Data Communication
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Lecture - 38
Multimedia Services

Hello viewers, welcome to today's lecture on multimedia services. The advancement of technology has led to the deployment of high speed networks and also tremendous research in the field of compression has led to development of powerful compression techniques. As a result the bandwidth required after compression is small and networks provide high speed broadband services. These two together has made many new applications possible.

In today's lecture I shall try to give an overview of the popular applications or services available today for multimedia applications. Here is the outline of today's lecture. First I shall give a brief introduction and then mention about the popular multimedia services. They can be categorized into three basic types.

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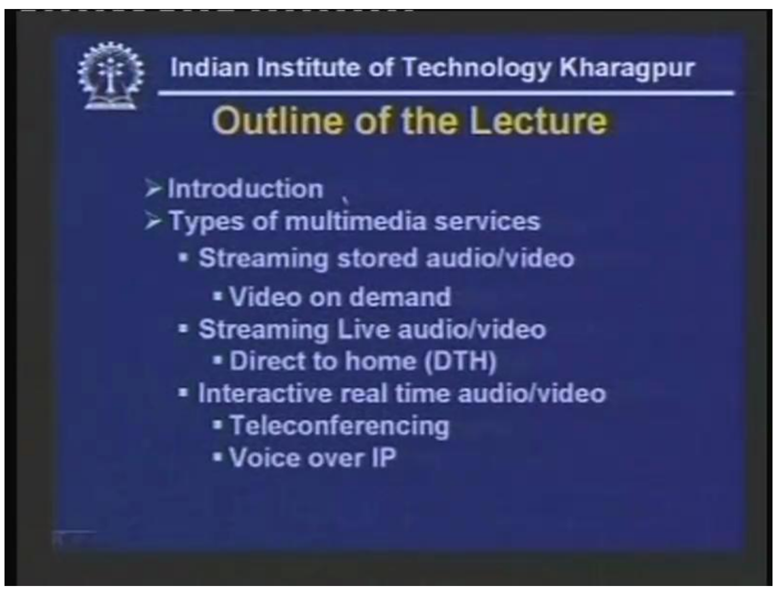

First one is streaming stored audio and video, video on demand is an application, streaming live audio and video, direct to home DTH and interactive real time audio/video for which I shall give examples from teleconferencing and voice over IP.

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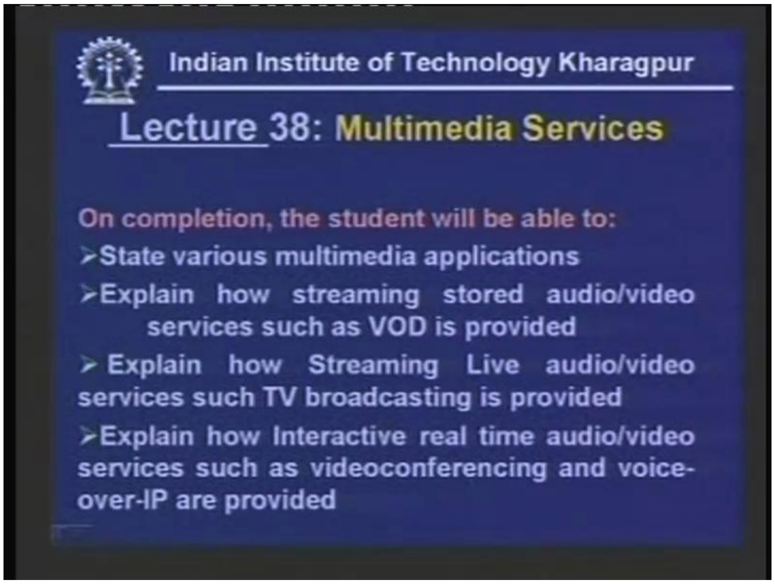

On completion of this lecture the students will be able to state various multimedia applications possible today, they will be able to explain how streaming is stored, let me emphasize on this term stored audio/video services such as VOD is provided, they will be able to explain how streaming live audio/video services such as TV broadcasting through internet is possible, they will be able to explain how interactive real time audio/video services such as videoconferencing and voice over IP are provided.

So here is the basic introduction. As I mentioned the deployment of high speed networks and reduction in bandwidth requirement because of compression has led to the emergence of diverse and many applications and this application can be broadly categorized into three types as I have mentioned streaming stored audio/video, streaming live audio/video and interactive audio/video and I shall give some examples of these application such as video on demand, broadcasting of radio/TV programmes through internet, interactive audio/video such as tele/video conferencing and voice over IP.

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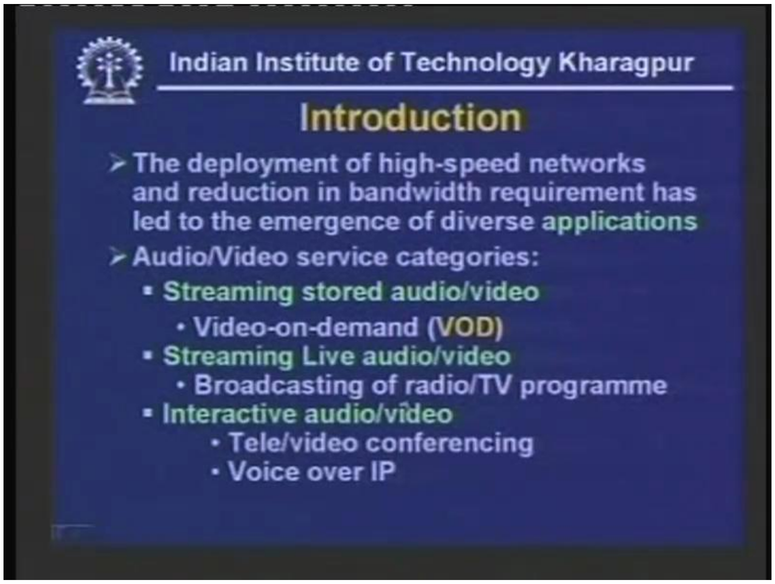

First let us focus on streaming stored audio/video. The simplest approach of giving this service is to download the compressed audio/video files just like a text file by client and then the client plays the file.

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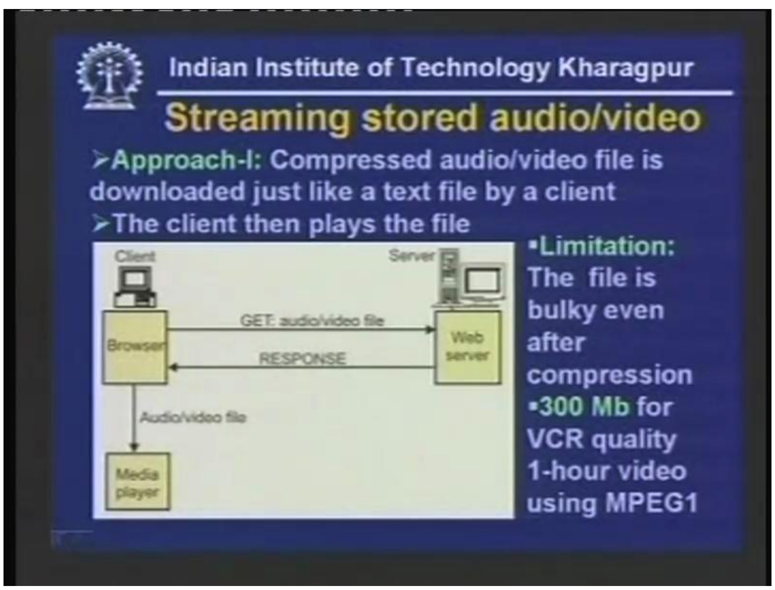

Here the schematic diagram is shown. Therefore, in the first step the audio/video files are requested by the client through the browser to the web server and in response the web server sends the file and after receiving the file obviously it has to be stored here this can be played in the third step with the help of a media player so it involves three basic steps; getting the file from the server by sending a request like GET and getting the response from the web server then playing it. Now in this the limitation is the file is bulky even after compression. As a result the file will be very large. For example, without compression the size of a one over MPEG-1 file is 600 megabyte, after compression it comes to 300 megabyte for VCR quality 1 over video using MPEG-1.

Obviously to download such a file it will take quite some time, it may take minutes some time hours depending on the size of the file and the bandwidth of the link through which the downloading is done so it will take time, it will occupy large storage space in the client system and then it can be played. These are the limitations of this streaming stored audio/video service. Here of course we are not really doing the streaming although I have written here streaming stored/audio/video service is being transferred then it is played but it is not a streaming service that way.

Let us now consider the second approach. In second approach the media player is directly connected to the web server for downloading audio/video file.

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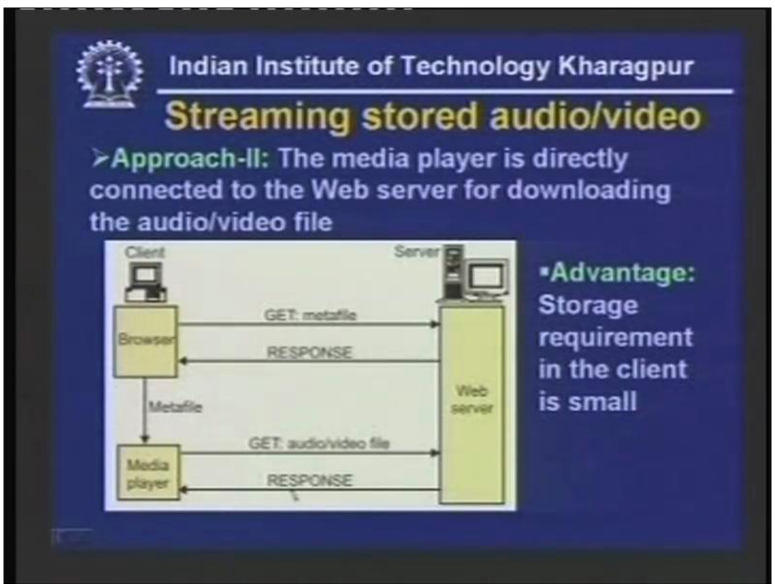

Here what is being done the browser again sends a request not really for the file but a metafile. The metafile contains information of the actual file so the web server sends that response to the browser. Then the media player uses that metafile to interact directly with the web server then the media player sends a request like GET audio/video file to the web server and in response the web server streams the audio/video file and that can be played here.

So in this case the storage requirement is very small. Here we are using essentially two different types of file. First is a metafile and the actual file that is being actually being played and performed in a streamed manner. So the web server streams the file and sends it to the media player as it plays. This is how it takes place. Now here the advantage is that the entire file need not be transferred before you can play it.

In the previous case we have seen it may take minutes or hours before you can play the file. in spite of the fact that you may have very large storage space you may have to wait but here it is not so, you can start playing after every short time and only a small portion will be transferred from the web server to the media player then the media player we'll keep on playing as it receives the actual file from the web server. So this is the advantage of this approach. So the client system need not have very large storage space in this particular case.

However, here the limitation is both the browser and the media player uses the HTTP for downloading the metafile as well as the audio/video file so all the steps will require the use of HTTP which runs the TCP. And as we know TCP is not really a very good protocol for the purpose of streaming. Why TCP is not very suitable? The reason is TCP first of all performs flow control, error control and as a result if there is some error the some kind of retransmission will be required, that is not supported or required for streaming because of high redundancy present in the files. So it is not really a very good approach that's why the third approach is used in which case a separate media server is used for downloading audio/video file.

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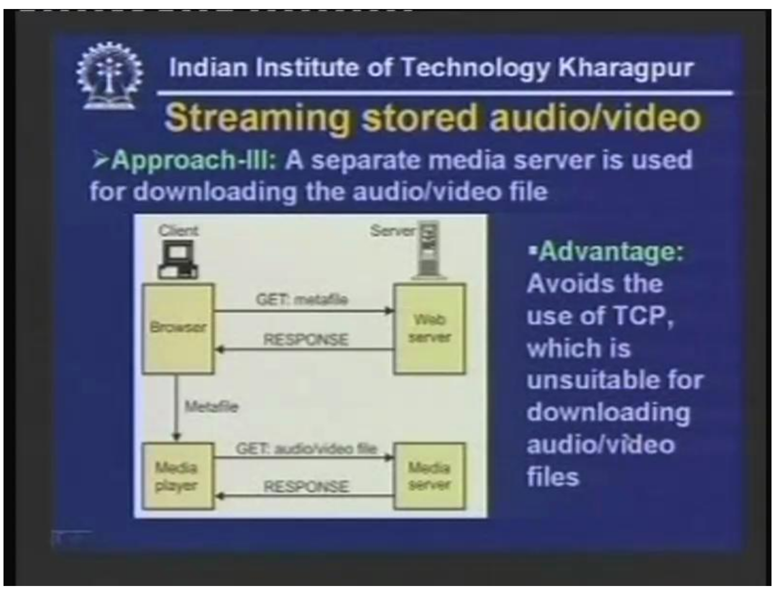

So in the first step the client through the browser sends a GET request for the metafile then the web server sends the response and that metafile information is sent by the browser to the media player then the media player communicates with media server in step four and five and then the media player communicates with the media server to perform the communication. In such a case it avoids the use of TCP which is unsuitable for downloading audio/video files. So here it uses UDP where UDP does not support flow control, error control and so on which is not really required here however UDP alone cannot do the job it uses other support protocols to perform this streaming. Hence, this is a better approach than the previous two and this is being commonly used for audio/video streaming.

Now let us consider the application of this streaming stored audio/video. The most popular application is the video on demand. We are all familiar with the stores which gives the CDs that is CDs of movies on rental basis.

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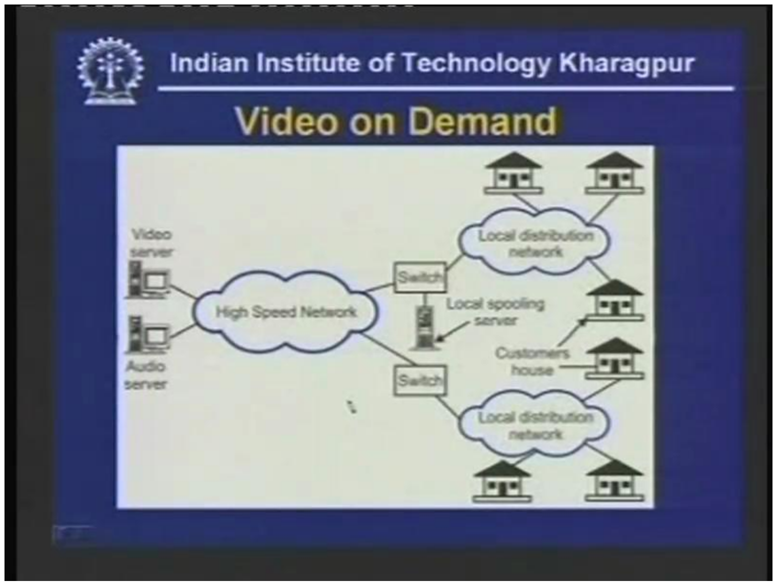

We take the CDs to our home then we can play them and whenever we want we can pause, we can stop or we can replay that is a part of the movie can be replayed. We want similar facilities through the internet. Instead of rental stores here we shall use the audio/video servers to do the job and the only requirement is that we must have a high speed network such as SONET or ATM and these links are usually fiber optic. We have already discussed SONET and ATM.

The servers are connected through high speed network and to some switches and with the help of some switches there is local distribution network and you can have some kind of local spooling server. Sometimes the files are transferred to the local spooling server and from that local spooling server it can be broadcasted or multicasted to a number of users. For example, we can use different types of local distribution systems.

For example, it can be LAN in some cases or it can be some other type of network that can be used for local distribution network. So this is the requirement for video on demand service. You'll require a high speed network and then it is transferred and it goes to the switch and then distribution is performed. Let me take up an example as a video on demand that is being done in IIT Kharagpur campus. Video on demand is being used here for educational purposes.

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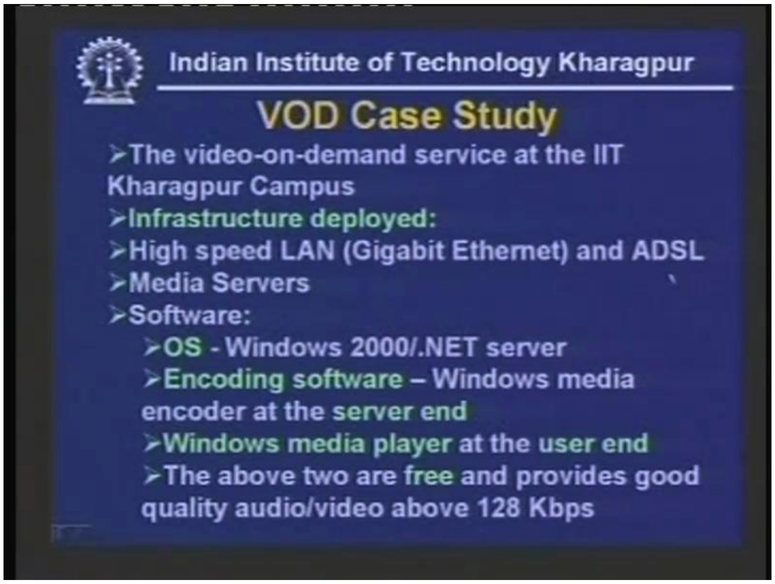

We don’t use it for commercial purposes but for educational purposes. Instead of movies here we use a number of video courses. There are about hundred video courses which are used as video on demand and throughout from the campus anywhere the students, faculty members anybody can access these video courses and there are 30 to 40 lectures in each of them each of one hour duration so that is being provided within the IIT Kharagpur campus. For that purpose you require some infrastructural facilities. Let us look at the infrastructure deployed in IIT Kharagpur.

First of all you require high speed LAN and for residences ADSL network. We require media servers these are the hardware required, of course you will require some software as well like operating system. There are several alternatives but here windows 2000/.NET server is being used then you require encoding software, here also there are several alternatives. Windows media encoder is being used here at the server end and at the user end windows media player is being used.

These are being used because these are primarily free and they provide good quality audio/video above 128 Kbps. Let us now focus on the distribution network in IIT Kharagpur campus.

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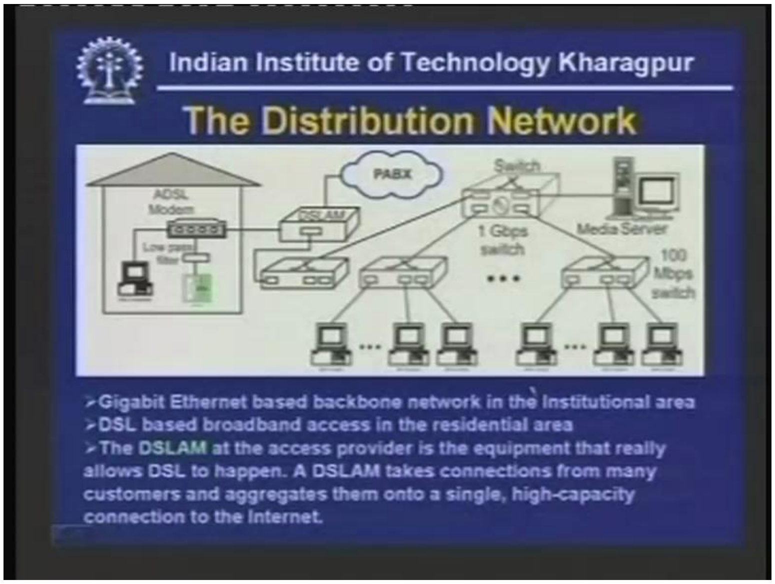

There is a Gigabit Ethernet based back bone network. This is the main switch the layer three Gigabit Ethernet switch which acts as a backbone. This switch is installed in the computer and informatics centre and from that place through optical fiber link various departments and even all the hostels are connected. So optical fiber links are going to all the departments and all the hostels where there are 100 Mbps Fast Ethernet switches so with the help of that it goes to different users or desktops.

However, for the residential area the requirement is different. This LAN facility is not extended to the residential area but with the help of DSL based broadband access the broadband services are provided to residential area and for that the technology that is being used is the DSLAM which is used as the access provider which is the equipment that really allows DSL to happen.

What it does is there is a box which is shown here DSLAM which is essentially a multiplexer, this DSLAM takes connections from many customers, it goes to twelve residences where there is ADSL modem. Then the signals from twelve customers are aggregated and are connected to 100 Mbps that means Fast Ethernet LAN. And on the other side to get the audio service it is connected to the PABX square where about 2000 PABX line is there so with the help of this, both audio and internet service is available.

The campus LAN it is connected to the Gigabit Ethernet switch which is the internet service provider to the residences. Then it comes through the residence to ADSL modem and in that ADSL modem one port goes to the desktop and another port through low pass filter goes to the telephone. So the telephone conversation which is restricted to usually 4 Kbps does not disturb the data communication for the internet access so both can take place simultaneously internet access as well as telephone telephonic conversation. So with the help of this broadband service available to the residences as well as to the institutional area that means the hostels, departments and so on this video on demand services available.

Now let us focus on the media servers that are being necessary. Media servers has to be little powerful in terms of the main memory and the hard disk storage. Media servers typically has got twenty Pentium 4 processor each with main memory of 1 GB and hard disk drive with RAID-5 which means you require five hard disks each of 147 Gigabit capacity.

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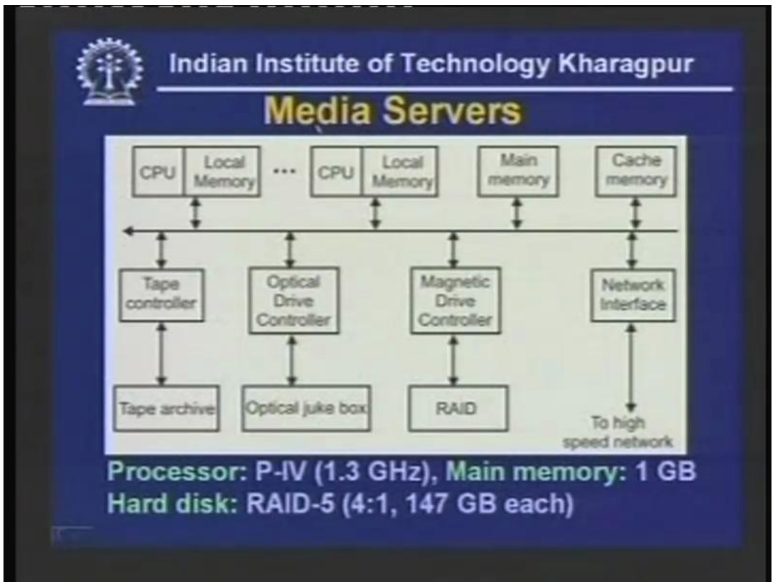

Although there are several alternatives like one can use tape, optical disk drive and magnetic disk controller most of the course material is stored in hard disk in IIT Kharagpur and others are not used but they can be used if necessary. Let us talk about some important issues related to this video on demand service.

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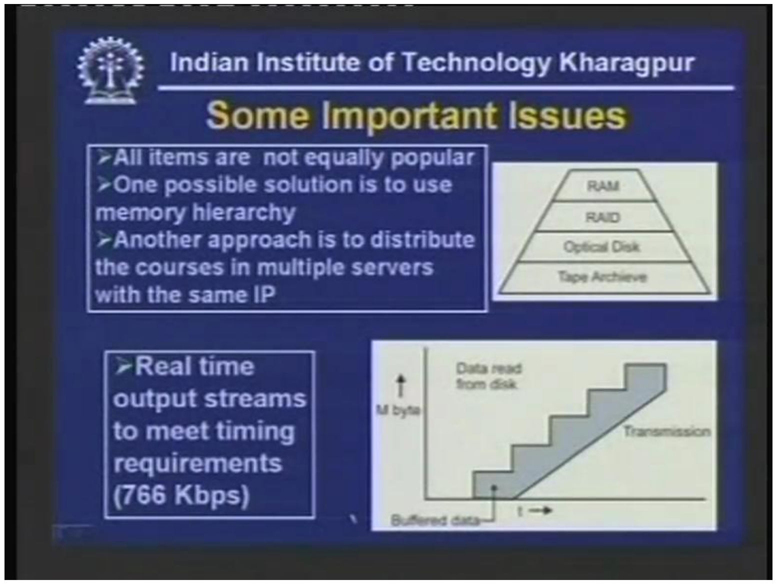

It has been observed that not all thousand movies are equally popular. Similarly if you have hundred courses each of them may not be equally popular, some are very popular, some are less popular, some are not at all used. So how do you provide better service for the courses which are more widely used? This can be done in two ways. One possible solution is to use memory hierarchy. As you know in computer the memory can be hierarchically organized.

At the top layer you have got the random access memory which is the costliest. So cost is higher for RAM, cost is lesser for this hard disk RAID then it is still lesser for optical disk and still lesser for tape archive. On the other hand, for capacity it goes in the reverse direction, RAM has lesser capacity than hard disk, optical disk has still higher capacity than the hard disks and tape archive can have still higher capacity so what can be done is some of the courses which are very popular can be stored in hard disk, lesser useful courses can be kept in optical disks and courses which are very rarely used can be stored in tapes. In this way one can hierarchically distribute different courses.

However, with time the cost of hard disk drives has come down significantly. Nowadays the hard disk drive cost is not very high so this alternative is not being used in IIT Kharagpur. Instead of that what is being done is instead of having one media server three media servers are used each of the same configuration with the same IP address so it is transparent to the user.

However, out of these three servers one server is loaded with lesser number of courses that is most popular courses are loaded in the server and the number of courses loaded in that server is fewer. In the second server the number of courses is little more which are usually accessed but not very frequently and in the third server you can put a large number of courses but the courses are not regularly used. So in this way using all the courses in hard disk using RAID-5 is being done and use of RAID-5 also provides you higher throughput which is necessary for streaming purposes.

As we can see here real time output streams has to meet timing requirements. For example, the kind of encoding that is being done here requires streaming at the rate of 766 Kbps to get flicker free display. Whenever you read some data from disks it is read in terms of sectors in discrete form.

On the other hand, whenever it is displayed it is done in a streamed way I mean continuous manner so you have to use some kind of buffering. This is how the data is being read so it is read and one sector is sent and another sector is read so this is the time required then another sector is sent then after reading another sector it is being sent here in this way it goes on. So, after buffering for some duration the transmission starts so here the play starts and you can see the play is taking place continuously. So in spite of the fact that the data is received in discrete form in terms of packets the transmission is taking place continuously to get flicker free display. These are the important issues to be remembered in the context of VOD service video on demand service.

I have discussed in nutshell the requirement for this video on demand service infrastructure. You require broadband network service then media servers then the operating system at the server end and windows media player at the receiving end. This is how the video on demand is provided for educational purposes in IIT Kharagpur. Now let us come to the next application that is your streaming live audio/video.

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We are very familiar with the radio stations, TV stations which are continuously transmitting audio and video signals and with the help of our radio receiver and TV we can tune to one of the stations, can watch or listen to radio songs or news and similarly we can watch the TV. But what we want here is we want to do it through internet. So, whenever we try to do it through internet similar kind of audio and video services we have to understand the requirement and see how it can be supported.

First of all just like the conventional audio/video service using radio and TV stations through internet it is also sensitive to delay and also retransmission cannot be done whenever we do it through internet. As I have mentioned because of lot of redundancy if one frame is discarded it does not matter but after this transmission if that frame comes back we cannot really play it then so retransmission has to be dismissed.

Also, we have to understand that in this case communication is really multicast and live it is not unicast as it happens in case of video on demand. Video on demand is essentially unicast from the media server to a particular user. Of course that media server is simultaneously giving service to a number of users at a time. However, it is usually unicast in nature. However, in case of streaming live audio/video it has to be multicast and live, it is not stored.

One example of streaming live audio and video is through satellite network. As you know the program material from studios are sent to uplink earth stations and through satellite this goes to different networks.

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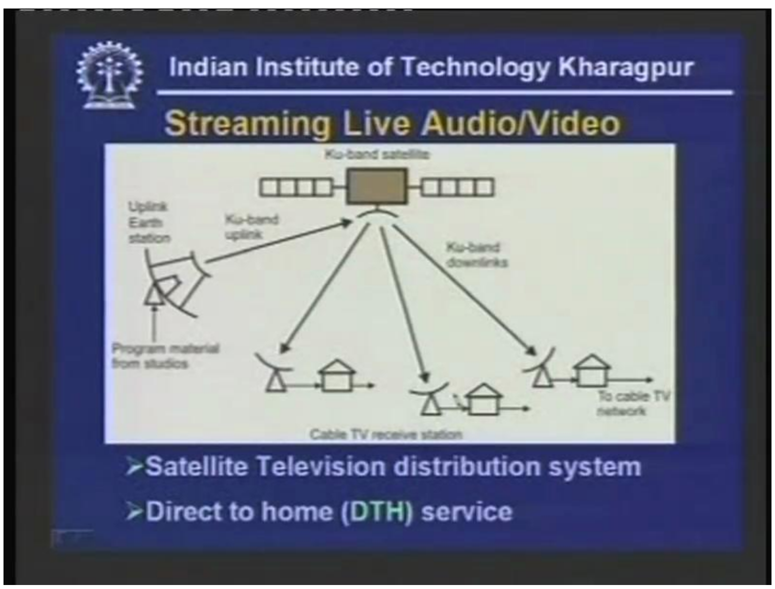

You have the VSAT antennas and there it is received and you can have cable TV network or you can have a transmitter like LPT low power transmitter or HPT high power transmitter which can broadcast these or it can be broadcasted with the help of the cable TV network.

As I mentioned earlier the cable TV network can also be used for distribution of video on demand but the use of cable TV for video on demand is not popular particularly in India. This is one of kind of service. So, through network it is distributed then either using LPT or HPT high power transmitter or low power transmitter it can be broadcasted. Another possibility is to use cable TV network for distribution.

Another new service has been introduced which is which is becoming gradually popular that is direct to home service. Here there is no need for cable TV provider or low power transmitter or high power transmitter is not involved. So, through satellite link each home will have a small antenna and that antenna can be used to receive this signal from satellite where we will require one set top box. With the help of set top box it can be played. The set top box is essentially a computer it has got the CPU, ROM, RAM and through input output port it goes to TV and remote control and in addition to that it has got the MPEG decoder.

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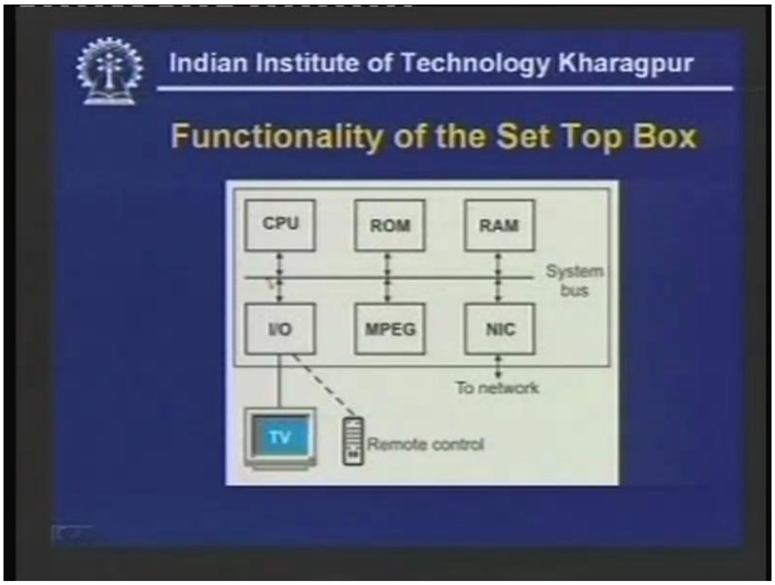

So the signal that is being received through the network, in this case it is a satellite network, it can be other types of network as well from the network it receives the compressed signal which is decoded by this MPEG decoder then through IO it goes to the TV and you can select different channel and you can configure it in different ways with the help of the remote control. So in residences you require a small antenna and a set top box with the help of which you can have this direct to home service which provides you this streaming live audio/video services.

Now let us consider the third service that is real time interactive audio/video. In case of real time interactive audio/video you have to first understand the requirement for real time interactive audio/video.

First let us consider the requirements then we shall discuss various applications and see how these services are provided. First of all let us consider how a client and a server communicate through some audio/video in real time and interactive manner.

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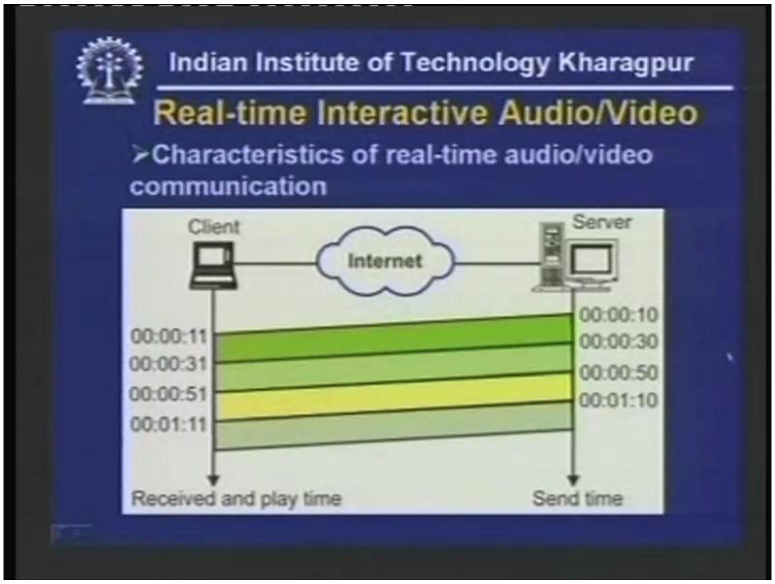

Let us assume four files are there each is holding audio/video information each of twenty second and this is being transmitted through internet and here it is assumed that the internet takes about one second to send the packet from one end to another. Of course there is propagation time here, one second is the propagation time and twenty seconds is the transmission time of the audio/video file. so the server sends the audio/video file to the client and where it is played. Let us assume the server has started sending the first file from 0 0 0 10k this is 0 0 0 0 ten hours, this is the hour, this is the minute, this is the second. So this is your hour, this is your minute and this is your second (Refer Slide Time: 32:40).

As you already know this time information is provided by the encoder. The MPEG encoder provides time stamping in each of these packets or files. So with the help of this time stamping the time is known so at this time this is being sent and the receiver knows what the starting time is. However, it reaches one second later so immediately after receiving it, it starts playing so as it starts playing it will play for 20 seconds and after 20 seconds it will start receiving the second packet or second message we can say which also has the twenty second information which comes in between 00 00 31 to 00 00 51 within this time it takes twenty seconds to receive here and it displays. That means play is starting one second later and then it is continuously being played here in real time. In this way four files are transferred in real time from the server to the client and it plays there continuously.

So here we have assumed that the delay is constant and it takes one time. So this one second delay of the network does not really matter because here the playback is starting after one second and then it is continuously playing it, so it is received and played, so the received and played time is shown here. but unfortunately the network does not have a constant delay and as we know the video files are having variable bit rate because of the compression of the different files can be of different amount this will lead to variable bit rate traffic and as a consequence the files that will be reaching or the messages or the information that will be reaching or the messages or the information that will be reaching at the client end will be different that means some variation in delay as it is shown in the next slide.

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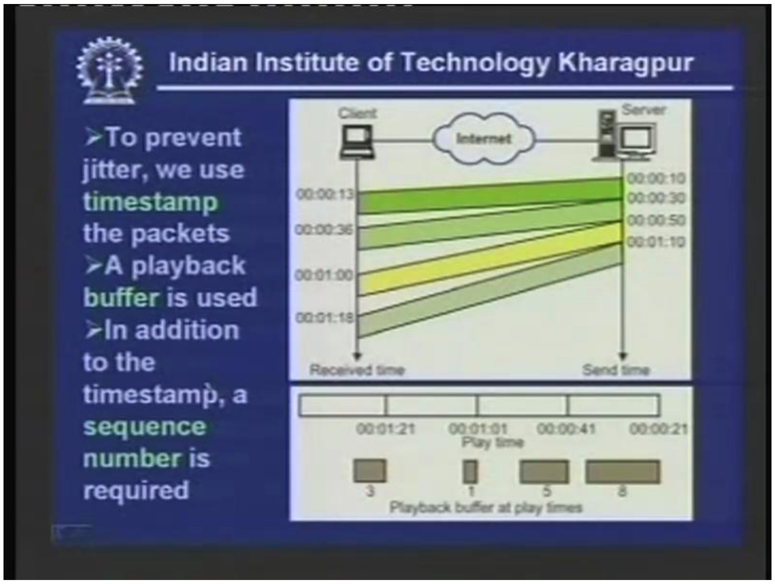

Here let’s assume the first file is having a delay of three seconds, this one (Refer Slide Time: 35:30) is having a delay of three seconds, the second file is reaching after a delay of about six seconds, third file is reaching after a delay of about ten seconds so this is three, this is six seconds, this is ten seconds and this one is reaching after a delay of about eight seconds. Now what you can do in such a case? In such a case in spite of the fact the receiver does not know what is the starting time of each of the packet. Since they are received at with different delay they cannot be played one after the other because of the variation in delay which is known as jitter. How can you overcome this jitter? Apart from the use of these time stamps the requirement is to use some buffer. What can be done is before starting these files they can be buffered for certain duration.

It is assumed that maximum delay that can take place is eleven seconds. So what is being done at the receiving end is that the first file is played after 21 seconds so 00 00 21, the sender is sending at 00 00 10 but the play starts at 00 00 21.

However, to do that it is necessary to store the information of eight second. What has been received in 8 seconds has a delay of three seconds so after eleven seconds you have started playing so eight seconds of information are to be buffered and then it starts playing. And as it plays the other message has come in and that too has to be stored.

Usually what can be done is some kind of double buffering can be used so the first file is stored in one buffer, second file is stored in another buffer, third file is stored in another buffer so that as the first file is played from the first buffer the information can come to the second buffer because of variable delay and it can reach here. So you can see that in spite of the variable delays of eight seconds, three seconds, six seconds, ten seconds it can be played without any jitter by buffering it and playing after eleven seconds and at different instants of time you can see when the playback play starts at different points this is the buffer required (Refer Slide Time: 38:49).

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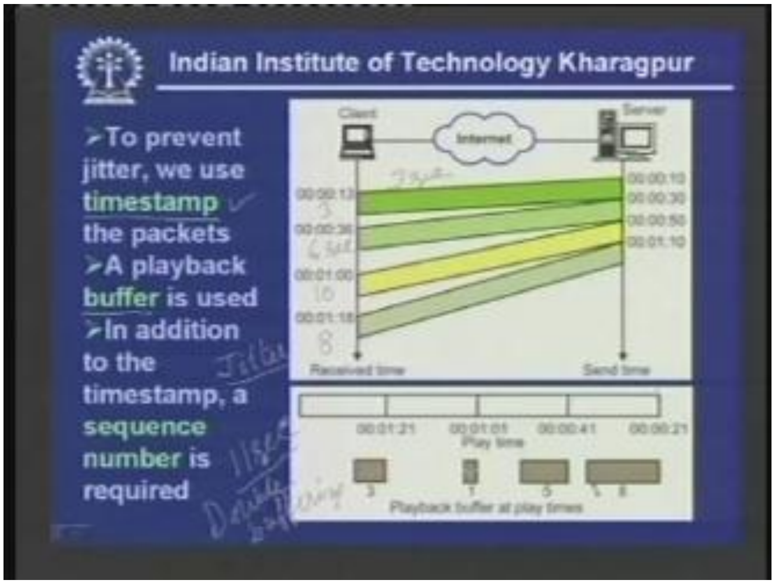

In the first case information of eight seconds is required, in the second case information of five seconds buffer is required, in third case information of one second buffer is required and so on. This shows that in addition to time stamping there is need for buffering. However, these two are not enough; it requires some kind of sequence number. what can happen is, suppose you have received a message starting with time stamping 00 00 13, second one you have received with time stamping 00 01 00 and this packet has been lost on the way this one has been lost on the way (Refer Slide Time: 39:40) so it has been damaged in such a way the other side the receiver side has not been able to recognize it, what will happen in such a case. So in such a case the receiver does not know that a particular message packet has been lost.

Therefore, to overcome that situation it is necessary to use in addition to these two sequence number. So, each of these messages are provided with a sequence number. That means this is provided with sequence number one, this is provided with sequence number two, this is provided with sequence number three, this is provided with sequence number four.

So, if the message with sequence number two is not received, that means if one is received and then three is received and two is not received then the receiver will know that message three has not been received. This is how by using time stamping buffer and sequence number it can be played one after the other in a continuous manner without any jitter it can be played and even as in frame reaches one after the other, suppose this frame

(Refer Slide Time: 41:04) reaches later than this one that also will be taken care of by this sequence number. So these are the facilities required for real time interactive audio/video sequence transmission and suitable protocol is to be used to support this.

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One application of this real time interactive audio/video is video conferencing. Using a network a camera and a head set people can interact as if they are talking face to face in a room. Nowadays if you want to hold a meeting you have to go somewhere to attend the meeting and possibly people from different parts of the country will come to attend the meeting. But nowadays busy executives don't have time to travel so in such a case one can do videoconferencing, it can be done for holding meetings, it can be done for holding interviews etc.

For example, a person sitting in Bangalore can take an interview of a person sitting in Calcutta or Kharagpur talking to each other and they can see each other as if they are talking face to face. These are the general applications; conducting interviews, holding meetings, setting up meetings or giving lectures, this has become popular in many universities.

The professor sitting in his room can broadcast a lecture and it can be viewed in different lecture rooms, may be not only in one campus but in several campuses or in different cities and it can be done in an interactive manner. This has become a reality nowadays.

Apart from this general application another very important application is in telemedicine. A person in a rural place can take the help or guidance of an expert doctor in a city with the help of this internet by using teleconferencing. So if teleconferencing is deployed in some places from there an expert doctor can give advice to a general practitioner for a particular disease or sometimes an expert surgeon can guide another surgeon from a remote place with the help of this teleconferencing. These are the various important applications emerging based on this teleconferencing.

Two types of video conferencing can be done. One is point to point conferencing which is basically a communication link between any two locations; another is multipoint conferencing which is the link between a variety of locations. So you will require not only point-to-point which requires unicast communication but also multicast where multiple users can talk to each other. Similarly video conferencing can be done simultaneously at different places.

Nowadays for example when we hold conferences there also we use this service. For example, there two program committees one in India and another in USA. The two program committee members of USA and program committee members of India can interact by using videoconferencing; can hold meetings for deciding which papers are to be selected for a particular conference and so on. So this has become very common and widely used. As I mentioned to support this real time audio/video service such as video conferencing it is necessary to have multicasting.

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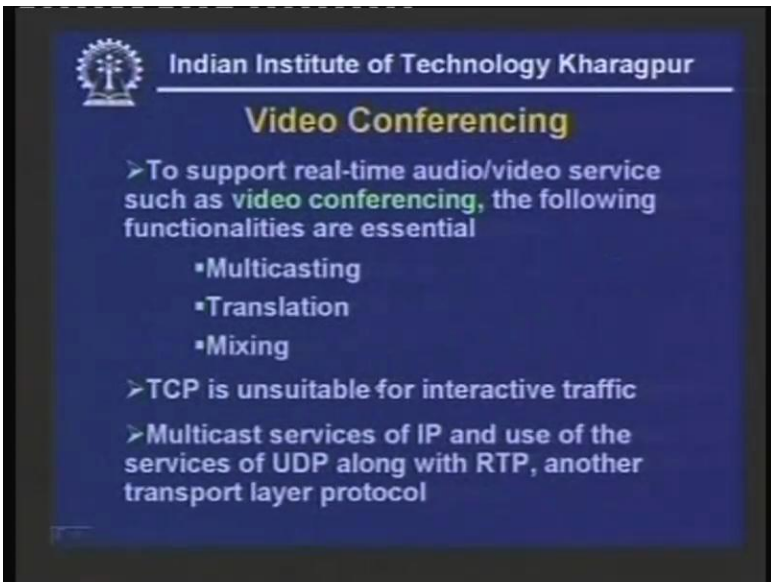

Sometimes you will require translation because all users may not be equally capable so in some cases the bandwidth of the signal has to be reduced so that some users can take part in videoconferencing or in some situation mixing has to be done, signals from a number of places are to be mixed then they can be aggregated and sent to another place so these facilities are necessary for video conferencing.

And as I mentioned TCP is not suitable for interactive traffic and we have to use multicast services of IP and use of services of UDP along with RTP which is another transport layer protocol. So UDP along with RTP is commonly used to support videoconferencing.

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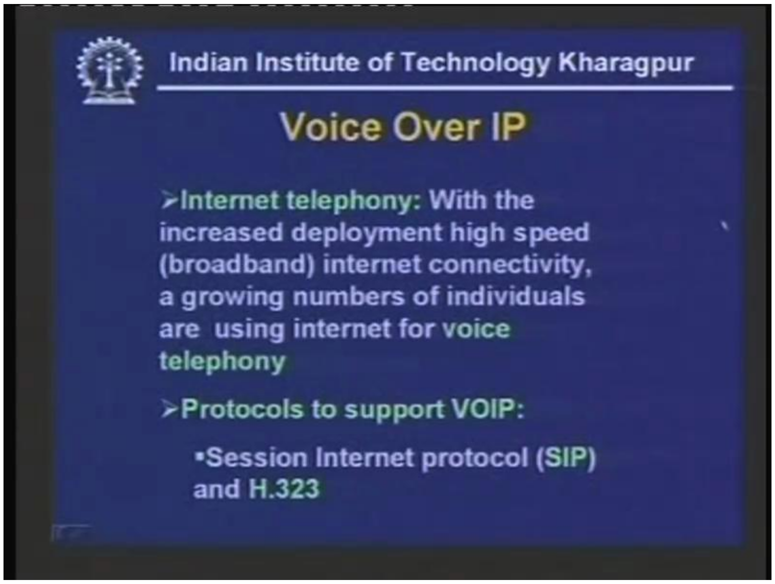

The last application which we shall discuss is voice over IP which is essentially internet telephony. You are very familiar with our telephonic conversation through Public Switched Telephone Network (PSTN) this is essentially a circuit switched network. now this similar kind of services are being provided through internet which is known as internet telephony and this has been made possible with the increased deployment of high speed internet connectivity and a growing number of individuals who are using internet. Because of this a growing number of individuals are using internet for voice telephony.

I shall discuss two protocols which have been developed to support voice over IP. One is known as SIP Session Internet Protocol and another is H.323 these two are used to support voice over IP. First let us consider SIP. SIP can send different types of messages like INVITE, ACK acknowledgement, BYE, OPTIONS, CANCEL and REGISTER. Each message has a header and body.

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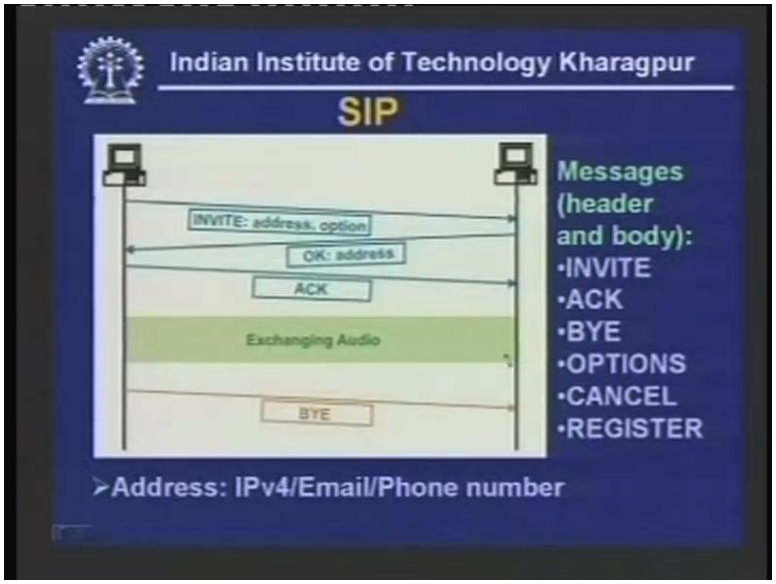

For example to set up a connection, here is terminal one and here is terminal two (Refer Slide Time: 48:16) they want to talk to each other so this terminal sends ‘invite’ message with address and various options bandwidth requirement and other things then at the other the other end the terminal responds with okay message and it also provides the address then this terminal sends the ‘acknowledgement’ message and then they can exchange audio. And after the audio conversation is over the initiator sends a ‘bye’ message to terminate the communication. So with the help of these three ‘invite and acknowledge’ is used to set up a connection, ‘bye’ is used to terminate a connection then ‘options’ are used to negotiate various options and ‘cancel’ is used to discontinue of a particular application. Let us discuss about ‘register’ a little later on.

There are different types of address options possible in case of this SIP protocol. For example, one can use IPv4 IP address or one can use email address or one can use phone number of course it has to be done using the SIP format. For example when IP address is used it has to be SIP followed by colon apal then at the rate of some IP address 14416.192.110. So this is one address format or it can be the email address so this part is same apal@cac.idkjp.net.in or it can be the phone number in the SIP format. So addresses can be provided in number of ways.

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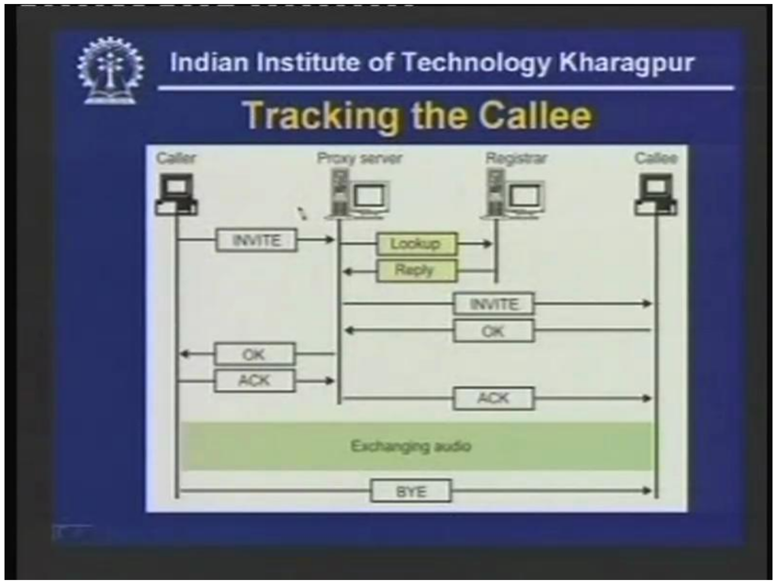

I was talking about the use of register. Particularly whenever we are using DICP then a particular user for example a callee may not have a permanent IP address. In such a case to track the callee the concept of register is used. What is being done, the caller sends the invite message which is sent to the proxy server using the email address it does not know the IP address then the proxy server with the help of a look up service sends the message to the register and the register is responsible for holding the information of each of the users so the register replies and provides the IP address. Then the proxy server sends the invite message to the callee using the same IP address and whenever it sends okay that is being communicated to the caller so proxy server and caller again communicate messages like okay message and acknowledgement message and finally acknowledgement is send by the proxy server to the callee and then the exchange of information can be done and when the exchange of information is over the 'bye' message can be sent by the callee.

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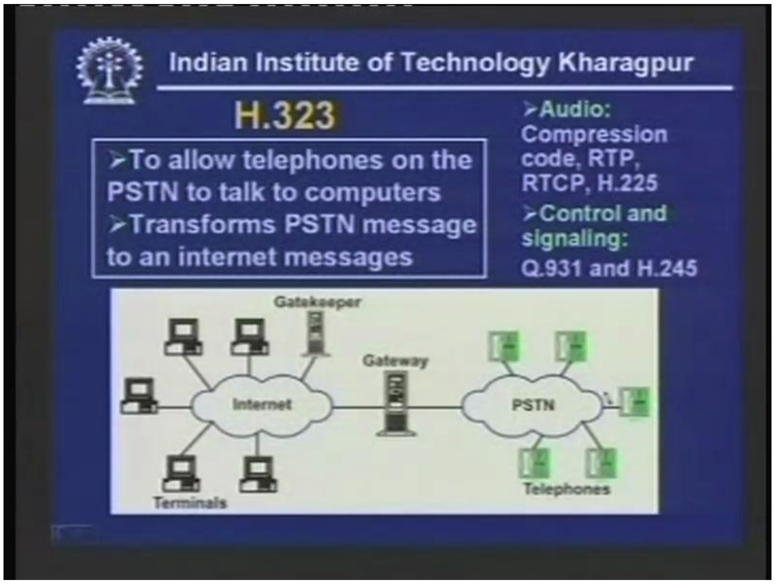

Another protocol available is the H.323. This is used to allow telephones on PSTN to talk to computers. In the previous case the communication was between two computers, here we want a telephone; an ordinary telephone has to communicate with another person on the computer. So here you see the Public Switched Telephone Network (PSTN) is linked to the internet through a gateway and a particular server known as gate keeper is required and to support this you require some protocols like compression code, RTP, RTCP, H.225 and for control and signaling Q.931 and H.245. here how it is being done, a terminal sends a message to the gate keeper which responds with the IP address H.225 message is used to negotiate bandwidth between the caller and the gate keeper then Q.931 is used to set up the connection in the same way then H.245 is used to negotiate the compression method that can be used and RCTP is used for management then Q.931 is used to terminate the connection. Thus you require a number of protocols to support this which is part of this H.323.

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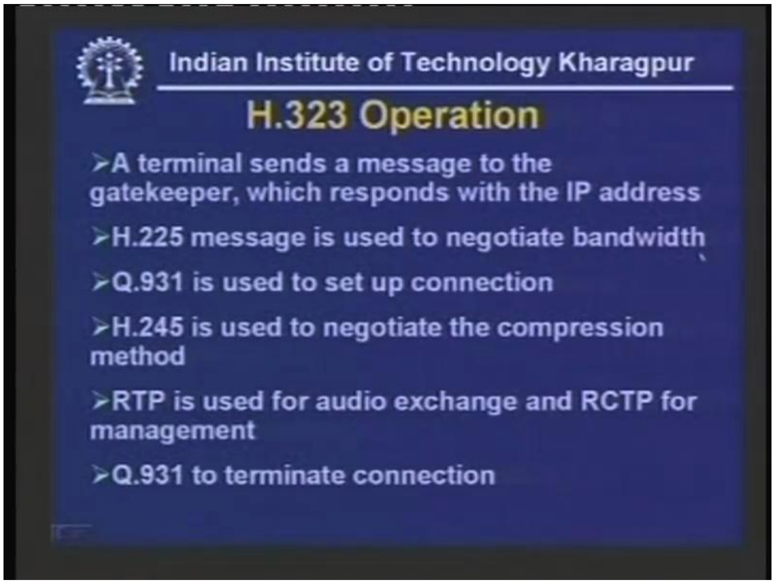

Now let us see what Skype is. Skype is the most voice over IP software service that is being used nowadays.

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Skype is a peer-to-peer VOIP client introduced in 2003 developed by KaZaa a US based company. It has become very popular possibly having the largest user base. With the help of this two people can speak with each other using handsets and microphones connected to their computers directly. It is free between any two computers.

However, whenever one wants to talk between a computer and PSTN line then of course one has to pay. Skype client can be very easily installed, within few minutes it can be installed and can be used. Skype has used astonishingly good video compressor providing very good quality audio, it also supports in addition to voice instant messaging, search and file transfer. Moreover, it uses encryption so as a consequence communication through is much secured.

Now it is time to give you the review questions.

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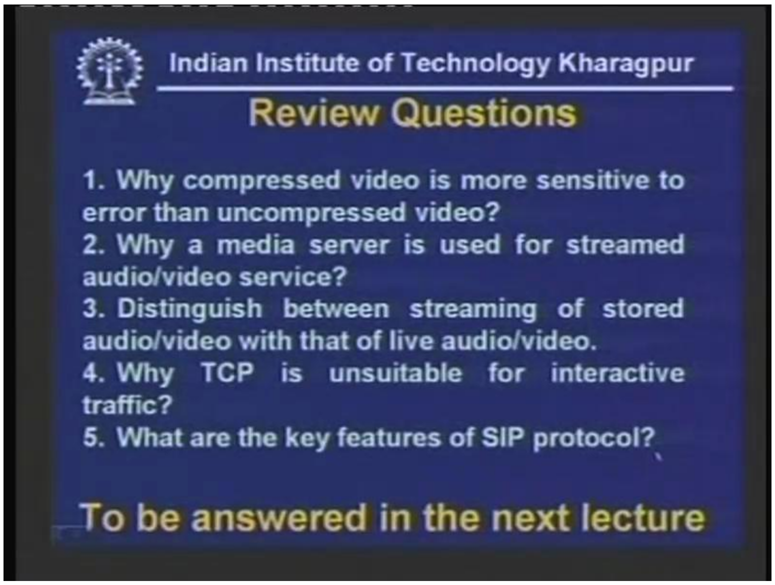

1) Why compressed video is more sensitive to error than uncompressed video?
2) Why a media server is used for streamed audio/video service?
3) Distinguish between streaming of stored audio/video with the that of live audio/video
4) Why TCP is unsuitable for interactive traffic?
5) Why the key features what are the key features of SIP protocol?

Answer to the questions of lecture -37.

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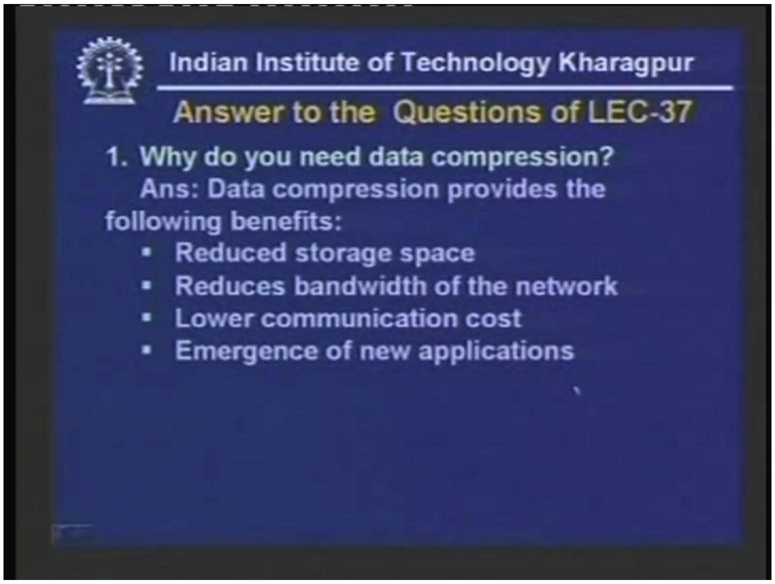

1) Why do you need data compression?

As I have mentioned data compression provides the following benefits; reduced storage space, reduces bandwidth of the network, reduce communication cost and emergence of new applications as I have discussed today.

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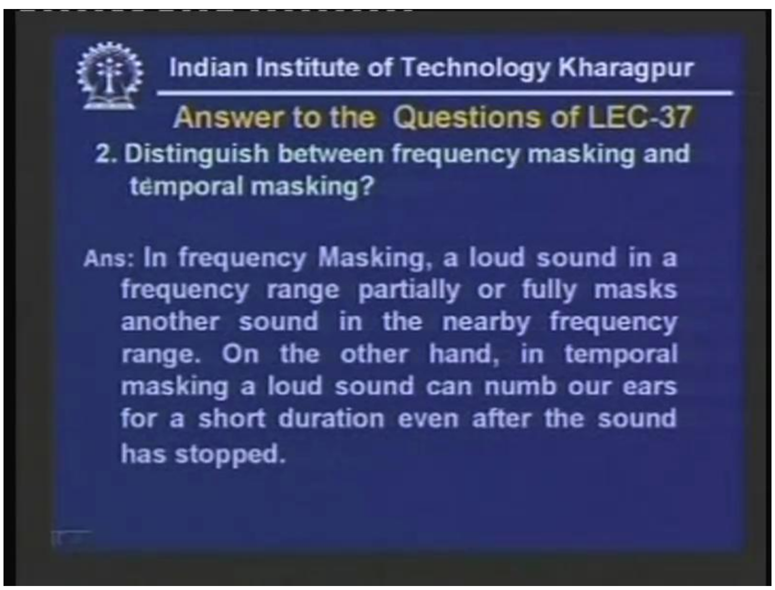

2) Distinguish between frequency masking and temporal masking?

In frequency making, a loud sound in a frequency range partially or fully masks another sound in the nearby frequency range. On the other hand, in temporal masking a loud sound can numb our ears for a short duration even after the sound has stopped. These are two differences used for audio compression in MP3.

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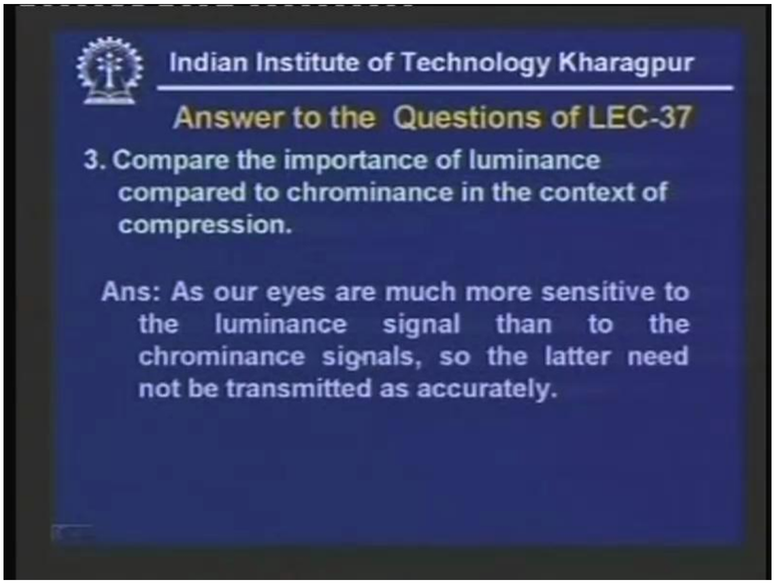

3) Compare the importance of luminance compared to chrominance in the context of compression.

As our eyes are much more sensitive to the luminance signal than to the chrominance signals the latter need to be transmitted accurately so different precision is used for luminance and chrominance as you have seen. Better resolution is used for luminance than chrominance because luminance is more sensitive our eyes so this facilitates better compression.

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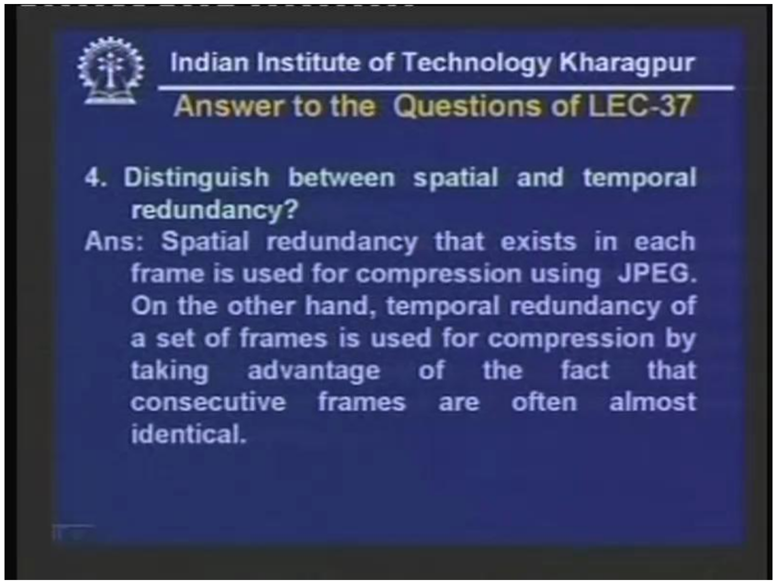

4) Distinguish between spatial and temporal redundancy?

Spatial redundancy that exists in each frame is used for compression using JPEG. On the other hand, temporal redundancy of a set of frame is used for compression by taking advantage of the fact that consecutive frames are often almost identical. That is used by MPEG and this is being used by MPEG.

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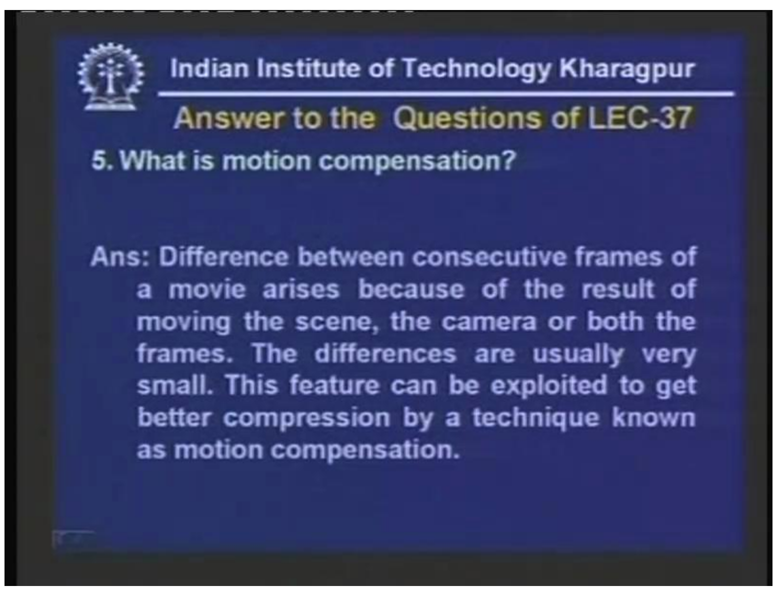

5) What is motion compensation?

Differences between consecutive frames of a movie arise because of the result of moving the scene, the camera or both the frames. The differences are usually very small. This feature can be exploited to get compression by a technique known as motion compensation as it is done in MPEG.

With this we have come to the end of today's lecture and also we complete our discussion on multimedia communication.

Thank you.