Digital Voice and Picture Communication

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Lecture - 35

**H. 323: Interworking with PSTN, Limitations and Solutions**

In this lecture we are going to see some more aspects of H dot 323 and one of that is its interworking with the PSTN. Then we are also going to see that what exactly are its limitations especially when we are talking about the career grade telephones and then what solutions have been proposed till now in order to overcome these limitations. So these three aspects we will be taking up in this lecture.

Already we have got some background about the H dot 323's capabilities and its functionality. Right from the very beginning, since the concept was thought of in 1995 .... 95 or 96 in order to go in for standards like H dot 323.

Right from the very beginning this aspect was carefully considered that with the PSTN it should have a very proper interworking because typically the call setup scenario could be from anything; like one can have a phone to phone; when I say phone I am referring to the PSTN phone so you may have the call setup scenarios or three different call setup scenarios are possible and these are: one is the phone to phone call setup; by phone I mean to say PSTN phone so it is not something that goes directly through the circuit switched network but we are referring to, that from one phone to the other it has to ultimately make use of the IP.

So IP will be used as a backbone or at least a part of the backbone will be the IP. But ultimately it is one PSTN phone to another PSTN phone. Then you can have an IP telephone or what you can say as the H dot 323 terminal equipment and from H dot 323 terminal equipment you can connect to a PSTN phone or you should be able to do the reverse of that also that means to say phone to H dot 323 TE. And really speaking there is also the fourth possibility that H dot 323 TE to another H dot 323 TE which is anyway taking place.
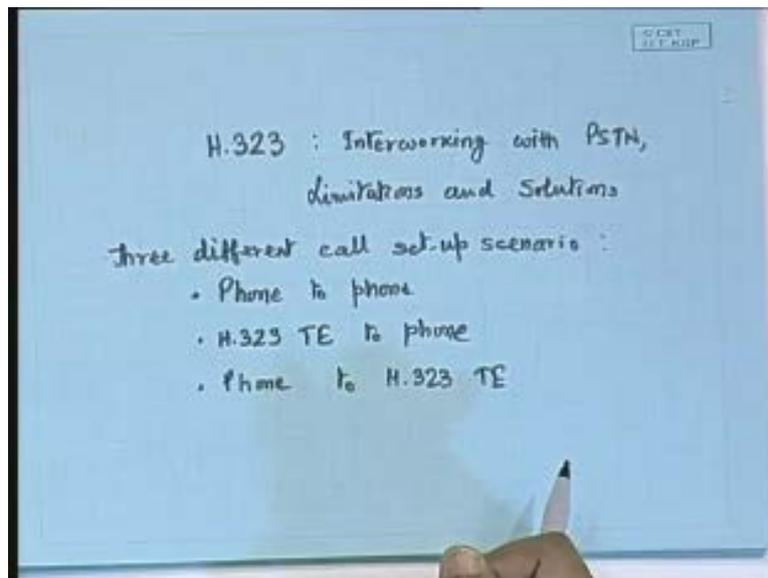
In fact, when it is H dot 323 TE to another H dot 323 TE there we are not at all involving the PSTN in anyway; it is entirely within the IP and H dot 323 yes definitely caters for that but the interworking is not an essential requirement and as far as the H dot 323 as far as the IP connections are concerned we also have other choices; it is not that only H dot 323 but even the Session Initiation Protocol or SIP which was developed in the 90s one can make use of the SIP protocol also as far as only the IP telephone to IP telephone contact is needed. But you really require a thing like H dot 323 whenever you are having the interworking with the PSTN.

Now, typically when one has to take care of this PSTN interfacing then a gateway has to provide several functionalities. Let us see what are the essential ingredients that one is going to have in the gateway.

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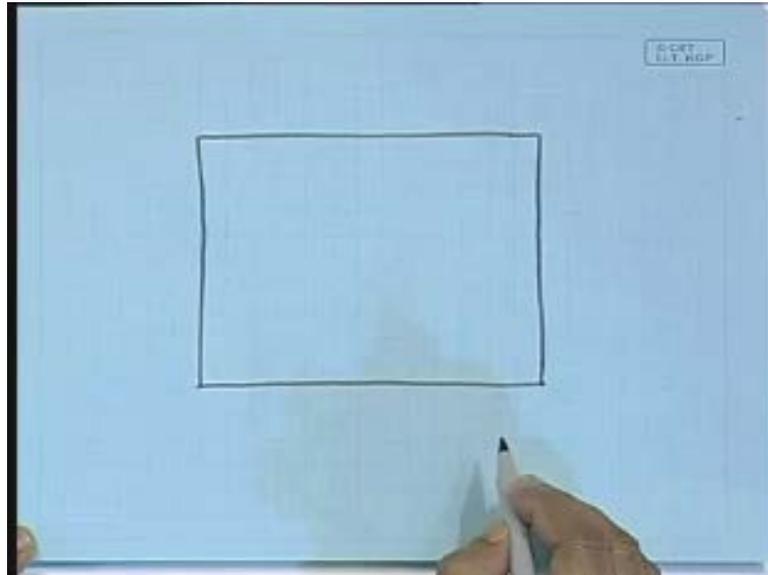
Say this is the gateway and we are drawing a big box for the gateway because it is really going to be big and it is this big which is going to be the problem for it. So, on one side we have the IP domain the internet domain and on this side (Refer Slide Time: 5:57) we have the PSTN domain and there should be some effective interfacing for this.

From the PSTN side let us first talk about the........... so there are two different things that the gateway has to do: one is taking care of the signaling aspects and the other is taking care of the actual media streams or the data which will be the PCM samples in the PSTN domain and which will be the RTP payload in the internet domain. So we should have a kind of a demarcation that part of this gateway will do the signaling job and part of the gateway would do the media stream transportation job.

Now if we say that if we segregate the two and define signaling plane; if the top half of this we define as the signaling plane and the bottom half we define as the media plane then what are the signals that we are going to receive at the gateway? One is the ISDN Q dot 931. Why specifically we are calling as ISDN Q dot 931 is because Q dot 931 is also there in the H dot 323. But here we are specifically talking about the ISDN interfacing part of the Q dot 931 which should be a part of the PSTN. And because we have got all these modalities phone to phone, 323 terminal equipment to phone or phone to 323 terminal equipment so that's why it can come from internet to PSTN or can go from PSTN to internet; both signaling as well as

the media streams, so that is why there should be a bidirectionality. And now within this block we must have a PSTN signaling interface. So this subblock that is what we are showing, it is the PSTN signaling interface.

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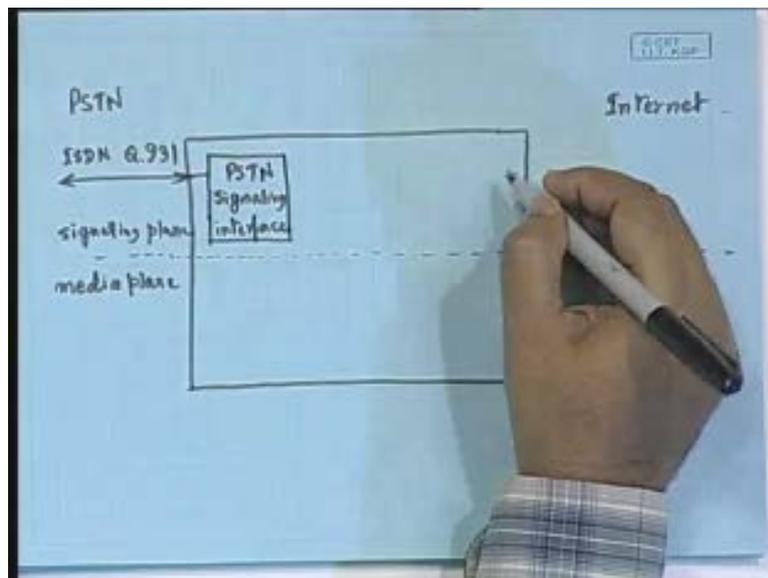
And in the internet side again, at the internet end we must correspondingly have a VoIP signaling interface and again a bidirectionality. So, in the internet the VoIP signaling has to be on the H dot 323 standard. That means to say that it has to include; what? It has to include the RAS, it has to include the Q dot 931 the part which is supported by H dot 323 and for the capability negotiation it must have the H dot 245 so this is needed. So we have the PSTN end and the internet end so supposing from an internet telephone one wants to communicate to a PSTN telephone so the internet telephone signaling would come through this H dot 323 channel, it will go to the VoIP signaling interface but there must be a conversion that has to be done so that it is compatible with the PSTN signaling interface because whatever signaling interface we are doing over here that signaling should be transferred at the PSTN end so we must have a block that does the signaling conversion. So in between this PSTN signaling interface and the VoIP signaling interface we have to have a signal conversion signaling conversion block.

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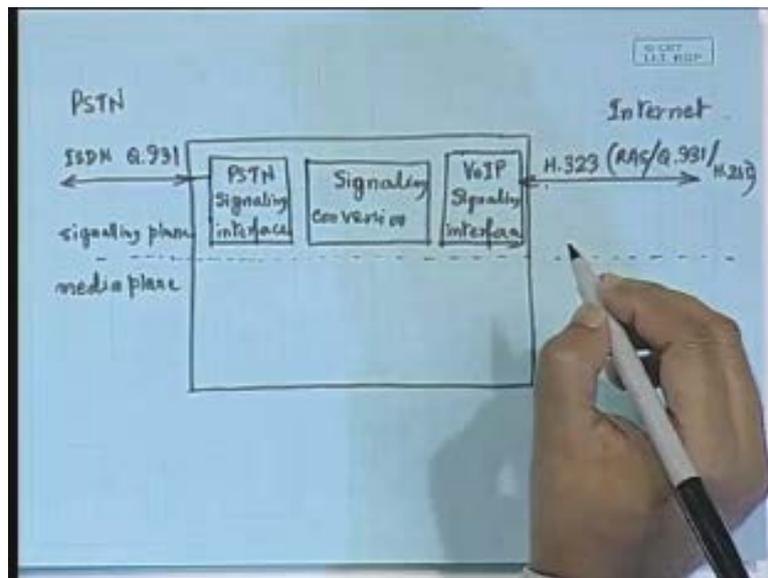
Now on the other side of it that is to say at the media plane side it is only dealing with the data because the top part is dealing with the signals, the lower part will deal with the data so here it will be the data will be a part of the RTP payload, so here we write as RTP because data will be in the RTP payload and again at the PSTN side we will be having either the encoded PCM samples that would come from the PSTN up to this media interface which we are going to have over here or the opposite also can happen.

When it is internet phone communicating to PSTN that time the RTP payload from the RTP payload we have to extract the data and then that data has to be converted to the PCM samples and then should be transmitted through the PSTN interface end. So there must be similarly the PSTN media interface on this side, so this subblock will be the PSTN media interface (Refer Slide Time: 12:21) and this block we will correspondingly call as the VoIP media interface.

So very similar to what we have for the signaling, here there must be a media conversion or media transformation; so there must be a media transformation which means to say that if you are receiving the PCM or rather the encoded PCM; definitely from the PSTN end you will be having compressed and encoded form of the PCM samples or whichever encoding you use the encoded samples will be coming through the PSTN interface

and what then one has to do is to decode that so there must be a decoder; within this media transformation there must be a decoder and that decoder would extract the samples and then it must have a corresponding encoder which will first encode this; because in the RTP payload what are you going to send; you have to make use of some encoding process like the....... I mean, if it is a speech then G dot 729 or G dot 728 whichever standard is being followed it has to be encoded into that; if it is video then any H dot 26 x form of encoding has to be done and it is this encoded bit-stream which along with its headers etc goes into the RTP payload.

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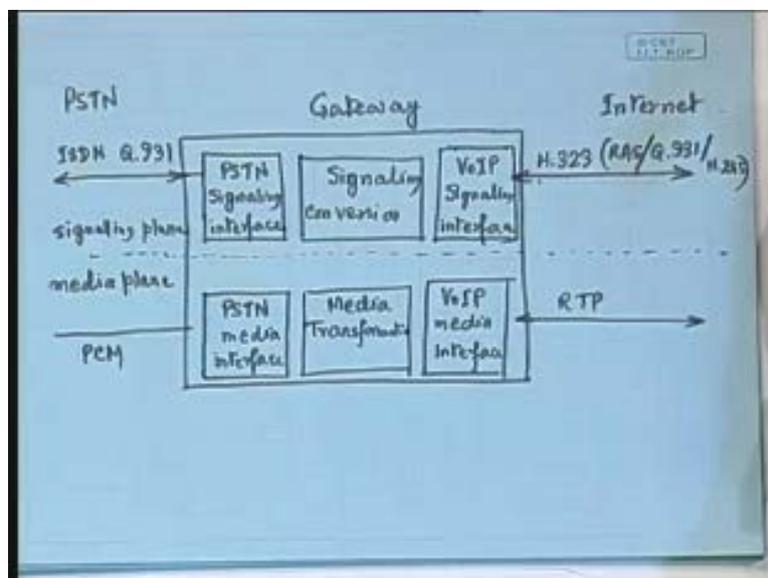
Hence, naturally there two things are involved: one is the encoding part and one is the decoding part; it is the decoding followed by encoding and this is also referred to as what is called as media transcoding. So transcoding is the requirement that we are having within this gateway and just a reverse type of a transcoding will also be necessary whenever we are having the data stream coming from the internet and has to go to the PSTN because in that case we have to extract the RTP payload, get the data, we have to decode the data because RTP payload contains again encoded form of data so we have to decode the RTP payload and then we have to encode it in the way this PSTN operates; the form of encoding we have to do so again it is a decoder followed by encoder but of a different form.

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In that case it was the PCM to H dot 261 or G dot 728 those kinds of conversion and here it is just the opposite conversion. So there should be a second form of transcoding that has to be there within this block. So essentially you see that there are six blocks but if you are taking this as one block and this as another block that means to say one is that........... a gateway has got one major block which we are combinedly going to call as the PSTN interface. So PSTN interface includes the PSTN signaling interface and PSTN media interface and likewise there is a VoIP interface that includes these both and then we have the signaling conversion and media transformation signaling conversion and we have the media transformation and on top of it the gateway also has to do the connection management that is coordinating the signal flow. So although we have not drawn that block explicitly but that should be the hidden controller within it so the connection management that also will be a part of the gateway.

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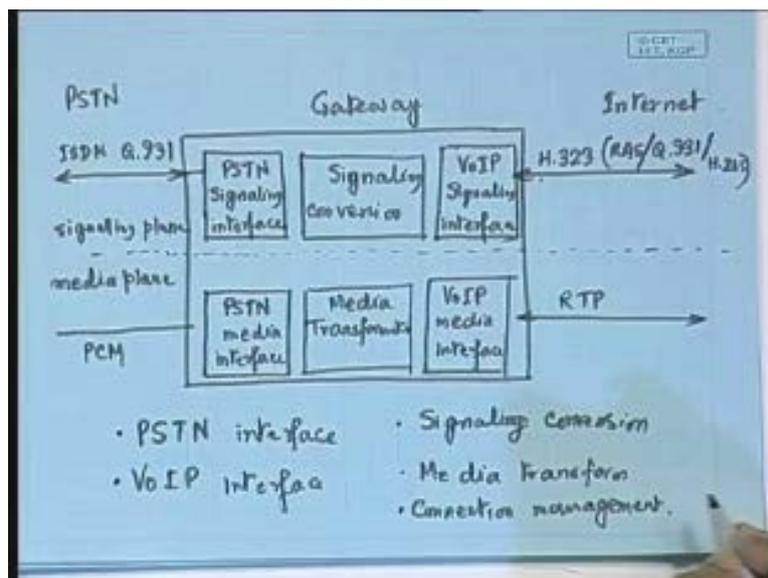
Therefore, naturally it is considering the interworking with PSTN, it is quite an involved gateway that we have to make and how that gateway has to be made?
One can use hardware, one can use software. now the first concept the initial concepts were that make everything within one box; go in for a monolithic solution, build a gateway where you have all these things incorporated and go in for a monolithic solution of that. There the difficulty that one has to realize is that it is too many things that goes into one block. Let us see that which block is more involved.

There are two distinct parts: one is the signaling part and the other is the handling of media streams. Now if it puts if these two combinedly puts a lot of stress on the CPU's capability and if there is a memory limitation, so depending upon the CPU capability and memory limitations this job would take considerable time and resource on the part of the CPU that controls this gateway and as a result of that the number of lines that you can handle is going to be much limited, the number of users who can simultaneously use that gets restricted and that may get restricted significantly as compared to the number of users whom you find in the PSTN.

So, as the career grade telephony gained experience certain limitations of H dot 323 was felt. We can list out these limitations. The first which we have been referring to is the question of scalability. H dot 323 gateways because of all these processing burden H dot 323 gateways can support only a few thousand lines whereas the PSTN whereas PSTN can support tens of thousands. So naturally this is a very big issue. And if internet telephone has to compete with the PSTN then naturally something should be done to overcome this limitation because we should not be putting too much of burden (Refer Slide Time: 20:41) on one single gateway by having an architecture like this single gateway.

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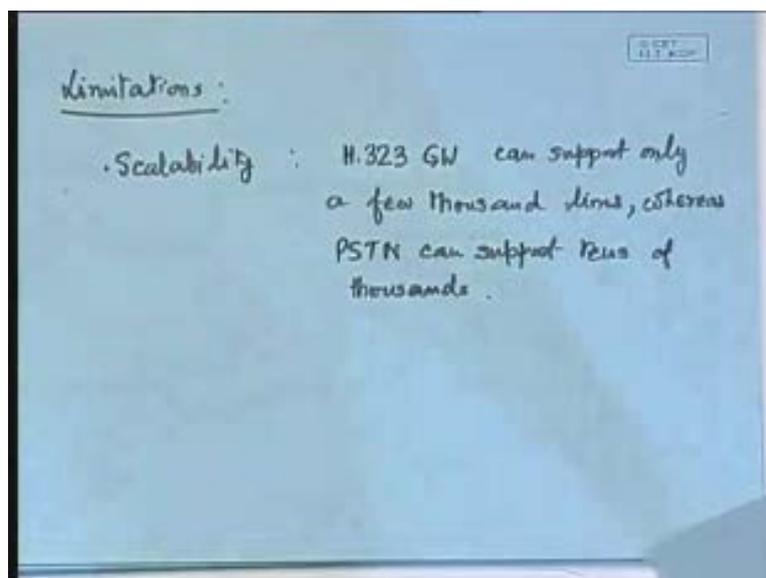
There is yet another capability that I should have talked of actually in the last class. Actually when the signaling started the signaling protocol developments actually started with the

PSTN and there <mark>as I told you, I told you already</mark> about the Q dot 931 that Q dot 931 was a signaling standard that the PSTN community followed but there is yet another protocol which exists in the PSTN and that is called as the SS7 the signaling system 7 so SS7 has got certain capabilities which the PSTN can make use of like say for example one of the capabilities that I can mention is the direct dialing facility.

You see, here when you want to dial or connect from one TE; say terminal number 1 is connected in one LAN and terminal number 2 is in a different LAN and you have to go through the gateway, so there you have to first dial the gateway and then once you are granted the gatekeeper grants you the permission then as the second step you have to dial the number; something like going through the exchange and it is a two-step dialing process, no direct dialing process is there. But that would have been possible had there been SS7 connectivity. So SS7 connectivity is there in the case of the PSTN which basically interconnects several switches through a protocol and that is the SS7 connectivity but there is lack of SS7 connectivity in the case of the H dot 323. So I should say the lack of SS7 connectivity. And as such this SS7 connectivity cannot be inserted into this architecture but we will see that how the modified architecture may take care of this.

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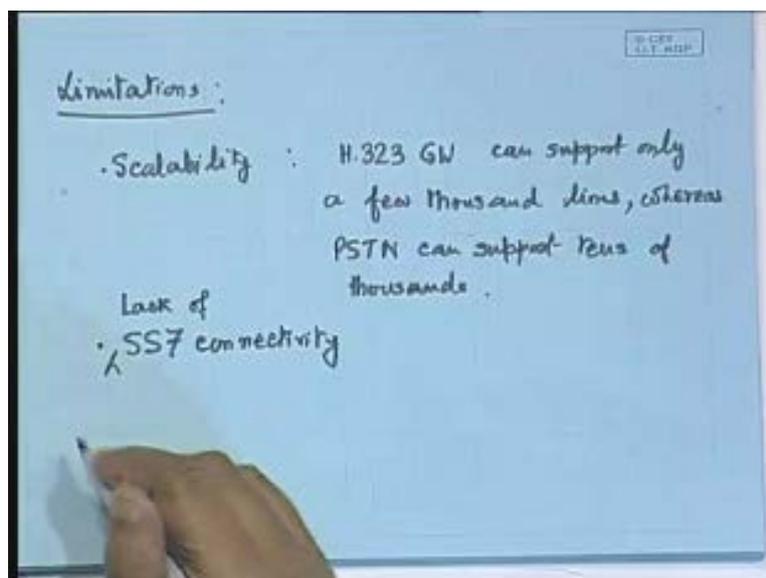
Then another is the reliability aspect or the availability. <mark>now you see that supposing in the now in</mark> In the case of the H dot 323 or in the case of the IP telephone rather you are seeing

that all the calls are actually through the gateway. Now what happens in case the gateway collapses? In case there is a malfunction in the gateway and the gateway packs off then what you do; and supposing it can it can happen in a middle of a call, even if you think of having redundant gateways you cannot switch from one gateway to the other at the middle of the call so naturally your call abruptly terminates because of the gateway failure.

H dot 323 does not provide anything like a failover mechanism. So H dot 323 does not have any failover mechanism when gateway fails in the middle of a call so that aspect should be taken care.

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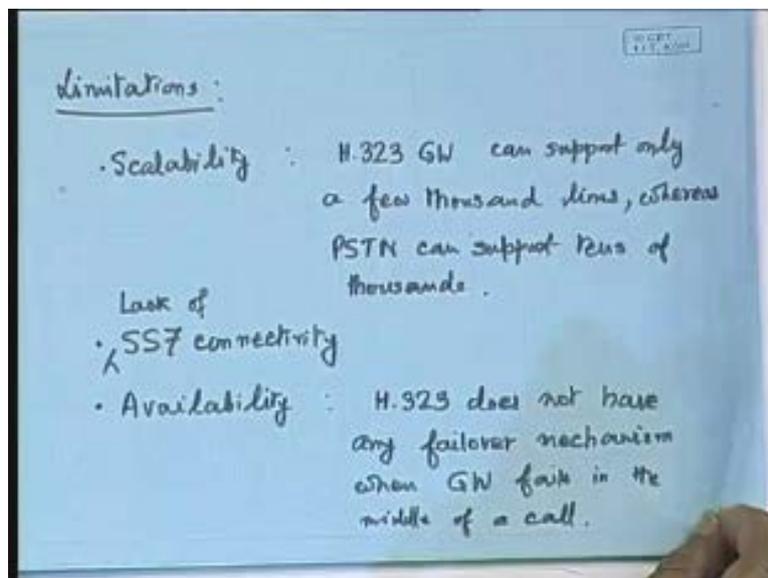
And then something I was mentioning during the lack of SS7 connectivity only that the user friendliness many user friendliness aspects do suffer because of the non-availability of SS7 connectivity; due to non-availability of SS7.

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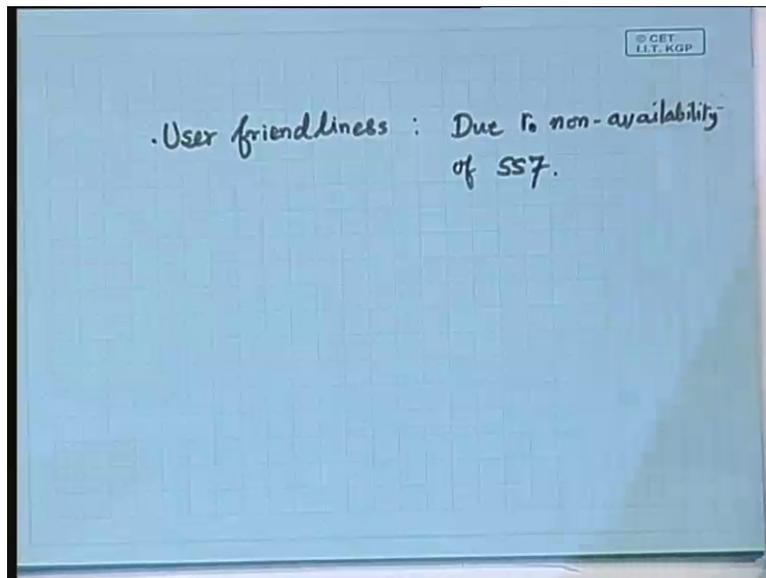
So naturally it is true that there has to be And why SS7 interface cannot be built? You can ask that why do not we build that. Well, as it is we are looking at a very burdened block or a very heavily burdened block and if you have to cater for the SS7 interface there that makes it further more complicated so definitely the efforts should be to off load this gateway.

Now you see that amongst these two aspects: the signaling conversion and the media transformation which one is going to be more computationally involved; what is your feeling after going through the different aspects of encoding, decoding; we have seen encoder decoder for all these. Definitely the media transformation is the most involved block; because signaling is after all what?

Signaling is a set of hand shaking it is okay. If you are having ideas about the hand shaking at one end then the hand shaking customs at the other end also can be............ if the hand shaking custom at both the ends are known then one can convert those hand shaking protocols easily that is not a big computational burden. But media transformation is definitely a very big computational burden because it is a full duplex computation that we are attempting.
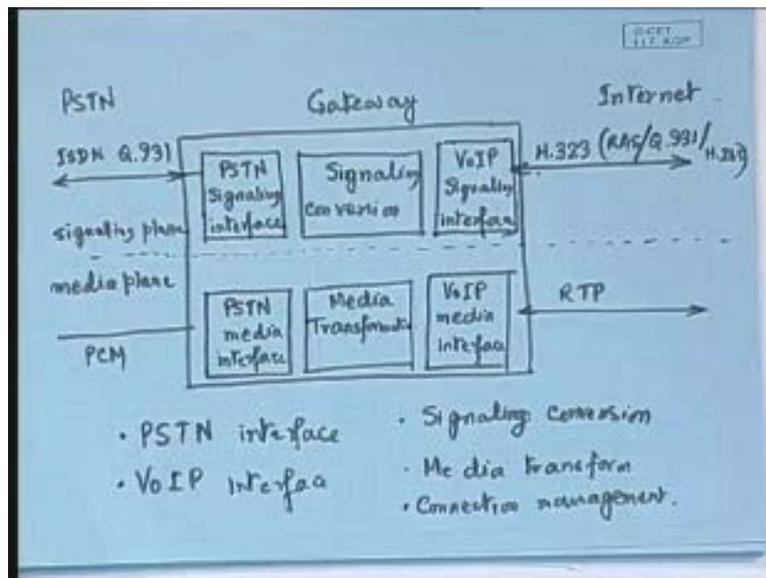
So this form of a transcoding that has to take place simultaneously from PSTN end to the internet end and from internet end to the PSTN end it has to simultaneously go on and that is why there should be some architectural breakup. One should not integrate the signal and media both together. But again the question is that if we only have the media gateway if we now split it up into a media gateway and a signaling gateway in that case is it enough; because you may say that okay, just now I said that media transformation takes the bulk and signaling takes small part of it then would not it be that the media gateway would still be congested and the media gateway will have to take care of the bulk media stream transportation from the PSTN end to the internet end and so on and vice versa.

Well, if the media is at least separated, if the media gateway is separated then at least one can think of an architectural solution where may be instead of one media gateway you can go in for a multiple media gateway or rather to say that trying to have some sort of a decentralized solution even within the media gateway also.

But of course such a kind of a decentralization would essentially require that there must be a controller who takes the effective control of the signal gateway and if there are more than one media gateways then the signal control also should be then the media gateway control also should be done. So there should be one block which should come in which should have an

interconnection between the signal gateway and the media gateway and that block we will be calling as the media gateway controller.

But building hardware for the media gateway controller is not the only enough thing because now there must be a proper communication between the media gateway controller and the media gateways which exclusively deals with the media streams at the RPT and also at the PCN domain. But actually speaking when this so So there should be a protocol that goes into this and this protocol is referred to as the media gateway control protocol. Lot of new breakthroughs had to be made in order to have in order to move away from the centralized solution of a single box gateway (Refer Slide Time: 30:58) to a multiple box or rather a decentralized system of gateways and that is what we are going to have.

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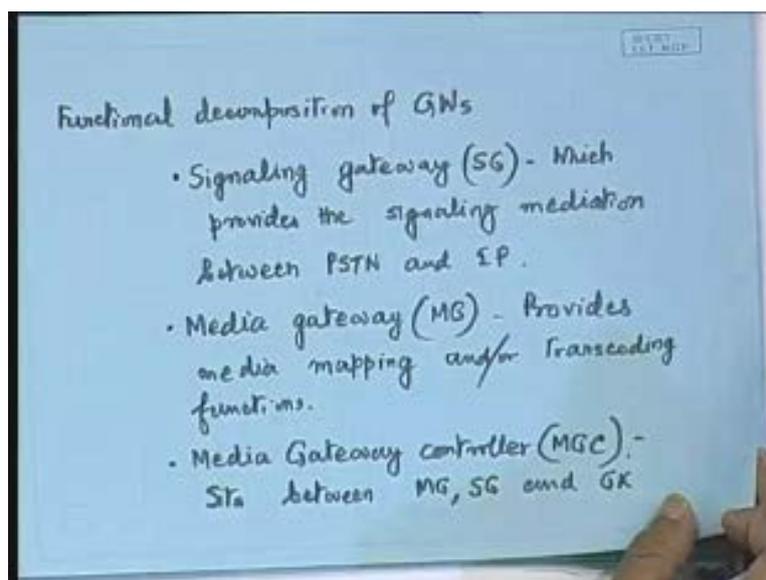
So the functional decomposition of the gateway we should have like this. We have the following elements. one is the signaling gateway or SG which provides the signaling mediation between the IP and PSTN and then one should have what I just now mentioned the media gateway what we can tell in short form as the MG that provides the media mapping and/or transcoding functions. The third one is the Media Gateway Controller or the MGC which sits between the so this sits between the MG the SG and the gatekeeper because ultimately there has to be a gatekeeper.

And block-wise it would be something like this that supposing we keep the signaling gateway or the SG over here so SG will have one link with the PSTN and then we keep a keep an MG and that MG will be connected to the H dot 323 TE (Refer Slide Time: 34:34) and then we have a media gateway controller MGC. So MGC receives the PSTN signals through this SG or rather interfaces the PSTN signals through this SG and on the IP side it interfaces........... so this is the IP side on the IP it is linked with the gatekeeper and the media gateway is actually linked with this MGC and this media gateway will ultimately have the PSTN interface; means for the PCM, for sending the PCM samples so this is the PSTN end and within this is the IP end.

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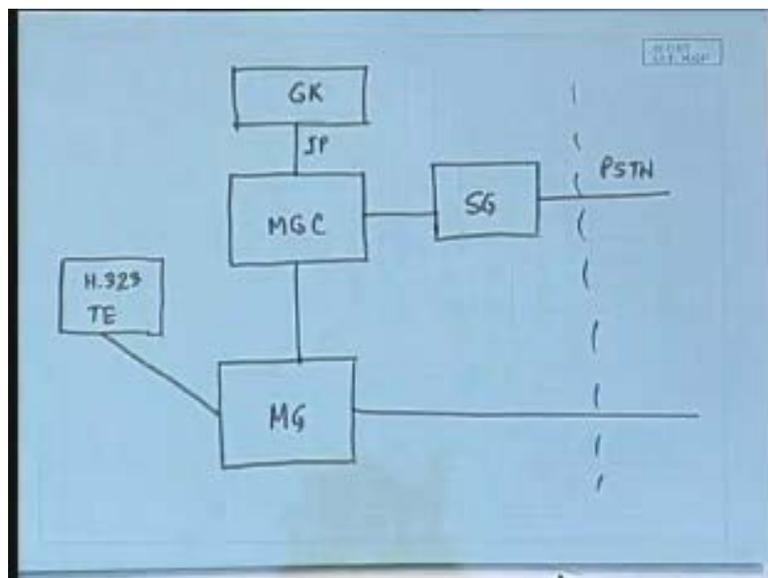
Now a single MG is definitely not the kind of a solution that we are looking for because single MG again will have lot of computational burden into it. So what one should do is to have multiple MGs means one MGC can control multiple MGs. If we have one MGC controlling multiple MG then one can have a one can have a system of virtual H dot 323; then one can have virtual H dot 323 and in this process one can increase the line capability because what you are going to do essentially is that you can have; with this MGC you can have the other MGs.
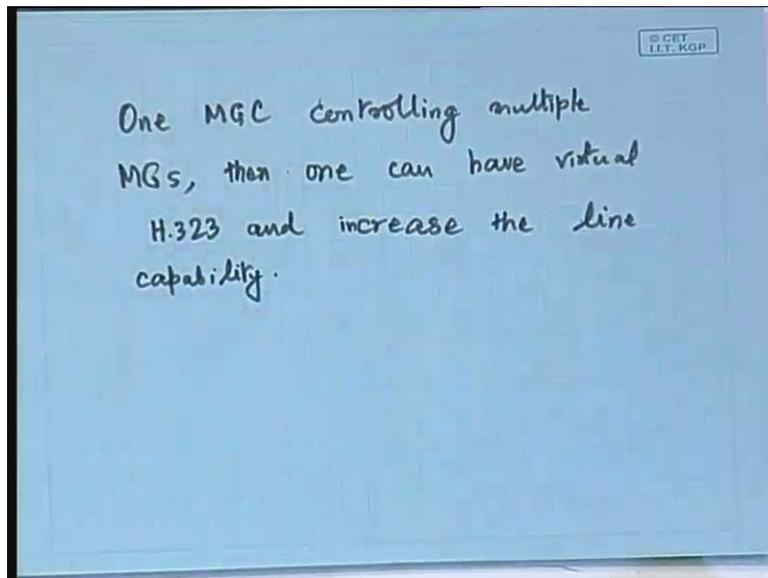
This is one MG, this is another MG like that (Refer Slide Time: 37:24) each doing its different different roles but ultimately there is some form of a multiplexing at the PSTN end so that the functions of these MGs are decentralized it is only through some protocol that it will be connected to the MGC.
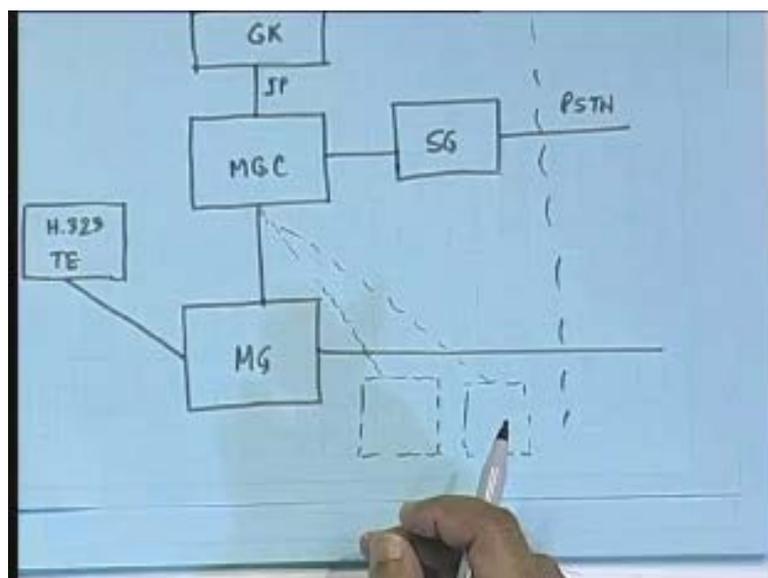
But even now also there is a risk that what happens when this single MGC backs off because we have not yet addressed the reliability aspect but we were talking that availability aspect or reliability is of definitely a prime concern. So, in order to have the availability one should have a kind of a redundancy that is built within it so there should not be one single MGC but there should be multiple MGCs and all these MGC's status all the call status etc will be anyway monitored by the gatekeeper. so that when the gatekeeper

So, if in the middle of the call there is any failure of the MGC which is handling the call in that case because the gatekeeper already has the information about the call history the call setup and so on so it knows that who is the source, who is the destination so that there can be a failover transfer; means immediately on detection of this fail which the GK will be able to do because as GK is monitoring this MGC so it can divert from one MGC to the other and the call can still proceed; even with failure of an MGC the call can still take place and this is going to be some kind of an advantage.

Therefore, in such a kind of case what one should have is that multiple MGCs should also control each MG. Because if you are connected to this MG and this particular MGC fails then the other MGC should be able to control. So this sort of a redundancy can be built in by having multiple MGCs and the decentralization can be achieved by having multiple MGs. So definitely there is a bit of architectural...... there is a good deal of architectural solutions which have been worked out in the recent times in order to take care of what do you say in order to take care of the limitations that we were talking of.

Thus, essentially we can address all these issues. And not only that, you see that what one can do is that within these MGs we can also workout the interfacing not with the MGs or rather because the SGs are now under burdened that we do not have a single gateway to take care of everything that is why in this SGs we can built in the SS7 capability also so that whatever difficulty we had because of the lack of SS7 connectivity this also can be taken care. So the limitations so all of limitations that we had talked of few minutes back that means to say the scalability, now we can see that the scalability can be improved by going in for multiple MGs and going in for virtual H dot 323 in order to increase the scalability.
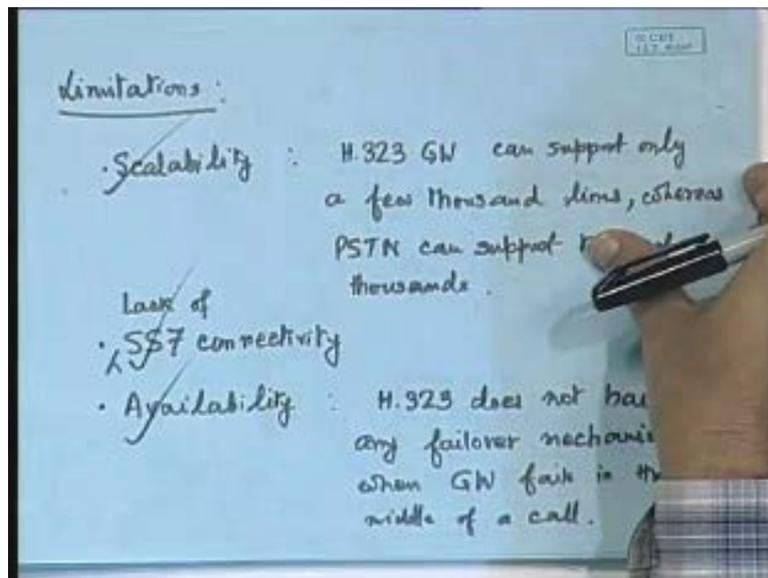
SS7 connectivity can now be taken care by building the SS7 interfacing within the SGs. Availability; yes, it can be ensured by having multiple gates by having multiple MGCs and the fourth point is of course the user friendliness that yes, if the SS7 connectivity is ensured then even one stage dialing should also be possible with the SS7 connectivity support.

But as I was mentioning that this form of an architecture definitely puts a new developmental process which is developing a protocol so that this MGC can communicate with the SG and the MGs and that protocol is referred to as the Media Gateway Controlled Protocol MGCP which is the Media Gateway Controlled Protocol but soon after the original development of this MGCP soon it was felt that to have these kind of capabilities like handling multiple MGCs for the sake of reliabilities and catering for the multiple gateways the original MGCP that was developed had to be upgraded soon and that is where the ITU and the IETF they worked jointly.

The IETF and the ITU they jointly worked and they came up with a standard which is referred to as the Megaco in the IETF parlance and ITU refers to this as H dot 248. So with the use of the Megaco or the H dot 248 it was seen that all these limitations could be overcome. That means to say that it leads to a kind of a feeling that only H dot 323 is not enough, we require some capability enhancements in H dot 323 using the H dot 450 what we were saying that the H dot 450 should sit on top of............... of course you can treat H dot 450

as part of H dot 323 already but it should be supported, it should go hand in hand with this H dot 248 standard. So together with H dot 323 this H dot 248 or Megaco should take care of the capability limitations that we were talking of.

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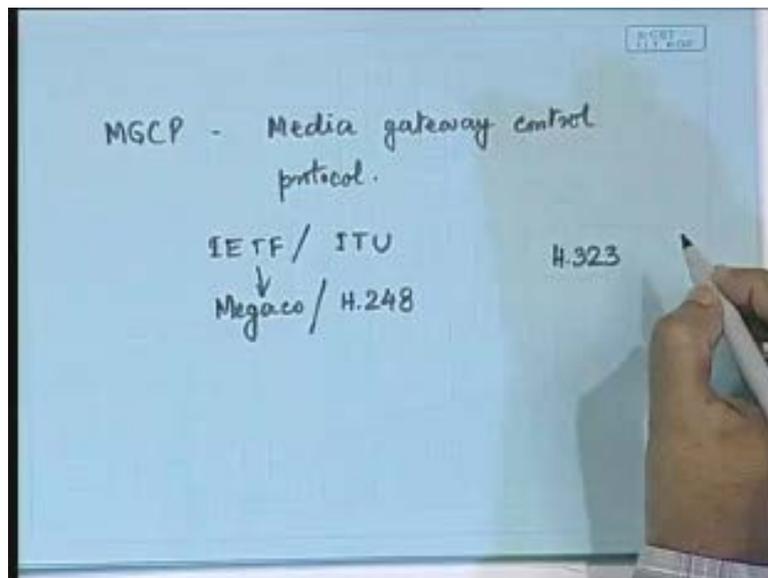
But then it was also seen that.............. so in in such kind of cases what happens is that the user terminals, the terminal endpoints are making use of H dot 323 whereas the media gateways and the signaling gateways they will make use of the H dot 248 standard so both will coexist together. but it was seen that many of the terminal equipment like say for example the cable modems or to say the ADSL modems, the modems also will be treated just like the way the terminal endpoints are treated but then for these why the calling should be done in the H dot 323 way; so there is also a simultaneous thinking that whether in future one can dispense with the H dot 323 altogether and rather use the H dot 248 itself for the signaling purpose as well as for the media stream interface. That is one part of the development.

The other area where the development process is on is with the SIP the Session Initiation Protocol. You see for an IP telephone whatever signaling is needed everything can be done by using the SIP protocol functions. SIP also will support the same type of signaling; only thing is that they will call in SIP parlance they will call it by different names but the basic protocol-wise there will be lot of similarities, the only thing is that SIP does not support the

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interworking with the PSTN. So again there if one finds out a solution between the interfacing or interworking between the SIP and PSTN then that also can pose as an alternative H dot 323 because SIP happens to be a very simple protocol.

In fact, in the next class we will see some aspects of the SIP protocol and also we would like to see some of the performance parameters that comes into place with the H dot 323. So H dot 323s multiplexing with several terminal equipment etc and its performance measurements these factors we will be seeing in the next class, thank you.