Digital Video and Picture Communication

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Lecture - 39

Video Conferencing: SIP Protocol

Today we are going to continue with the video conferencing. In the last class we had seen some ISDN-based configuration which is essentially a circuit switched technology and in this lecture we will just show you some one or two other alternative techniques of connections and then we go over mainly to the session initiation protocol or the SIP which is the protocol that one uses in order to establish a video conferencing session. The setup as well as the tear down they are defined by a protocol.

In fact, one can use the H dot 323; especially whenever one needs the interface between the PSTN and the IP. SIP is purely an IP-based protocol. It does not assume or does not necessitate any a priori connection setup be it a real connection or a virtual connection. This is what we are going to cover today and before we go over to the actual topic let me make a mention about one or two important references which you can go through in order to learn further about these techniques and whatever I have covered or going to cover are mostly based on these papers.

This is the first paper that you can refer to is that about from the ((Burguds)) ((00:02:33)) paper which I have already referred to you that is regarding the voice over IP; even for voice over IP as well as for the video conferencing you can refer to the paper by Hong Liu and Petro Mouchtaris. Both of them are from Telecardia Technologies. In fact, Telecardia Technologies are the ones who had participated actively in the development of the H dot 323 and this is Voice over IP Signaling. This gives you fairly detailed idea about H dot 323. Some of it you can get from Burguds paper and also from this one H dot 323 and beyond.

So when they talk about beyond they also cover the SIP protocol which I will be discussing mainly today. And this paper appeared in IEEE Communications Magazine. This is not a transaction paper this is a magazine paper so it should be easy to understand for all of you;

the page number is 142 to 148 and it appeared in IEEE Communications Magazine October 2002 no October 2000 this is October 2000; and in the same issue; actually this is a special issue on the Advanced Signaling and Control in the Next Generation Networks that is the special issue October 2000 and in the same issue you should refer to a paper by H Schulzrinne.

In fact, Schulzrinne is from Columbia University; they had contributed to the development of SIP and Jonathan Rosenberg. The title of the paper is The Session Initiation Protocol. They are going to talk in details about the SIP and it is the Internet-centric Signaling so the same source that is IEEE Communication Magazine October 2000; the page numbers for this one is 134 to I think it is the paper just before this so 134 to 141 is this one that is on the SIP and then followed by this from page number 142 just the next page onwards you will get Voice over IP Signaling: H. 323 and Beyond.

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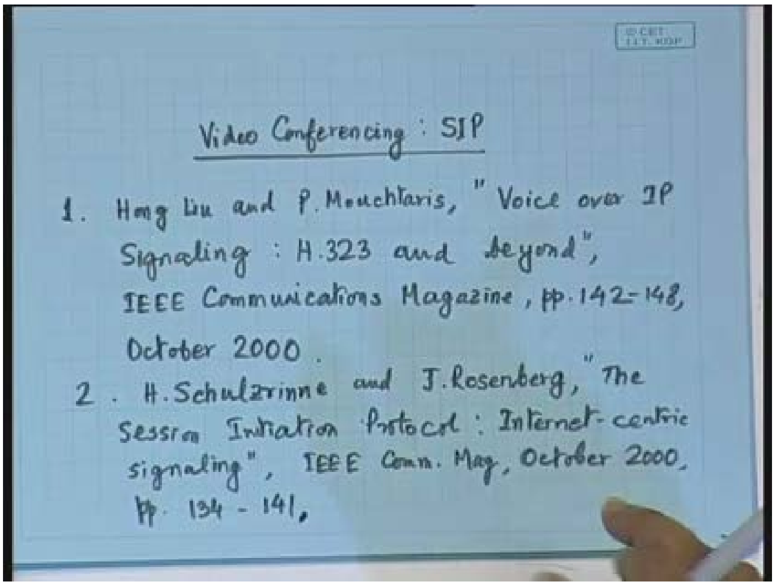

I suggest that you should refer to these two references in order to get a better feel about these techniques. So let me go over to the discussions about the voice over the ISDN-based video conferencing. In fact, I was showing you two alternative configurations mainly: one is the one that uses the small office configuration or where we use the BRI method of connections. The other is that where we can have a medium office configuration which would use large number of terminal endpoints and it should be in turn connected to the access switch and the access switch will be connected to the PSTN network using the PRI interface. So the PRI would actually be shared by all the terminal equipment which are there within the establishment. The idea is that one can ask for or negotiate for any bandwidth within 1.544 megahertz. So, if bandwidth is available one can go in for a high bandwidth video conferencing in fact and one has to reserve the resource accordingly. This is what is possible using the PRIs.

PRI has got some distinct advantage as I mentioned about the about much less cable requirement because whatever cabling you have to do is mainly within the office but outside the office it is only the PRI which carries the traffic. And then obviously less cabling means improved reliability and the dynamic bandwidth allocation which was not possible in the case of the BRI type of interface because there we had to use to dedicated BRI links to that and the ability to make hard bandwidth calls and redundancy. So these are some of the issues which gives the alternative configuration and advantage.

What I want to show you is mainly is large office configuration and in the case of large office configuration for the ISDN video conferencing essentially what we look for in a large office is that the system should not be down because of internal factors as well as for external factors.

Now one thing which you see there is that if supposing the access switch that fails; in the case of the ISDN video conferencing if the access switch fails in in that case the whole system comes to a standstill so we should have some redundancy at the switch level. And not only that it comes to the question of network also that supposing you are registered with one company one telephone company and there the cable gets down due to some reason some weather conditions some natural disasters or something like that because of which one company's telephone network is down and if you have the connectivity with another company then you can also have some kind of a redundancy so that one can establish an alternative path.

In fact, we can think of a configuration like this that say we have a switch over here (Refer Slide Time: 10:21) and then this is what is external to the switch that means to say that outside the establishment and within the establishment we will be having a number of terminal endpoints which are connected to this switch in a very similar way what I have already shown and these are all the internal PRIs so these are the internal PRI and here we will be having the external PRI external PRI and this is connected to the PSTN.

So up to this it is nothing new to us because we had already seen this type of configuration. But in order to have a redundancy what we should have is that not all these terminal endpoints we should be connecting to this switch. But in fact, we should have a connectivity with another switch and some other terminal equipment of the same establishment may be connected to the second switch so the second switch serves as a redundancy and you can use the external PRI for that and use a separate PSTN network; means if this is provided by company A (Refer Slide Time: 11:40) this may be coming from company B so that even if company A’s network is down you have company B’s network running and in fact, these switches can be internally connected together using PRI link.

Therefore, the advantage is that if supposing PSTN A is down then you get the full connectivity in the sense that PSTN B is up so in that case all the terminal equipment which are there in this segment as well as which are in this segment can make use of the PSTN B network and vice versa; in case of PSTN B failure it can go through the PSTN A.

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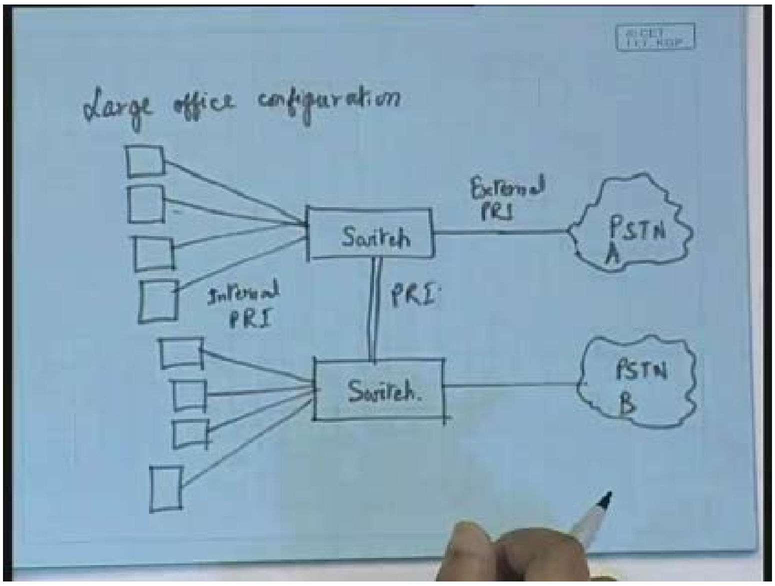

So this is the large of its configuration that one can have building in some amount of redundancy in the configuration. And in fact, when it comes to enterprise to enterprise connectivity one can not only use the PRI and go through a PSTN network but many a times for very what you can say for very urgent and for very strategic communication one may not be depending upon the telephone network or rather the establishments from one office to another.

Like say for example; if we are talking of a big banking organization supposing their one office is located at New York and another office is located in London and this New York office and London office then to have their communication their video conferencing, voice over IP and all t things they want their links to be up so what they should do is that they should not only have the PSTN network but as a backup they should have the T1 link which will be dedicated to the organization and they can connect within themselves.

So if it is the organization's branch office located at a place connected to an access switch over here and then we have the PSTN network and then it goes to another location that means to say their another office their another branch and this is internally connected like this and in addition to PSTN one can also have a dedicated T1 link and T1 link as you know that T1 link would offer you a bandwidth of 1.544 megabits per second which is fairly good enough for the video conferencing. So in fact, your video conferencing can still be up by using this dedicated T1 link.

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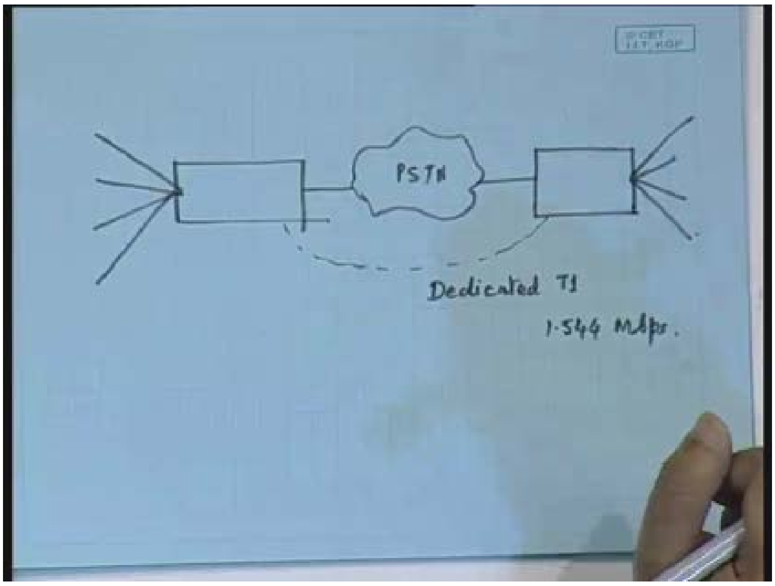

So, many a times the big banking establishments, insurance companies they can use this sort of a dedicated links and also especially in today's context people need to have their connectivity even if they are in the ISDN connectivity they should also have a connectivity with the IP so they should have a gateway or rather what you can mention as the IP to ISDN gateway so that they can have the connectivity with both IP as well as the ISDN. So there one can have configuration like this.

Using the IP oblique ISDN gateways and video bridges: in fact, I mean, in the last class I was showing you about the different types, typical examples of connections of the MCU's or what is known as the multipoint control units; they can realize like I have shown you the broadcast type of configuration I can show; I had also shown you the one-to-one connectivity then also shown you the combination where several streams can be put together into one bundle stream and then distributed to others.

So all these things the MCUs are also referred to in many of the literatures as video bridges so do not get confused by the term because many a times while reading the journals or references you may come across the term video bridges so do not do not think that it is it anything new, video bridges are of course the same as the video MCUs. In fact, what you can have as the IP/ISDN gateways and video bridges the configuration could be like this that (Refer Slide Time: 16:55) supposing you have a switch that is the same access which that I am talking of and to this you are having the terminal endpoints and then you have connectivity with the PSTN network and then you can have links which are connected to a video MCU; this is a video MCU and the video MCU could also be connected to a LAN.

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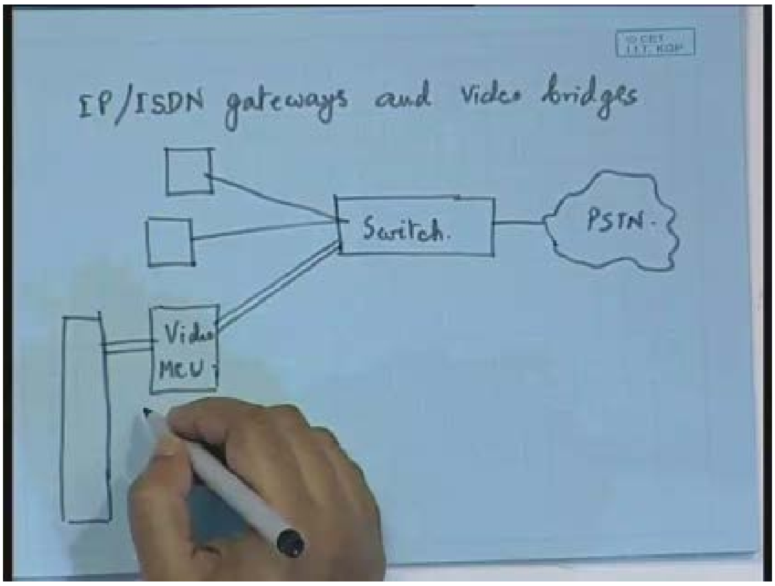

Now, because the MCU is the ultimate controller you need the MCU connected to this; not only eventually it should be connected to these terminal endpoints but also to the LAN. This is where I have LAN and then we should have an IP to ISDN gateway IP/ISDN gateway and the IP to ISDN gateway would again be connected to the switch so that one can establish ultimately a connectivity between the PSTN and the LAN.

In fact, more and more organizations are today preferring with this mode of configuration because although many companies who require a video conferencing they had invested lot of money in building up the ISDN video conferencing facility. But today seeing the growth of the IP-based video conferencing people also cannot suddenly switch over from the ISDN to IP so what they build up is the IP/ISDN gateways and try to have a connectivity with the IP as well.

So, in order to have the have a proper connectivity with the IP what you essentially require is an efficient signaling technique. In fact, signaling techniques we have already come across when we talked about the H dot 323 and especially within H dot 323 we also studied the signaling protocol which is actually used in the ISDN. But in this case for the session initiation protocol or SIP we are primarily concerned with the advanced signaling and control functionality of multimedia services in the IP network. And it is not only used for advanced signaling and control although that is purpose that it is used for advanced signaling and control over the IP networks but it can also be used for things like instant messaging, generating alerts, event notifications so, for all these things the same SIP can be used.

In fact, SIP when I describe it you will be finding that it is a very attractive kind of protocol just like the way the http is. In fact, the http the hypertext transfer protocol it is a text-based protocol; all the signaling they are text-based and very similarly for the session initiation protocol also it is a text-based transfer of messages although it is strictly not a transfer protocol in the sense that it is only used; I mean, such text-based signaling is only used for the connection setup but once the connection has been established the rest of the things or rather the media streams have to go through the RTP.

Thus, now primarily although you will find a lot of similarity between the features of the SIP with that of the PSDN or ISDN signaling that we have already studied, the main and the major difference that you will come across is that unlike the PSDN or ISDN this signaling technique does not require any reservation of the resources and it never assumes any a priori connectivity be it a real connection or a virtual connection. This is one major aspect that it does not reserve any resource that is one of the major feature does not reserve resource or assume real or virtual connections.

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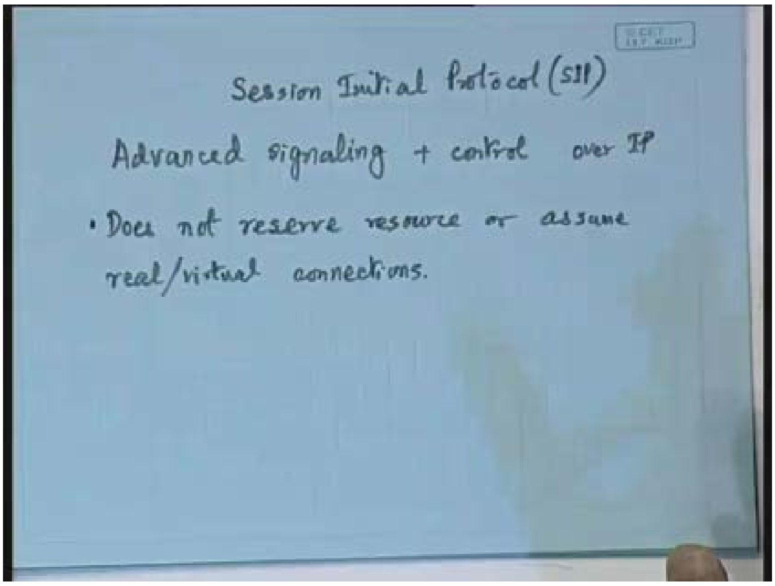

SIP: this being protocol architecture has to necessarily use some other architectural support some other protocol architectural supports as well. So what are these that the SIP architecture this includes the following protocols.

One is the RTP which is used for the transportation the actual media packets transportation sorry transportation and then it uses another protocol which is called as the RTSP the full form of it is the Real Time Streaming Protocol Streaming Protocol and this is for on-demand media streams and then it uses the MGCP you must have forgotten MGCP I mentioned few classes back that is the Media Gateway Control Protocol MGCP and then the SIP also uses....... I mean, for the multimedia sessions description it uses what is called as the Session Description Protocol Session Description Protocol in short form it is called as the SDP, so SIP uses the SDP for the description of the session.

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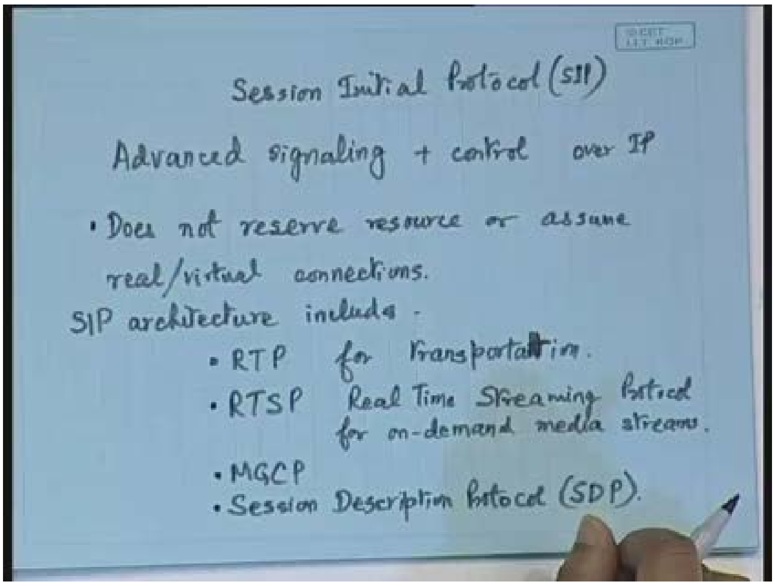

When I was saying that SIP as a protocol has got lot of similarity with the http is not only the text based signaling but also one of the major similarities which you will be finding is the usage of the logical address rather than the IP address or the specific location specification; no specific location specification is actually required; one can one can use the resource that is to say that one can use a logical address to do that and in fact, the usage of logical address the greatest advantage that you come across for that is that the users in the IP network they can have a mobility just like the way that if you have an account in the Google mail Gmail.com or if you have an account on the Yahoo or any other domain where you have a web mail facility then no matter wherever you maybe in the world if you have any internet access anywhere you can check your mail because your mail ID will remain the same universally and you can login to this and you can get your emails. Also, the people who are mailing they will not be mailing you at any specific destination they will just give your mail ID they will know your email address so they will just give your mail ID and then your mail will reach you no matter wherever in the world you are. So you have mobility.

And the same thing is expected whenever you are going to have a video conferencing because naturally I mean, you cannot expect that in order to have a video conferencing I will be sitting at one particular place. Most of the times I may be in my office but even then also I may have to travel out of office, I may have to leave the station and go somewhere else and even then also I should be having all the facilities to enter into any video conferencing session. So it should permit me that. That is why instead of specifying the physical location it is the logical addressing scheme that works just like the way we do it for the internets and for the emails.

SIP as a protocol is not very old. in fact, it was established in March 1999 and this protocol was actually published by the IETF the Internet Engineering Task Force I have already mentioned about them they are the body who have proposed all these protocols or standards for the internet users; and since its publication in March 1999 there has been at least five or six major modifications which have been done and in fact, the SIP is also an evolving protocol, at least in the initial years it was no evolving.

Now if you want to know about the details of the current SIP activities one of the very useful resources that I can suggest for you is the Columbia University website. You have to refer to http://www.cs.columbia.edu and you should see their page on the SIP so that will give you a fairly updated idea about what are the SIP activities taking place now.

Now just to give you a brief overview about the SIP there are four logical types of entities that participate in the SIP. The first is what is called as the user agents because the source or the destination they are referred to as the user agents so user agents are the ones that initiate the requests and also the ones which receive the request and participate. Hence, user agent’s job is to initiate request and serve as the destination.

Then we have another important logical entity which are called as the registrars. In fact, it stipulates the usage of registrar servers because the users must be registered within a domain; just like the way it is there for any internet applications that you are registered with one internet service provider and this keeps track of the users within their domain, so keeps track of users within their domain; like for example: iitkgp.ac.in you are registered within this domain so no matter all who are present here as participants as the contact participants they are all registered with the iitkgp.ac.in and see that this is very important because even if at one point of time you are not directly connected to the iitkgp.ac.in server you should be locatable.

I mean, if you have a webmail facility over the iitkgp.ac.in you should be locatable. Your mail can be accessed from any other server to which you are temporarily registered; you are temporarily having a login or you are having a login with another server another ISP even then also you should be able to keep the contact. So registrar keeps track of their own domain and that is the purpose.

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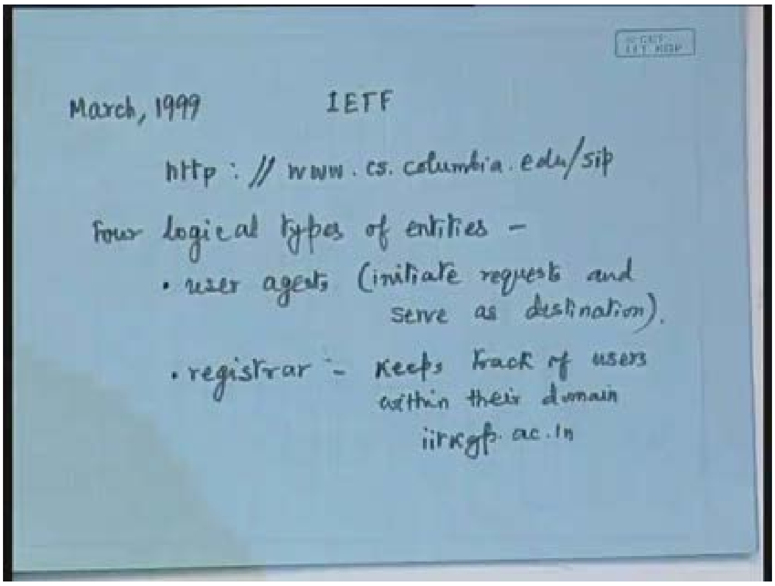

Then another entity is obviously the proxy. Proxies as you know that these are the application layer routers. In fact, you all if you have gone through the networking courses you must be knowing that what the job of the proxy is. In fact, all these connection establishments take place in the client server mechanism. that the user When the user places a request, your user agent places a request then the user agent accesses the client and user agent places a request to the proxy because you are your first step of connectivity is the proxy so you are connecting to the proxy and then the proxy is actually forwarding your request to the end destination or the proxy may contact another proxy in order to locate the destination.

Thus, what the proxy does is that the proxy establishes the connection and then the proxy is actually............. proxy is an entity which can serve as the client when it is requesting or forwarding the request to the destination. Then other than proxies we also have what is called as the redirect servers.

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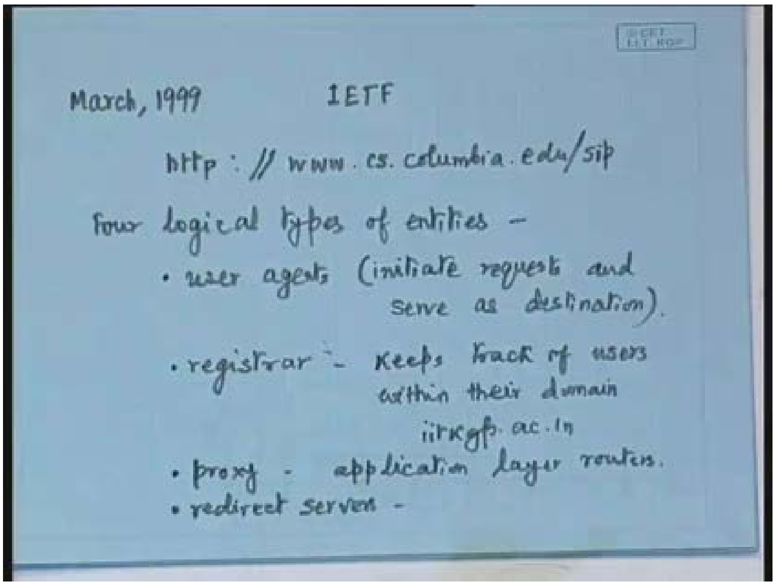

The redirect servers actually................ the difference between the proxies and the redirect servers is that the redirect servers will receive the request and it will return the location of another SIP user agent and then it will be your botheration to establish the connection with that user agent. It does not establish the connection on its own whereas the proxy whereas the proxy forwards that proxy makes a request whereas the redirect server only informs you about the location.

This is just like you just call up an organization then if you if you come across any very friendly operator or you come across somebody who is not the ultimate person to whom you want to speak to.................., if the person is friendly he or she will tell that okay I am connecting to his number so he will make the he or she will make the connection and that will facilitate you to talk to the end party. But at other times the person to whom the call goes he may just tell you that please dial this telephone number this is not the telephone number of the user to whom you want to connect but his or her telephone connection number is this although he could have connected I mean, just taking little bit of extra pain he could have transferred the call but some people they do not want to transfer the call but just tell you the numbers so that you can ultimately contact. Therefore, the redirect servers do like that that the redirect servers will tell you the SIP address of the destination but do not establish the connection.

Now, typical SIP session actually goes through like this that we have user agent who originates the call and user agent is connected to a proxy and this proxy is in turn may be connected to several other proxies. I am showing this with a dotted line means there may be more than one proxy or just one proxy maybe enough.

Supposing if you are contacting if you are let us say abc@iitkgp.ac.in and you want to contact somebody who is xyz@iitkgp.ac.in that means to say that both the user agents are in the same domain one having the login ID as abc and other is having the login ID as xyz so they can be connected by the same proxy and there only one proxy is enough because the domain name is the same and the last proxy to which the end users are connected one can have several user agents who will be there as the end destination so these are the different user agents.

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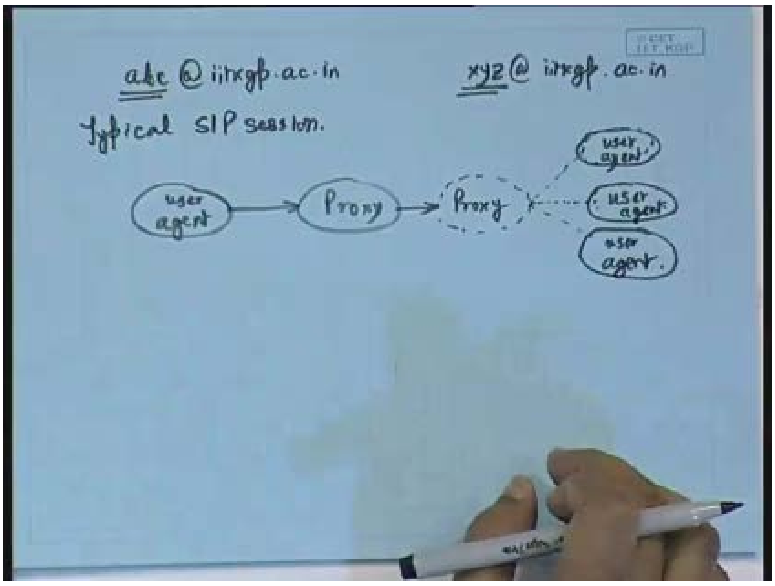

This is a typical SIP session which would involve these and there are five or six request methods. So the request methods that one uses request methods for SIP some of the SIP user's request methods I am mentioning. One is the register; register is a request method that allows either the user or a third party to register contact information with an SIP server. Then the initiation of the calls signaling sequence is done by the message which is called as the invite. The invite message initiates the call signaling sequence. We will see an example that will make it very clear. Then we have the acknowledgement when the actual end-to-end connection is established and CANCEL. And to terminate a session there is another message which is called as bye. So BYE terminates a session.

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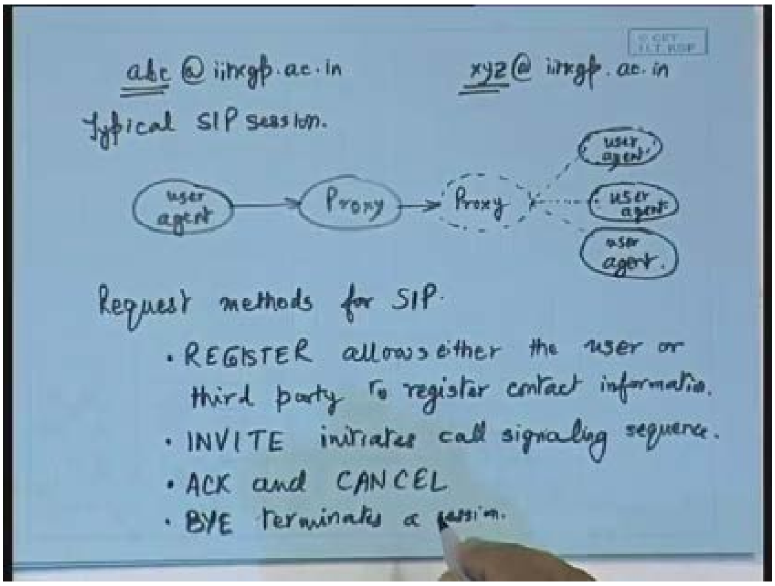

Just to show you a typical example let us say that connectivity with the single proxy; say abc@iitkgp.ac.in connected to xyz@iitkgp.ac.in so only one proxy server is there which is at the iitkgp.ac.in. This is a proxy server, this is an SIP proxy; of course one can use the same computer as an SIP proxy and also for the IP proxy and supposing there is one party who is here and there is another party say this is abc and let us say this is xyz (Refer Slide Time: 40:54) and we will show the signaling sequence like this just in the form of the timing chart that we prepared last time and abc has got a telephone; telephone means we assume that this is an SIP telephone which will be having a handset and you can have a voice communication through that and also you have a screen may be a small LCD screen through which you can see the other end's picture also. So it is a video conferencing facility in the sense that it is a typical kind of video conferencing that one can have; portable video conferencing equipment where on a desktop phone you are having the handset as well as the LCD screen to see the pictures.

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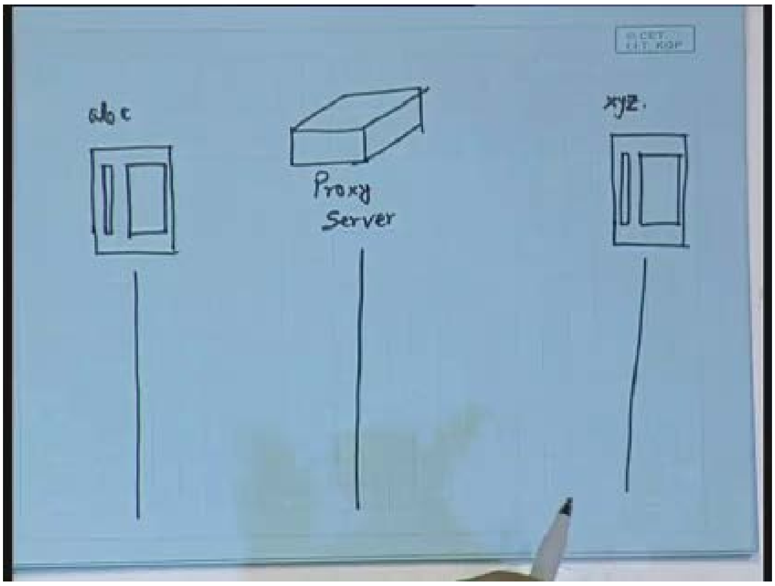

Now both are connected to the same domain and supposing abc wants now to communicate with xyz; now xyz is having an IP address; xyz is connected to a computer where there is an IP address for them but abc need not have to remember that IP address; who remembers IP addresses after all. Now he may be either remembering the mail ID of xyz or since they are in the same organization they may not be even instantly remembering their mail ID, they may be remembering just the telephone number. So either the mail ID or the telephone number is good enough in order to provide a logical address.

Supposing xyz’s telephone number happens to be 9876 that is the internal telephone number that xyz is having, so abc can just say 9876, he can just dial 9876 from his phone from his SIP phone and then abc's SIP phone has got an address book where it will convert that 9876 into 9876 at let us say iitkgp.ac.in and then it will forward this to the proxy server; then the proxy server has got a base a database where it can find out that okay 9876 is corresponding to xyz then it can actually get the IP address of...... I mean, it can locate xyz and that xyz may be that time logged in to the same proxy server and then it can locate xyz and it will have the IP address for that so it can forward the call to xyz. So the sequence of signaling would go through like this that there will an invitation so it says an invite message and this is the first message that goes so every message is having some kind of a message identifier.

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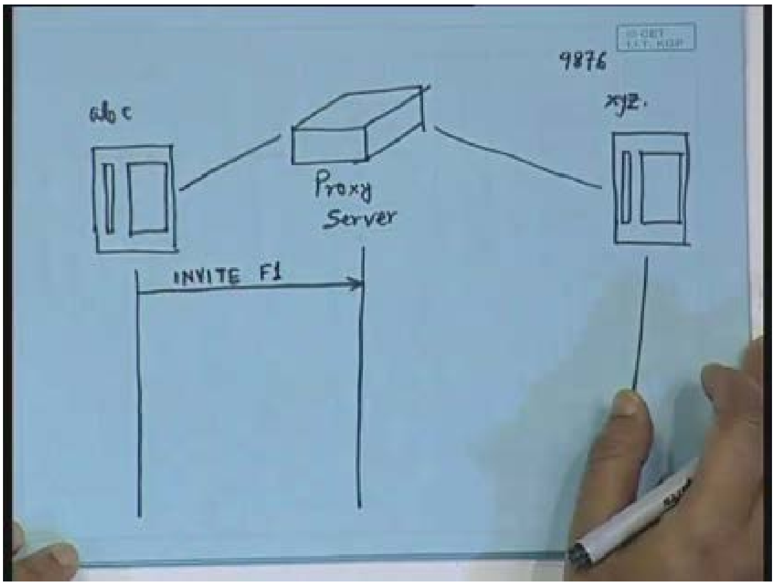

Let us say that we just say invite F1; F1 is a kind of a message identifier; the first message is F1 and we will call the second message as F2. So what will be the second message? The second message will be that from the proxy server to the end destination there will be an invitation. So now the proxy server invites xyz. So this is invite and this is the message F2.

Now, when it sends this invitation the proxy server may not be sure that whether the connection can be established or not but at least it sends an acknowledgement to the source saying that it is trying, so trying is another signaling message that it can give, so trying and this is say the message F3 that it is trying.

Now every message is also having a three digit number just like the way you have even for http error 404 the requested URL could not be found so that is error 404 so like that every message has got say three digit number, say it could be 100 could be the type of the message so 100 trying F3. And now with the invitation now xyz's telephone has to get an alert, it has alert the user xyz so the alerting is done through a ringing and when the connection is established that time the ringing.............. so the ringing is not only in this telephone but the ringing has to be echoed back to the source. So let us say that this is message 180 so ringing this is F4 and this ringing will be forwarded to the source so here also we will be having a ringing.

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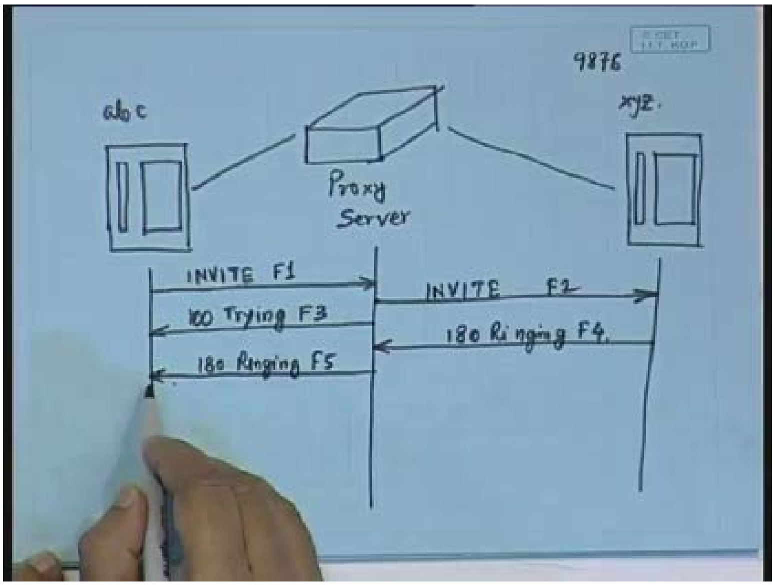

So now abc knows that xyz's phone is ringing and now xyz lifts his or her handset or presses the speaker phone button then there will be.............. and when the actual button is lifted or the handset is lifted and the actual dialogue is going to commence then you are going to send a message called okay, say this is the message F6 and again this will be forwarded by the proxy server. So what the proxy server is doing is that it is once acting as the client then again acting as the server and then this becomes 200 OK F7.

Now this is; the signaling part is already done and in fact, I should not have drawn this line that long. Now the calling party that is abc will send an acknowledgement. So this is an acknowledgement. now this acknowledgement so Now an end-to-end contact is there and in fact, abc's telephone now knows the IP address of xyz. Wherever xyz is located xyz's IP address is now known and then the acknowledgement is sent and then they can exchange their messages or the data streams using the RTP media stream. This is the RTP media stream communication that will take place and then one can and when the terminal and when the session ends that time it has to send a BYE signal so BYE comes from the called party to the caller and then the BYE is again acknowledged by saying the message OK. So this is 200 OK which will go from the abc to the xyz and then the actual session will get terminated.

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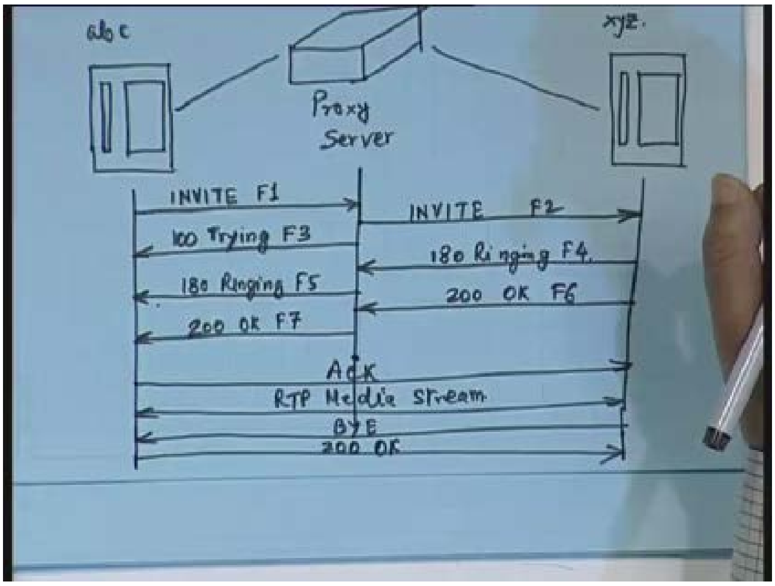

Therefore, this is the so this is a typical example using one proxy server and when one uses multiple proxy servers then it is just an extension of this and where multiple proxy servers could be required is that when especially the domain changes.

Supposing abc@iitkgp.ac.in wants to communicate to xyz@iitb in IIT Bombay say iitb.ac.in so now the iitkgp.ac.in they have to identify the proxy server of iitb.ac.in and then only the connection can be established. So, for iitb.ac.in which abc will give as the email ID that it is xyz@iitb.ac.in, now the proxy server has to look for the DNS name lookup that what does that iitb.ac.in actually means in terms of the IP address and then the DNS happens to find out the IP address and then the call can be forwarded to the IP to the actual IP address of the iitb.ac.in which will be the proxy for iitb.ac.in. So it goes from iitkgp.ac.in proxy to iitb.ac.in proxy and then goes over to the xyz. so this is This is just an example of the SIP. In the next class I will be concluding the other aspects of SIP. So for this class, thank you.