Digital Voice and Picture Communication

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Lecture - 33

VoIP Signaling: H dot 323 Protocol

....................introduced the basic ideas about the voice over internet protocol and the fundamental aspect that we had seen is unlike PSTN the Public Switched Telephone Network which is essentially a circuit switching system which requires a connection to be established beforehand and hence it is costlier because you need the entire connections to be reserved for the call duration and vis-a-vis is that the voice over internet protocol became popular because unlike the circuit switching here you are using a packet switched scheme in which case you are sending the digitized voice data in the form of packets over the network so it does not require the exact connection and the packets can be delivered according to the packet switching protocols and there we found that because it is a real-time requirement we have to use the real-time protocol supported with the UDP; we cannot use the TCP/IP because in this case this is being a real-time there cannot be any retransmission request or the acknowledgements, these things we cannot do under the real-time anyway.

Now one of the points which we touched upon was the aspect of the end-to-end delay and the end-to-end is a very important parameter whenever we are communicating between two terminal equipment. One terminal equipment at one place and the other end we are having that and there may be a very larger geographical distance of separation between them, it could be within a campus, it could be within a city, it could be between the city in a nation, it could be even from one country to another, it could be even one continent to another so the amount of delays would depend upon several factors.

We have seen that even in the circuit switched telephones also, in PSTN also we experience a significant amount of delay especially whenever we are looking at the very long distance communication, whenever we are talking across the continents there we can feel the amount of delay which one gets in order to get the response and sometimes that really spoils the real-time conversational effect. Now it is there in the circuit switching and in the case of the packet switched networks it is even larger.

In the case of circuit switched what we do is that when a large number of users are involved and large number of channels have to communicate we adopt a thing like the time division multiplexing so a PSTN employing TDM that experiences delay but a PSTN but if we are, instead of PSTN we are using the VoIP using the internet as a backbone there the end-to-end delays could be even larger but there is definitely a degree of specification or degree of tolerance which is mentioned which is defined in the standard so that the real-time conversational effect is preserved.

And as I was telling you, I think you appreciate now that the amount of end-to-end delay is a factor of several things and one of that is the buffering. And we have seen that if we want to reduce the bandwidth and make a low cost for it; in fact the reason why we are going in for the VoIP is the low cost because we are not reserving the resources, we are not reserving the transformation lines during the call but we are just transmitting the packets so because of that it gets much cheaper so we should that that is the reason we should also try to use a low bandwidth channel let us say something liking in G dot 729 what I was mentioning there 8 kilobits per second is quite a standard one, but again as you go in for lower bit rate there the amount of delays or the amount of buffering that you require that is also considerably larger because you need to accumulate a sufficient number of bits before you can transmit it as a packet so the packet accumulation time that is also very important and that contributes to a large delay. But let us see that what is specified.

In fact what is specified for ninety percent of the conversational application is that I think I made a mention to you that the end-to-end delay is something like 150 milliseconds, it is specified to be 150 milliseconds. Of course 150 milliseconds of delay that may not be possible, I mean, it may not be possible to restrict the delay to 150 milliseconds whenever we are talking of the calls from one nation to the other, one continent to the other things like that so there the delays would be higher.

But a typical delay component would be something like this. So we can just make a quick look at what are the different delays that we have and there is some delay budget which is a rough specification; do not think that it is a hard and fast mandatory that this has to be fulfilled; so it is the on-net budget if we say that how much of delay budget will be there, we are expressing the times in milliseconds and the delay source. So we are considering different sources of delay when we are using a standard like say G dot 729.

It is G dot 729 and basically we are referring to 8 kbps codec 8 kilobits per second so 8 kilobits per second you can take to be the bit rate and just see the different delay components. The first one is the device sample capture that means to say that whenever you are capturing how much of delay you encounter from the device itself device sampling capture so there it is 0.1 milliseconds. The encoding delay, this is of course a bit involved thing, forms nearly 10 percent of the total delays is contributed by the encoding delay which basically means the algorithmic delay and the processing delay that you encounter and that is typically 17.5 milliseconds and what is essential in the case of the VoIP is that because here the packetization at the source and depacketization at the destination these two processes are mandatory so that is why you are doing the packetization and depacketization delay depacketization delays. This is also very significant, 20 milliseconds; this is something which will be avoidable in the case of the PSTN because there you are not using any form of packetization.

Then you have move to output queue or queuing delays; the queuing delays up to output; this is not much, queuing delays is typically 0.5 milliseconds, then the access or uplink transmission delay so access link access link transmission delay, this could be maximum up to 10 milliseconds and what is unpredictable is the backbone network transmission delay. So this is the backbone network transmission delay.

So, although we are saying a typical delay of 150 milliseconds when we add up all these things so backbone network transmission delay, that being unpredictable we cannot really comment on that so let us say that we just put something, let us say that we put a number x we keep it undefined, let us say that x milliseconds is the backbone network delay and then the so this is access transmission delay for the uplink and this is again the access link transmission delay access link transmission delay for down. So this is for down (Refer Slide Time: 11:01) and down also would be of the same order; up is 10 milliseconds, down is also 10 milliseconds, input queue to the application this is not much; in fact it is 0.5 milliseconds only. Another very prominent area, prominent source of delay is the jitter buffer because as I was mentioning that delay jitter is very important here because whenever especially whenever you are using a multiparty conferencing, there from the end-to-end delays that can vary from one terminal to the other and that gives rise to something like delay jitters and the delay jitter; in order to compensate for delay jitter what you have to do is to use a delay buffer you have to use jitter buffer and jitter buffer could give you 60 milliseconds.

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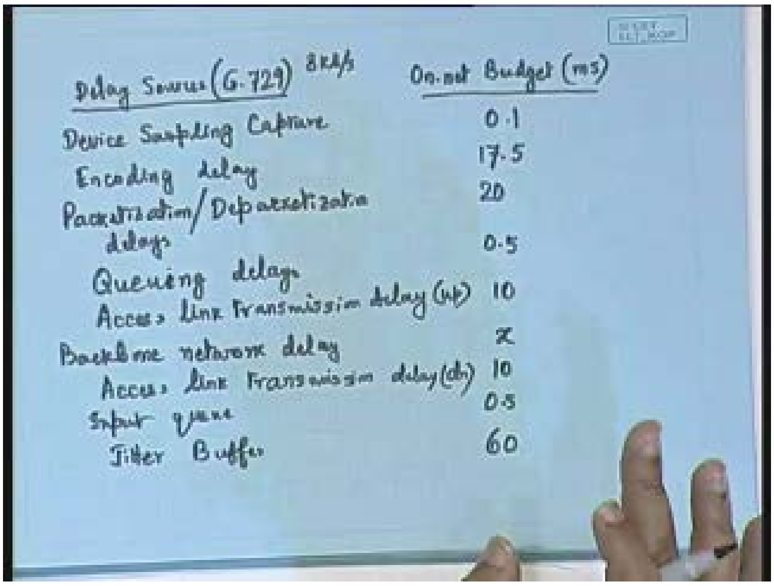

Again this is a very typical figure, this would depend upon the bit rate so may be that at 8 kilobits per second we have specified the 60 milliseconds to be the delay but this may change depending upon your bit rate of application this could be higher this could be lower and the decoder processing delay this also you have to consider. But decoder processing delay is not significant because it is the encoder which is always the most involved thing so this is decoder delays which is 2 milliseconds and the device playout delay at the output the device playout delay this is of the order of 0.5 milliseconds so these are the different components.

And now when we add them together, in fact after addition we will be finding that with this, I mean, considering this jitter buffer to be 60 milliseconds and all the other parameters taking their typical value it comes out as 121.1 plus x. Now this x is the backbone network delay. So obviously if we specify that the total delay cannot exceed 150 milliseconds as is the case with the typical applications. But in the specific cases like the very long distant calls and whenever it is across the continents, there, obviously the network the backbone network delay you cannot keep it restricted to something like 30 milliseconds of figures because here we require close to 30 milliseconds 28.9 to be very precise if you want to maintain it at 150 milliseconds but that will not be possible. That will be possible for within country network also, that would be possible for national networks to say but for international networks you have to tolerate delays which are longer than that.

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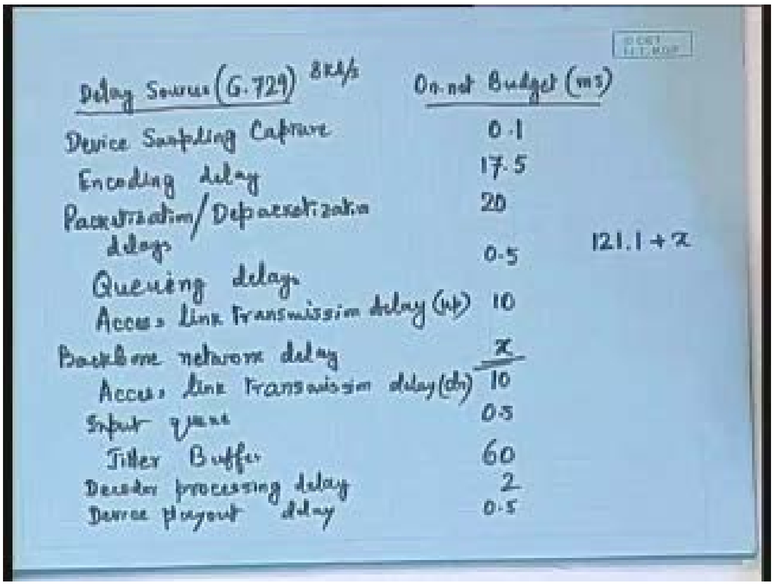

Now these are So this is one very important aspect. In fact when we tell about the advantages of the voice over internet protocol vis-a-vis the PSTN we also have to remember that delay is something which goes in our disadvantage. This delay is significant. Especially things like this packetization depacketization delay, then this jitter buffer delay these are some of the delays which are very typical to VoIP only you will not find them in any circuit switched network. But still VoIP is popular again because of the cheap mode of transmission; you do not require the resource reservation entirely.

Now VoIP has gone through several stages of development. In fact, before I go further I should have given you some introductory reference. In fact, one of the papers that you can refer to is by B. Goode Goode, the name of the paper is "Voice-Over Internet Protocol" voice over internet protocol and this is a kind of a tutorial paper; you can say this appeared in the Proceedings of IEEE. in fact proceedings of IEEE you will be finding that mostly on the very contemporary topics articles which are of tutorial nature where lot of basic introductory things are explained this you should consult, Proceedings of IEEE, Volume 90, number 9

September 2002 so please consult this so many of the aspects that we had discussed you can find over there. But today our topic will be going into little more involved aspect of the voice over internet protocol and we will be talking about the voice over internet protocol signaling.

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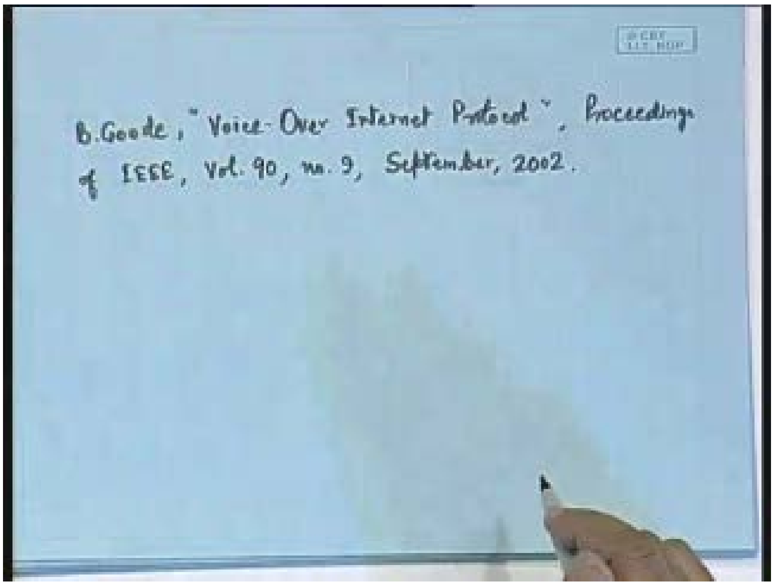

In signaling the protocol of the standard that is widely accepted or rather the protocol which has been specified by the International Telecommunication Union ITU is the H dot 323 protocol. We will talk about this in the present lecture. Basically what we mean by signaling? Signaling is, mind you, not a concept that is very typical of the voice over internet protocol. In fact the signaling is there even for the ordinary PSTN networking also that has to go through certain kind of signaling.

And what for signaling is used?

To tell you very broadly, what you want to do is basically some kind of protocol for a call set up and is likewise for the call teardown. When you want to build a call, when you want to end a call everything should be followed in a protocol. In fact, there is nothing that is unusual about it. Even if it is not a telephone, I mean, whenever you are talking to somebody in a bit of formal way, not your friend whom you are meeting on the corridor or in the dining room, so there you can directly without following any protocol also you can make a you can start a conversation. But even then also when you want to call a friend at least you have to shout with his name and then he has to look at you, he has to respond, he should be prepared to listen to you and then you begin your communication something like that. But again the degree of protocol that would vary that to whom you are trying to communicate.

When you are trying to communicate with a teacher naturally your protocol becomes somewhat different. If you have to now, okay, to the teacher also you can raise your hand, you can ask something and then when the teacher is ready to answer to your question he finishes some specific point and then he comes back to you that okay what is going to be your question; some kind of a permission you get and then you ask the question. Again you see that even for a formal way of ending a conversation that also requires a protocol because you do not just say something and go out; I mean, you may do it in a very informal way but when you want to do it in a bit of formal way you always say that see you then or bye these are the some of the common protocols that you follow.

The same thing is there even for the call establishment, call setup and call teardown. I am not saying that not only it is there in our conversational process but even before the conversation actually begins that means to say that we send the voice packets over the network just one terminal has to communicate with another terminal in that formal manner before the actual transmission of packet can begin through the RTP.

So RTP we know; we understand that when we are transmitting the voice packets that time it is a real-time transmission of voice packet so we have to transmit ultimately through the RTP. But the establishment of the call, I mean, all the things that go on before a call is actually started that requires lot of signaling process and it is there in the process of the PSTN Public Switched Telephone Network and it is very much expected that a very similar signaling should also be followed as in the VoIP.

Actually this aspect was not really addressed at the very beginning but soon when the need was felt that in order have an interoperability of the products; because what happened is if we look at the developmental process of VoIP we will be finding that the initial development; forget about the pre-commercial aspects that when the VoIP was restricted mostly to the University research and all, but when the application actually started coming in with the PCs, there the VoIP's were initially implemented as a proprietary protocol which was adopted by the PC vendors. So naturally if it is a proprietary protocol of the PC vendors there you are restricted to using the same kind of PC for the communication and the interoperability is lost.

So realizing that, in order to ensure the interoperability this standard was actually enforced for the communication between the terminal equipment.

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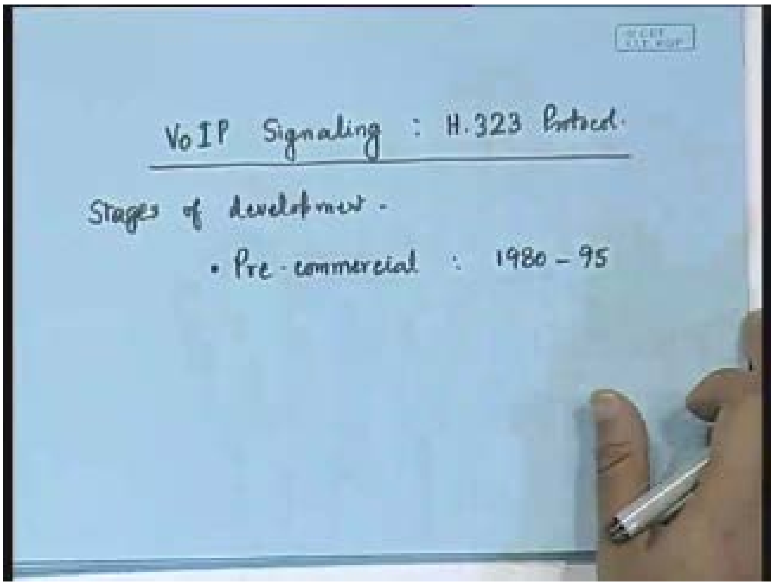

In fact the H dot 323 protocol it is not that something which was entirely developed from the scratch. If you look at the architecture of H dot 323 you will be finding that it is actually an umbrella of different protocols which are already existing. So, as I was telling you about the stages of development the different stages of development that the VoIP has gone through, roughly we can in the chronological way we can broadly divide into three stages: One is what is called as the pre-commercial stage, the very early development started taking place way back in 1980 and till 1995, so this was the time when you can say that it was more in the concept preparation stage and the University research domain mostly. But some standardization efforts were also done realizing the future or would-be needs of voice over internet protocol. because you see, when the need was felt, again we are talking of a time, I mean, consider the latter aspect of this period 80 to 95, by 90 by 85 to 90 the JPEG standard was formulated, within a couple of years the MPEG standard was also formulated, MPEG-1 came into being. So when this standard started coming and at the same time the VoIP development was conceived, people could dream that at least a few years down the line the packet switching is going to dominate the internet community in such a big manner that in future days, whether to talk of voice or whether to talk of video images everything would be through the internet and realizing that it was felt that the real-time protocol has to be there;

you cannot say that everything could be just on an acknowledgement and a retransmission for lost packets, you cannot have that all the time. So realizing the future potential, during this stage only the Internet Engineering Task Force the Internet Engineering Task Force you must have heard about this body IETF, IETF formulated different standards and one of that is for the so they had a subgroup, IETF had a subgroup which is called as the Audio/Video Transport Audio/Video Transport or in short form it is called as the AVT. So it is the AVT subgroup of the Internet Engineering Task Force, so this is the Internet Engineering Task Force.

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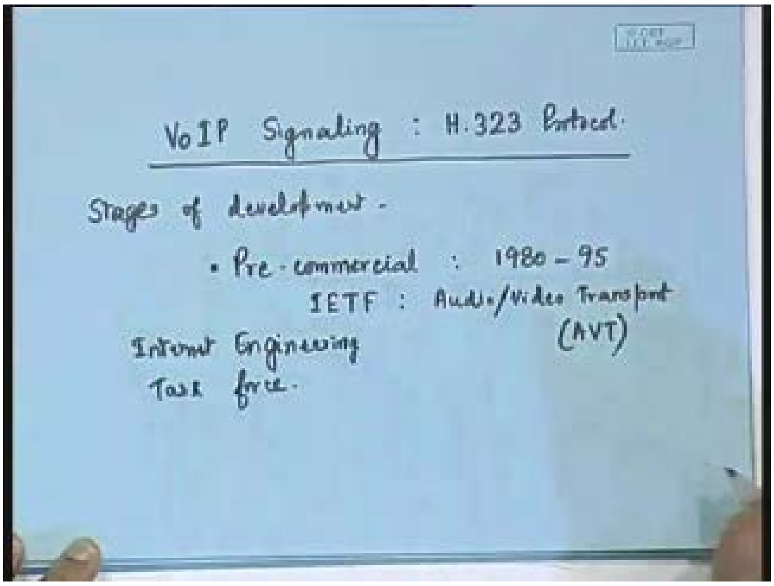

Now the AVT team, they realized the real-time transmission requirements of that and it is during this time that they developed the RTP protocol which you know is really a great thing to do because unless RTP protocol had been there it would not have been possible to ever realize the dream of the voice over internet. But at the same time when the University research was going on, people also felt the requirements of the multiparty conferencing. And in fact by then, when we are talking of 80 to 95 in very small scales, people also started doing the multiparty conferencing or the video conferencing, multimedia conferencing, mostly the video conferencing. But the video conferencing what was done during that time was not through the packet switched network but it was through the circuit switch.

In fact, for video conferencing or multimedia conferencing using the circuit switching the standards were already established; there were standards like the H dot 320 and later on the H dot 324; may be H dot 324 is not really contemporary to this few years down the line H dot 234 was developed. But let us talk that even H dot 320 was there.

And another aspect which was developed for the purpose of the multimedia conferencing is what is called as the Session Initiation Protocol Session Initiation Protocol. This was really a great thing to be developed, the SIP, because SIP forms the core of the forms the core protocol for the multimedia conferencing or the video conferencing. Naturally all the preparations were done during this 80 to 95 stage; we had the SIP, we had the RTP, we had the multimedia conferencing standards like the H dot 320 at least and then came the second stage of development and the second stage of development actually was PC centric.

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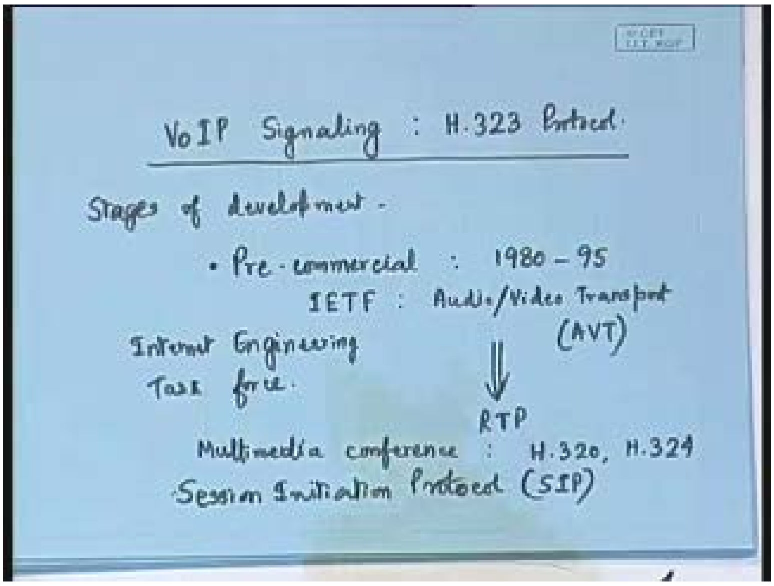

So this was PC centric and as I was telling you that it is during this time that means to say that we are talking of the time 1995 to 1998 when the PC vendors they wanted to make this VoIP popular by adopting proprietary products. But proprietary protocols definitely could not be the answer because then the interoperability is lost and that is where the that is when the ITU the International Telecommunication Union had to intervene and they had to conceive of the signaling protocol in the form of the H dot 323 which we are going to discuss in details now.

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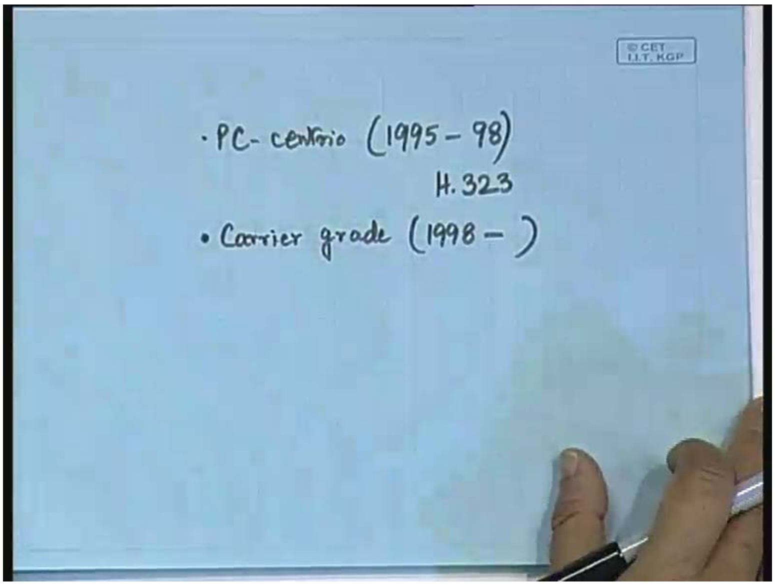

Then the third stage or rather to say you can still say that that is under the development and lot of efforts have been already carried out on that that is the carrier grade VoIP. Just like the way; see, for the PSTN you have the carrier grade PSTN in the sense that where you can have you can have the bandwidth partitioning bandwidth allocations to the different subscribers and the way you are finding today the mobile phones are working. So catering to the large number of users which of course the H dot 323 standard per say could not do it, so naturally here we are looking at beyond something beyond H dot 323.

So we will be talking about the carrier grade VoIP little later, maybe I mean in a smaller scale we would restrict because we want to talk more about the signaling aspect and there especially this H dot 323 standard.

Now, before we go into the H dot 323 standard let us see that what is the kind of the network environment that we are talking of.

Primarily, in order to have the H dot 323 would operate on different zones of the LANs. Let us say that there could be one local area network, there could be another local area network and then might be so this one LAN and this is another LAN and these two LAN may be connected by a wide area network. Let us say that here we have a WAN to connect these two LANs.

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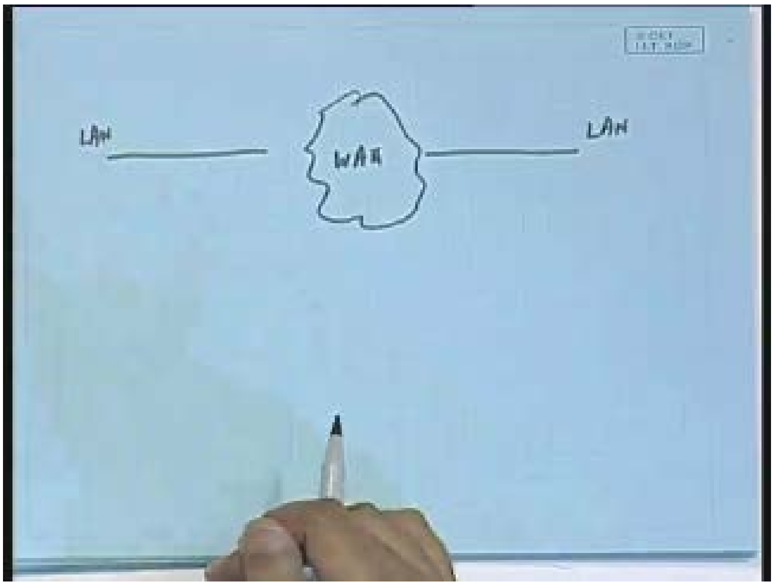

And actually speaking there will be such multiple WANs multiple LANs fully interconnected to each other. But when such kind of network structure we are talking of then there are certain essential things that every network........... we should call this as a zone, so say we typically consider a zone-1 of LAN and another zone-2 of LAN (Refer Slide Time: 32:07).

Now zone-1 of LAN will be having some terminal equipment. These terminal equipment are nothing but the end points means where the user is located and the audio or video transmission has to take place. And likewise in zone-2 also we consider some terminal equipment. There may be large number of terminal equipment. Now let us say that typically we want to establish a connection or rather we want to transmit voice from this TE to this TE or back in a conversational mode we want that.

Now, other than these two we also need several other components. One of that is that since we are communicating between two different LANs and we are migrating from one network to the other, definitely what is essential for us is to use a gateway. So this zone will have a gateway (Refer Slide Time: 33:12), this zone will also have a gateway. You can say something like this that supposing there are two buildings, in within one building there are many terminals which are available, in building two also there are terminals but between or let us let us not talk about terminals s let us talk in the usual way that okay we have one person in building one wants to communicate to somebody in building number two. then what that person has to do is to go out if he wants to deliver a message; let us say physically when he wants to go and deliver a message; say telephones are not available and he wants to just go and deliver a message; he has to come out of the main entry door so that main entry door is nothing but the gateway. So there will be gateways.

Hence, even if the LANs are different the gateways have to take care; gateways have the standard so that whatever the terminal equipment wants to communicate to another the language of the way in which it should be done through the WAN and maintaining the standard that will be maintained by the gateway.

Then we also have a........ for the multi-conferencing facility multiparty conferencing facility we should have what is called as the multiple control unit Multiparty Control Unit or we can say in short form as the MCU. Even if MCUs are not there, still a two way conversation will be possible but when you want three or more than three parties doing a conversation then there you would be requiring an MCU.

Now gate is needed, terminal is needed, you want multiparty then MCU is also needed but just see that even in buildings also when you just have a gate when you have an unmanned gate you will land up in lot of problems; you require a security at the gate; what for is because you do not want that everything in the building should be accessed by anybody, there should be something like an authorization.

We talk about security in that respect. But even that security what we want is not somebody who just get up and say Salaam and then allow you to enter but it should be something like an intelligent security in the sense that when you want to communicate from one terminal to another you have to make a request to that gatekeeper that yes you would like to make a communication, that gatekeeper will give you the call admission, that gatekeeper will allocate the bandwidth for you. So in other words, it is the gatekeeper who should have the full control.

In fact, this gatekeeper concept was not there initially. As long as the gateway is there to interconnect the zones and as long as the terminal equipment are there it should have been enough. But whenever we are talking of the commercial usage of VoIP, there, the gatekeeper should play a major role that even things like the access control billing all these things have to be done by using a gatekeeper. So we are keeping the gatekeeper or in short form we are saying as GK also in the network.

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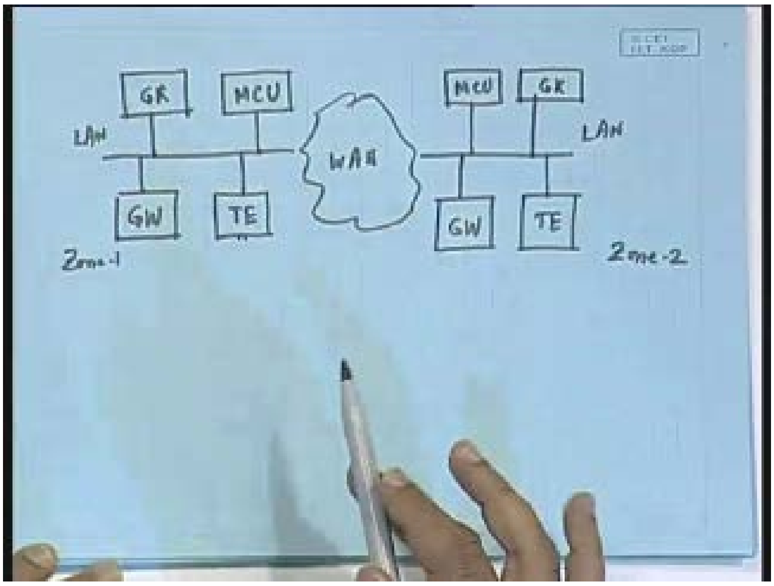

These are the essential components. Now there could be multiple number of TEs, there could be even multiple number of gateways because you may use separate gateways for different WANs and not only that; one point which you have to remember is that, for the normal internet gateways that one uses for the data transmission they are the LAN to LAN connection or LAN to WAN connection these are the only important aspects. But here this gateway has to also consider the interworking with the PSTN. Because the very fact that we are using the VoIP does necessarily mean that not only we should be able to communicate with somebody who is connected in the LAN but also with somebody who is in the PSTN and vice versa; somebody in the PSTN also should be able to communicate with somebody who is having an internet telephone with him. So because of this aspect this gateway what we had seen, the gateway for the just plain and simple data communication network, here the gateway should be little more involved to take care of the PSTN to the internet interface that also should be available.

Therefore, when such PSTN to internet conversion is there naturally there is there has to be a signaling conversion, there has to be a message conversion and everything. Now this is more or less the architecture, so this is the architecture of an H dot 323 network. But what is the essential ingredients of H dot 323; let us consider these aspects.

As I was telling you that H dot 323 is essentially an umbrella of some protocol. In fact, it is an umbrella of four protocols and what are these four protocols?
One is what is called as the registration admission and status. You can say gate pass that the gate keeper has to check your gate pass so that is by the registration admission and status or called as the RAS. In fact RAS is an accepted protocol and when RAS is to be used mandatorily you have to use a GK the gate keeper.

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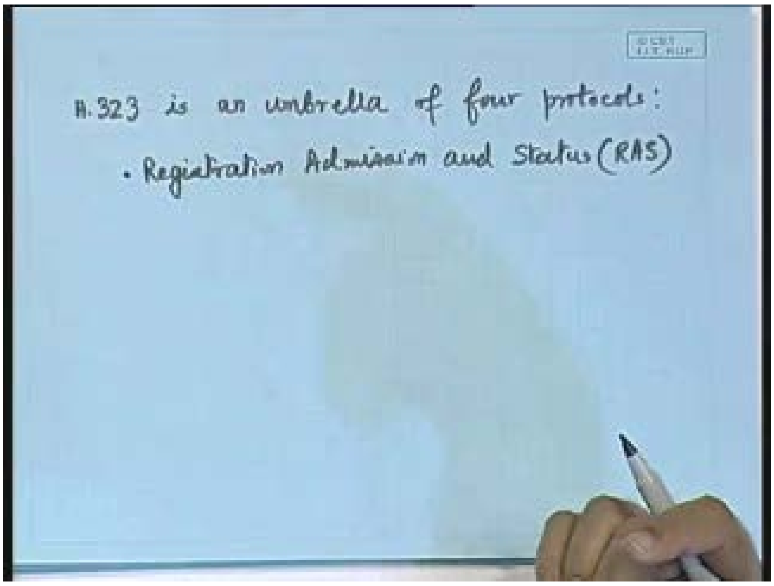

The earlier versions of voice over internet protocol did not use a gate keeper so there RAS was not an essential component but when H dot 323 came in that time the RAS was also considered to be one of the basic ingredients. Then, of course, registration admission is only for taking care of the call admission and bandwidth allocation etc all very preliminary things. but then, also the actual signaling protocol like you are wanting to set up, then the acknowledgment that you are granted the resource to use it so these signaling protocols are already existing; even prior to H dot 323 these signaling protocols were existing for the PSTN and in fact there the standard that was used is Q dot 931 so Q dot 931 basically contained a signaling protocol over the PSTN and ISDN.

But in this case the Q dot 931 when we want to apply it over VoIP naturally it has to consider that what are the signaling that would be necessary for the VoIP because specifically in these case we are having a gatekeeper sitting in between so naturally the signaling protocol will go through some differences because we are having another party that is sitting in between.

So Q dot 931 as we perceived as we had in the PSTN or ISDN needs extension and that is addressed in the Q dot 931 what is adopted in H dot 323. So definitely it is something enhanced than the Q dot 931 of PSTN/ISDN.

Then there is a requirement for the connection control and for connection control the protocol that is existing for the ITU is the H dot 245. This is the connection control protocol. And, by connection control we mean to say that basically it does what is called as the capability negotiations between the end points.

You see, the terminal end points what we are talking of in the H dot 323 architectures they could be having differing capabilities because there are a number of standards that is there for the audio codec so which standard of audio coding are you using and then which standard of video coding are you using all these different standards are existing and the capability again what is the capability of the terminal it has to be negotiated based on that so one terminal equipment wanting to communicate with another terminal equipment has to go through a process of enquiring about the capability and then allocating the resources accordingly so this is controlled by the H dot 245. And of course one aspect which has to be essential is a real-time transport protocol. So the RTP which is already an essential ingredient for transmission of the actual packet that is also coming under the H dot 323 umbrella.

H dot {323 uses the existing protocol and adds to make the VoIP more towards a commercial application and towards the sophistication. These are the four that are essential ingredients which are attached plus there are some enhancement capabilities which I will be talking of in future.

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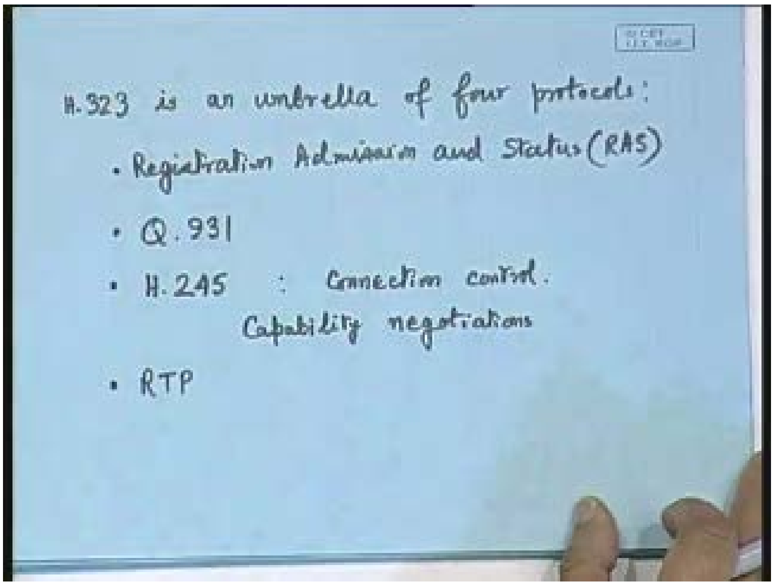

Now these protocol relationships you can have a total look at what are the how these protocols should be related to each other. So let us get a pictorial view of that. Say that in the matter of the control you have the some control protocol, you have got some data protocol, you have protocols for the audio and video that means to say the audio and video coding and then you have the AV control the audio/video control and especially this has to be a real-time control and the type of controls which we were talking of in terms of the gatekeeper the gate keeper controls so this control means that here we are referring to as the signaling control the signaling plus the connection a signal plus connection control; so controls at different levels, control at the signal and connection we will be talking of over here and here it will be audio/video control and here it will be gatekeeper control.

Therefore, for control over the signal what is the standard that we are using? I have mentioned it just now that the signaling controls are specified under Q dot 931 and the connection controls are given by H dot 245. For data there is an already existing protocol. In fact data does not come directly under the H dot 323 but in a typical transmission scenario you have to use the data and it is T dot 120.

For audio/video naturally you require the codecs and these are the different codecs that you are using for the audio. For audio the codec standards followed by the ITU is G dot 7 series

7xx so like that we have the G dot 729 and all such different versions. And also we talked about the H dot 26x standards like the H dot 261 263 264 these are for the video.

But of course these are the coding standards. This is about the encoding but then whatever bit stream you are generating that has to be put through the real-time transport protocol. So naturally RTP should come immediately after this G dot 7xx or H dot 26xx and for the audio visual control there has to be a real-time control protocol and that is called as the RTCP it is the real-time control protocol and for the gatekeeper we already mentioned that it has to be the registration admission and status protocol or the RAS protocol. These are the basic things that we will be requiring at the higher level and at the transport level we require the use of the TCP or the UDP.

Now whichever are not real-time; you see the signaling, signaling has to be a very robust one; signaling control or the connection control for the negotiations of the capability negotiations or the integrity of the data; everywhere the integrity is important and until and unless you begin the actual voice communication or the actual video communication your real-time aspects are not yet coming. So there you can afford, for these protocols you can afford to put them into the network using the TCP whereas the other parts like this actual audio/video packets which has to be transmitted through the RTP and this RTCP or even this RAS these have to be done through the universal datagram protocol or UDP so this should be UDP/RPT and then finally the last layer has to be the IP layer below this (Refer Slide Time: 49:34).

So either it is TCP/IP taking care of these protocols or you should be or it should be UDP/IP taking care of the RTP and above RTP the codecs. This is the total pictorial scenario so this you can say as the protocol relationships in H dot 323.

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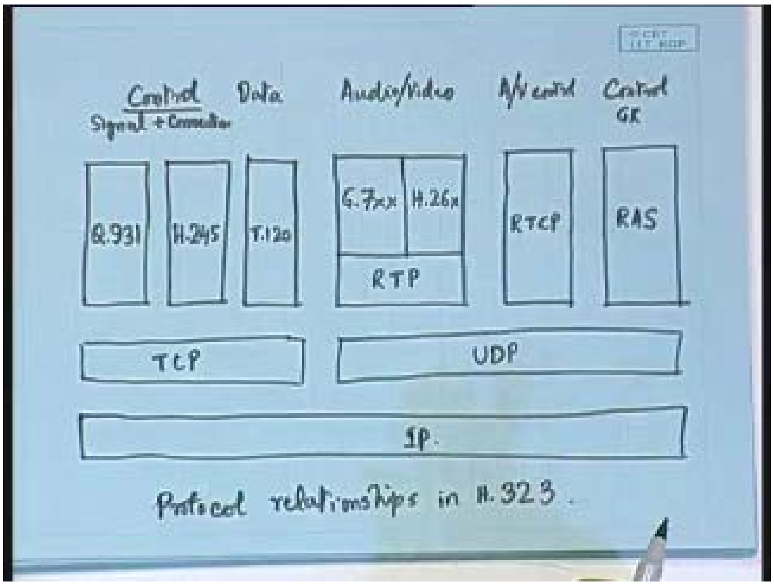

Now I think it is very essential at this stage to know about how the actual that what are the different stages that one goes through in a process of call.

In fact, for an H dot 323, call to materialize and end successfully end there it has to go through seven phases. So H dot 323 call goes through seven phases and what are the seven phases; the very first phase is what is referred to as the.......... we can say the phase and then we can talk about the concerning protocol for that. So the first phase is the call admission. That means to say that asking for permission through the gatekeeper to make or receive a call. Even for receive a call also you have to seek the gate keeper’s permission. And at the end of this phase the endpoint or the terminal equipment receives the Q dot 931 transport address.

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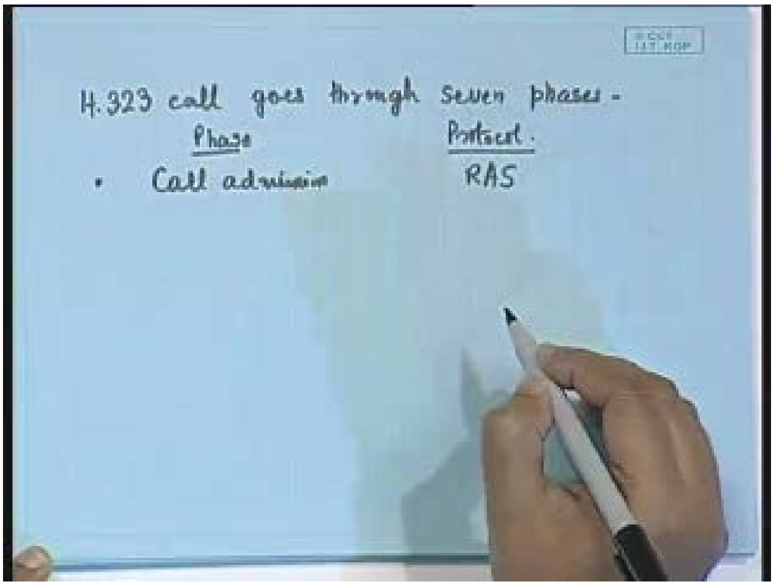

Now which is the protocol that should support this call admission?

It is the just now I mentioned the RAS protocol so it is RAS protocol and that resides in the GK. So RAS protocol in the GK has to take care of the call admission. Then comes the question of call setup. You are permitted; when know that you are permitted, the GK tells you and gives you the transport address then you do the call setup; what protocol should be used over here? Yes, Q dot 931 Q dot 931 has to be used because Q dot 931 is the signaling protocol that is used and then once the call setup is initiated then there should be the capability negotiation. And capability negotiation we already said that the protocol for that is H dot 245. After the call setup it goes to the capability negotiation for which the standard is H dot 245.

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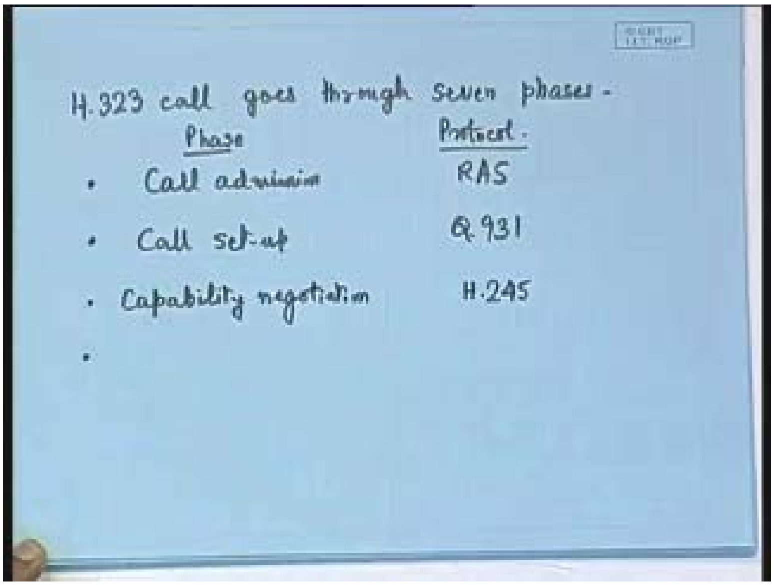

And then once the capability negotiation is successful then the actual conversation begins. This is the stable part of the call which can continue as long as you want to talk to your friend; you can maintain the stable call and during this, what should be the protocol; what should be the only protocol that comes in is the RTP. At the lower layer it is the RTP, upper layer of course the codec is important whether it is G dot 729 or H dot 264 whatever.

Then once the call is over you have to go throw the reverse process; the call has to close down so you have to say the resource reservation that you had done in terms of the codecs and all these things that has to close down. So there is a channel closing and this also goes through the capability negotiation process, so H dot 245 for the reversal the channel closing. And then the opposite of the call setup in the telecommunications parlance is called the teardown called teardown so this is the ending process and ending also has to be associated with the proper signaling sequence which will be given by Q dot 931 and then the final saying goodbye this is by the call disengage process and this call disengage process there we have to just disengage ourselves after satisfying the GK; we have to inform the GK that yes the call has ended.

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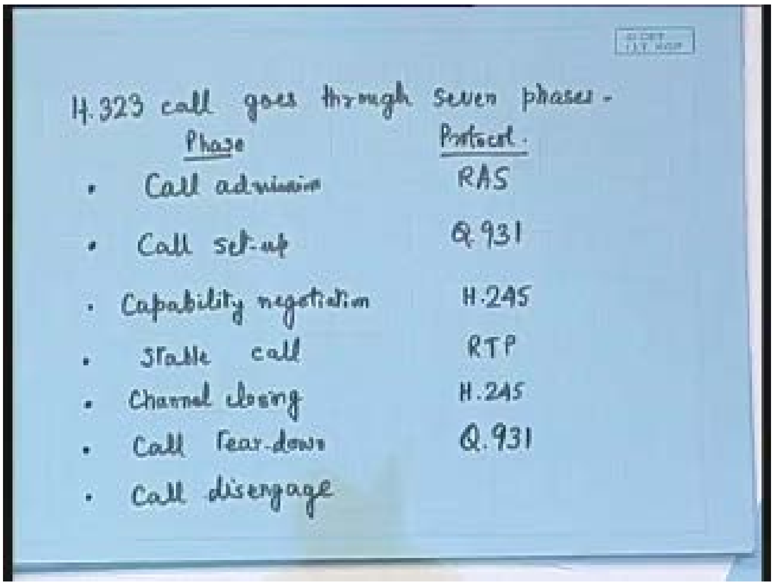

Now GK has the monitoring. So, if GK has to record our call or if the GK has to note down the times for which the communication was used; for billing purpose the GK can use that so the GKs information goes primarily through this RAS. This is where GK is mandatory.

I was telling you that GK is not always mandatory; for very simple systems it is not mandatory and in fact like it may appear to you that as if to say the connections is quite a time consuming process. There is a methodology for fast connect also in which case you can integrate this call setup and this capability negotiation and even the same thing for channel closing and call teardown they can be combined together in what is called as the fast connect protocol. This is a variant.

Anyway we will talk about some further aspects of H dot 323 in the coming class; till then, thank you.