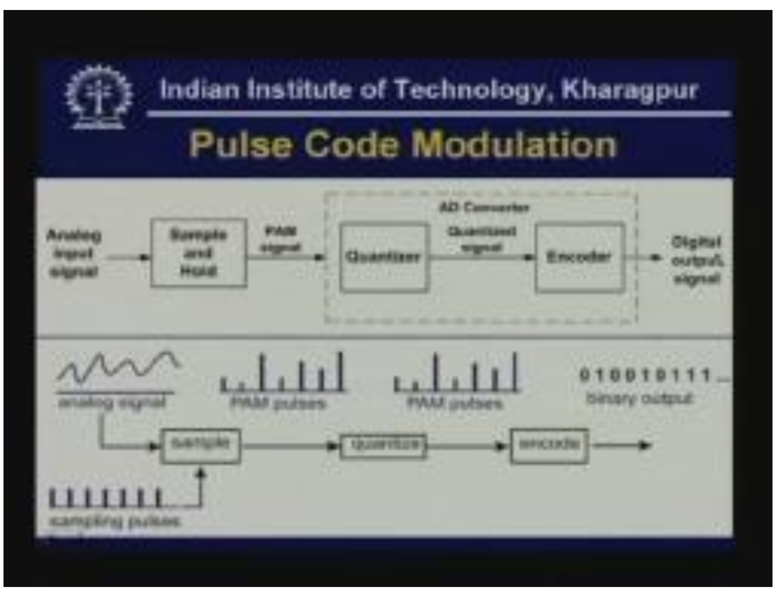

So this is the analog signal, this is the sample pulses and this is the PAM signal (Refer Slide Time: 30:12) and after quantization and encoding you get the binary output the digital data that you are getting which can be line encoded for sending through the transmission media. So this is the transmitter part which generates digital output signal from the analog input signal and then this digital input signal can be transmitted through the transmission media after line coding which will be received at the receiver and again you have to perform the reverse process.

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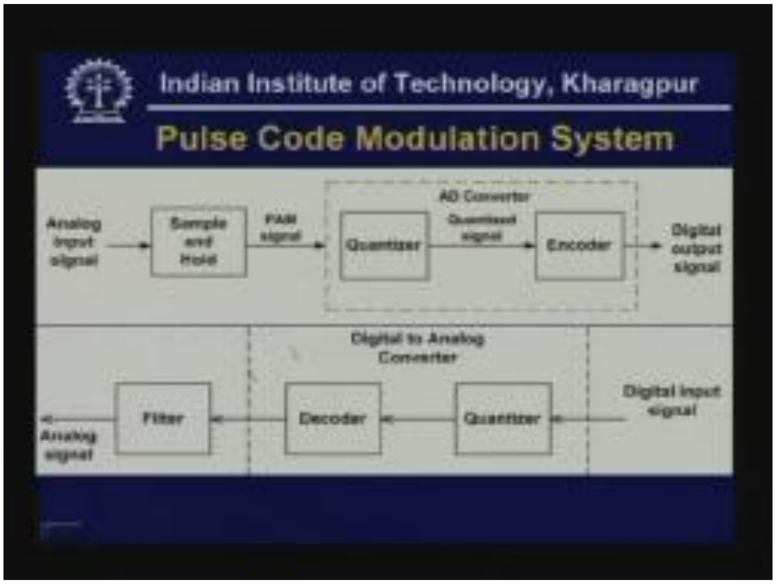

The reverse process first is done with the help of digital to analog converter or DA converter and the D-A converter will generate analog signal which has to be filtered. After PAM we got the quantized values so that has to be smoothed with the help of the filter and we shall get the analog signal recovered from the system. So this is your receiver and here is your transmitter, this side is your transmitter. So on this side and this side is the transmitter and this side is the receiver as I have shown.

As an example let us consider an input signal which is voice which has a bandwidth of 4 KHz and as I said the sampling frequency has to be at least twice the highest frequency in the input signal otherwise a problem known as aliasing error will occur. You may be asking why the sampling rate has to be more than twice the maximum signal. If it is not done that way it will lead to you a kind of error known as aliasing error.

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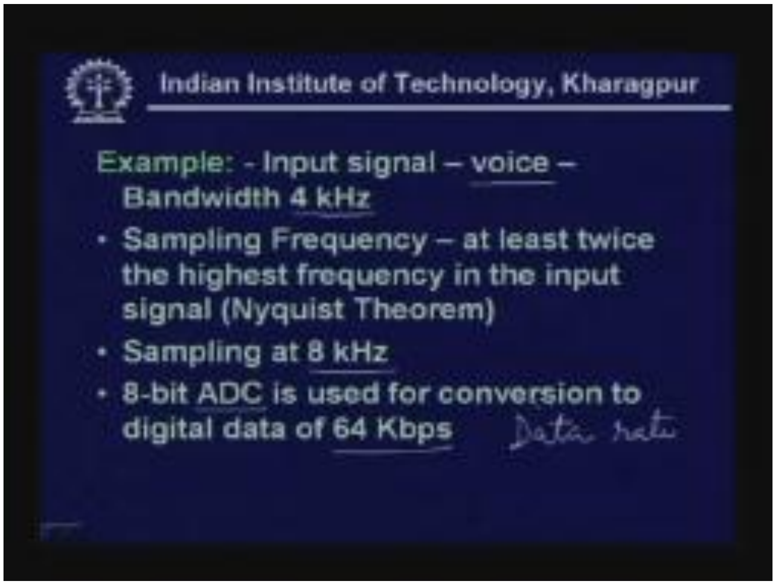

Sometimes the signal is band limited before doing the sampling to avoid aliasing error. so the 4 KHz bandwidth signal at sample at the rate of 8 KHz and then you are using an analog to digital converter and using an 8-bit A-D converter each sample value is converted into 8-bit. So each sample value that means in one second you have got 8 k sample values and you multiply with 8-bit to generate a digital data having 60 Kbps so data rate is 64 Kbps. So you see here a voice of 4 KHz band bandwidth has got a 64 Kbps data rate whenever you convert into a digital data, this is the data rate. This has to be properly encoded by line coding and then sent over the transmission media.

As I mentioned whenever you use an analog to digital converter of suitable number of bits quantization error is introduced, essentially it depends on the step size. Suppose you are using 5V the maximum is from 0V to 5V and quantization level is 8-bit that means 256 levels so the step size is essentially 5V by 256. So any intermediate levels between this 5V/256 will be translated into those discrete values and that will lead to error. And even when the input signal is of low amplitude or high amplitude we are encoding by using the same step size and we are not doing conversion in a different way. Therefore as a result with the constant of a fixed number of levels the situation can be improved using variable step size.

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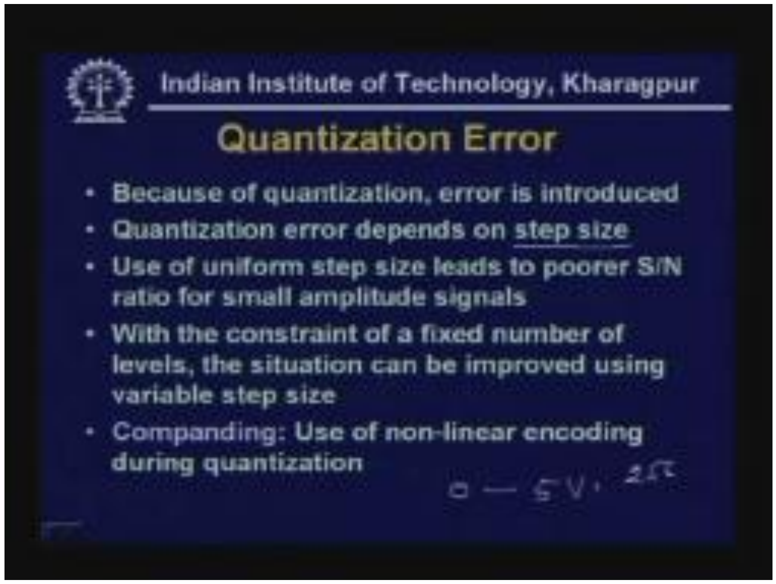

Because of the fixed step size the signal to noise ratio is more for high level signals. Unfortunately for small amplitude signals the signal to noise ratio is poorer. How that can be overcome? That can be overcome or improved by using variables step size by a technique known as companding. It is essentially a technique by which we are introducing non-linear encoding. that means whenever the signal amplitude level is low then we are using smaller step size so that we get the better accuracy and on the other hand when the amplitude level is high step size can be high. That is precisely what is being done in this case. For that purpose we can use a non-linear circuit to convert the signal in this form.

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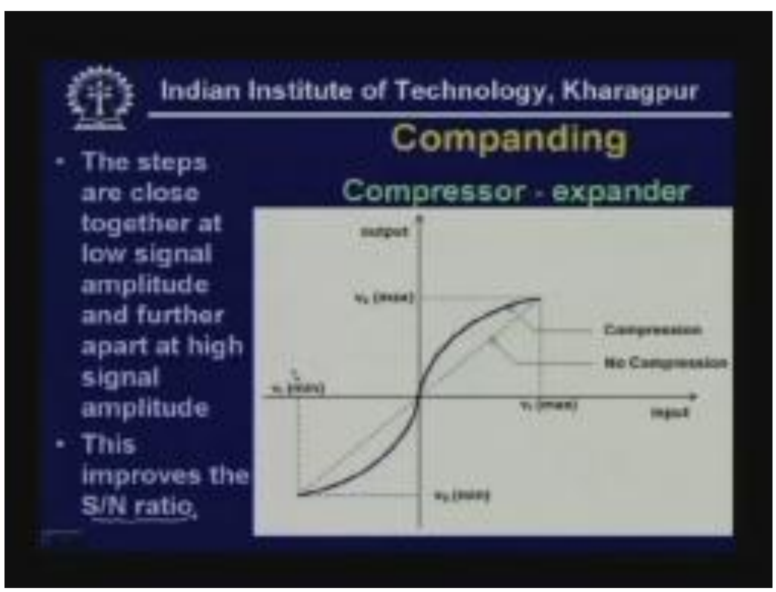

So here as you see (Refer Slide Time: 35:41) for low amplitude values whenever you are not using compression this particular curve is linear. For all values smaller and higher values the slope of this curve is same in this particular case.

But whenever you are using compression then you see that this slope is more than whenever you have got higher values. That means slope is more for smaller values than higher values. Then after passing through this non-linear circuit you can apply the signal to the analog to digital converter and this will avoid the problem that we have already discussed.

So what we are doing is the steps are close together at low signal amplitude and further apart at high signal amplitude. This improves the signal to noise ratio particularly when the signal level is low. So, companding is the technique which can be used to improve the signal to noise ratio.

Now let us look at the limitations of Pulse Code Modulation PCM technique that we have already discussed.

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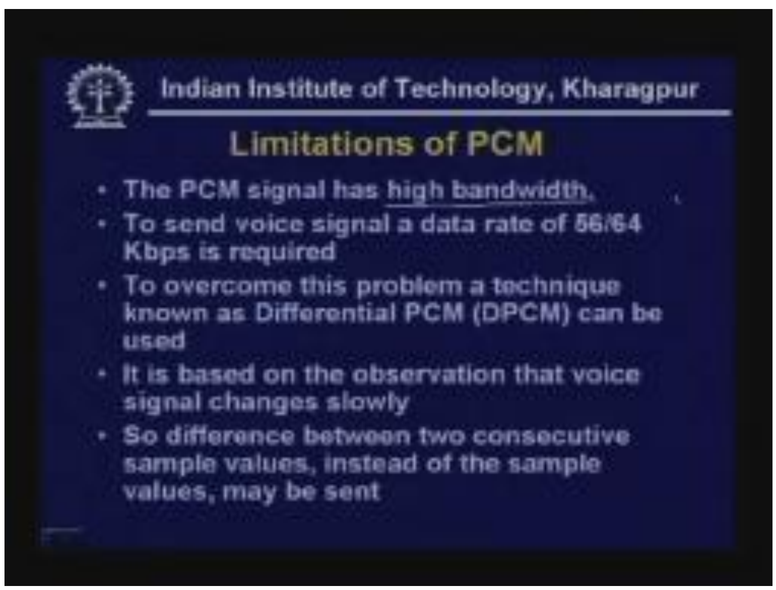

Pulse code modulated signal has got high bandwidth as you have seen. Say 4 KHz analog signal will require a bandwidth of 4 Kbps.

For a voice signal you require a data rate of 56 if you are using 7-bit encoding or 64 Kbps if you are using 8-bit encoding. to overcome this problem a technique known as differential PCM can be used. A differential PCM can be used whenever you are using this PCM technique. But this is based on the observation that whenever you are using PCM that is sending analog signal we have observed that the signal is changing very slowly. Since the voice signal changes very slowly why not send the change rather than the absolute values that is the basic idea of this differential PCM.

Instead of sending the absolute values the difference between two consecutive sample values instead of the sample value may be sent, this is the basic idea of differential PCM. Since the signals are changing slowly the difference will have small values so the number of bits required will be less so bandwidth will be less so that's the basic idea of this differential PCM. And you can use either analog technique or mixture of analog and digital technique or purely digital technique to do this. It can be done in one of the three ways to generate differential PCM signals.

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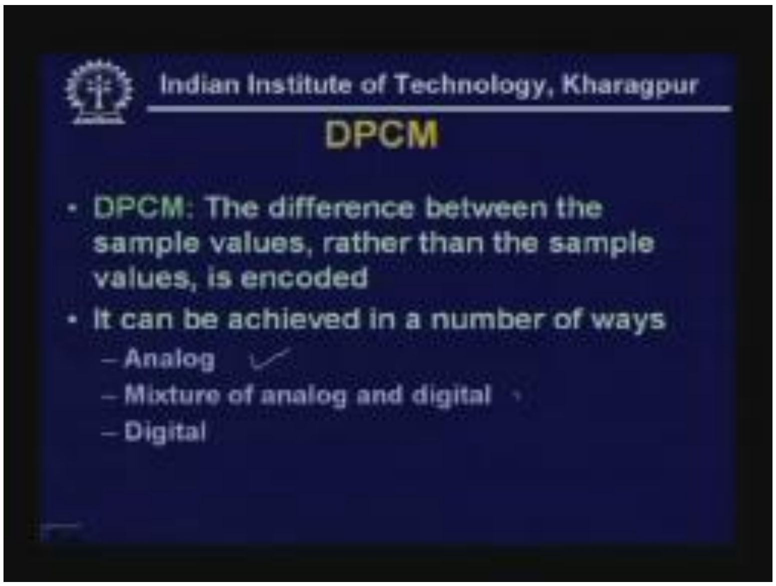

However, a very special case of differential PCM is DM which stands for Delta Modulation. In what way it is a special case of differential PCM? In differential PCM what we are trying to do is we are sending the difference between two consecutive sample values. Now what we are doing is we are reducing the difference to a single bit. We are sending either 0 or 1. That means if the difference is positive that means if the next sample value is more than the previous sample value we shall be sending one. And if the previous sample value is more than the present sample value then we shall be sending 0. So instead of sending 8-bit per sample we shall now send only one bit per sample. That's the basic idea of Delta Modulation. So depending on higher or lower we are sending output one or 0 and we are sending one bit at a time so as a consequence the bandwidth will be much less as you can see here.

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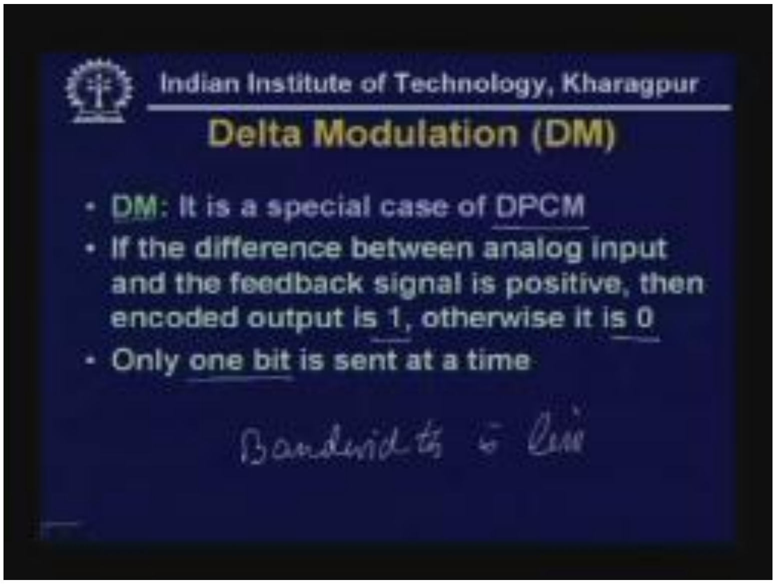

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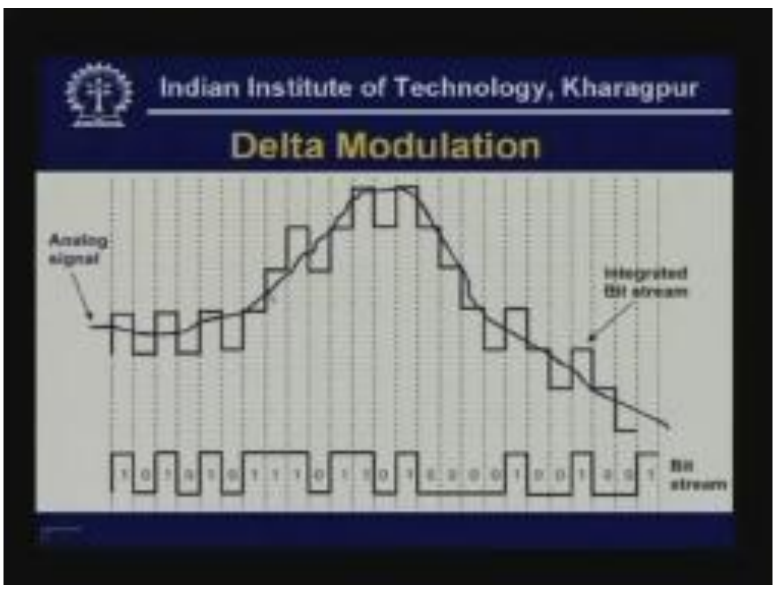

This is the example of the Delta Modulation. here is the analog signal changing very slowly as you can see and this change in being monitored and if it is increasing that means the previous sample value is less than the previous sample value you are sending ones and whenever it is less we are sending zeros. Therefore we are sending one bit at a time so the bit stream that you have to send is 1 0 1 0 1 0 1 0 1 1 1 and so on and you can see here that per sample we are sending only one bit so the bandwidth is less as you can see in case of Delta Modulation.

You can use a circuit like this (Refer Slide Time: 41:00) an analog comparator, here is the analog input and with the help of this clock we are finding out whether this comparator output is 0 or 1 and depending on that 1 was 0 and with the help of that this is essentially some kind of integrator or adder which regenerates the reconstructed signal. That means analog waveform that you are receiving is reconstructed here and with one bit unit delay you are feeding it so that the previous value is compared with the present value, here is the previous value and here is the present value (Refer Slide Time: 41:45) these two are compared with the help of this comparator. And with the help of such simple circuit you can generate the delta modulated signal. Here you get the zeros and ones the binary output which can be sent through the transmission media after suitable line coding.

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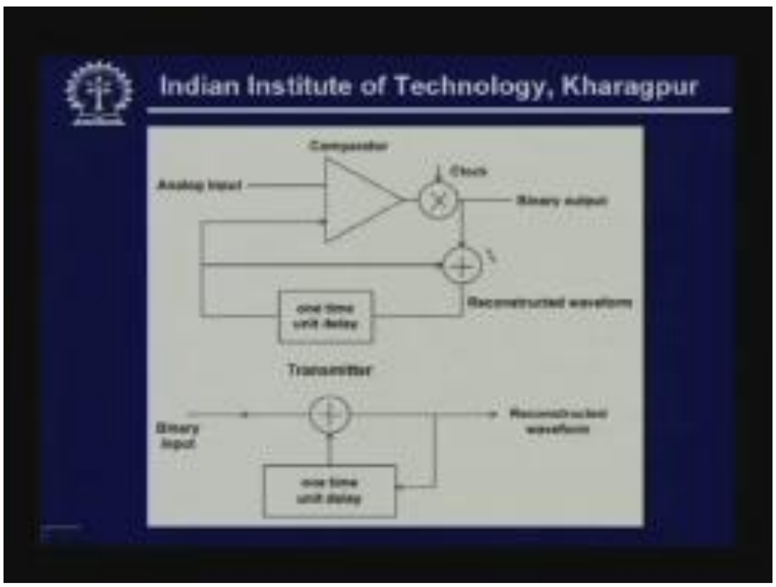

As you can see here the receiver is also very simple, here is the simple adder, the previous value after one bit delay is added with the binary input and then the reconstructed waveform is obtained here. So we can see here that Delta Modulation does not require complex hardware. Here are the advantages of Delta Modulation.

Simplicity of implementation: Each sample is represented by a single binary digit as you have seen which makes it more efficient than Pulse Code Modulation technique.

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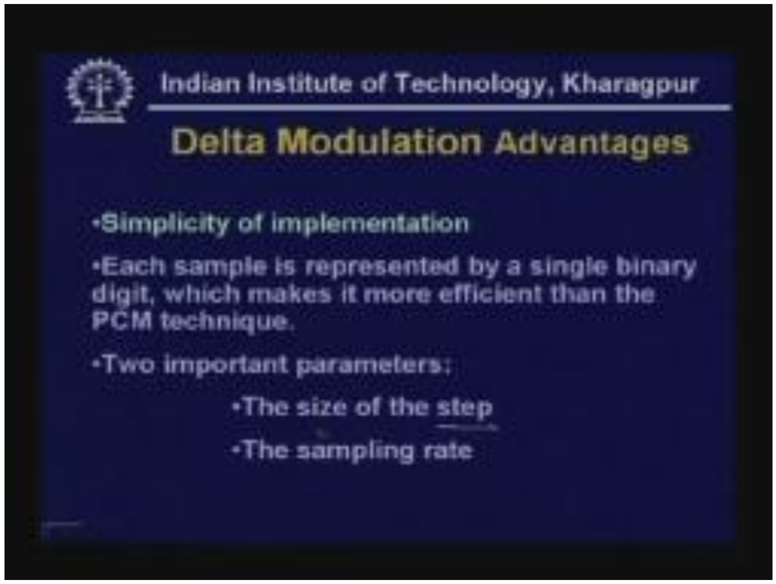

However, there are two important parameters; the size of the step and the sampling rate. These two play a very important role.

This fixed step size leads to overloading. What do you mean by overloading? As you can see here if the analog is signal changing very slowly we can see that the data that you have to send will become 0 1 0 1 which has to be sent alternately which leads to overloading. That means it is slowly changing.

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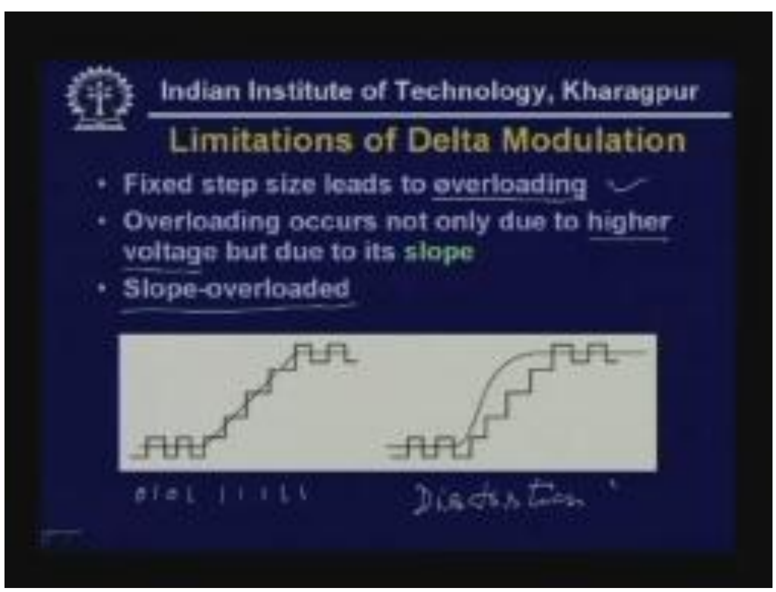

If the step size is such with the input changes within the step size then alternately you have to make 0 1 0 1 this is known as overloading. And overloading occurs not only due to higher voltage but also due to slope. Higher loading means whenever it exceeds the limit then it is overloading. Moreover error arises whenever the slope is fast then we shall call it slope overloading as you can see in this diagram. Here this kind of slope is getting converted properly. As you can see we shall be sending 1 1 1 1 and 1 consecutive ones.

But in this particular case the sampling rate is such that it is not able to follow the first changing signal. This is leading to slope overloading. And as you can see here whenever you reconstruct the signal from this received signal we will not get whatever is transmitted that means whatever is received so this leads to distortion. This distortion can be overcome. First of all distortion is arising because of fixed step size and the sampling rate and that can be overcome by using Adaptive Delta Modulation.

This technique can be used to overcome the problem of overloading particularly slope overloading. What we are doing is when the step sizes are controlled in such a way that step sizes are small when the signal changes are small and the step sizes are made large when the signal changes are large.

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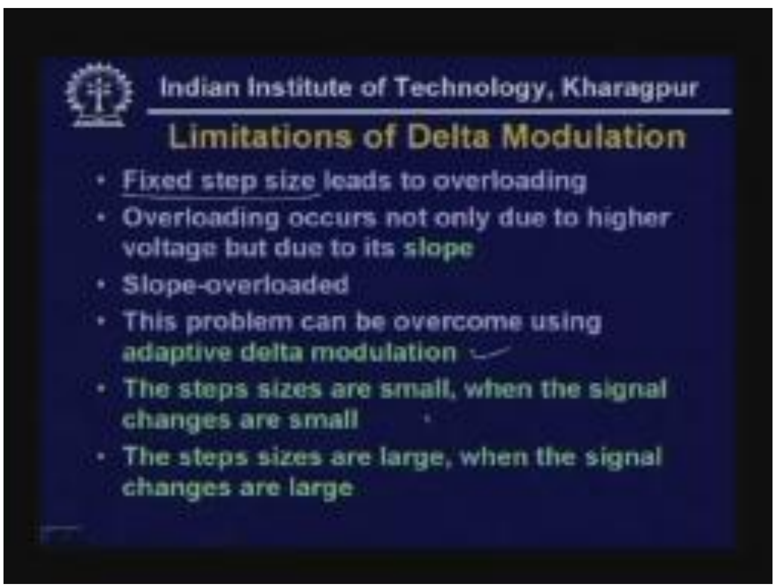

So what we are doing is the step size is not made fixed. When the input changes slowly the step size is made smaller and when the input changes rapidly then it says the step size is made larger. That is how it is able to overcome the overloading problem. So here we shall compare the Pulse Code Modulation with the Delta Modulation.

We have seen that for voice signal with 256 quantization level that means AD converter has got 8-bit using 8-bit ADC the data rate is 64 Kbps and this requires a channel bandwidth of 32 KHz. we know that channel bandwidth is 2B so from that the channel bandwidth has to be 32 KHz. Moreover it requires more complex hardware. That means the PCM does require higher bandwidth and more complex hardware to generate the pulse code modulated signal.

On the other hand the Delta Modulation does not require higher bandwidth. However, there is a point. Point is does Delta Modulation give you good quality signal or good quality reproduction at the receiving end?

Unfortunately it has been observed that to obtain comparable quality a sampling rate of 100 KHz is required in case of Delta Modulation to send voice signal. as you have seen in case of pulse code modulated signal the sampling rate was 8 KHz and then after converting into Pulse Code Modulation by using 8-bit encoding you were getting 64 Kbps. On the other hand to get the same quality of performance in terms of intelligibility and quality the sampling rate has to be 100 KHz otherwise you don't get that much quality.

So we find that ultimately bandwidth was the key parameter in favor of Delta Modulation and it is only possible whenever we compromise in quality and intelligibility. That means if we compromise in quality and intelligibility then Delta Modulation requires lesser bandwidth.

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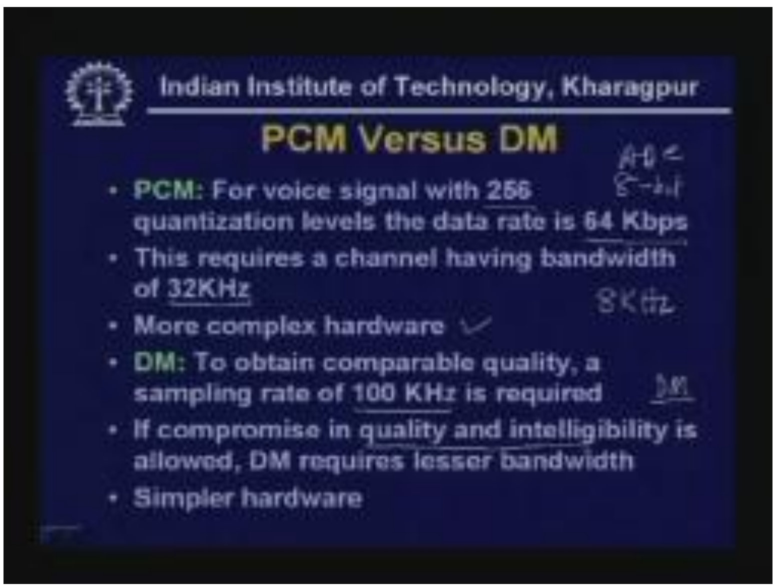

As you have seen Delta Modulation requires much simpler hardware compare to Pulse Code Modulation. So we find that Delta Modulation has some advantage, Pulse Code Modulation has some advantage but whenever we want quality and we don't want to compromise on quality and intelligibility Pulse Code Modulation is preferred. That is the reason why in modern telephony you will find Pulse Code Modulation is widely used rather than Delta Modulation.

But whenever we can compromise on quality and intelligibility we can use Delta Modulation. So it’s a trade off, whether you want lesser bandwidth or higher quality. So depending on these two we can choose either PCM or DM. But nowadays better quality is more important that’s why the PCM is preferred.

We have discussed various techniques for encoding digital data to digital signal and also for encoding digital analog data to digital signal so both types of encodings we have discussed. Let us look at some examples or application examples. We have discussed Manchester encoding. We shall find how Manchester encoding is used in Ethernet LAN.

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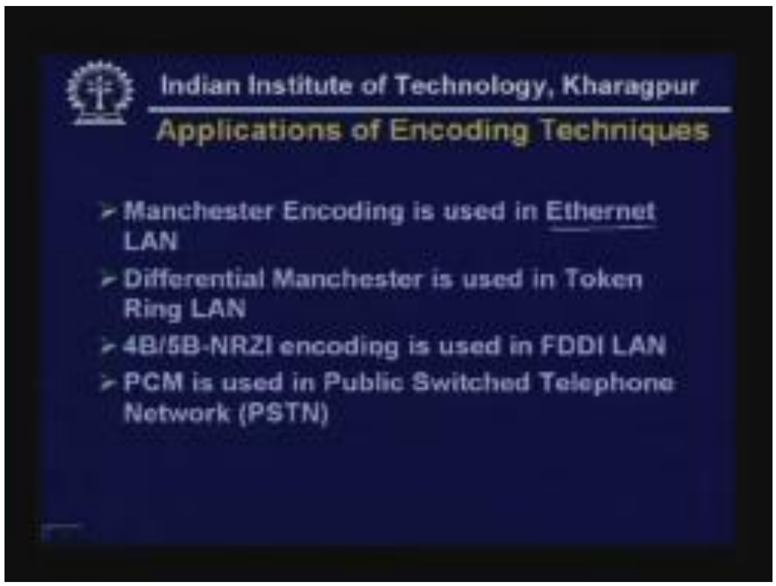

Differential Manchester encoding is used in token ring LAN. We have already discussed this. Then we have discussed the 4B 5B block coding and using NRZ I line encoding that is used in FDDI Fiber Distributed Data Interface LAN. So we are finding that the digital to digital conversion is widely used, those encoding techniques are used in LAN technology. On the other hand that Pulse Code Modulation technique is used in Public Switched Telephone Network or PSTN.

We shall discuss all these applications later on in detail. With this we come to the end of today’s lecture. We have some review questions to be answered in the next lecture.

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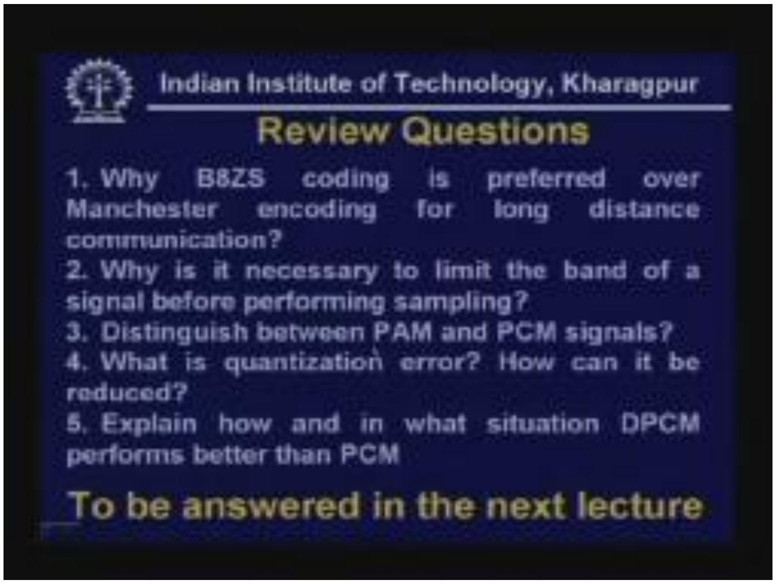

1) Why B8ZS coding is preferred over Manchester encoding for long distance communication?

2) Why is it necessary to limit the bandwidth of a signal before performing sampling?

3) Distinguish between PAM and PCM signals. What is the difference between Pulse Amplitude Modulated signal and Pulse Code Modulated signals?

4) What is quantization error? How can it be reduced?

Obviously we are referring this question with respect to PCM.

5) Explain how and in what situation differential PCM performs better than PCM.

The fifth question will be answered in the next lecture.

Here are the answers to the questions of lecture-7.

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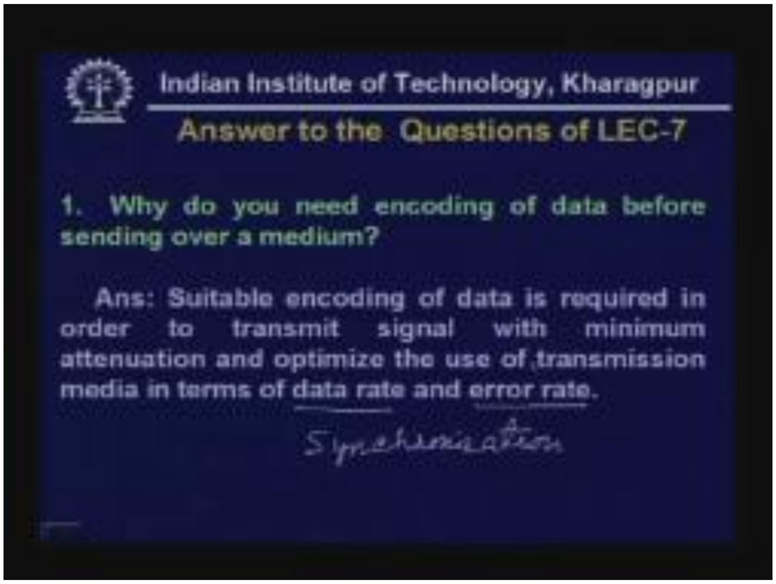

2) Why do you need encoding of data before sending over a medium?

Answer is: suitable encoding of data is required in order to transmit signal with minimum attenuation and optimize the use of transmission media in terms of data rate and error rate.

So apart from that I should add synchronization. That is another parameter which is achieved with the help of suitable encoding.

2) What are the four possible encoding techniques give examples.

The four possible encoding techniques are digital data, digital signal which are used in transmitters. Second is analog data to digital signal used in codec, codec stands for coder decoder, third is digital data to analog signal used in modem and the fourth is analog data to digital signal example is telephone system.

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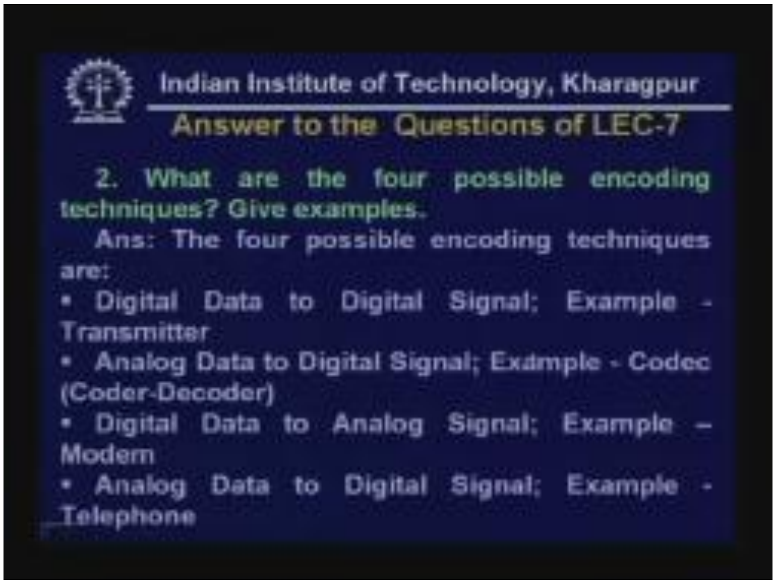

3) Between RZ and NRZ encoding technique which requires higher bandwidth and why?

RZ encoding requires more bandwidth as it requires two signal changes to encode one bit. We have seen that per bit two signal changes are present. Obviously RZ will require higher bandwidth than NRZ.

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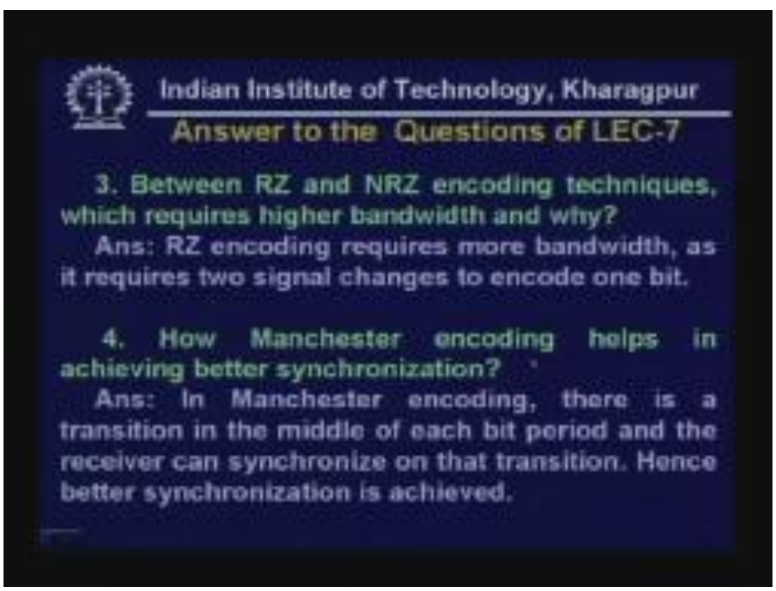

4) How Manchester encoding helps in achieving better synchronization?

In Manchester encoding there is a transition in the middle of each bit period and the receiver can synchronize on that transition hence better synchronization is achieved in Manchester encoding.

With these we conclude our discussion on digital transmission. In the next lecture we shall discuss on transmission of analog signals, thank you.