Data Communication
Prof. Ajit Pal
Department of Computer Science & Engineering
Indian Institute of Technology, Kharagpur
Lecture No # 9
Transmission of Analog Signal-I

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

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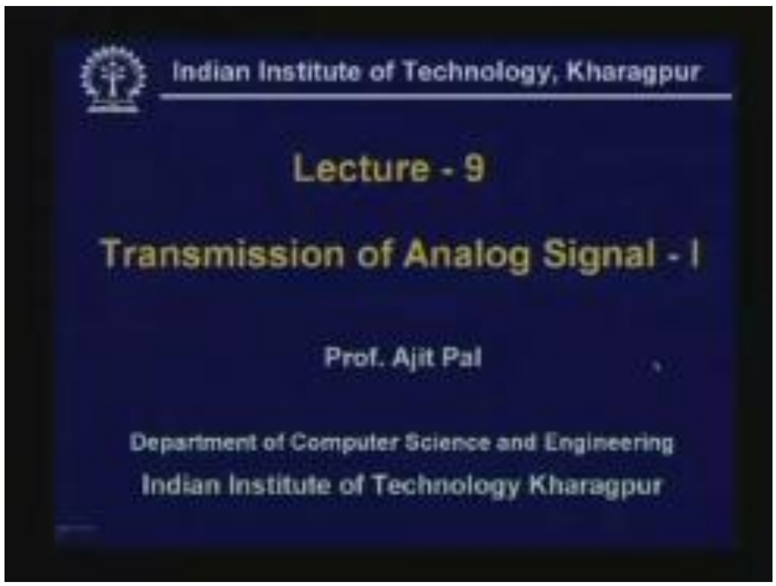

In the last two lectures we have discussed about the transmission of digital signal. there we have seen how analog and digital data can be encoded into a digital signal form which can be transmitted through the media transmission media various types of transmission media and we have seen how the encoding is done so that the bandwidth of the signal matches with the bandwidth of the transmission media so that it passes with less attenuation, less distortion and also it provides you necessary signaling for synchronization, error detection and other purposes.

Now we shall focus on the transmission of analog signal. In this lecture we shall cover the following topics: First we shall discuss why modulation? It is essentially an introduction to this particular lecture, the need for modulation. Then we shall consider various modulation techniques. And in this particular lecture we shall primarily focus on amplitude modulation.

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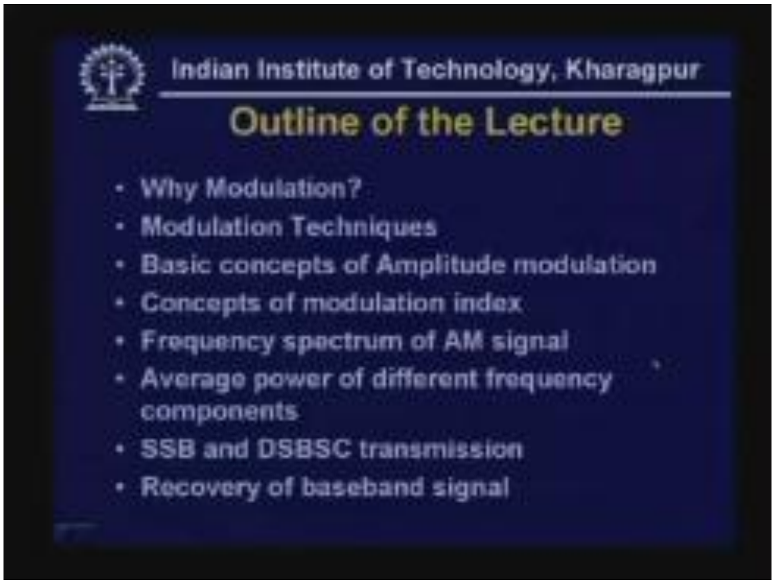

So we shall introduce to you the basic concepts of amplitude modulation then we will see the concept called modulation index which will be introduced to you, frequency spectrum of AM signal, average power of different frequency components that is whenever a signal is modulated it generates different frequency components, and we shall also see what is the average power of different frequency components. Then we shall consider some special situations like Single Side Band SSB and DSBSC Double Side Band Suppressed Carrier Transmission. Finally we shall discus about the recovery of baseband signal.

After attending this lecture the students will be able to explain the need for modulation, they will be able to distinguish between modulation techniques, they will be able to identify the key features of amplitude modulation then they will be able to explain the advantages of SSB and DSBSC. Finally they will be able to explain how the baseband signal can be recovered from the received signal.

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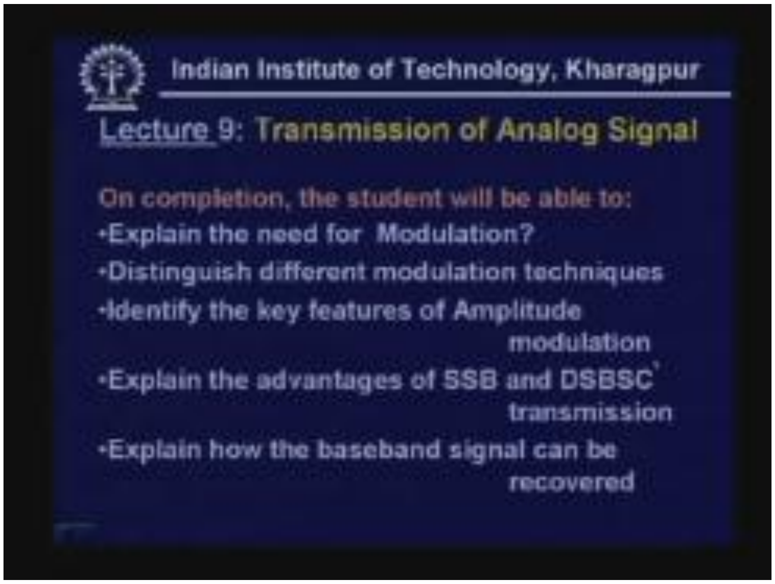

So this diagram gives you the basic scheme of analog data to analog signal. Here as you can see we have used one analog data as an input to the analog to analog conversion system.

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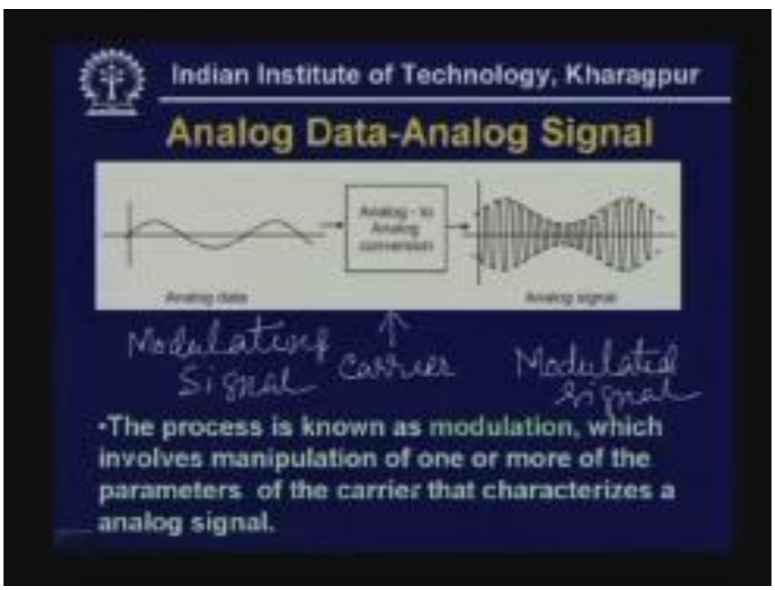

Here apart from applying this analog data another auxiliary signal usually a sinusoidal signal is applied to this analog to analog converter known as carrier. And this carrier is essentially a sinusoidal signal and as you know a particular analog signal whenever it is a sinusoidal there are three important parameters by which the signal can be characterized that is the amplitude, phase and frequency. Either one of them individually or a combination of them are modified to generate a signal which is known as modulated signals. So here is your modulated signal and this process of applying a signal to be modulated is known as modulating signal.

Modulating signal and carrier are applied to the modulator and this process of conversion from analog data to analog signal which involves manipulation of one or more of the parameters of the carrier that is amplitude, frequency of phase that characterizes the analog signal is known as modulation. So this process is known as modulation. And you may be asking why modulation is necessary?

When we do modulation one important operation that is being performed on the signal is known as frequency translation. What it does is it translates the signal from one region of frequency domain to another region, it's like this. Suppose this is the representation of the signal in frequency domain so here you have got your f (Refer Slide Time: 6:44) and obviously in this you have got the amplitude. Now suppose you have a signal with frequency range say f_1 and f_2 so this is the range of frequency and we can say that this is how you can represent it.

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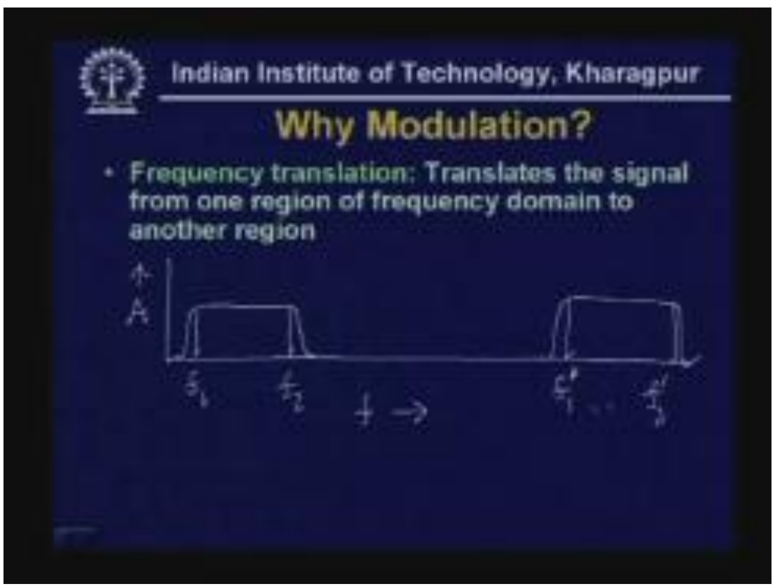

You have frequency components from f_1 to f_2. Now by modulation this signal can be translated to another frequency range say f_1 dash to f_2 dash to this so it can be translated to another frequency range. Usually this frequency range f_1 dash or f_2 dash is much higher than f_1 and f_2. So this is how this can be translated. However, the information content of the translated signal is such that the original signal can be recovered from it.

Now you may be asking what is the benefit of this?

One important benefit of this is that you will be able to use an antenna of practical size. So whenever you do translation say baseband signal to higher frequency can be transmitted through a bandpass channel using an antenna of smaller size. Suppose you are trying to send 1 KHz signal so what is the wave length for this? If this is the frequency what is the lambda value? As we know the wavelength will be 300 000 m, we have already discussed about it.

Now whenever you are trying to send a frequency of 1 KHz the wave length is 300 000 m and obviously the antenna has to be comparable to this size. So obviously an antenna of this size 300 000 m is impractical you cannot do it.

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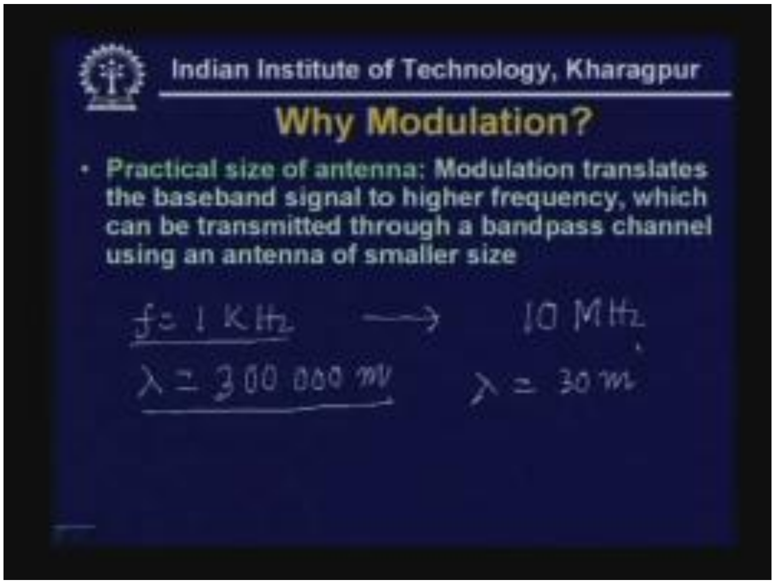

However, if you translate it to a frequency say 10 MHz whenever you translate it to 10 MHz this will correspond to a lambda that is equal to only 30 m so obviously it is quite possible to have an antenna of 30 m. So after translating this to a higher frequency then you can have an antenna of smaller size and then this modulated signal can be transmitted very easily using smaller size antenna. And of course at the other end after receiving it you have to do demodulation which we shall discuss later.

So this is the first benefit of modulation, we will be able to have an antenna of practical size.

What are the other benefits? Another important benefit of this modulation is narrow banding.

Suppose you are trying to send a frequency range of say high fidelity audio frequency range which has frequency range of 20 Hz or 20 KHz so you can say hi-fi audio, obviously you send music and other things. So you see the ratio of the highest frequency to lowest frequency is quite high. That means highest frequency is 20 KHz and the lowest frequency is 20 Hz. The ratio between highest frequency to lowest frequency is thousand.

Obviously if you design an antenna for this frequency then it is not at all suitable for this frequency. Or if you design an antenna for higher frequency then it will not be suitable for transmission for lowest frequency. So a single antenna will not be able to cover or able to transmit both the signals effectively and efficiently.

Now suppose you modulate it using 1 MHz signal the carrier frequency is 1 MHz. Now this frequency is translated to 1 MHz that is your 10 to the power 6 plus 20 Hz and this one becomes 20 into 10 to the power 3 plus 10 plus 10 to the power 6 Hz.

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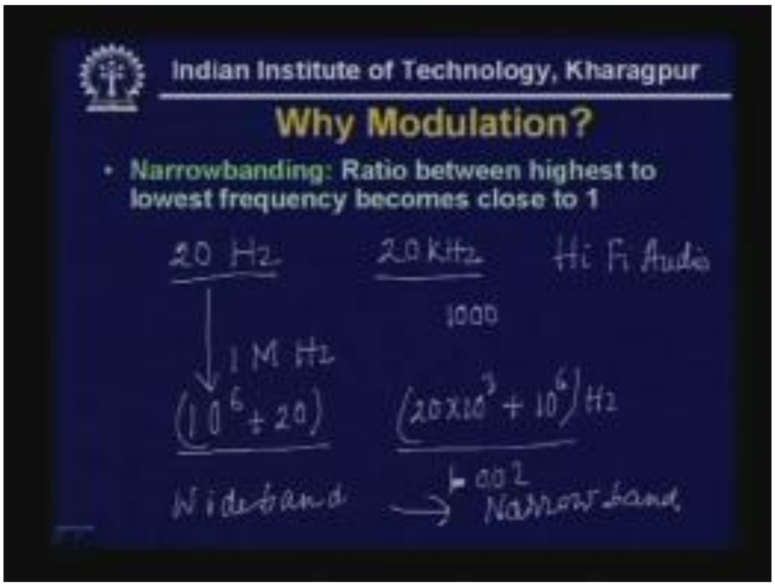

Now if you take the ratio between the two the ratio will be only 1.002. That means if you design an antenna for highest frequency that will be able to send the lowest frequency very easily because as you can see the ratio between the highest and lowest frequency is only 1.002. In other words a signal antenna will be able to transmit both the frequencies efficiently and effectively. This process is essentially narrowbanding.

Here we are converting a wide band signal wide band not in terms of absolute values but 20 Hz to 20 KHz their frequency is not really very high frequencies. However, you may consider it as a wideband in terms of the ratio of the highest frequency to lowest frequency. Now it is translated into a frequency of very narrow band and narrow band having the ratio of highest to lowest is only 1.002. This allows you to transmit very effectively by using suitable antenna. Therefore this narrow banding is another benefit of this modulation.

Third benefit is multiplexing. as you have seen whenever you do the multiplexing suppose here is your original signal having bandwidth say 0 to let's assume this is f_1 to f_2 where f_1 is very close to 0 so this signal is usually called baseband signal.

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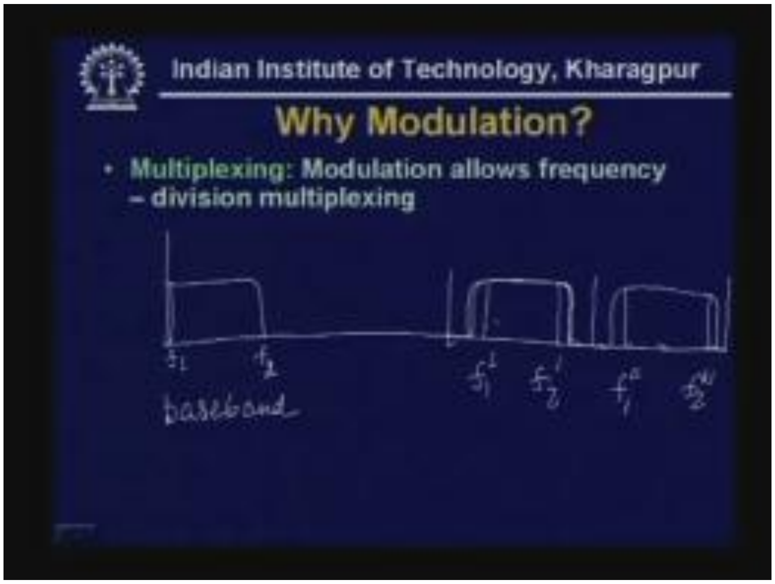

Now you translate to a frequency from f_1 dash to f_2 dash. Now this is converted into a bandpass signal which can be transmitted through a bandpass channel. Now another frequency of the same range can be translated to frequencies such as f_1 double dash to f_2 double dash. Now both of them can be sent through the same channel. As you can see here you can be separate them with the help of bandpass filter at the receiving end.

In other words this allows you to send these two separate signals simultaneously through a transmission media. This process is known as multiplexing and this allows frequency division multiplexing.

As we can see here the frequency range or the bandwidth available is divided like this, this is one part, this is another part and so on so in this way you can divide it into a number of frequency range which can be sent through the transmission media then at the other end the receivers will be able to separate them by suitable filtering. So this is another benefit of this modulation signal that is multiplexing, it allows you multiplexing.

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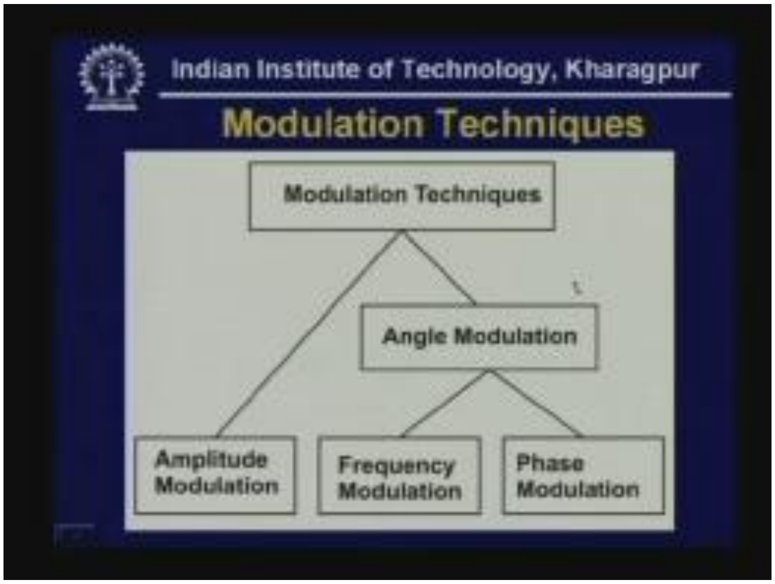

Here are the various modulation techniques that is possible. As I told you will be able to modify one of the three parameters amplitude, frequency and phase. Whenever you vary the amplitude we get amplitude modulation that means amplitude of the carrier, the second alternative is known as angle modulation where you are modifying the frequency that means the frequency of the carrier is modified based on the signal to be sent or modulating signal or it can be phase modulation. These two together are known as angle modulation.

In this lecture we shall primarily focus on amplitude modulation. So here you see we are giving an example of a signal. This is a baseband signal of low frequency. In this case it is a sinusoidal signal and this is your carrier. And here as you can see the amplitude of this carrier has been modified with the help of this baseband signal. That means now here the amplitude is maximum so here you have got maximum amplitude, here it is minimum so you have minimum amplitude so we get a carrier frequency with time varying amplitude this is known as modulated signal.

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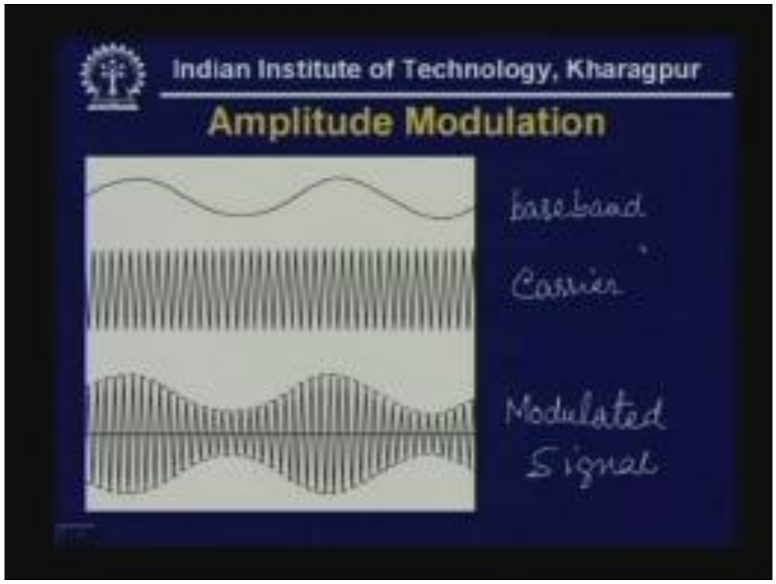

Let us see how it is done. The waveform is represented by this equation $e_m(t)$ is equal to E_m cos (2pi f_mt).

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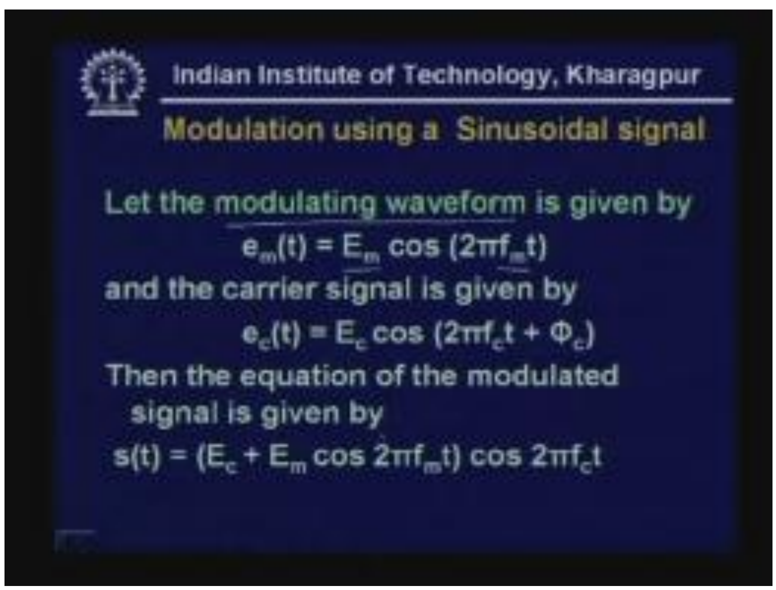

Here we are considering the modulating waveform as a sinusoidal wave with frequency f_m having maximum amplitude E_m and the carrier signal is represented by $e_c(t)$ is equal to E_c cos (2pi f_c t plus phi$_c$).here you see that the maximum amplitude is E_c it has got frequency f_c and a phase difference phi$_c$.

Then the equation of the modulated signal can be given by s(t) is equal to (E_c plus E_m cos 2pi f_mt) cos 2pi f_ct so this is the signal which is generated after modulation. And there is an important parameter called modulation index which is represented by m.

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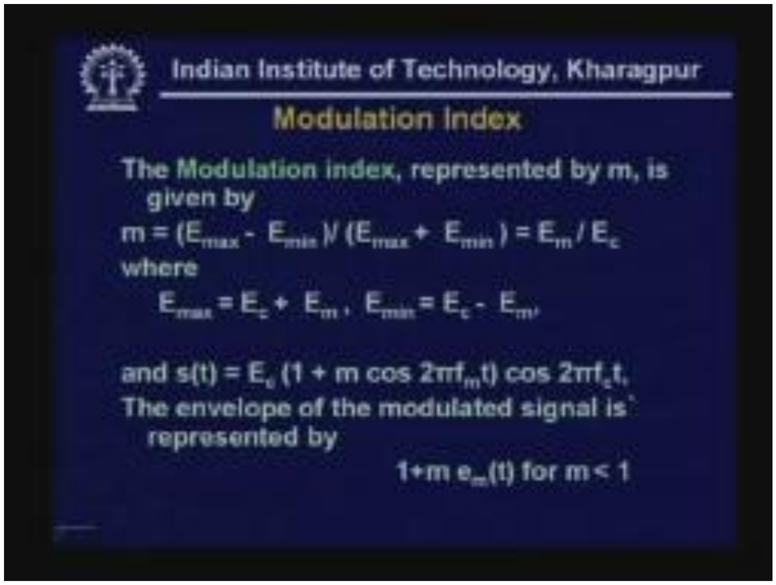

The m is equal to (E_{max} minus E_{min}) by (E_{max} plus E_{min}) is equal to E_{min} by E_{max} if you look at this diagram here you see this is your E_{min} and this is your E_{max} of the modulated signal (Refer Slide Time: 18:26). Now if E_{max} and E_{min} are the maximum and minimum values then modulation index m is equal to E_{max} minus E_{min} by E_{max} plus E_{min}. As you can see here E_{max} is equal to E_c plus E_m that means here this maximum amplitude will be equal to E_{max} is equal to E_m minusE_c.

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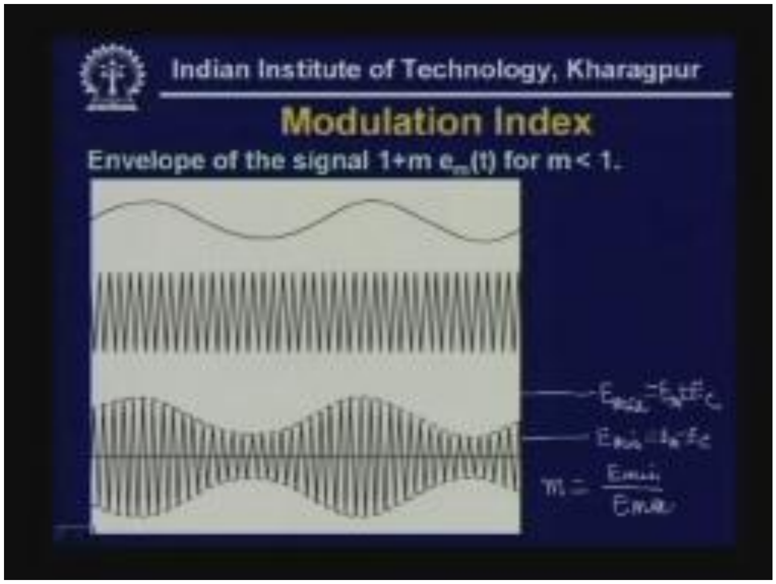

Then modulation index that we get m is equal to (E_{max} minus E_{min}) by (E_{max} plus E_{min}) E_m by E_c. That means in this case we get m is equal to E_m by E_c. In this particular case E_m is equal to given by the maximum amplitude of the modulating signal and E_c is the maximum amplitude of the modulated signal.

Now this kind of waveform you get whenever the value of m is equal to less than one. That means the maximum amplitude of he modulating signal is less than the maximum amplitude of the carrier then we get a waveform like this.

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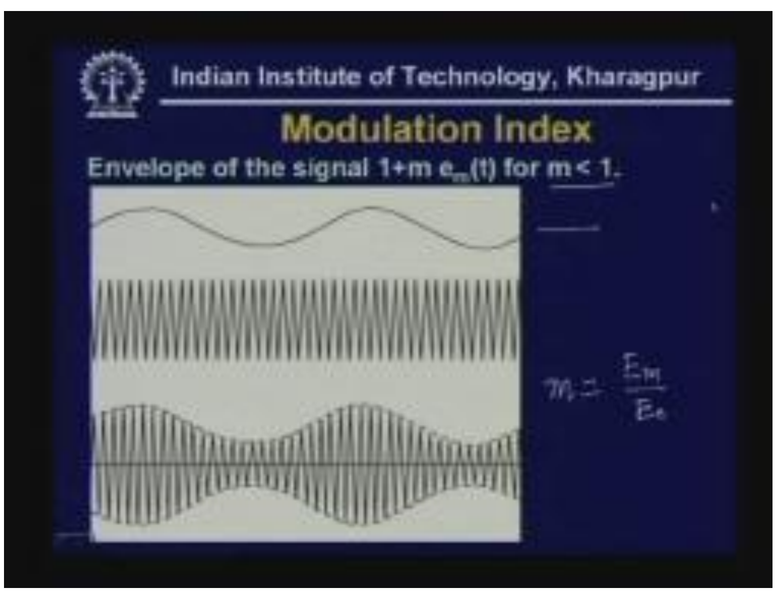

Now let us see what happens whenever you increase the modulation index. As you increase the modulation index or the value of m as you can see here the value of E_m has been increased compared to the previous diagram.

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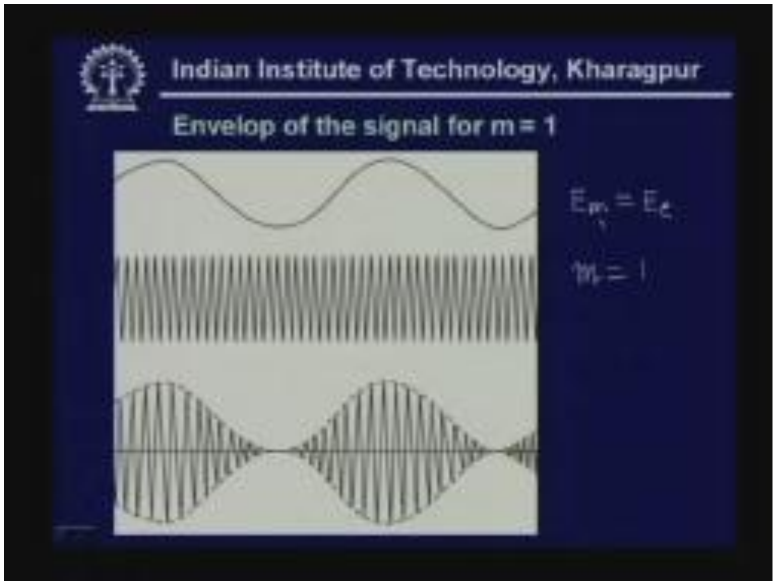

So here what has been done is E_m has been made equal to E_c. So in that case what will what is happening is you are getting the value of m is equal to 1 because m is equal to E_m by E_c.

So, for m is equal to 1 as you can see in this case this signal amplitude of this and amplitude of this is same then the difference is 0and the maximum value is $2E_c$ so it varies from 0 to $2E_c$ so in that case we get the maximum modulation. That means this is the maximum permissible modulation possible because as we shall see in the next slide when you get m is equal to greater than 1 here the value of E_m has been made greater than E_c. So when E_m is greater than E_c then value of m is greater than 1. And in such a case we get a waveform like this.

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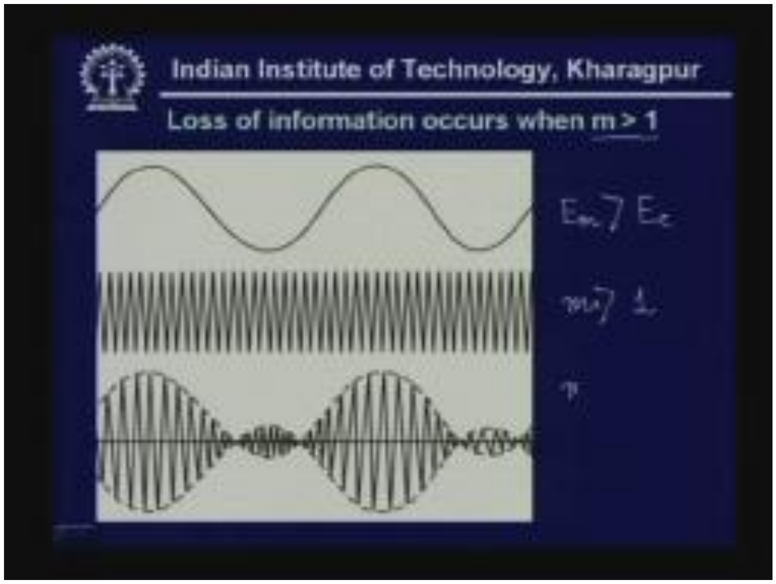

And it can be shown that whenever m is greater than 1 then it is not possible to recover the signal at the other end. That means recovery of the signal will not be possible whenever the modulation index is greater than 1. Let us now look at the frequency spectrum of the modulated signal.

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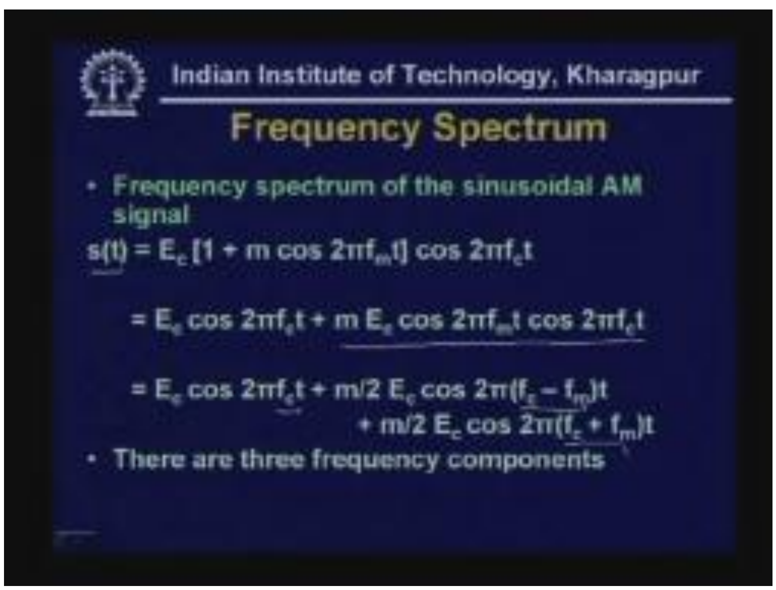

Let us consider modulating using a sinusoidal AM that means the modulating signal is a sinusoidal signal. And in that case the modulated signal will be equal to s(t) is equal to E_c [1 plus m Cos 2pi f_mt] Cos 2pi f_ct which can be expanded to E_c Cos 2pi f_ct plus m E_c Cos 2pi f_mt Cos 2pi f_ct. This particular term can be represented by this form; so m by 2 E_c Cos

2pi f_c minus f_mt plus m by 2 E_c Cos 2pi f_c plus f_mt. So we notice that there are three frequency components. One frequency component is f_c, another frequency component is f_c minus f_m and third frequency component is f_c plus f_m. So after modulating a sinusoidal carrier with the help of a sinusoidal modulating signal we get three frequency components and their respective amplitudes are shown in this diagram.

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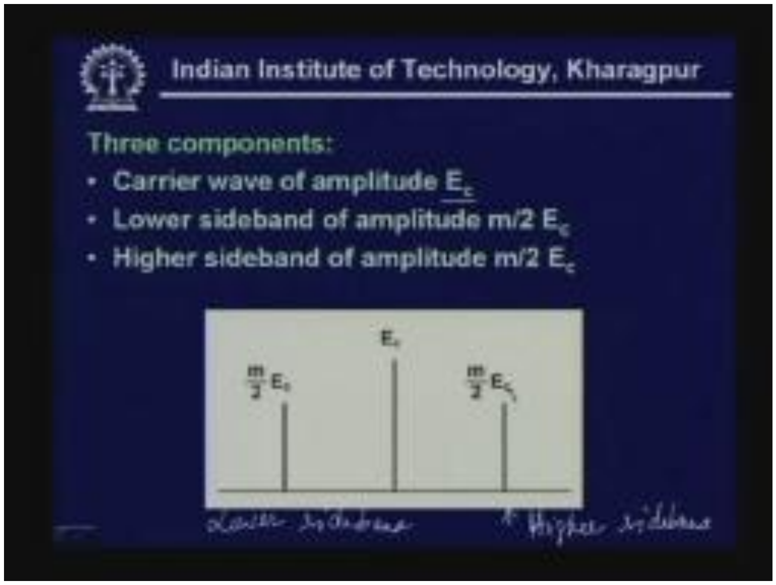

Here we see we get a carrier of peak amplitude E_c, we get lower side band this is known as lower side band (Refer Slide Time: 23:30) this is your carrier and this is your upper side band or higher side band. So this higher side band has an amplitude mc by two Ec so m by 2 E_c, E_c and m by 2 E_c these are the three signals having different amplitudes. So E_c cos 2 pi f_ct and M by 2 E_c cos twice pi f_c minus f_mt so these are the three frequency components.

Since AM is as you know the ratio of E_m by E_c essentially this will be equal to Em by 2 and this will be also equal to E_m by 2. That means the side band frequencies will have amplitude that is equal to Em by 2 because M by 2 into E_c will be equal to as you know m is equal to E_m by E_c so E_c will cancel out and that will give you E_m by 2 (Refer Slide Time: 25:02) that means the amplitude of the side frequencies will be equal to E_m by 2 which is dependent on the amplitude of the modulating signal.

Now what is the effect of this? Let us take with the help of an example.

Suppose here a carrier of 1 MHz and peak value 10V is modulated by 5 KHz sine wave having maximum amplitude 6V determine the modulation index and frequency spectrum. So in this case what is the value of m? Value of m is equal to 6 by 10 where $E_m$is equal to 6 and E_c is equal to 10 so this is 6 by 10 that is 0.6 obviously this is less than 1 that is very good, we get good quality signal which can be recovered. That means original signal can be recovered from the modulated signal that is being received.

Now, what will be the range of frequencies? Hence 1 MHz means 10 to the power 6 minus 5 KHz 5 into 10 to the power 3 to 10 to the power 6 plus 5 into 10 to the power 3.

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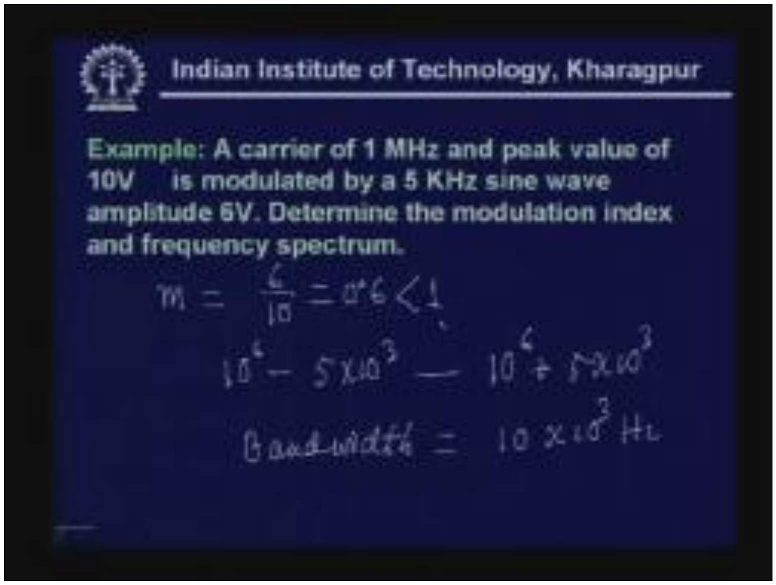

This will be the range of frequencies and bandwidth is equal to is difference of the two that means equal to 10 into 10 to the power 3 Hz or the bandwidth is 10 KHz. So you see that bandwidth is not very high twice that of the modulating signal. So this is shown in this particular diagram.

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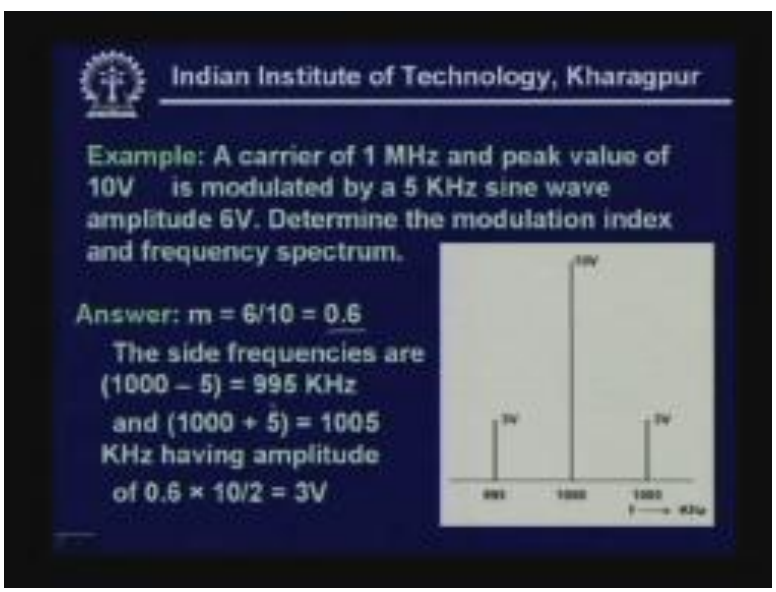

Here we see the modulation index as we calculated 0.6 and the side frequencies are 995 KHz and since we are modulating the sinusoidal other frequencies are not there we shall get three frequency components 995 KHz, 1000 KHz and 1005 KHz.

So these are the frequencies and here we get the three amplitudes of the frequency components, the carrier will have 10V and the side bands will have 3V. This is the frequency domain representation of the modulated signal. Obviously the time domain representation we have already seen. The time domain representation will be somewhat like this (Refer Slide Time: 27:38). On the other hand the frequency domain representation as we can see will have something like this that means three spectral components (Refer Slide Time: 27:47).

So now let us consider the bandwidth whenever the modulating signal is not a simple sinusoidal wave. We are now modulating the signal with the help of audio signal. And in that case the audio signal has a bandwidth of B_m. From almost 0 to B_m is the bandwidth of the audio signal. Now the modulated signal will have bandwidth starting from f_c minus B_m so here is your f_c minus B_m to f_c plus B_m and as you can see bandwidth is equal to 2 B_m that is the bandwidth of the signal. So the bandwidth of the modulated signal is two B_m as it is written there.

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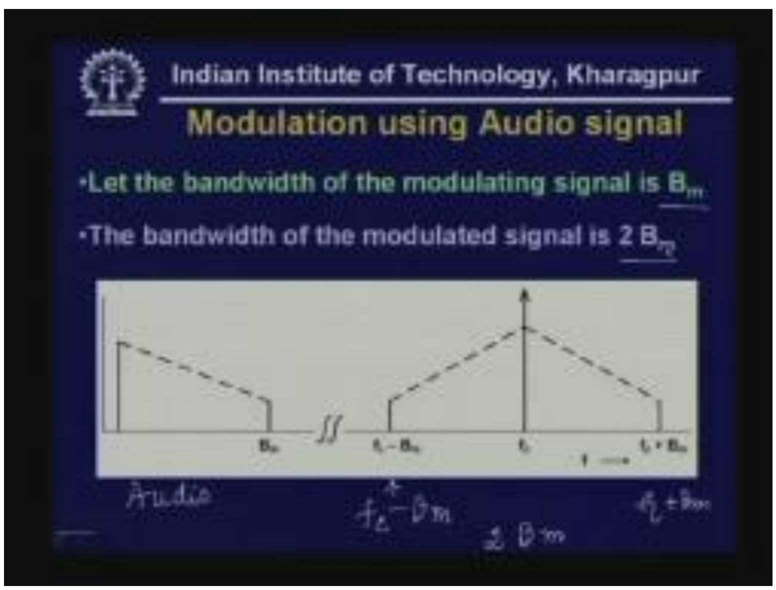

So you see that after modulating a carrier of frequency f_c we get frequency translated signal having twice the bandwidth of the modulating signal. Thus modulating signal has bandwidth Bm so you get twice the bandwidth and of course the amplitudes of different frequencies will be based on the modulation index as we have seen. Now how much power will be associated for transmitting this signal? It is very important to understand the power required for transmission of the analog signal. As we have seen there are three different frequency components. Thus each of the frequency components will require power for transmitting through the antenna.

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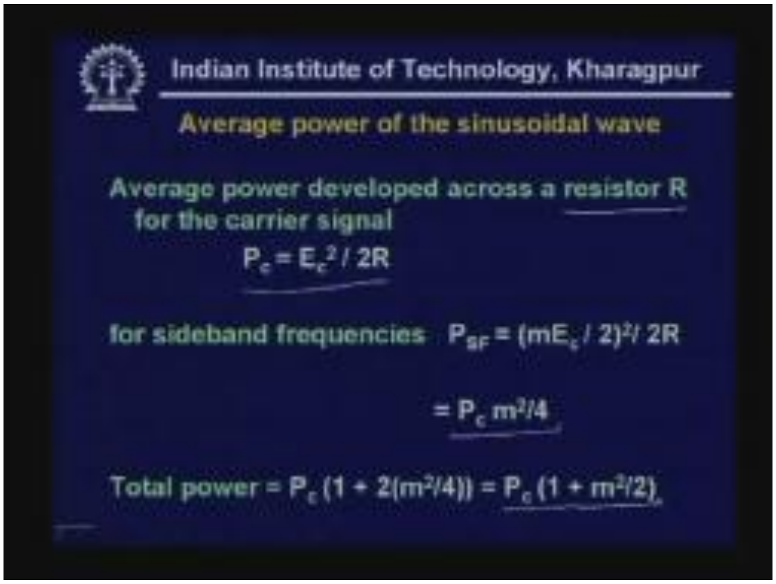

So let's assume that the power is developed across a resistor of value R. Then for the carrier the average power is equal to E_c square by 2R. This is for sinusoidal wave (Refer Slide Time: 30:07) this is voltage m is E by 2 so that is ((30:18)) square by 2R so here it is Pc into M square by 4. Since there will be two side bands to calculate the total average power you have to add the power required for transmission of the carrier and also the power required for transmission of two side bands. So if you add up you get the total power required for transmission that is equal to Pc into 1 plus m square by 2.

Now one very interesting observation from this is that maximum power is required for transmission of the carrier. We have seen in the previous diagram you can see here it is Ec square by 2R and as you can see depending on the modulation index the side band power as you know usually m is less than equal to 1 so this value will be always less than one fourth. That means you are using less than one fourth of the power to transmit one of the side bands. That means half of the power is transmitted to send the two side bands less than half and half of the power you are using to transmit the carrier signal.

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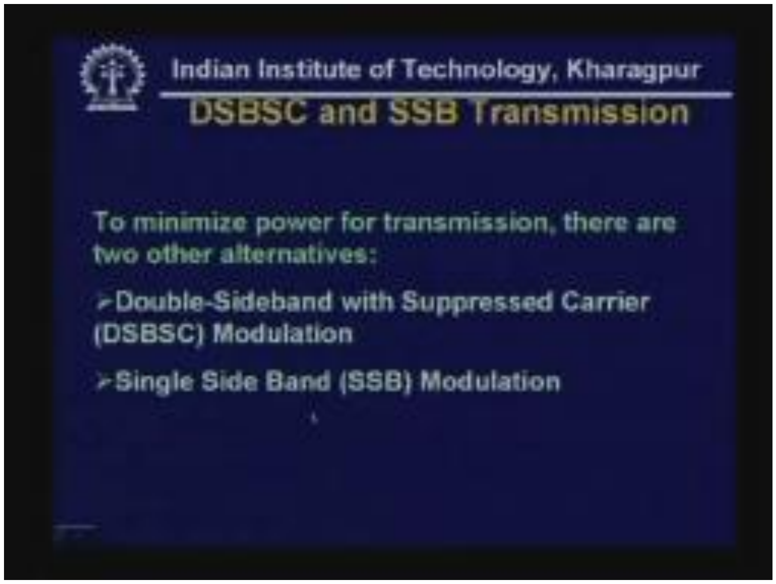

Now for recovery purpose do you really need all the three frequency components. it has been found that we can recover the original signal even if even when the carrier is not transmitted. That means even if we have two side bands then we can recover. So that has lead to what is known as Double Side Band with suppressed carrier modulation. So transmission is possible by suppressing the carrier and by sending only the two side bands.

Another alternative is you can send one of the two side bands. So the original signal can be theoretically recovered. There are some practical aspects that we shall consider later. However, it is possible to recover the modulating signal from one of the two side bands. So if we can send only one of the two side bands and can recover the original signal that is modulating signal that means we shall be able to perform transmission with minimum power.

The power required for transmission power required to derive the antenna will be much less. So this has led to two different types of modulation; one is Double Side Band suppress carrier modulation. This utilizes the transmitted power more efficiently than Double Side Band AM transmission. The normal transmission is known as DSB AM where we are transmitting both the side bands as well as the carrier.

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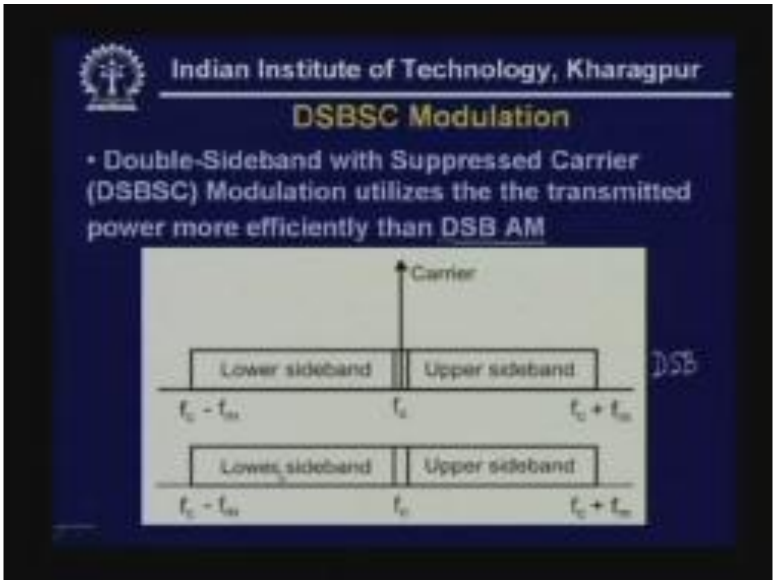

This is your normal DSB transmission Double Side Band transmission. Now the carrier has been removed and as we can see here we have got only two side bands. This is your DSBSC (Refer Slide Time: 34:04) Double Side Band with Suppressed Carrier transmission so you see the carrier is not present. To transmit the signal we will require only less than half the power which is possible by using Double Side Band transmission. so this is the primary advantage of the DSBSC Modulation. However, it has some disadvantage in recovering the signal which we shall discuss later.

As we have mentioned another alternative is to use Single Side Band Modulation. So, in Single Side Band Modulation as you can see you can send either the upper side band or the lower side band. So, in this particular case as we have seen the power required for transmission is not only one fourth less than one fourth I will say so if the modulation index is less than one obviously the power requirement will be less than one fourth and not only you will be able to transmit with lesser power but the bandwidth requirement is also reduced.

Here you see for upper side band bandwidth is f_c 2 f_c plus f_m and for lower side band it is f_c minus f_m to f_c. So whenever you have bandwidth trench you have to multiplex many signals then Single Side Band is the solution. So as you can see the modulated signal can have only bandwidth of B_m and as a consequence this is very efficient in terms of bandwidth as well as in terms of energy that is required for transmission.

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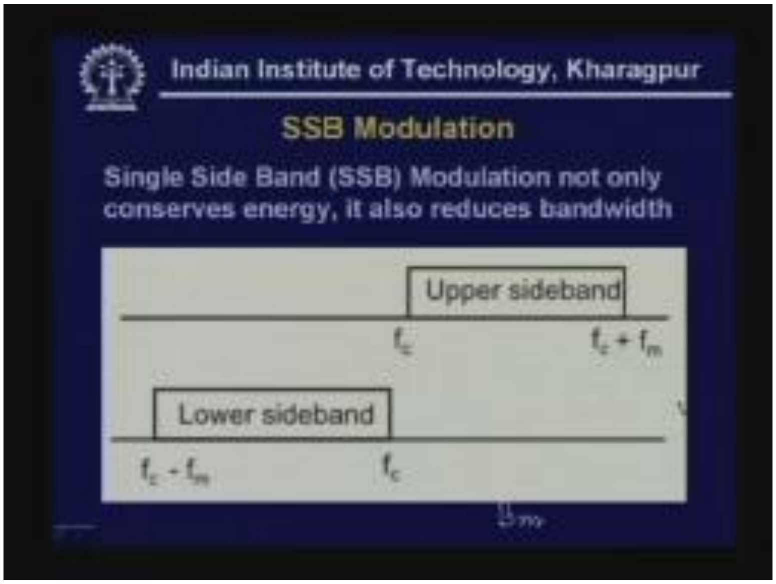

Now let us focus on the recovery of the baseband signal. How do you recover the baseband signal at the receiving end? Obviously we are transmitting signal with the sole purpose of getting back the modulating signal. That means if you are sending audio signal then at the other end we would like to get back the audio signal and they started in its original form.

How that can be done? One approach is by multiplying the signal second time. As you can see let the baseband signal be empty and after multiplication with the carrier the signal is now converted to m(t) Cos 2pi Cos W_ct where W_c is 2pi f_ct. This is the modulated signal m(t) Cos W_ct is that is being transmitted. Now we multiply this with the help of using the carrier signal. So, if we multiply the original signal that is your modulating signal second time with the carrier signal and first time we multiplied with the carrier signal to get the modulated signal now you are multiplying second time so let's see the effect. We get m(t) Cos square W_ct which can be expanded to m(t) 1 by 2 plus ½ Cos2W_ct. So we find that she has got two components one is m(t) by 2 another is ½ m(t) Cos2W_ct

So you observe that the baseband signal has reappeared here. However, not only the baseband signal has reappeared but you have got two other frequency components like 2f_c minus f_m and 2f_c plus f_m. If you expand this 1 by 2 m(t) Cos 2W_ct you will get two frequency components these are 2f_c minus f_m and 2f_c plus f_m.

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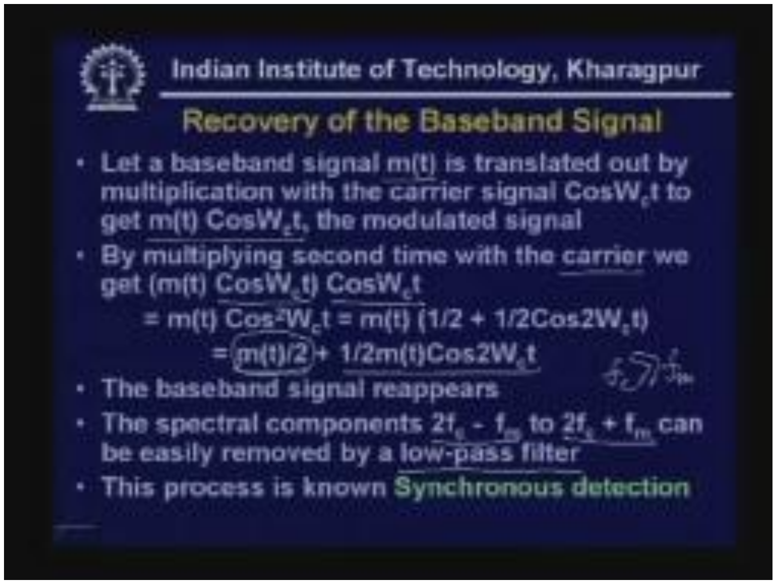

However, normally f_c is much greater than f_m. As a consequence these spectral components $2f_c$ minus f_m to $2f_c$ plus f_m can be very easily removed by using low pass filter. That means if you use low pass filtering then you can get back only the baseband signal so this approach is known as synchronous detection. That means whenever you multiply the modulated signal second time by using a using the carrier you get back the baseband signal after filtering out the other high frequencies such as $2f_c$ minus f_m and $2f_c$ plus f_m and this process is known as synchronous detection. Now there is one important limitation of the synchronous detection. the important limitation is the signal $Cos2W_fc$ has to be precisely synchronous.

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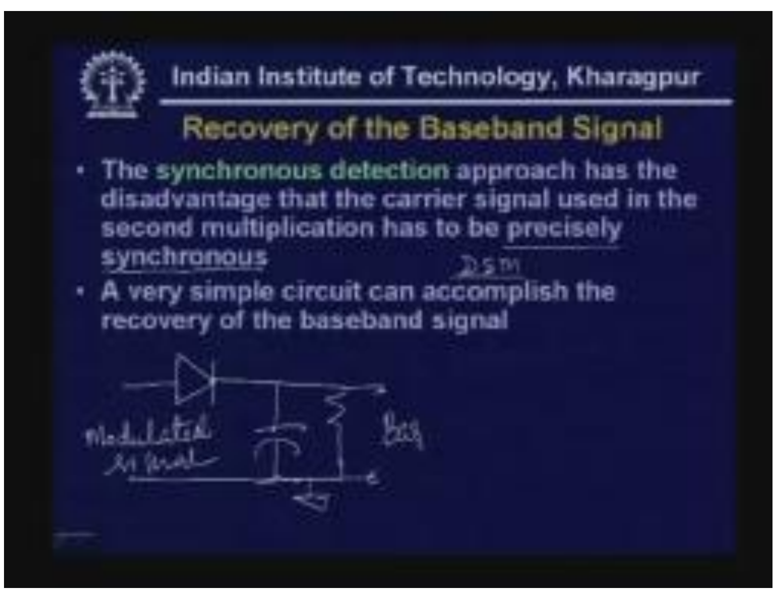

So the synchronous detection approach is straight forward. But it has a disadvantage that whenever you do the multiplication then that carrier signal which you are using for multiplying second time has to be precisely synchronous. That means there should not be any phase difference. If there is any phase difference then you will see that after multiplication the signal that you get will not be the baseband signal.

Question is, how do you get the synchronous signal? You have to use costly hardware to generate precisely synchronous carrier signal at the receiving end. That's why it is better to have normal Double Side Band modulated signal. So if you receive the carrier that carrier can be used to regenerate or generate a synchronous carrier which can be multiplied to get back the baseband signal.

However, if you don't send the signal, if you use DSBSC then it will be very difficult to generate the synchronous carrier at the receiving end. So this is a very difficult process. On the other hand if we use that original DSM signal it can be recovered very easily by a simple circuit like this (Refer Slide Time: 41:40). You can use one diode, a capacitor and a resistor and this can be grounded. So across this you can get back. So here you apply the modulated signal, here you will get back the baseband signal. This type of simple circuit can be used if we use DSM signal. That means there is no need for synchronous detection, you can use this kind of simple circuit as it is shown here.

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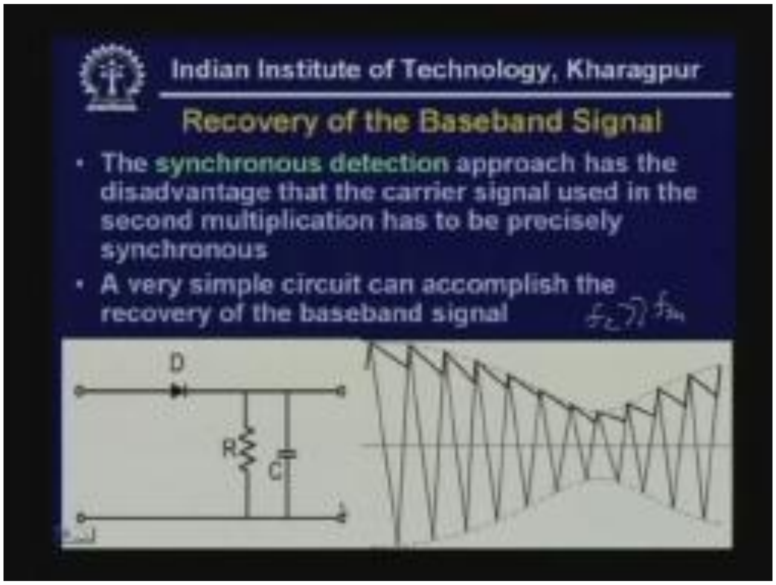

A diode, resistor and capacitor combination can be used to get back the original signal. As you can see here the value of R and C can be chosen in such a way that it will follow the carrier. That means you will get this particular curve which is essentially the baseband signal. And since this frequency is very high (Refer Slide Time: 42:55) f_c is very high than fm this will be quite smooth and you will get a single that baseband signal recovered here by using this kind of simple circuit. This is known as detection with the help of a diode and this is commonly used in many situations.

Of course the modulated signal received at the receiving end is greatly attenuated. As you know the signal is passing through a long distance and the attenuation is proportional to 1 by distance square. And as a result if the receiver is at a long distance the signal will be highly attenuated. Not only that the atmosphere is always generating some noise when there is lightening, spark and various other atmospheric disturbances. so at the receiving end the signal to noise ratio is very poor and also the signal level is very low.

Moreover, there may be many other channels adjacent to the signal. That means we will be using some kind of frequency division multiplexing. Thus a number of channels will be transmitted and they can be very close, side by side, so in such cases recovering the signal may be little difficult.

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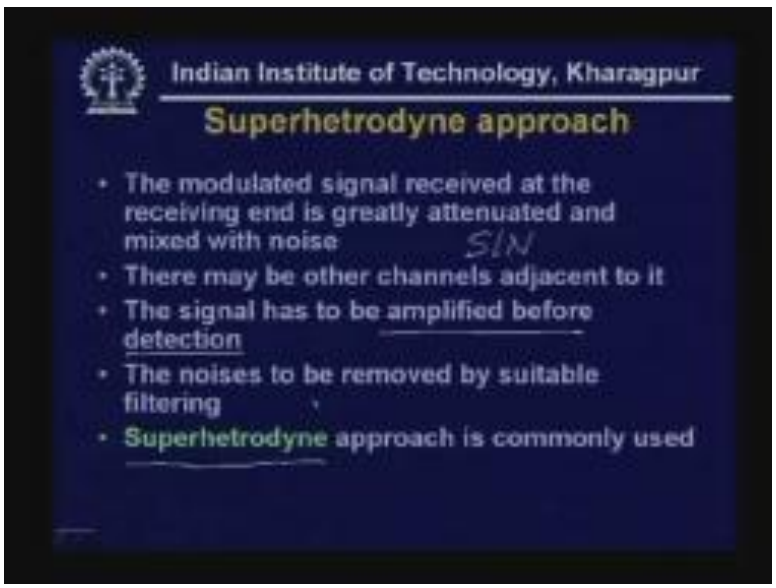

So, whenever the signal is highly attenuated and you have got many channels adjacent to it. The signal has to be amplified before you can do detection and the noises are to be removed by suitable filtering. And for this purpose one approach that is commonly followed is known as superhetrodyne approach.

Let me explain the superhetrodyne approach with the help of the block diagram that is commonly used.

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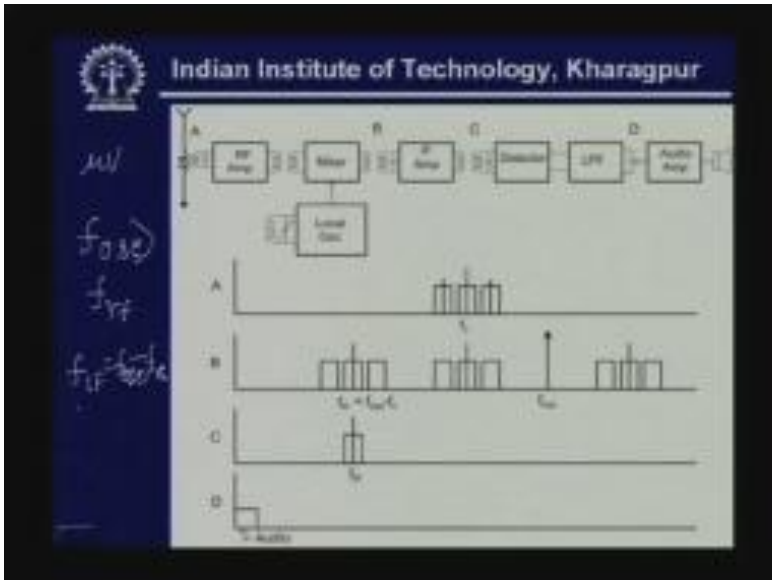

Here you have got an antenna, this is the complete receiver circuit (Refer Slide Time: 45:16), here is your antenna where you are receiving signal of the order of may be microvolt or less, then there is an RF amplifier. RF amplifier is essentially amplifying the carrier signal so this is tuned to the carrier frequency. So here there is a tune circuit, tune amplifier that is amplifying the RF carrier.

Then in the superhetrodyne technique what you do is you use a local oscillator having frequency Fose. This local oscillator frequency is greater than greater than the frf, it is greater than the RF frequency. So this local oscillator frequency is mixed with the amplified carrier signal amplified RF signal and whenever you do that as you know the signal that we have received is shown here which has the carrier frequency and two side bands.

Now whenever you do that mixing with the help of a local oscillator which has higher frequency than frf that's why it is called superhetrodyne. if this oscillator frequency less than the rf then it is not superheterodyne (Refer Slide Time: 46:47) and whenever you multiply a mix with f_c and this modulated carrier you will get frequency spectrum like this, you will get intermediate frequencies equal to fose minus f_c and fose plus fc. Then you get frequency fif is equal to fose minus fc and also fose plus fc these two will be there.

Now, what you do is you simply filter that fif. That intermediate frequency is filtered with the help of intermediate frequency amplifier. So intermediate frequency amplifier not only it filters out this frequency component but it also amplifies it. So it does the filtering and amplification together and then it is applied to a detector. So here it is applied to a detector that detector will generate the baseband signal. Thus the baseband signal and of course that RF the radio frequency noises can be filtered out with the help of low pass filter as shown here then you will you will be applying to an audio amplifier.

So, after detection and low pass filtering you get back the audio signal as it is shown here which is then amplified and applied to a loud speaker.

This is a typical AM receiver that we use in our houses; this is a common household item.

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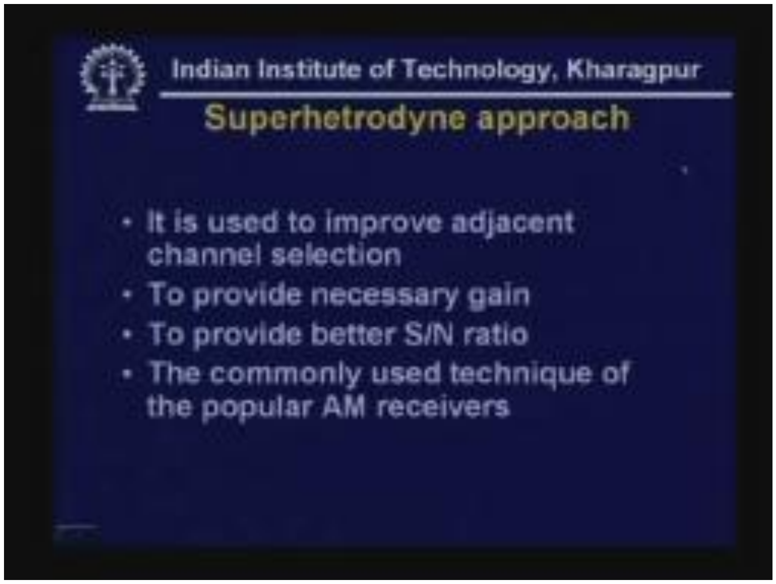

Superhetrodyne approach provides us a number of benefits. First of all it is used to improve adjacent channel selection, it provides necessary gain because as you can see amplification is done at different stages. So here it can be microvolt then it is here millivolt mVwhenever you do the detection and here it can be of the order of Volt. So microvolt to Volt a gain of about 1000 is performed as it goes from RF to audio. Then it provides a better signal to noise ratio because of the filtering and also there is tune filtering where only the carrier frequency is received so the noise is rejected and we get good quality signal. This is the commonly used technique of popular AM receivers.

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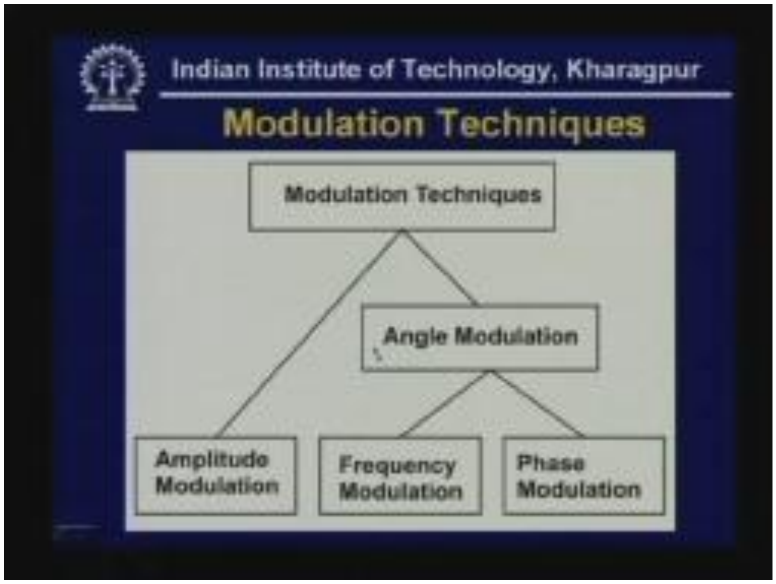

We have discussed the amplitude modulation technique, we have discussed what modulation index is, the frequency spectrum, the power required for transmission and we have also discussed how the signal can be recovered. And in the next lecture we shall discuss about the angle modulation and as I have mentioned it has got two different versions; frequency modulation and phase modulation and this is how it will look like, there will be no change in amplitude but frequency is modulated.

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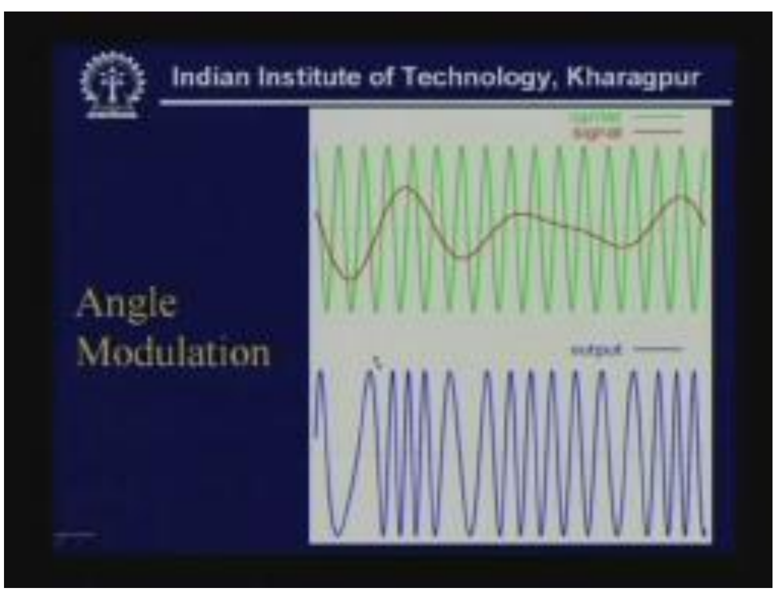

Here are the review questions based on this lecture.

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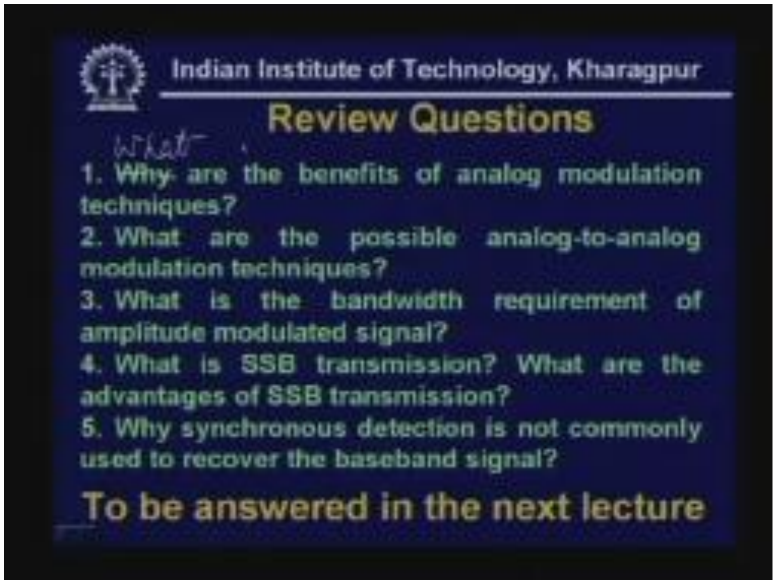

1) What are the benefits of analog modulation techniques?

2) What are the possible analog to analog modulation techniques?

3) What is the bandwidth requirement of amplitude modulated signal?

4) What is Single Side Band transmission? What are the advantages of SSB transmission Single Side Band Transmission?

5) Why synchronous detection is not commonly used to recover the baseband signal?

Now it is time to give the answers of the previous lecture.

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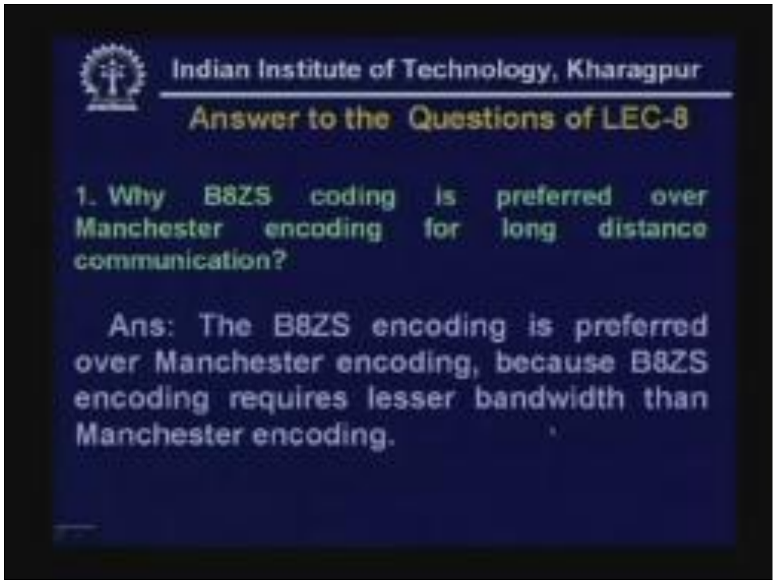

1) Why B8ZS coding is preferred over Manchester coding for long distance communication?

As you know the B eight ZS encoding is preferred over Manchester encoding because B8ZS encoding requires lesser bandwidth bandwidth as we have seen is equal to the bandwidth of the baseband signa. But the Manchester encoding has bandwidth about almost twice the baseband signal. So we see that B 8ZS has lesser bandwidth which is useful that’s why it is preferred for long distance communication.

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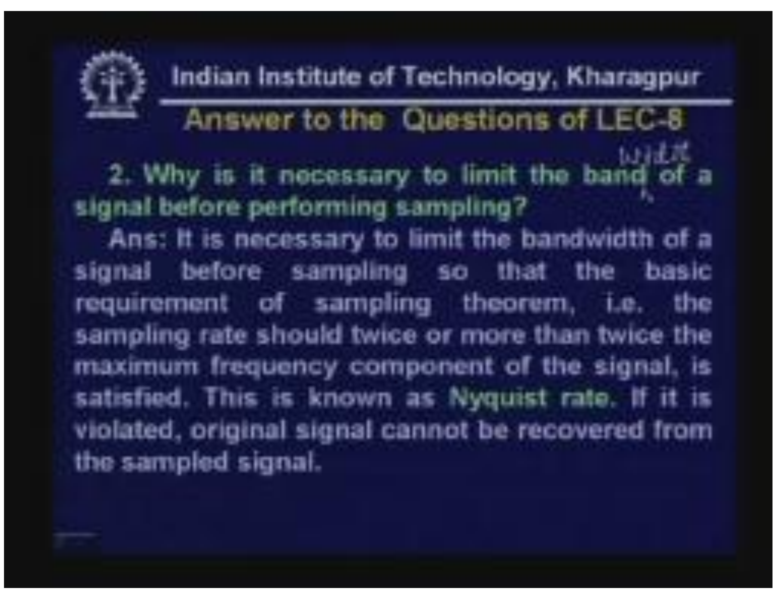

2) Why is it necessary to limit the bandwidth of a signal before performing sampling?

It is necessary to limit the bandwidth of a signal before sampling so that the basic requirement of sampling theorem that is the sampling rate should be twice or more than twice the maximum frequency component of the signal that is satisfied. This is known as Nyquist rate as I have already discussed. If it is violated original signal cannot be recovered from the sampled signal so it will suffer kind of distortion which is known as aliasing error.

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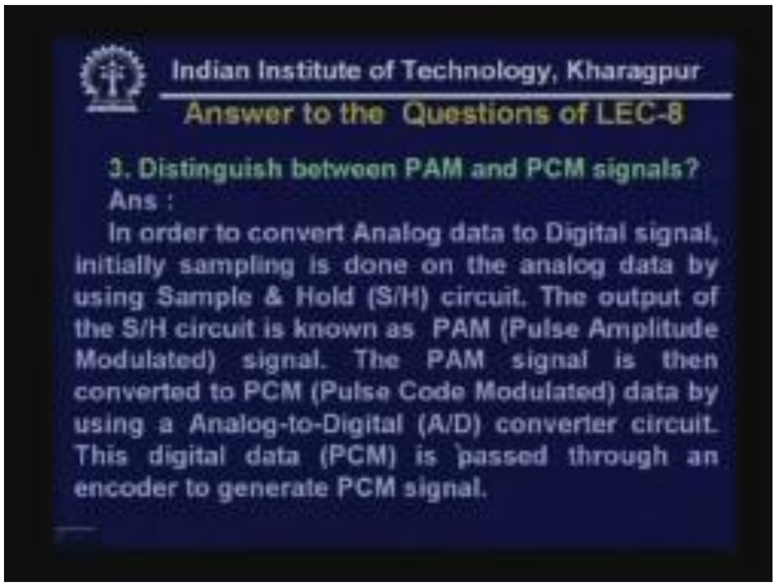

3) Distinguish between PAM and PCM signals.

In order to recover or in order to convert analog data to digital signal initially sampling is done on the analog data by using sample and hold circuit. The output of the sample and hold circuit is known as PAM signal. The PAM signal is then converted to PCM signal. So PAM is essentially an intermediate step to get the PCM signal. After you have got the PCM signal an analog to digital converter is use to quantize the signal then you use an encoder to generate the PCM signal, line encoding is done. Thus PAM is essentially an intermediate step for generating PCM signal.

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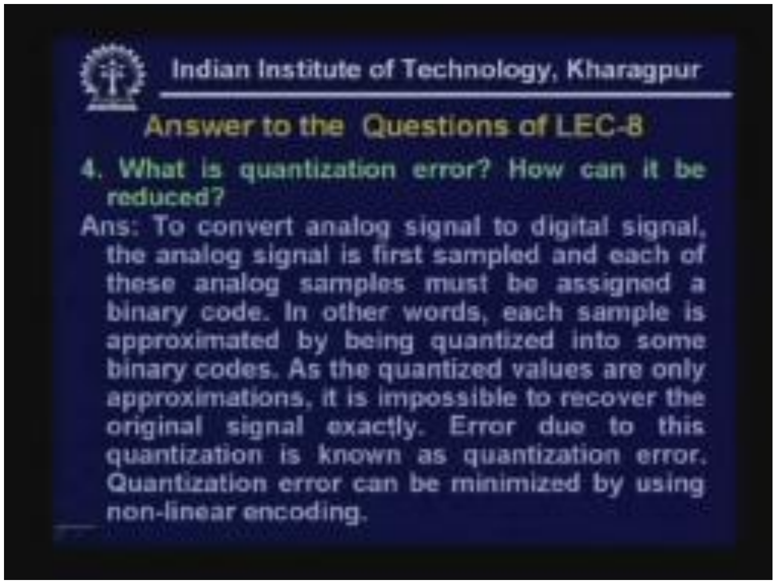

4) What is quantization error? How can it be reduced?

To convert analog signal to digital signal the analog signal is first sampled and each of this analog samples must be assigned a binary code. In other words each sample is approximated by being quantized into some binary codes as we have already seen. As the quantized values are only approximations it is impossible to recover the original signal exactly and this leads to quantization error .and quantization error can be minimized by using non linear encoding as we have already discussed.

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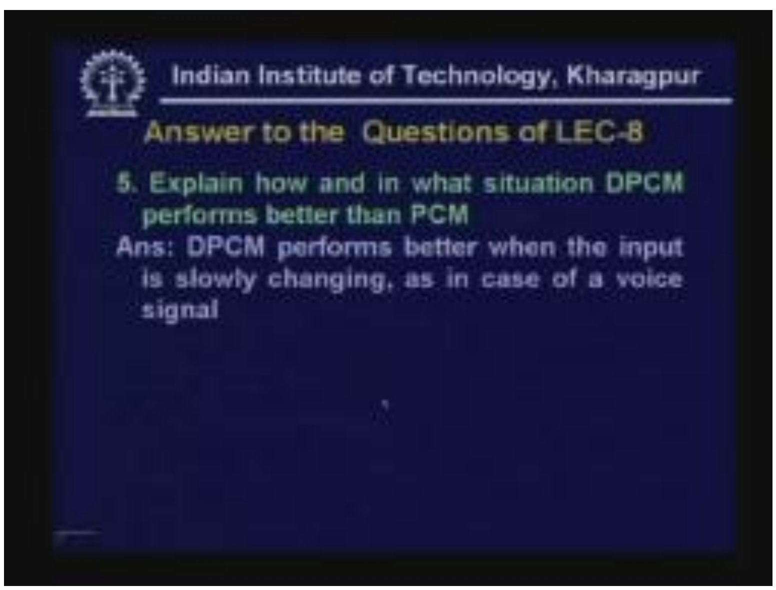

5) Explain how and in what situation DPCM that is differential PCM performs better than PCM?

DPCM performs better when the input is slowly changing as in case of voice signal. As you have already seen whenever the signal is slowly changing then the differential PCM will require very small number of bits. And as we know the delta modulation is the extreme case where you require only one bit. However, if the signal is not slowly changing then this approach cannot be used. So, only when the signal is slowly changing this can be used, the PCM has a better performance than PCM. So with this we come to the end of today's lecture, thank you.