Data Communication
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Lecture 36
Multimedia Networks

Hello and welcome to today's lecture on multimedia networks. Multimedia communication is considered to be the holy grail of data communication.

No lecture series or course on data communication is complete without a discussion on multimedia communication. I shall cover the various aspects of multimedia communication in three lectures. This is first lecture on multimedia communication and in this lecture I shall go for the following topics.

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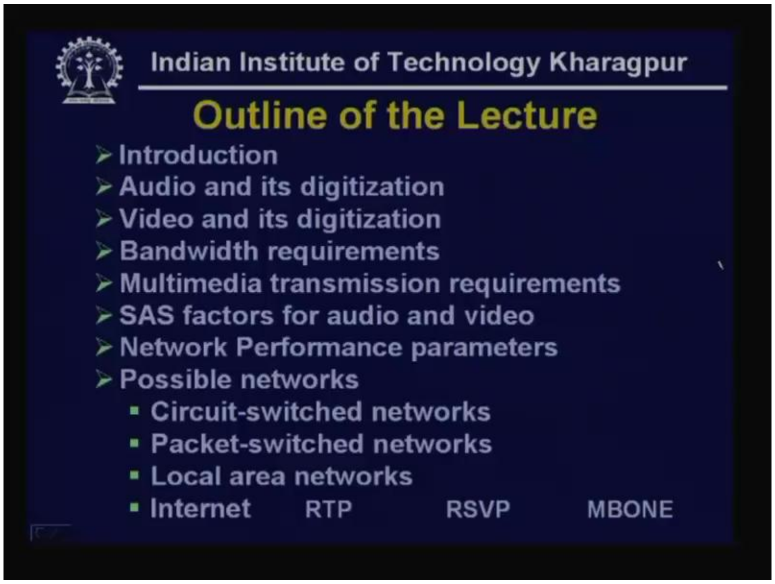

First I shall give an introduction about mentioning what is multimedia, then we shall consider how audio and video which are the two important multimedia which can be digitized, then the bandwidth requirements after digitization, then we shall discuss the multimedia transmission requirements and SAS factors for audio and video and based on this the network performance parameters required for transmission of multimedia will be discussed and we shall explore the possibility of multimedia communication through different types of networks such as circuit-switched networks, packet-switched networks, local area networks and internet.

And as we shall see that there are many limitations we have to overcome by adding some functionalities or protocols such as RTP, RSVP and MBONE which we shall cover in this lecture.

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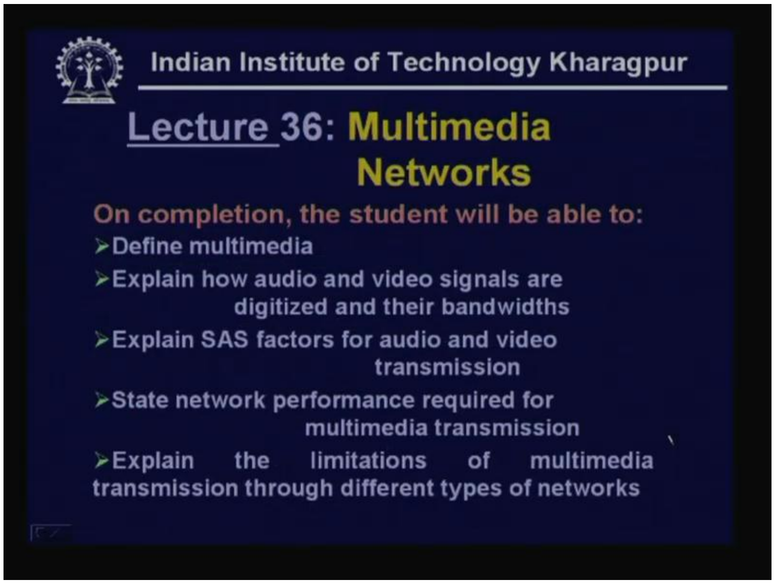

On completion the student will be able to define multimedia that means what you really mean by multimedia, they will be able to explain how audio and video signals can be digitized and what their bandwidths are, they will be able to explain the SAS factors, I shall go into the details of this for audio and video transmission and they will be able to state the network performance required for multimedia transmission and then they will be able to explain the limitations of multimedia transmission through different types of networks as I mentioned.

Now let us come to the definition of multimedia. Multimedia stands for more than one continuous media such as text graphics, audio, video and animation. In this context this presentation that I am making can be considered as multimedia presentation because I am using text, I am using graphics and I am using animation.

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However, commonly when two or more continuous media are played during some well-defined time interval with user, interaction is considered to be multimedia and the most demanding of them are audio and video. So we shall primarily focus on audio and video although text, graphics, animation are also considered as part of multimedia.

What you mean by audio?

By audio we mean what we hear through our ears, video is what we see through our eyes. So let us now see how we can get the digitized audio signal because we have to communicate through computer network which is digital in nature in most of the situations now.

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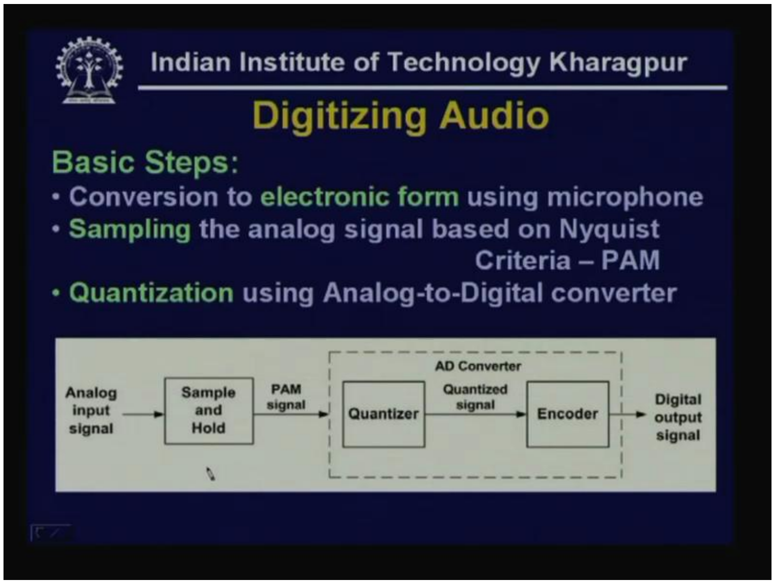

As a consequence the analog audio signals are to be digitized. First let us deeply consider how it is being done. As you know the sound waves are converted to electronic form using microphone. Microphone is used to convert sound waves in our ear to electronic form which is obviously analog in nature that analog signal is converted into digital form by using several steps.

First thing to be done is sampling. so you have to do this sampling of the analog signal based on Nyquist criteria and we get what is known as PAM Pulse Amplitude Modulated signal which we have already discussed earlier then another an step is performed which is known as quantization. Quantization is performed with the help of analog to digital converter and we get a digital output in terms of zeros and ones. So here it gives you how it is being done for a sine wave, here is the sine wave (Refer Slide Time: 5:55) so the sine wave is sampled and sampling frequencies have to be twice the maximum frequency that means 2 f max where f max is the maximum frequency of the signal and after sampling here it is quantized because you have to represent the digital value by using some discrete numbers and then these are converted into digital form by analog to digital converter.

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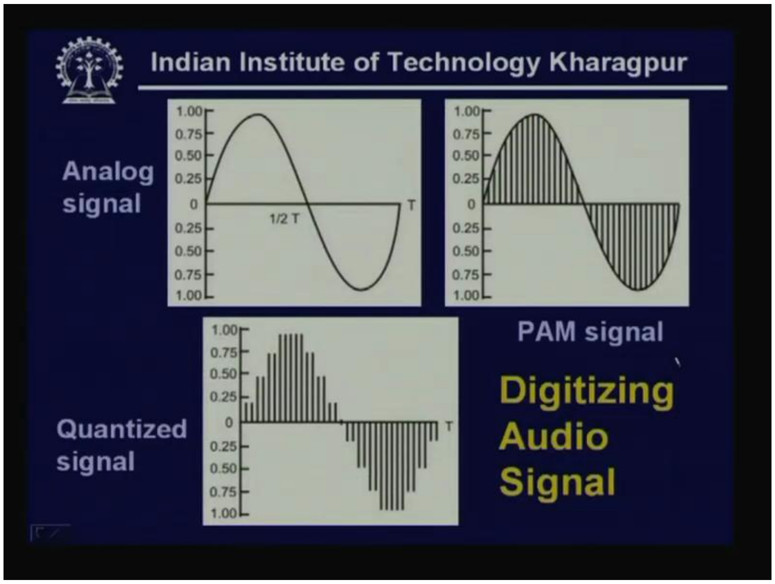

Let’s assume here considering voice and since the voice frequency component remains up to 4 KHz we can convert it to digital form by sampling at 8 KHz.

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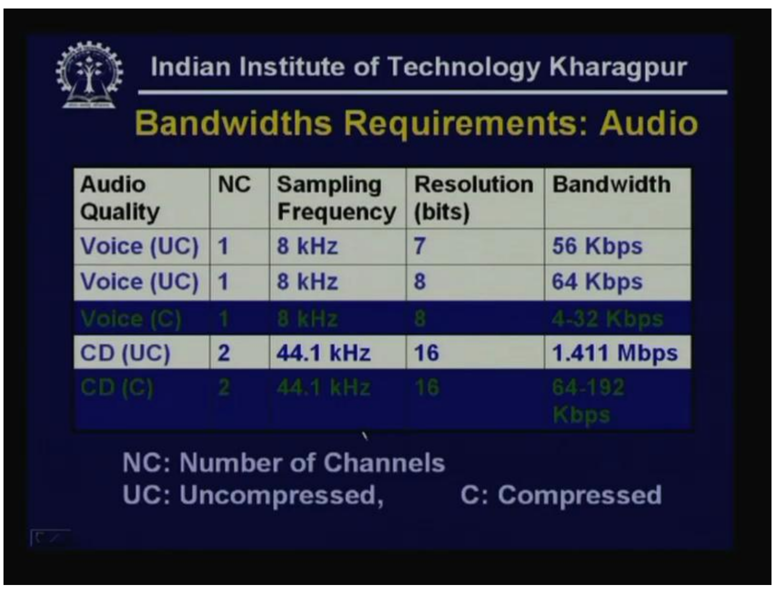

So when you sample at 8 KHz and use a 7-bit analog to digital converter the bandwidth required for communication of this digital voice is 56 Kbps. On the other hand, using sampling frequency of 8 KHz and a resolution of 8-bit that means ADC has the resolution of 8-bit we get a bandwidth of 64 Kbps.

Whenever we have to record music on CD the higher sampling frequency has to be used because music can have frequency components as high as 20 KHz that is the maximum limit of work here. So as a consequence the sampling frequency is 44.1 KHz and for resolution if we use 16-bit then it gives you 1.411 Mbps where a quite high data rate is generated. Of course whenever we record in CD usually it is stereophonic so there are two channels and in case of voice it can be a single channel that’s why we have to multiply the data rate by 2 to get 1.411 Mbps.

And normally as we shall see this uncompressed data both for audio and video has got higher bandwidth so some compression is used so that it can be efficiently communicated through the transmission media or the network. In this case, for example, whenever you compress voice that 64 Kbps bandwidth becomes 4 to 32 Kbps depending on the compression technique and nature of the voice used.

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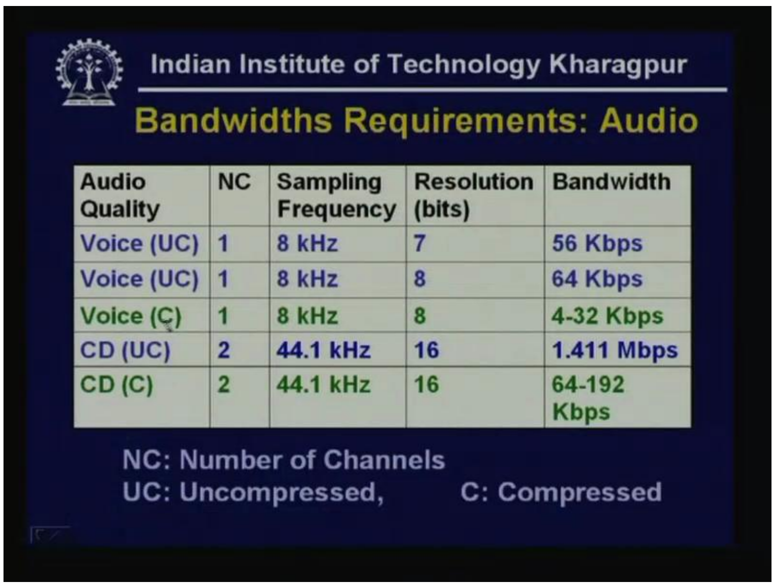

Similarly, the CD data whenever it is compressed from 1.411 mega bits it comes down to 64 to 192 Kbps which can be transmitted at a much lower cost. So compression is indispensable as we shall see not only for audio but particularly for video it is more important.

Now let us focus on video. The best way to understand electronic video signal is to understand how we get a picture from a digital camera. Nowadays all of us are familiar with digital camera, the digital camera has got electronic sensors that is called Charge Coupled Devices which is nothing but some semiconductor sensor which converts different levels of light to electronic signals of different amplitude so the light intensity is converted into some electrical signal of different amplitude.

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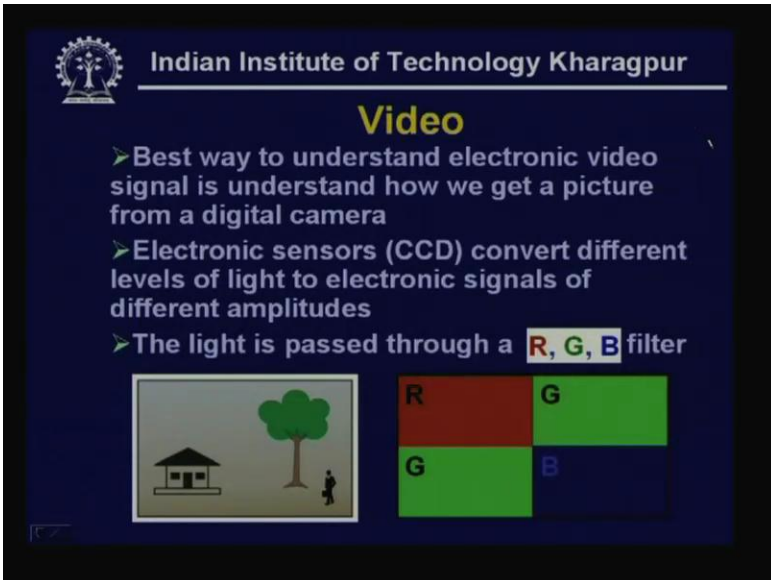

If it is color then the light is passed through an RGB filter as it is shown here and then you get three different components red, frequency and green from the same video. For example here you have got an image and this image can be converted into a digital form. And whenever we consider video it can be considered as a sequence of frames. for example, this is one frame, this is another frame and when these frames are displayed fast enough we get the impression of motion, as you know we use the retentivity property of our eyes whenever the frames are flashed on the eye at the rate of say 50 frames per second we don't identify the changes that's means it becomes continuous to us that's why to get flicker free display the frames are repeated fifty times per second so that is how we get video so essentially it is multiple frames per second.

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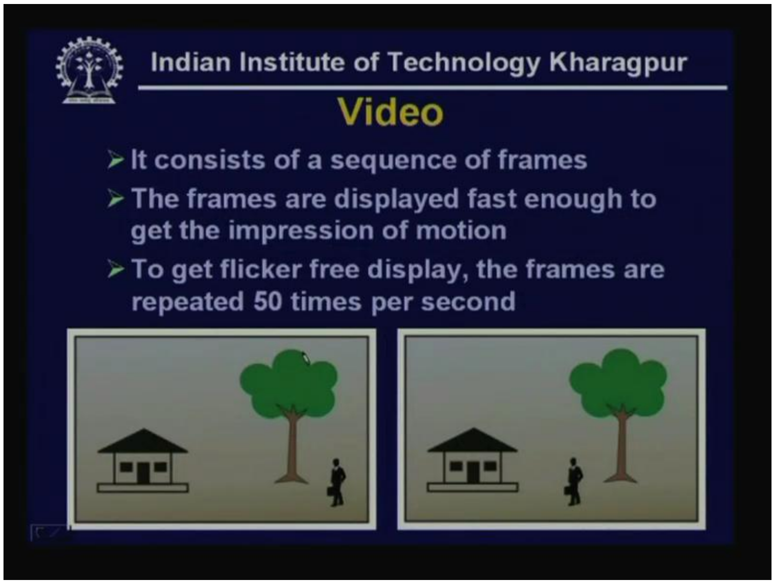

How do you digitize it? What is being done is each frame which is an image which is divided into small grids called pixels and for black and white TV grey level of each pixel is represented by 8 bit.

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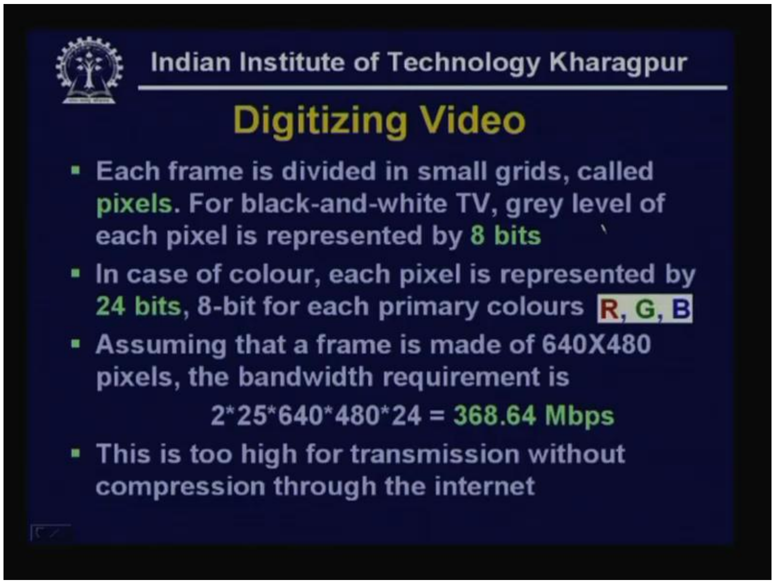

So if you represent by 8-bit then each pixel will give you 8-bit data, in case of color each pixel is represented by 24-bit so 8-bit for each primary colors so it is 8-bit for R red, 8-bit for G green and 8-bit for B blue components. In that case for each pixel you'll require 24 bits.

For example, let's assume a frame is made of 640 by 480 pixels which is essentially used storing video signals in CD, the bandwidth requirement for this is 2 into 25 into 640 into 480 into 24. Actually it is repeated 25 frames per second two times and these are the number of pixels and here is the number of bits per pixel which gives you 368.64 Mbps so quite a high data rate is necessary and this is too high for transmission without compression through internet. So compression is a must particularly for video. Here an example is given. This is a frame and it is divided into 640/480 pixels which you get from a digital camera.

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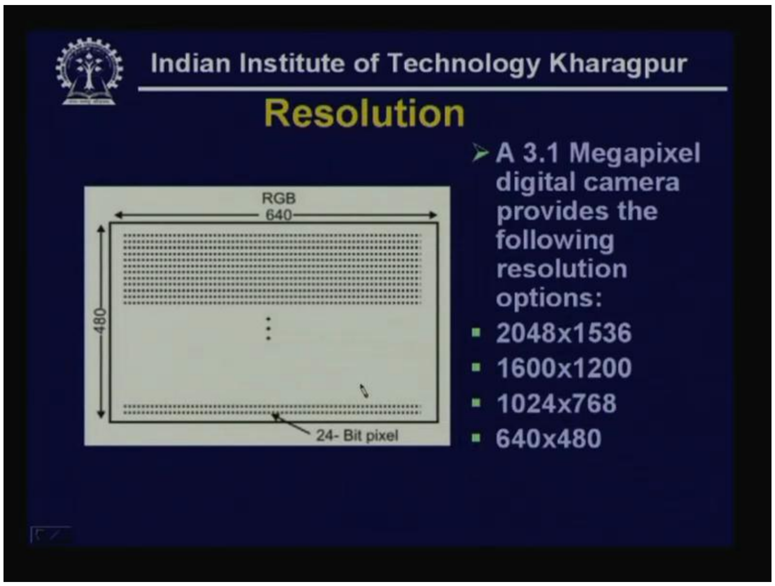

For example, a 3.1 mega pixel camera digital camera provides the following resolution. In the highest resolution you get 2048 into 1536 pixel that means in this direction 2048 and this direction y direction you get 1536 and if you want the lower resolutions then it can be 1600 into 1200 or 1024 into 768 or it can be 640 into 480.

Now, for different applications we want different resolutions. For example, for high density TV, HDTV the number of pixels required is or horizontally it is 1920, for vertical it is 1080 and using 24-bit as the resolution and 60 frames per second that gives you 2986 Mbps so it is quite high and it cannot be sent by using most of the networks.

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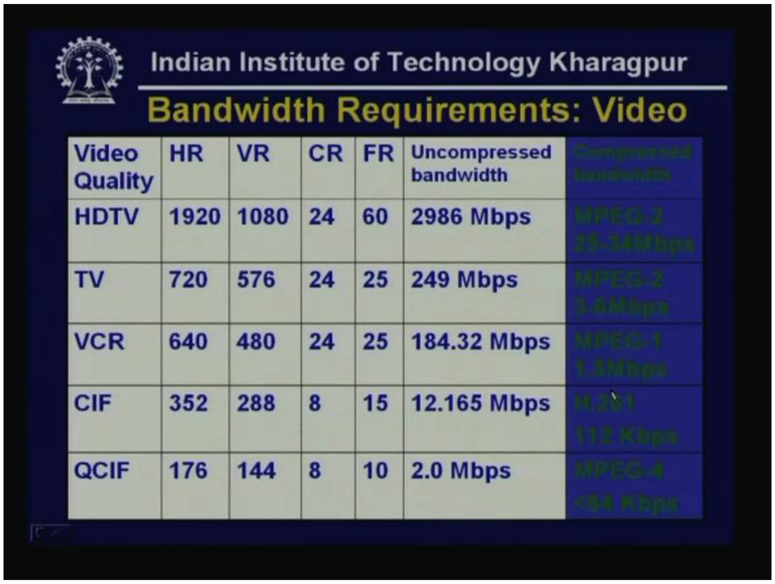

For our standard TV you require 720 into 576 pixels and for each pixel giving 24 bits and 25 frames per second gives you 249 Mbps that is quite high. For VCR it is 640 into 480 and again by using 24-bit and with 25 frames per second we get 184.32 Mbps. There are two other standard which are CIF and QCIF which are primarily used for video conferencing where the data rates are small, I mean the pixel number of pixel is used is smaller or of lesser bandwidth, for CIF it is 352 into 288 and 8-bit is used for giving resolution and 15 frames for per second gives you 12.156 Mbps. And for QCIF for low resolution multimedia and video conferencing 176 into 144 is the frame and 8-bit per pixel and 10 frames per second gives you 2 Mbps. Thus it varies from about 3000 Mbps to 2 Mbps for different applications, so this requires compression and with compression from 2986 Mbps it gets converted into 25 to 34 Mbps this can be transmitted through many computer networks using Mpeg 3.

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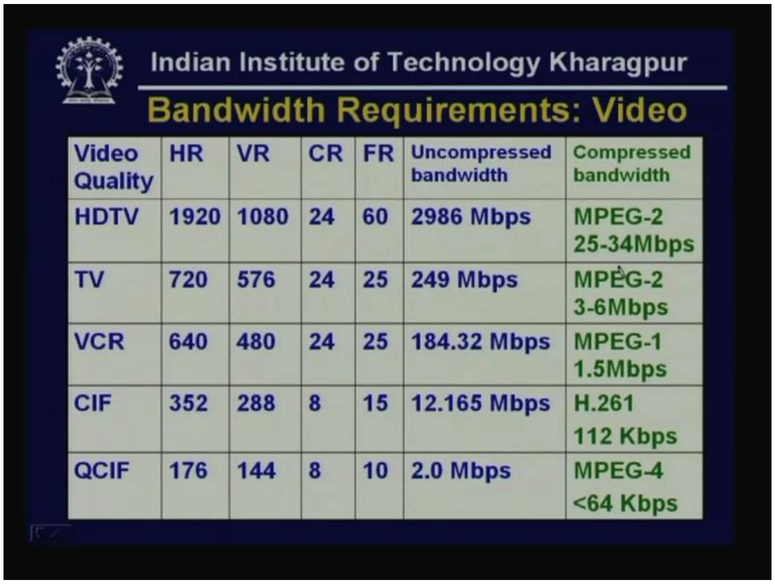

Video Quality	HR	VR	CR	FR	Uncompressed bandwidth	Compressed bandwidth
HDTV	1920	1080	24	60	2986 Mbps	MPEG-2 25-34Mbps
TV	720	576	24	25	249 Mbps	MPEG-2 3-6Mbps
VCR	640	480	24	25	184.32 Mbps	MPEG-1 1.5Mbps
CIF	352	288	8	15	12.165 Mbps	H.261 112 Kbps
QCIF	176	144	8	10	2.0 Mbps	MPEG-4 <64 Kbps

Your TV quality signals which requires 249 Mbps in uncompressed form gets converted to 3 to 6 Mbps by using Mpeg 2 technique and VCR quality signal requiring 184.3 Mbps in uncompressed form can be converted into 1.5 Mbps by using Mpeg 1 so the bandwidth required is greatly reduced.

For CIF you can have h.261 compression technique to give you a data rate of 112 Kbps and for QCIF by using Mpeg 4 you can generate less than 64 Kbps data rate. Hence, after compression the bandwidth is becoming much smaller which can be communicated through many networks that we have already discussed.

Now let us look at the qualitative requirements needed for multimedia transmission. For identifying qualitative requirements two things are necessary. First one is the response of the human ear which is to know the frequency range.

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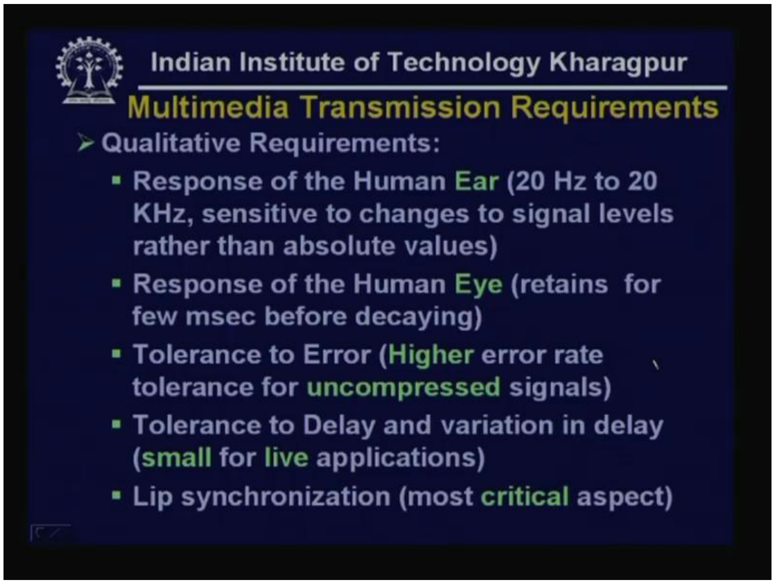

As you know our ear can hear from 20 Hz to 20 KHz, which is the range of frequencies our ear can hear. Of course but dogs and other animals it can be higher frequencies. One important property of our ear is it is more sensitive to the changes of the signal levels rather than the absolute values. That means the absolute values are not that important but the changes are more important so it is more sensitive to changes rather than the absolute values. So this characteristic of the ear has to be utilized or exploited when we do multimedia transmission.

Similarly, whenever we look at the response of the human eye we have to utilize the property of our eyes. Whenever some images is passed on eye it retains that information for few milliseconds before it dies down that is called the retentivity of the eye and that can be exploited for compression and later on for communication. Then there is some tolerance of error. There will be some error, error will take place as you communicate through some network and particularly higher error rate tolerance is there for uncompressed signal and whenever it is compressed then it is less tolerant to error.

Later on as we discuss the compression techniques in the next lecture it will be evident why whenever an error occurs it is less tolerant for compressed and it is not that much tolerant for uncompressed signals.

Tolerance to delay and variation in delay: Here also as we shall see there is small tolerance for live applications. So whenever it is not live application then you can tolerate larger delay or a variation in delay.

Possibly the most critical aspect is the lip synchronization. Normally voice and video these two are recorded separately compressed separately and transmitted separately but it is necessary to have synchronization so that there is lip synchronization. When somebody is singing then lip synchronization has to be done with the audio and if it is not there it will look very odd that's why this lip synchronization is the most critical aspect for multimedia communication. so the audio and video signals are to be synchronized. Later on we shall see how it is being done.

The performance parameters for multimedia communication is expressed in terms of a parameter known as synchronization accuracy specification factor SAS factor and this SAS factor is specified with the help of four parameters and essentially it specifies the goodness of synchronization.

In multimedia essentially two continuous streams of data is coming; one for audio another for video then it needs to be synchronized. Now, how good the synchronization is is being expressed with the help of the SAS factor which is represented by four parameters.

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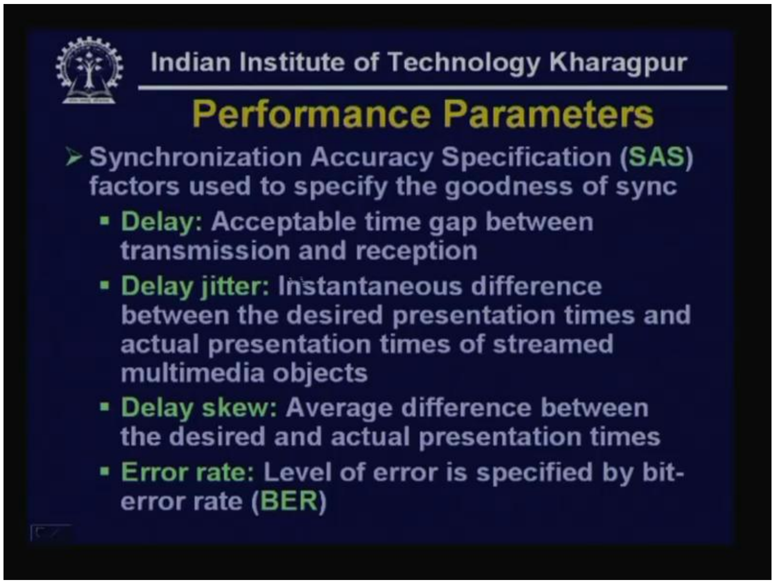

One is delay. Delay is essentially the acceptable time gap between transmission and reception. Here you have got a transmitter, it is passing through a network and it is going to the receiver. This is your receiver, and this is your transmitter (Refer Slide Time: 21:20) and obviously there will be some delay depending on the communication media. If it is satellite it will have long delay, if it is local area network the delay will be very small so depending on the network the delay will vary.

The second important parameter is delay jitter. Delay jitter is essentially the instantaneous difference between the desired presentation times and the actual presentation time of the streamed multimedia objects. Let us see with the help of this diagram.

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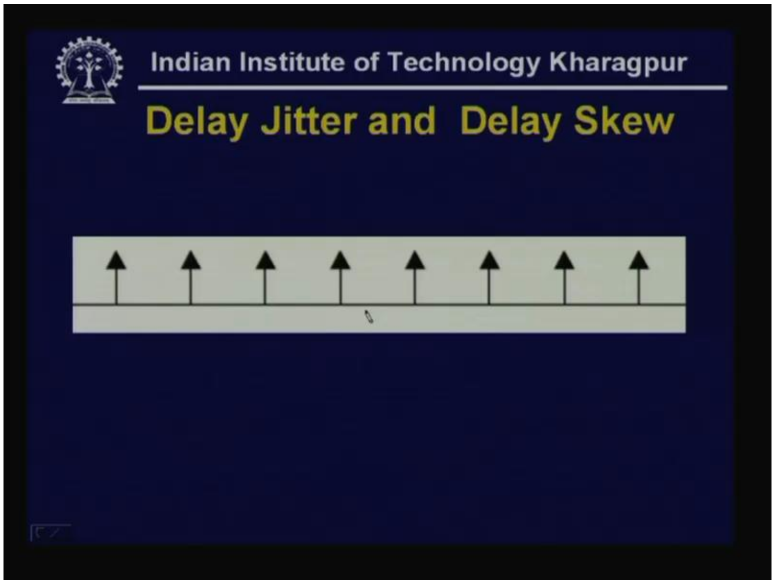

Suppose these are the presentation times for different multimedia objects, now because of the variation of delay it may reach here or it may reach here or it may reach somewhere here or it may reach somewhere here (Refer Slide Time: 22:09) so at different points with respect to the original or actual presentation times. This is called the delay jitter. Delay jitter represents the variations in delay, the instantaneous difference between the desired representation times and the actual presentation times of the streamed multimedia objects.

On the other hand, the delay skew is the average difference between the desired and the actual presentation times; it gives you the average value. For example, here immediately after synchronization it reaches here, then it reaches little later or here it reaches still little later here, it reaches still little later here, it reaches still little later and so on. that means the skew the difference is increasing the average value is increasing with time so in this direction you have got your time so this is being explained by the delay skew, this parameter specifies the delay skew.

Fourth parameter is the error rate. Some error is committed when the digital signal goes through the transmission media or the network which s represented by the bit error rate. That means the number of bits in error bits in error and total number of bits that give you the bit error rate. So, with the help of these four parameters the synchronization accuracy specification is specified.

Now let us consider it in case of audio. In case of audio particularly for conversation one-way delay should be 100 to 500 millisecond range and should not be more than this which requires echo cancellation.

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Then the delay jitter has to be ten times better than the delay. For example, if the delay is 100 milliseconds then the delay jitter should be less than 10 milliseconds so it should be ten time better than the delay. Then lip synchronization should be better than 80 milliseconds that means the time gap between the audio objects and the video objects should be less than 80 milliseconds and bit error rate should be less than 0.01 for telephone that is voice and less than 0.001 for uncompressed CD or less than 0.0001 for compressed CD. Therefore, the bit error rate requirement is more for compressed CD than the uncompressed CD.

Now let us look at the SAS factors for the video.

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For video the delay and jitter requirement is less than 50 milliseconds, for HDTV High Density TV, it is less than 100 milliseconds for broadcast quality TV, it is less than 500 milliseconds for videoconference. And error rate is less than point 0.000001 for HDTV, less than 0.00001 for broadcast TV and less than 0.00001 for video conference. So, for HDTV the bit error rate requirement is more stringent than the videoconferencing.

Now, let us focus on the traffic characterization parameters. Traffic characterization parameter arises because of the variability of the bit frames.

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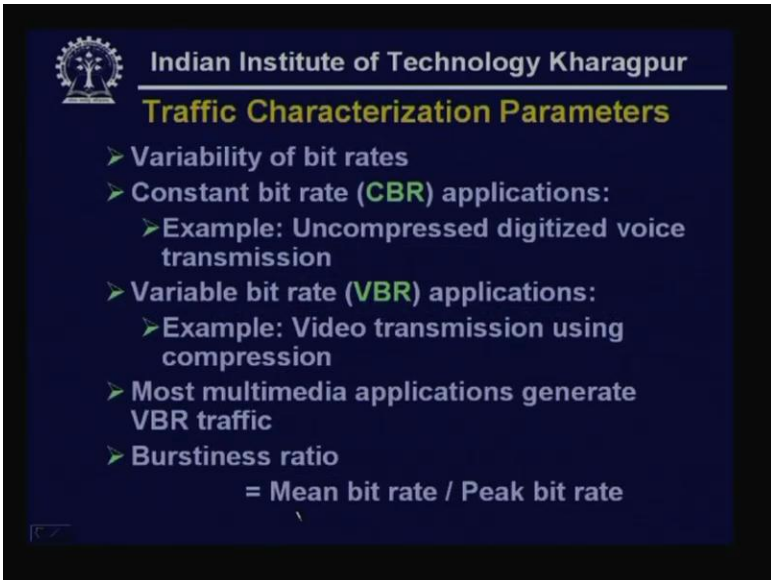

It is divided into two categories. One is constant bit rate applications, for example, uncompressed digitized voice transmission. Whenever voice is transmitted or video is transmitted in uncompressed form it gives you a constant bit rate so that is being explained by constant bit rate. On the other hand, whenever audio or video is compressed different parts of the video or the different parts of the audio will not be compressed by the same amount, the compression ratio will not be same for different parts of the audio or video so as a consequence it will generate variable bit rate so video transmission using compression leads to variable bit rate.

And particularly we'll see that most multimedia applications generate VBR traffic that is Variable Bit Rate traffic. This variable bit rate traffic is the cause for burstiness in the traffic and this burstiness ratio is expressed as the ratio of min bit rate by the peak bit rate. That means the min bit rate which is the average bit rate by the peak bit rate. So this ratio gives you the burstiness of a particular application. Now, having discussed the parameters required for multimedia transmission now let us focus on to the networks. The performance of networks is specified with the help of network performance parameters or NPPs. again here you have four parameters first one is the throughput.

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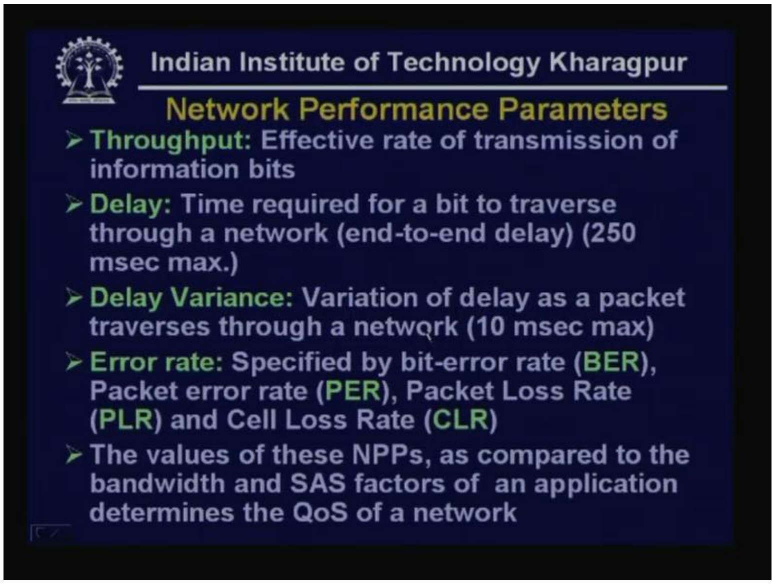

Throughput is the effective rate of transmission of information bits. For example, Ethernet has the data rate of 10 Mbps. However, although the data rate is 10 Mbps the throughput is much less; it can be 3 Mbps you may be asking why. The reason for that is in Ethernet as you know there may be collision and because of collision there will be retransmission or delay in transmission so because of all these things the overall throughput the rate at which the data will be delivered is much less than 10 Mbps so that's why throughput can be lesser than the data rates.

Then comes the delay, delay is the time required for a bit to traverse through the network, it is essentially the end-to-end delay. As I explained end-to-end delay is important and depending on network you are using it can be different.

For example, for satellite communication the round-trip delay is quarter of a second, for LAN it will be very small and for other wide area networks WAN it will be less than one quarter of a second but definitely it will be much higher than the local area networks LANs. And maximum that can be tolerated for network performance parameters is 250 milliseconds that is specified and delay variance that is the variation of delay as a packet traverses through the network. This can happen because of various reasons. whenever an application is sending a data it has to be packetized, there will be transmission time and there will be propagation time, there are three factors; packetization, transmission time and propagation time and because of these three parameters and their variations in some cases it will be store and forward type and because of all these things there will be delay variance and this variation of delay is expressed as the delay variance and it should have maximum value of 10 milliseconds.

Then error rate is specified in various ways. First one is bit error rate which is essentially the number of bits in error per unit time then packet error rate which is the number of packets in error per unit time and packet loss rate is the number of packets lost per unit time and cell loss rate is the number of cells lost per unit time. So, depending on the network it can be packets, it can be cells etc. For example for ATM it will be number of cells sent per unit time and the number of cells in error is expressed as cell error rate.

These values of network performance parameters are compared to the bandwidth and SAS factors. So the bandwidth and SAS factors of the application of multimedia transmission have to be compared with the values of the network performance parameter to determine the QoS Quality of Service of the network. So Quality of Service of the network will be determined by comparing these two. So, quality of service has to be identified and better the quality of service good is the network.

Let us now consider different types of networks and see how capable they are in for multimedia communication. First we shall focus on circuit switched networks. We have already discussed different types of circuit switched networks. The first one and the most popular one is the Public Switched Telephone Network or PSTN.

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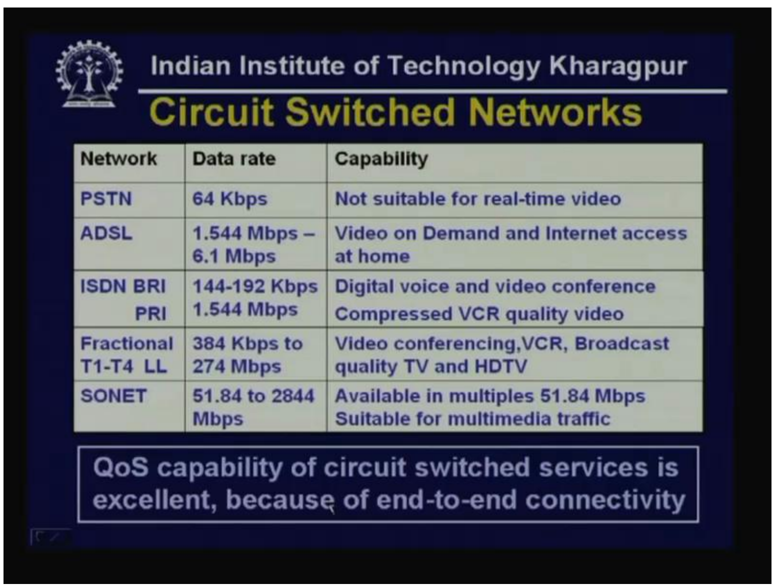

We have already come out of the era of plain old telephone service which was analog. Now in most of the places it has been replaced by digital that’s why I am telling PSTN Public Switched Telephone Network which gives you data rate of 64 Kbps. But definitely this bandwidth or data rate is not enough for real-time video so this is excluded for multimedia communication. However, we have seen that by ADSL which is the broadband service based on the public switched network t can give a bandwidth data rate of 1.544 Mbps to 6.1 Mbps over a small distance so this can be used for video on demand and also for internet access at home.

So this ADSL which is based on PSTN can be used for multimedia communication. Then comes the ISDN, ISDN has got two different interfaces BRI Basic Rate Interface and Primary Rate Interface giving the data rate of 144 to 192 Kbps or 1.544 Mbps respectively and this BRI interface is suitable for digital voice and video conference and PRI is suitable for compressed VCR quality video.

We have not discussed ISDN so let me very quickly go through the ISDN and give you an overview of ISDN before we come to other circuit switched networks.

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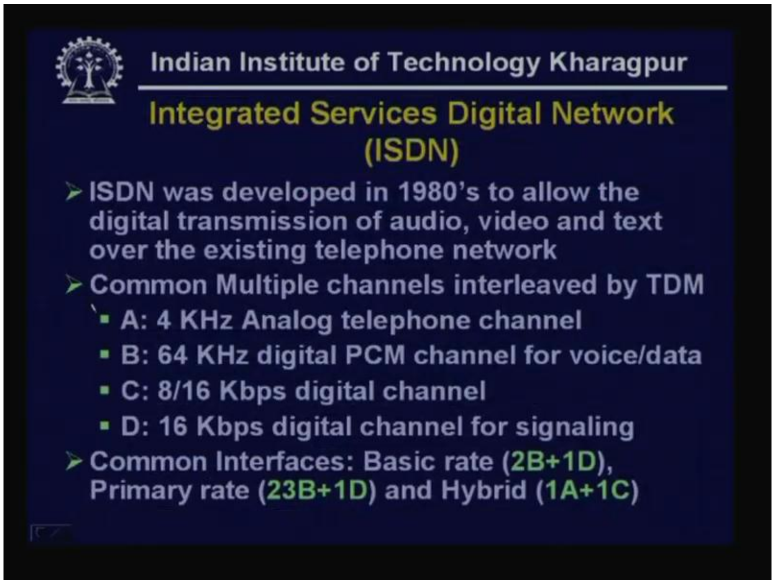

ISDN was developed in 1980s to allow the digital transmission of audio, video and text over existing telephone networks. So the basic idea was to convert into digital form instead of keeping it analog. It used common multiple channels interleaved manner by using time division multiplexing. There were four popular types of channels. First one is A type which is 4 KHz analog telephone channel, B type 64 KHz digital PCM channel for voice and video, C type 8 or 16 Kbps digital channel then D type 16 Kbps digital channel for signal, this is primarily used for out of band signal which we have discussed in detail.

As I mentioned there are basic types of interfaces of which two are more popular. Basic rate gives you 2B type and 1D type so these three channels are interleaved and in primary rate 23B type and 1 D type are interleaved together and in hybrid 1A and 1C these two are interleaved together. So this hybrid one is considered as replacement of this pure and old telephone system namely the analog one. So here it gives you the two types of interfaces.

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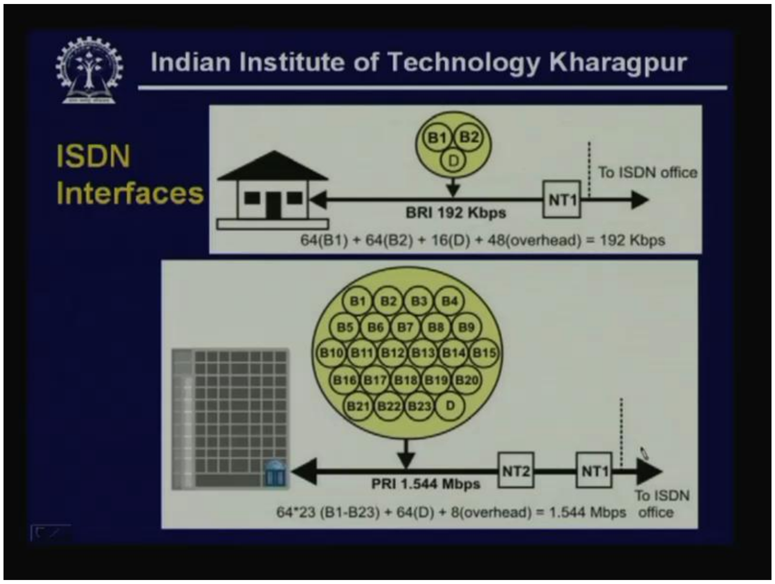

This is your BRI basic rate interface primarily used for home and small business houses. Here as I mentioned 2B type 64 kilobits and 1D type 16 kilobits are provided, of course there is some overhead of 48 Kbps with total data rate of 192 Kbps and this will require one network interface at home and from the network interface it goes to the ISDN office or ISDN telephone exchange.

PRI was developed for little bigger business houses which gives you 23B channels and 1D channel of 16 Kbps. These two together (Refer Slide Time: 37:18) and all these together gives you a data rate of 1.544 Mbps which can give you many advantages and multimedia transmission.

This will require two interfaces NT2 and NT1. NT1 is the same kind of interface as it is used in BRI it is essentially the interface which goes to the ISDN exchange and NT2 can be a PBX which will do some kind of demultiplexing to a number of places. So, these two are necessary for interface intra ISDN exchange.

Now let us go back to our previous slide. With the help of the PRI you can get compressed VCR quality, video you cannot have the TV quality video but compressed VCR quality video transmission is possible by using PRI. Then you have lease line having data rates varying from 384 Kbps to 274 Mbps for fractional t_i to t_1, t_2, t_3 and t_4 giving you different types of capabilities starting from video conferencing, VCR quality video, broadcast quality video and HDTV for different types of lease lines. Then we have already discussed SONET which is based on optical communication giving you bandwidth data rate varying from 51.84 to 2844 Mbps and it is available in multiples of 51.84 Mbps and definitely it is very suitable for multimedia traffic.

Now, the circuit switched network gives you very good quality of service capability because of circuit switched service which is excellent and because of end-to-end connectivity. So a circuit is set up and after the circuit is set up then whatever delay is there that is constant so delay jitter is not present so a digital pipe is available and continuously data is available as a result the quality of service is very good for the circuit switched networks. Now let us focus on the public switched networks. As we discussed the public switched networks has got two options. One is the virtual circuit type, another one is datagram service type.

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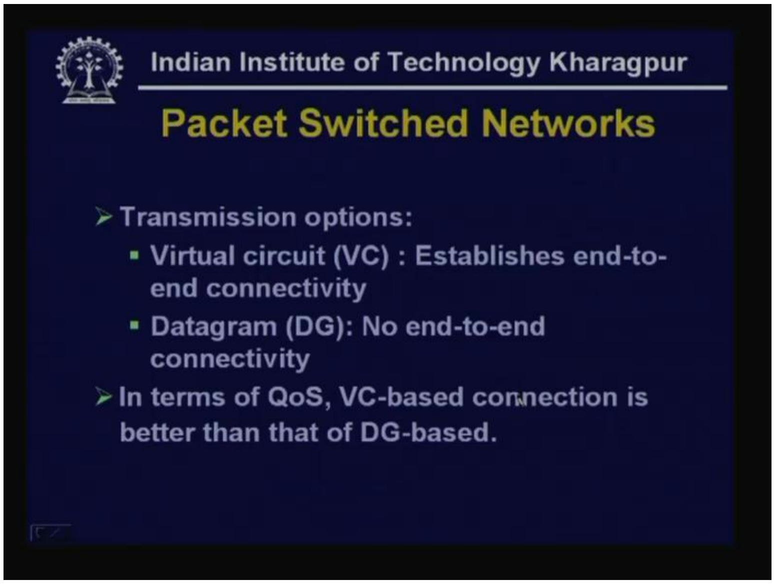

So in case of virtual circuit type it establishes end-to-end connectivity before data communication and all the data goes through the same path or same route which is not true in datagram service. As a result there is no end-to-end connectivity. Obviously in terms of quality of service the VC based connection is better than the datagram based connection. So, for multimedia communication we shall prefer this virtual circuit based packet switched network than this datagram type connectivity because it gives you better quality of service.

Here are the examples of packet switched network.

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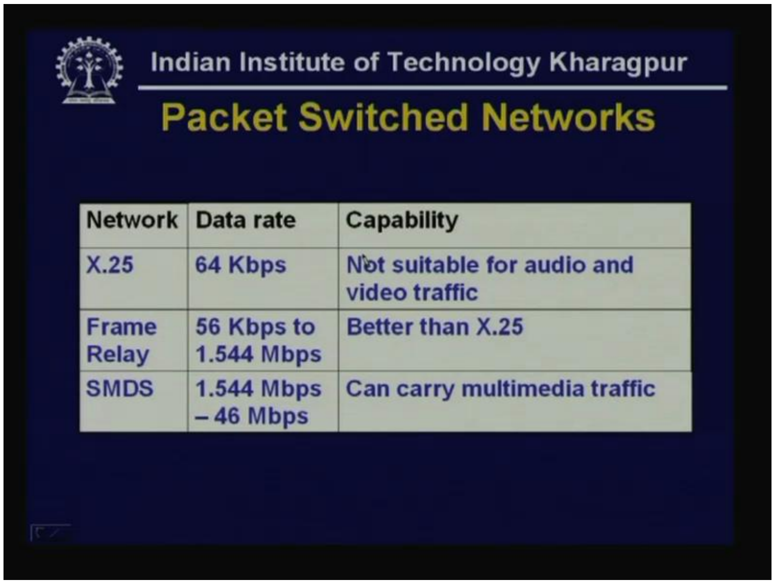

We have already discussed X.21 which gives you data rate of 64 Kbps obviously it is not suitable either for audio or video traffic. On the other hand, frame relay which gives you a data rate of 56 Kbps to 1.544 Mbps is better than X.25 so for some applications this can be used. On the other hand, SMDS which give you much higher data rate 1.544 Mbps to 46 Mbps can carry multimedia traffic. Let me very briefly consider why X.25 is not suitable because it gives you error detection and correction for every hop and as a consequence it involves lot of overhead and delay.

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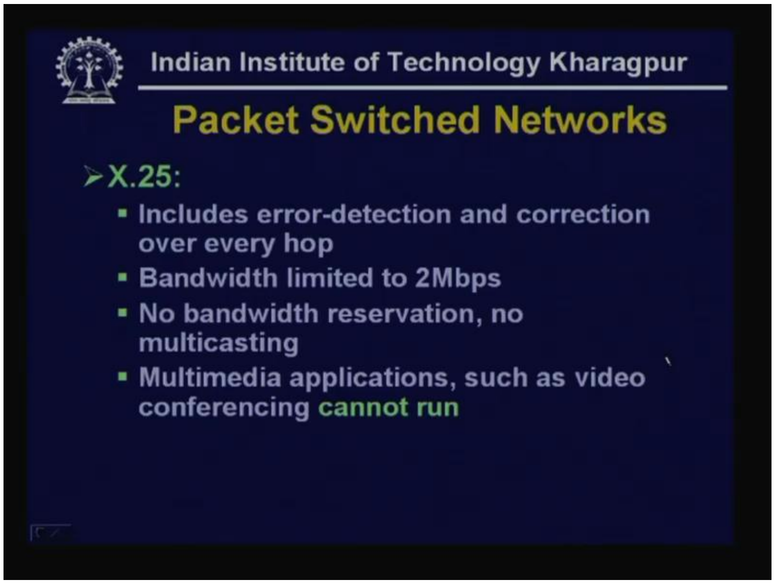

Since the bandwidth is 64 Kbps and nowadays higher bandwidth are also available but those limitations are still there and it cannot afford to bandwidth reservation, it cannot do multicasting either which is very important in case of multimedia communication so multimedia communication cannot run on X.25.

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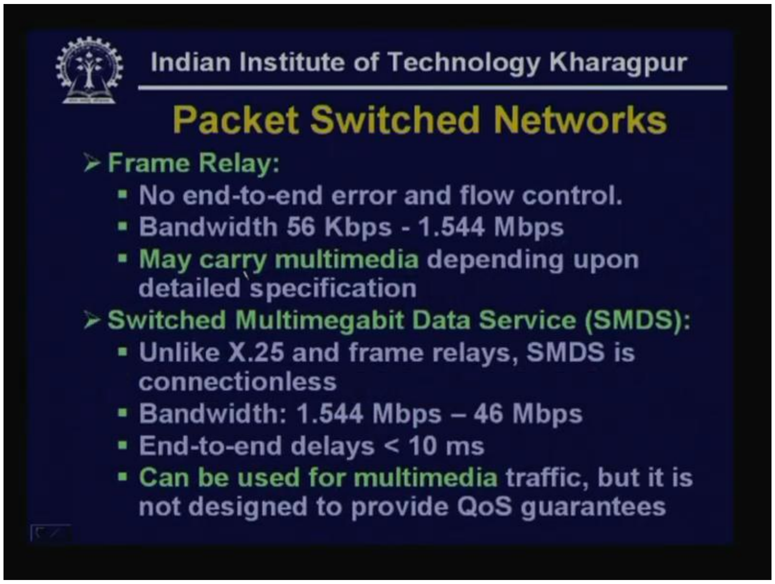

For frame relay again there is no end-to-end error and flow control as a consequence it is better suited for multimedia communication having higher bandwidth so it may carry multimedia depending upon the detailed specification of the multimedia application.

Now I mentioned about SMDS which is Switched Multimegabit Data Service. Unlike X.25 and frame relay SMDS is connectionless. So as a consequence because it is connectionless it can be used for multimedia traffic but it is not designed to provide quality of service. However, it gives you smaller delay less than 10 ms and so on so it can be used for multimedia traffic. Now let us focus on local area network. We have already discussed various types of local area networks, here is the summary.

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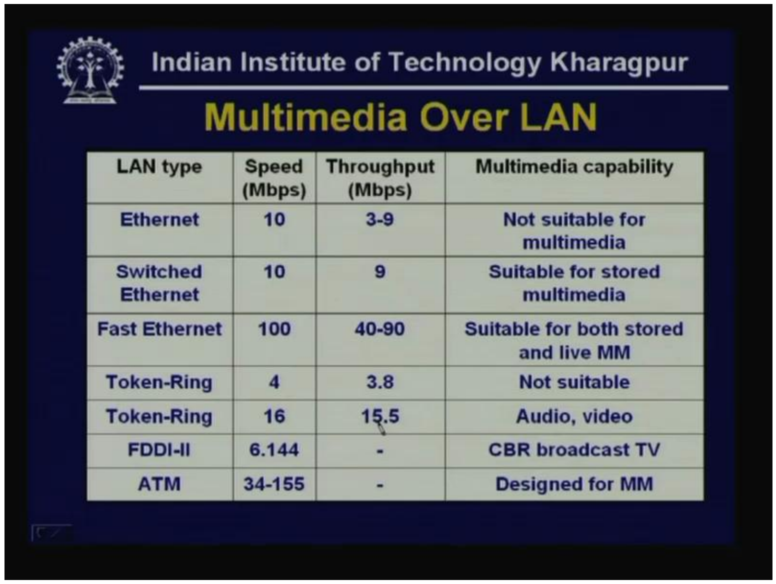

LAN type	Speed (Mbps)	Throughput (Mbps)	Multimedia capability
Ethernet	10	3-9	Not suitable for multimedia
Switched Ethernet	10	9	Suitable for stored multimedia
Fast Ethernet	100	40-90	Suitable for both stored and live MM
Token-Ring	4	3.8	Not suitable
Token-Ring	16	15.5	Audio, video
FDDI-II	6.144	-	CBR broadcast TV
ATM	34-155	-	Designed for MM

First one and the most popular one is the Ethernet which has a speed of 10 Mbps but as I said the throughput is much smaller than the actual speed so throughput can be can vary from 3 to 9 Mbps and obviously it is not suitable for multimedia and because of non deterministic nature of traffic a particular packet may suffer very a large number of collisions. Not only that a particular packet or a frame may get discarded. If it suffers a large number of collisions there is a possibility that a packet will not be delivered. As a consequence Ethernet is not really suitable for multimedia traffic.

On the other hand, switched Ethernet gives you speed of ten Mbps but it has much higher throughput because the bandwidth is not shared in this case and as a result it gives you higher throughput. Also, it is suitable for stored multimedia communication. We shall discuss about the different types of multimedia applications such as stored multimedia, live multimedia and interactive multimedia later on. Hence, it is suitable for these types of applications.

Then comes the Fast Ethernet which gives you a speed of 100 Mbps and throughput in the range of 40 to 90 Mbps so it is suitable for both stored and live multimedia. Nowadays most of the LANs are having Gigabit Ethernet backbone along with the Fast Ethernet distribution system. So in hierarchical system you that backbone band backbone network is by Gigabit Ethernet and other parts at the lower level you have got Fast Ethernet networks, in such cases this live multimedia communication is possible in such types of local area networks.

So far as the older rings are concerned the token ring with speed of 4 Mbps and throughput of 3.8 Mbps is not suitable, token ring with 16 Mbps data rate or speed having throughput of 15.5 Mbps can be used for audio and video communication, FDDI-2 with 6.144 Mbps can be used for constant bit rate broadcast TV and ATM which is based on BISDN broadband ISDN is very suitable for multimedia traffic and it is actually designed for multimedia traffic which is based on circuit switching concept and it gives you a speed of 34 to 155 Mbps. As a consequence ATM is very suitable for multimedia communication.

Now let us focus on multimedia over network which is most popular and important. Unfortunately like many other technologies the internet was not designed to carry multimedia traffic. so when the internet was designed and evolved in those days multimedia traffic was not non-existent it was essentially for data communication, communication of emails and other information and not really for multimedia at most some graphics so it was not really designed for multimedia traffic.

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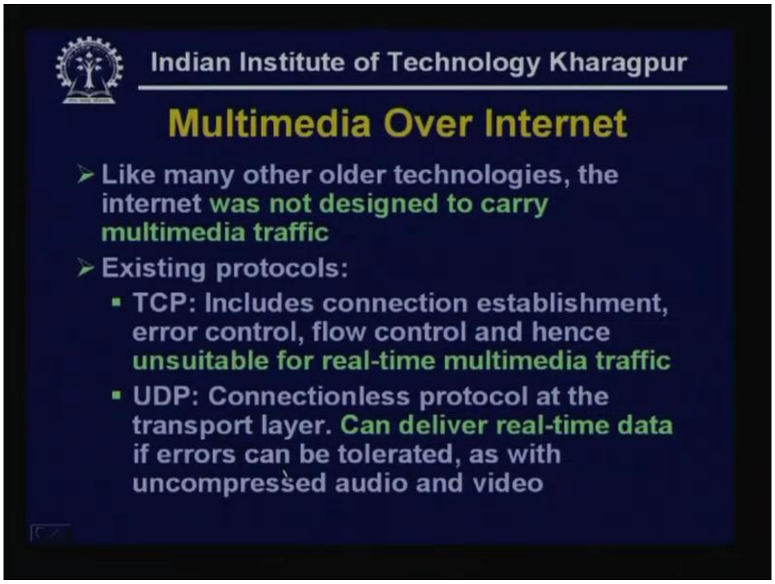

And as we know we have discussed two protocols TCP and UDP. TCP includes connection establishment, error control, flow control, congestion control and so on and we have seen that all these things put lot of overhead and leads to delay and delay jitter so as a consequence TCP is not suitable for real-time multimedia traffic. On the other hand, UDP which is connectionless protocol at the transport layer can deliver real-time data. However, as you know that UDP is unreliable there is no error control facility. As a consequence if error can be tolerated as with uncompressed audio and video this UDP provides an alternative for multimedia communication.

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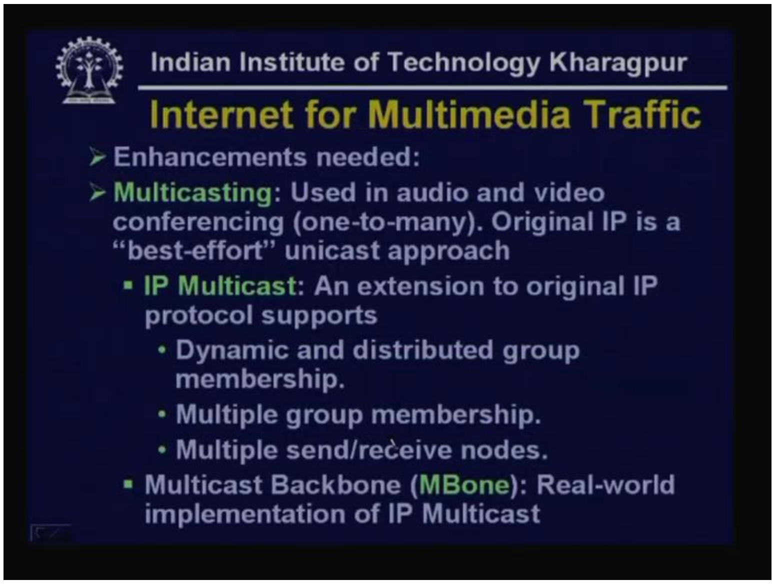

However, you will require a number of enhancements. You have to add some more functionalities and protocols to make the internet multimedia ready. Let us see what are the enhancements and what are the additional protocols that can be used to make the internet multimedia ready or multimedia enabled.

First one is multicasting. This is one very important requirement for multimedia. Multimedia is usually not between two persons, it is not for one to one communication, it is essentially for one to many. As a consequence it is essential to have multicasting feature. Unfortunately original internet protocol is the best effort unicast approach; here there is no concept of multicasting. So you have to add something on top of this which is known as IP multicasting which is an extension to original IP protocol to support dynamic and distributed group membership, multiple group membership and multiple send receive nodes. So IP multicast protocols or feature has to be added on top of IP to enable internet protocol multimedia. Another important function is known as multicast backbone which has evolved or has been developed to support multimedia to give you real-time in implementation of IP multicast.

Let me give you an overview of what you really mean by multicast backbone.

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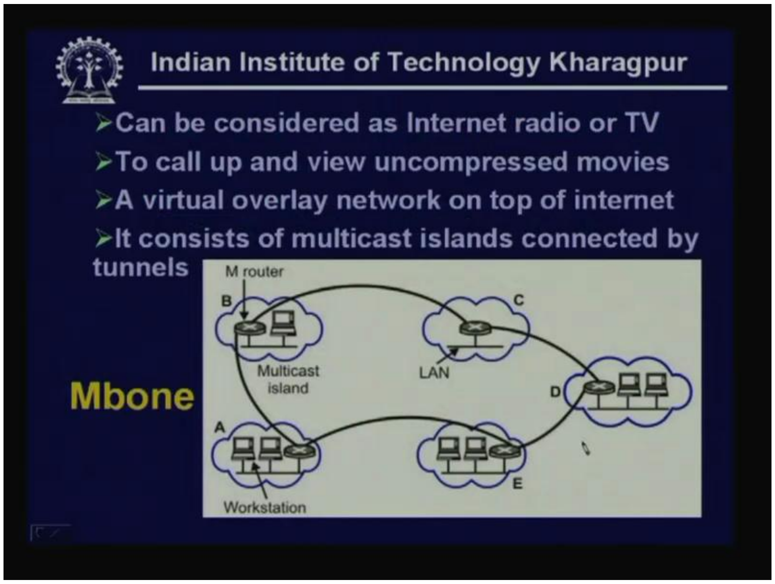

So it can be considered as an internet radio or TV. You are already familiar with the broadcast radio and broadcast FM. There are radio station and TV stations which continuously broadcast radio and TV signals and with the help of our receivers and television sets we can tune to one of the stations and get the radio and TV signals that we get through air but here we would like to get through internet. How it is being done?

An user will call up and ask for a particular service may be he will ask for a particular movie or some particular type of music so it will view uncompressed movies and how it is being implemented is it is implemented in this way, it is a virtual overlay network on top of internet. So the existing network is there and on top of that there is virtual overlay as shown in this diagram, it consists of multicast islands connected by tunnels. So here is a multicast island, it can be a local area network as it is shown here or it can be some other types of networks which have the multicasting facility so each of these islands has got multicasting facilities in them. These islands are connected by what is known as tunnels with the help of these m bone routers and through these routers they communicate with each other through to give the connection through the network.

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Then other enhancement required is UDP. UDP is more suitable for TCP for interactive traffic but it requires services of RTP a special type of protocol real-time transport protocol and application protocol designed to handle real-time traffic, it uses an even numbered UDP code and the next number used as a companion protocol for real-time control protocol. So RTP is used for data communication and RCTP is used for control signal communication. RTP provides end-to-end transport services to real-time audio, video and simulation data, however it does not give you QoS guarantees.

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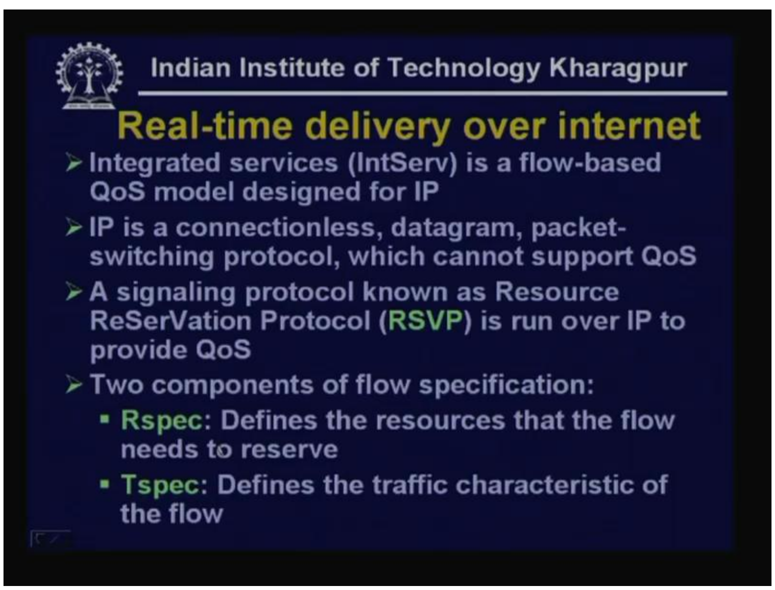

Another important protocol that is been developed is known as RSVP. It gives you a signaling protocol over IP and it runs over IP to provide the necessary Quality of Service. It has got two components for flow specification. Actually we'll require some flow based QoS model so two specifications are there; one is Rspec which defines the resource specification required by the multimedia application and Tspec which defines the traffic characteristic of the flow that means whether it is constant bit rate, variable bit rate etc. With the help of this RSVP protocol the resources can be reserved along the path by using path messages and reservation messages which are sent by the receivers and on each node RSVP attempts to reserve the sources to be able to provide the requested quality of service and it can run over IPv4 and IPv6 and it is scalable and allows dynamic group membership and routing changes.

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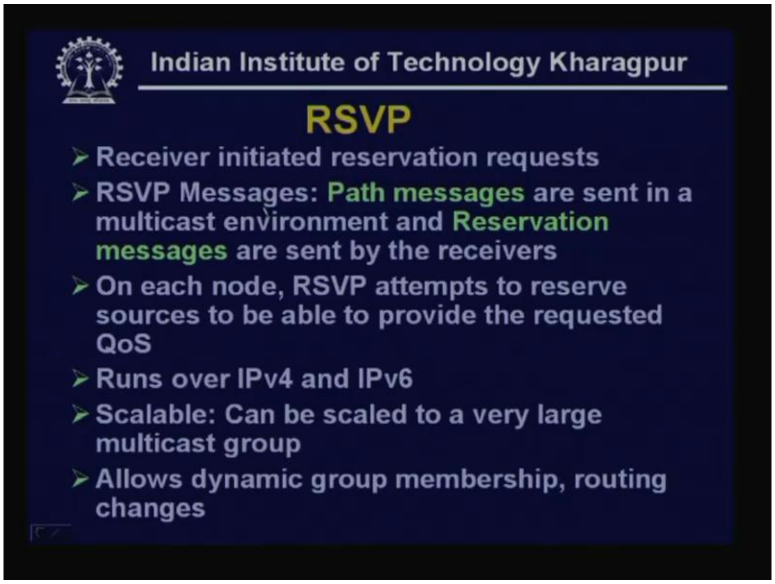

Let's see how it really works so you see from the sender the path messages will go to different receivers then the reservation is done by different receivers by sending their requests.

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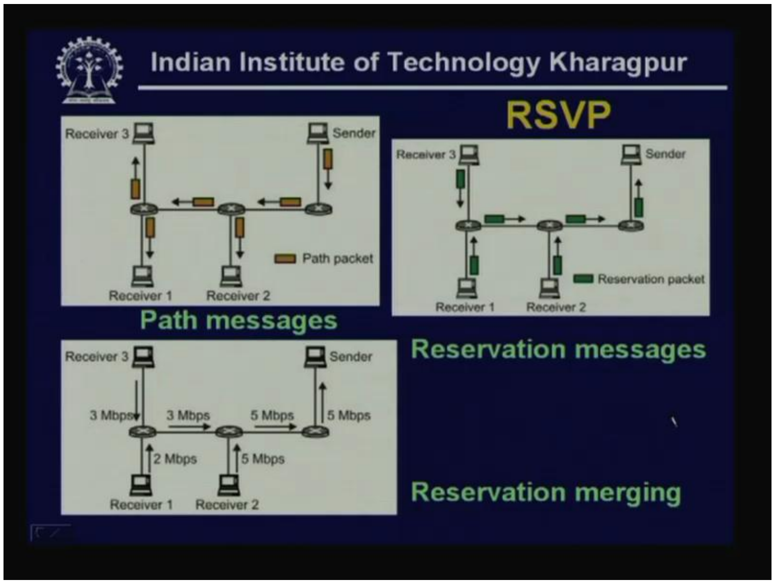

However, it also allows reservation margin for example from receiver 3 and 2 the bandwidth reservation is 3 and 2 which can be merged to have 3 Mbps and see this 3 and from receiver 2 this 5 Mbps are merged to get 5 Mbps bandwidth. So 5 Mbps is the merged bandwidth which is the required by the sender this is how the bandwidth can be reserved.

Now let me give you the review questions based on this lecture.

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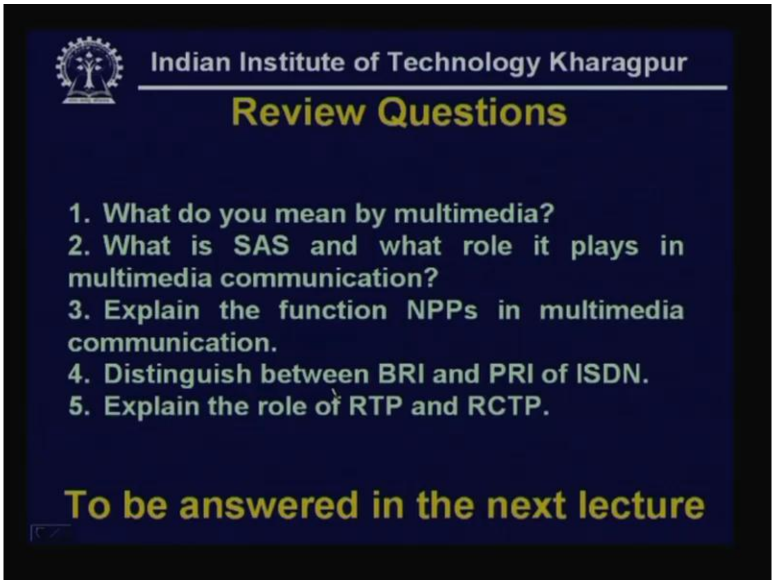

1) What do you mean by multimedia?

2) What is SAS and what role it plays in multimedia communication?
3) Explain the function of NPPs in multimedia communication
4) Distinguish between BRI and PRI of ISDN
5) Explain the role of RTP and RCTP.

Now it is time to give you the answer to the questions of lecture – 35.

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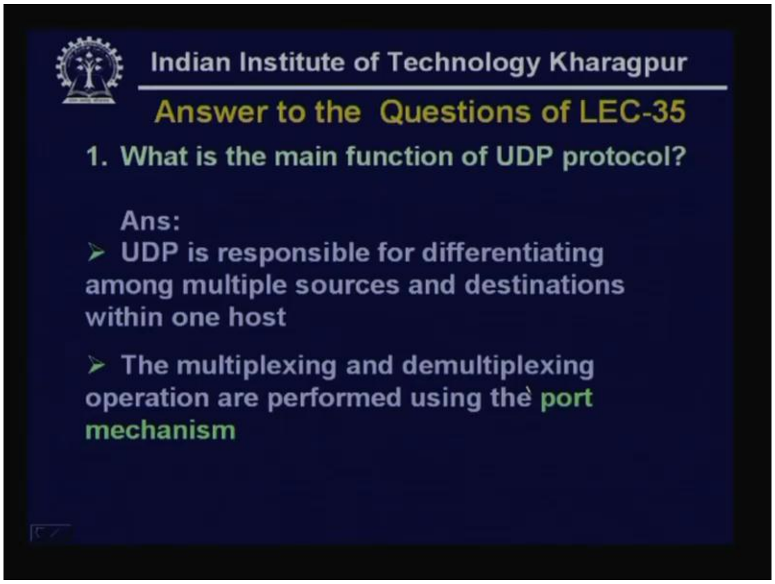

1) What is the main function of UDP protocol?

UDP is responsible for differentiating among multiple resources multiple sources and destinations within one host the multiplexing and de multiplexing operations are performed using port mechanism which I have explained in detail in the last lecture how multiplexing and de multiplexing is done

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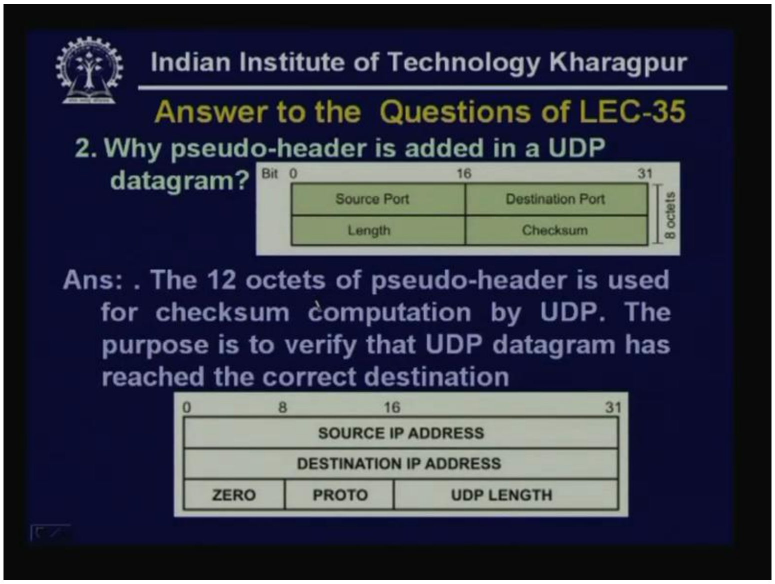

2) Why pseudo header is added in a UDP datagram?

As you know UDP header has got the information of source port destination port but it has not got no information about the IP address so a pseudo header is added a twelve octet pseudo header is used for checksum computation by UDP.

The purpose is to verify whether the UDP datagram has reached the correct destination and destination at this are essentially the source IP address and destination IP address so this pseudo header is added for computation of checksum in case of UDP.

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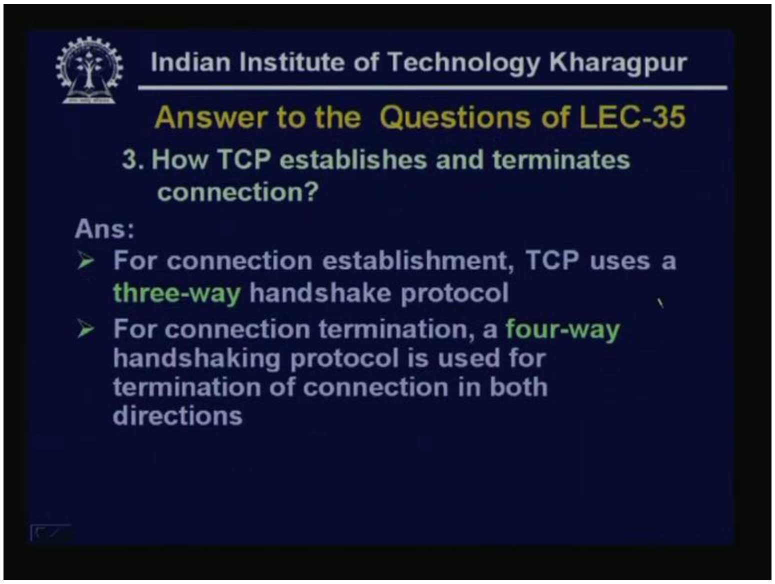

3) How TCP establishes and terminates connection?

For connection establishment TCP uses three-way handshaking protocol as I have explained in detail and for connection transmission termination four-way handshaking protocol is used for termination of connection in both direction

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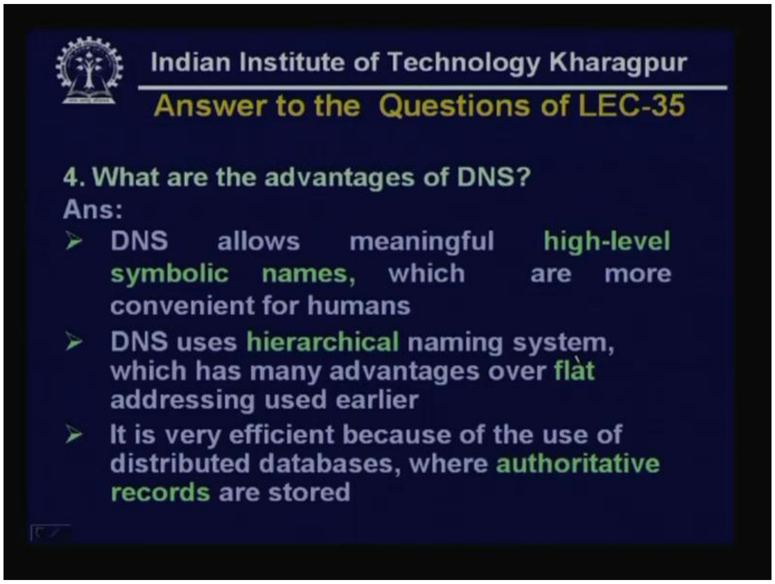

4) What are the advantages of DNS domain name system?

As I mentioned DNS allows meaningful high level symbolic names instead of IP addresses which is more convenient for humans and it uses hierarchical naming system which has many advantages over flat addressing used earlier. It is very efficient because of the use of distributed databases where authoritative records are stored.

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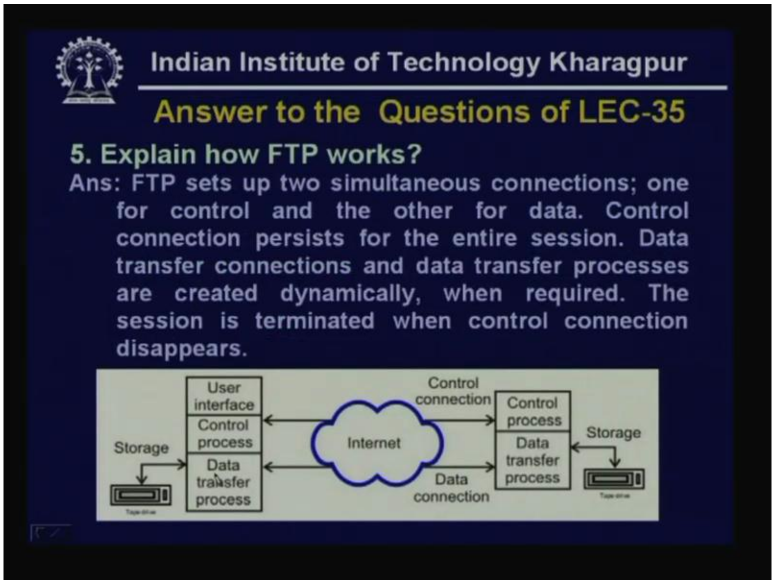

5) Explain how FTP works?

FTP sets up two simultaneous connections; one for control and another for data. Control connection persists for the entire session. Data transfer connections and data transfer processes are created dynamically as and when required.

The session is terminated when control connection disappears. This is shown in this diagram. Again we can see how the control connection and data transfer takes place from one place to another through internet.

With this we come to the end of today's lecture and in the next lecture we shall discuss about the compression techniques which are indispensable for multimedia communication.

Thank you.