Digital Switching
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Lecture – 31

So, we will go ahead with whatever I was describing yesterday, and now basically we have to look at what will be the elements usually participating in a VoIP system. So, first of all let us talk about voice over IP. We will explain this thing onto other kind of media streams also parallelly. I think another one which is I know because I have designed that system is live class rooms over internet, for example, where millions of students or trillions of students can actually attend a lecture. Now you can even build up that kind of system, but it is not only audio video; there will be two party audio video, it is possible, multiparty audio video is also possible. And there is possibility of actually using live chat; there is possibility of tracking how many people are there.

There has to be somebody who will be master, who will be controlling, okay, and people will join the session; they will leave their session; they will raise their hand; they will ask questions. They I might actually conduct a live poll in a classroom; I might give a demonstration or lot of interactivity which is possible. Now this is altogether different kind of media streams which are available. Now when the SIP or the session initiation protocol was designed, it was you have to design a protocol; forget what is SIP. Suppose, if we have to design a session, basically a mechanism by which any kind of this media session or interesting session can be created among any number of people, okay.

So, the idea is basically how this can be created. I need to build up an infrastructure through which this kind of setups, this kind of sessions can be identified whether they are existing or not existing, and then people can join these sessions. So, as far as this whole infrastructure is concerned, it is only for initiating you into the session. Once you have joined the session, you are into that; then this protocol or this mechanism may not work. There might be some other protocol to which you will be handed over.

Very simple example is a conference call, where there ten people who are in a conference, and these ten people; when the conference usually is on, there will be a master who will be always controlling the conference. He will say, now you can talk or you can talk, okay, somebody who will be a floor manager. So, floor management system is not part of the design for SIP; through SIP, you will be joining this thing, and SIP will then inform you this is my floor management mechanism. There will be a separate protocol for that.

Now when you will talk, you will be using some kind of audio streaming live audio streaming; if it is a video call, may be video is streaming also, but then what kind of video thing will be encoded or transmitted at what rate? What will happen if there is congestion? SIP does not bother about it. So, SIP will only make sure that people join the session appropriately, and after that, the two peers will negotiate with each other and will do whatever they want, and they will communicate.

So, technically what we are doing is we are now building up a peer to peer system. So, telephony through peer to peer, but this is fine in a futuristic system. Skype is a peer to peer; when two Skype clients are there, they talk; they communicate directly into end, but usually there will be problems in the network. So, sometimes peer to peer technically is not possible. So, it require sometimes what we call intermediaries which have to be because of the network issues like using private IP blocks or when you are behind a proxy or when you are behind a net. So, it becomes a problem. So, you require some intermediary nodes, okay.

So, we call them reflector in some of the designs. Some of the design these are known as super peers or super nodes, okay. At some places they are known as proxies, but these do exist. Now what will be the elements? The whole world will not be VoIP in one day; you have to coexist with the existing PSTN, and PSTN telephony also has to be connected; that is another requirement.

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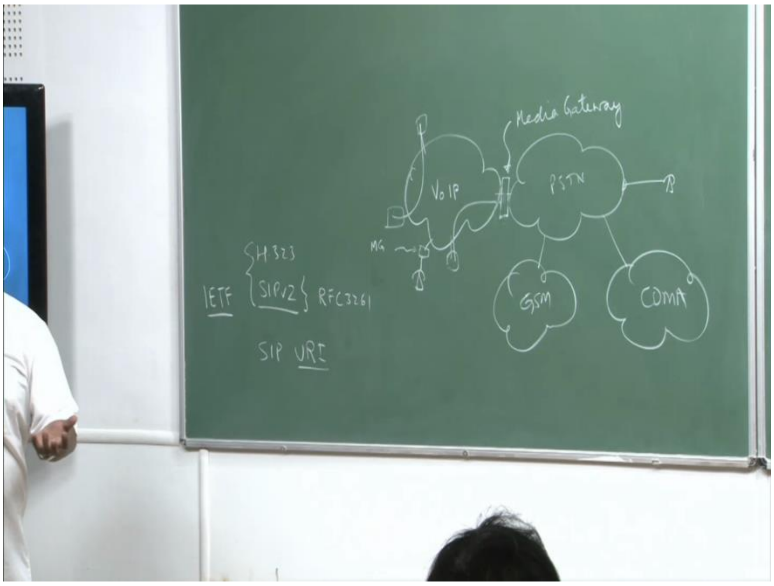

So, you will have a world where we will have an internet; we will have voice over IP clients. So, I am now moving one step further, and I am now including my PSTN network also into the system, okay, and I will also have a PSTN network. PSTN is public switch telephony network. It is a conventional circuit switch system in whichever telephones are actually connected, and you also will have GSM; you might actually have CDMA. Remember, these are also separate networks; they actually overwrite. So, this blob might be partitioned into two parts interconnected through a conventional PSTN, but this actually technically is a separate network, but it does gatewaying with the PSTN, okay. That is why your landline can call to the mobile phone.

Now this kind of blob is also going to come into the picture, and good thing is that this requires only internet protocol. So, whether you are using in the lower layer an IMAX or you are using 3G does not matter. So, far it provides IP connectivity, I can use VoIP system. Now how you will connect to the PSTN? I am not bothered about this. So, far if I can connect to PSTN, I can connect to all these; that is my assumption. So, there is no direct peering requirement between this and this, but this also can be done, but the same device which I am going to put here can actually do this job, okay.

So, I need a gateway here; this is technically known as media gateway. So, when you look from this side, all these phones will look like their VoIP phones; when you look from these sides, all these soft clients, we call them soft clients because most of the VoIP telephony clients are nothing but software. They can be put on embedded system and we call it a VoIP phone. You can actually buy these phones, and most of them actually all smart phones have a built in capability of becoming a VoIP phone, for example; it is just by loading software, okay. So, this media gateway does this translation activity.

Now one way is that whatever be the number of phones on this side, all of them can be mapped here as VoIP phones and I can run multiple instances for signaling, for doing peering bits for voice transport or video transport between any of this phone to here, and then there the translation will be done. And of course, there will be limited capability; if it is a conventional PSTN, only voice will be there. You might be using a packet switch system all the way for transporting your voice; you might be using a different codec.

When you come on to this side, this circuit switch system; you will be transmitting raw analog audio it is possible or if this can be that 64 kbps, what we call this audio which is there TDM stream that can be transmitted and it can be converted to analog at the edge exchange and then your phone can be connected to this, but usually these devices are costly. So, in fact, people usually want to use their own conventional phone, because they always are habitual with that; even to use that kind of things, some people might connect a small box. This is also known as media gateway and then intern connect their conventional phone on to this, okay. This is also pretty popular system.

So, you actually feel as if you are all having a conventional phone; you still dial number and everything, but whatever you do, this blocks captures all the signaling. It has a built in processor, and it does all the signaling everything which is required by VoIP and can connect to outside world actually. In fact, there has been a pretty popular system. I have actually seen that in companies who are calling from here to United States is pretty costlier if you actually use a PSTN phone.

Of course, now it is that cheap; the reason is because most of the operators take the call from your PSTN line. They actually convert it into a VoIP call, and through a media gateway, they will transport all the way to the United States or whichever country you want. And from there, they will again convert it through a media gateway into their PSTN network. So, long call communication is not through circuit switching; it is through packet switching. So, efficiency is pretty high, and call cost can be very small, but interesting thing is that I do not know whether you have seen; I have seen this in

India. People whose relatives are living in United States, they usually send them a small box, okay.

So, there are many companies who sell them, and it is basically a box in which on the one side you can connect the phone; it is a media gateway. You bring that box, attach in your ADSL modem or Ethernet, okay. And once that is done, only thing you should not have a proxy in between or netting router should not be. Netting router is fine, but proxy should not be there in between, because it is working over a UDP tunneling or TCP tunneling TCP proxy. But if it is STTP based, transport is not supported in that box, but now more advance boxes may be supporting that. So, even it can go behind a proxy if you can configure proxy user end proxy password.

So, what happens? The moment you connect this phone, this phone the moment that box gets activated will try to register to a server that indexing server of the operator in United States, and its IP connections, one can register there. It registers there; in that registry, it will register this particular address as this particular IP address of this box as nothing but what we call to the phone number. So, the phone which you are having in India actually is having a number which corresponds to US number. So, in US, people will think it is a local number, but actually the phone number lies in India. The call is routed through VoIP, because registrar for this phone is sitting in United States.

So, it has given a different number, but you need not have phone number as I told last time yesterday class. It has to be some user Id at the rate domain name. So, if the Vonage is providing this thing; I think this is one company which does that. So, this prefix is automatically headed by that phone and then there is a number which is US number which is your choice when you buy that box, okay. Now signaling people have built a proprietary. So, let us come to the features of the signaling and what all will be required. So, today what I have added is this one extra element which was not there earlier, the media gateway.

So, I have to handle this media gateway; I have to handle these clients through this media gateway; I have to handle the soft clients. So, a Skype also when you are making a call, actually is exactly doing same thing. Skype talks through their Skype network; this is the Skype clients. There is a media gateway which Skype has put at lot of places, okay. In

India, I think it is in US they have, but I think India they do not have a gateway, because when I dial from Skype client to India, I do not get an Indian telephone number, okay.

So, the only problem is the caller Id in that case. So, when you dial with the Skype to a phone number, you can do it actually if you have got the credit. So, once you do that. So, technically this media gateway is running as Skype client. Now just remember is a proprietary system which must have been built by Skype. Most of the media gateways come on this side with the signaling interface which is either h dot 323 or SIP. There are only two standards which are very commonly used. SIP also has two versions. What we will be talking about is this RFC 3261, okay.

This is the ITUT standard and mostly it is kind of still in use, but most of devices people want only this one. This is an open standard by IETA brought to origin, okay. Now this SIP v 2 is only handling session initiation or session creation basically enjoining the session process, and how you will get to know that to whom to contact? For example, what is your phone number; how do you know your friends phone number? That friend must have given it to you, or you must have found it on a website, or it must of come in mail thing, the bottom of the mail. In the same way, these you arise SIP you arise; we call SIP universal resource identifier. So, phone number the equivalent is known as SIP URI; URI stands for universal resource identifier; SIP stands for session initiation protocol, okay.

Now this is not the only thing; there are many other set of protocols which are required along with this. So, once I will come to this how the SIP actually handles the session setup. Once the session setup is done, this client will be talking directly to this usually. Usually in most common practices this will be done in this way, but also there is a requirement. For example, tomorrow if this SIP server or that indexing server or a registrar will be managed by, say, BSNL and Tata telecom or reliance, everybody get a license. So, far it is not there, but may be in a year or half year down the line, we will actually have those systems running in India also. Currently, it is not permitted.

What is permitted in India is only this kind of system, okay. Only thing which is only for call centers I think it is permitted to actually have VoIP system from conventional telephony to VoIP conversion that is permitted, because they run call centers. For example, for US if you are running a call center, this is a US number on which the people are going to make the call; in that call actually is all the way routed over internet all the way to India. So, those numbers which is their panel actually is the US number which is there, but Indian operators just provide them connectivity, and they have to actually keep on giving the reports.

One of the biggest problem in this case is you cannot tap the signals. If two peers connect directly, you cannot tap; if they both agree on some encryption algorithm an the payload itself is encrypted, even if you tap the packets, you cannot figure out what is there inside. Skype is I think it is a pay in a problem for Indian defense as well as home ministry everywhere, because they cannot tap those things. And you might have heard that during that Bombay stuff, they were actually using a Skype, and nobody was knowing what they are talking. And somebody from across the border was monitoring through his Skype only.

So, it is still a big problem Skype trapping; unless Skype company itself basically cooperates, but probably now it is viable, because Skype is taken over by Microsoft; Microsoft has an Indian office. So, Indian government can throttle them and make sure they cooperate. Earlier it was not possible. Skype was not a company in India. There was no incorporation here but now it is. So, I do not know what is the current situation? So, ministry must be taking care of that; tapping is not possible. So, you require a different kind of mechanism. This is usually not announced, but when you have actually put a tender whenever you buy equipment, it is a mandatory condition, and it is adversible that you should put it.

You never know when you will you yourself may require it or somebody else in the government may ask for it actually this particular scenario. We require what we call intermediary gateways. This is again a special kind of media gateways, nothing great. So, what will happen is I will give an example if for the timing forget SIP server. So, there is an indexing server running. I call it this is a more generic name actually indexing server. So, most of the peer to peer systems will have either distributed or a centralize indexing. So, I am assuming it to be a centralized indexing server here. So, when actually this guy want to talk to this person, assume it is like a Skype kind of client or Google chat kind of client. It will talk to this person and say I want to talk to so and so people.

So, everybody may update their status here. So, when this person wants to talk to this guy. So, this guy will inform him that somebody wants to talk and if he agrees to talk to this person. So, it will transfer its IP address to this person, and its IP address will be transferred to this guy. So, they both know each other, and then they will start talking to each other. So, this security somebody else will not be able to spoof, because the port numbers which are assigned for every communication; there are dynamically changing. So, which port number will be used for communication is assigned by this guy registrar.

Once those are done, they can just keep on communicating on that basis and can boot a stuff or more media streams, but this is dynamic; there is no static things. So, they have to keep on informing the port number at which this is active to the register or indexing server. Now if I want to implement a taping system how it will be done; I want to tap a certain number. So, I require then additional server; the way it will be done is this indexing server will know its IP address; this knows this IP address. This will also talk to this third guy your media gateway. You still create duplicated entities which can create kind of a conference call. Every packet which is passing through this filter is duplicated and can be routed to something some third party.

So, usually what will happen is this guy will be informed the IP address of this person; this guy will be informed another port number and IP address of this. So, this guy actually will see this IP address and port number as x, okay, and this guy will see that is this IP address and port number. So, when they will be talking, this media will be passing through this intermediary. An intermediary can be directly controlled not by indexing server; there will be a command which has to be given by your knob or the operator's command center. And they will control this gateway, and they can make duplicates of all the packets and can route to another client where the call actually can be listened parallelly or it can be recorded; it can be tapped actually. This is what is the tapping principle?

In conventional telephony also tapping does not mean that you go physically and put your wire and listen what the other persons are talking; this is not the way it is done. So, once we just inform, when the call is being routed through a switch; it is just being a copy gets created, and copy is also routed to another number through again a switching path. So, that is how in Delhi there is actually joined wing of three forces which actually does this particular job; all calls are can be routed to them without any issues, okay, and from any switch in India.

So, these are just remotely configured from the knob and any call which is passing through, they will just make duplication; it is routed. So, this either they do it or I think the police actually does it in some cases when the call tapping is required. And of course, one category is doing through mobile phone; mobile phone has a weak encryption. So, if you are in the same cell where the transmission is happening, you can listen to those slots, and that is the another way of doing it, okay. But you need to know that how the call is from which number to which number; at least one of them need to be known, okay.

So, this is the tapping mechanism which will be implemented. So, usually this kind of media gateways again need to be controlled, and these are not done through SIP; SIP only initiates in sets of decision. So, for doing this media gateway control, you need to do something more; only SIP cannot do this. So, lots of other kind of commands sets are required.

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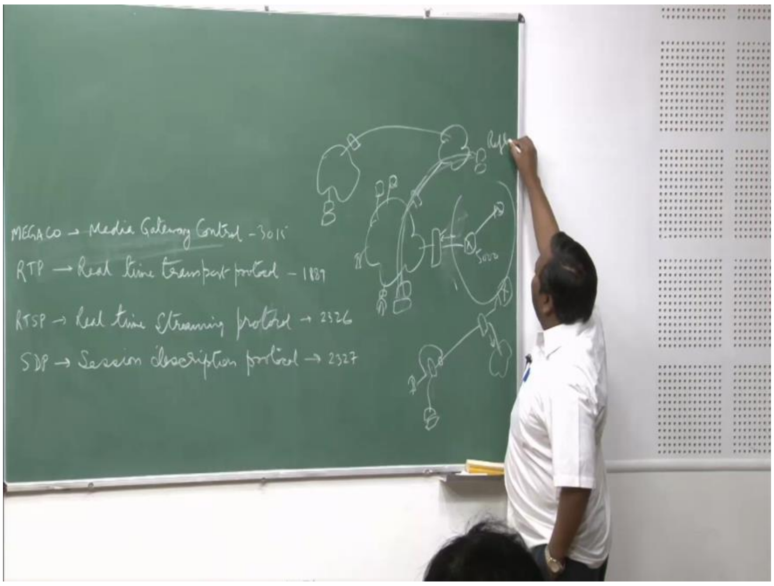

And we use a protocol called MEGACO for this. So, complete system is not SIP; complete system is many more things actually. I will just list most of them which are used. Then once we actually are doing a peer to peer communication between two end points, the voice has to be transported; it is a real time streaming. So, at time when you are playing certain things has to be fixed. So, time gap at which we have taken the samples and the recording side, it has to be played back in a similar fashion on the receiver side, okay.

So, this requires a real time transport protocol; this is again independent of SIP. SIP does not bother about it actually, how the media is handed. So, you will require also RTP and I think I have their RFCs numbers which are there. So, this is RFC 1889 3015. Now RTP is for real time communication or control basically, but for the streaming media where packet loss is okay. Once a while it is lost, it is lost; it does not matter. If you want to require a liability, you require the real time streaming protocol. This is usually for media, audio and video not for the control signals; RTP is for control signals.

So, RTSP real time streaming protocol will also be part of the design. This one requires an RFC of 2326. Now when we will actually use SIP, authorization authentication everything will be taken care of; the two peers can talk, but what SIP actually does is provides a mechanism by which an object, we call it opaque object. SIP never looks into what is there inside that. So, it is nothing but a file. This file is simply transported to the right person; he also sends back the file in the back. Based on this, these guys will share the information, okay.

So, they will share what is my IP address; what is my port number at which you should connect. Honestly speaking this indexing server does not tell in SIP case. Once the SIP call is setup, these guys through this route will inform that this is my IP address and port at which you should connect. This guy will also tell this is my IP address and port at which you should connect. So, all the signaling will be routed through indexing server or SIP server we call it SIP proxy; that is the term. So, that format of the file which will be transported through SIP is known as session description, and technically it is not a protocol. It only gives the format of the file through which a session can be described; any generic session can be described.

It is the format of the file; it is a text format. It is not x symbol a schema, okay. Now it is, of course, everything will tend to use x symbol schemas, because they are more generic, but this is a purely text based system which was built or evolved over the time actually. And we call it SDP; the term is session description protocol, but it is not a protocol as such, and this one is 2327. So, system which we are going to actually have another two months in IIT k will actually we based on SIP v 2. We will have our own media gateway huge number; 6000 analog lines will be terminating here.

So, there is no exchange; there is no switch on this side. Remember this is the way I am talking about PSTN. So, there will be conventional switches. In our design from here itself we will actually have 5000 wires going all the way to the each individual phone which is there. So, battery power will also be given by this equipment. Conventional telephony battery power is provided by exchange, but why telephony? It is not provided by the exchange.

It can be; there is a I triple e this a.2.13 I think that is what you have twelve. One of those standard is there power over Ethernet, POE we call it. So, if you are most of the time what will happen is how you will connect your VoIP phone to the IP network. Most of the time it will be Ethernet port which will be used; You can actually use wifi, nobody stops; you can use wimax, nobody stops. You can have any other kind of physical layer or data link layer; you can actually use on top of it. So, far it provides sufficient bandwidth. In fact, when negotiation of the codec will be done through this SDP thing which will be transported through SIP mechanism; that time itself bandwidth constant will be taken care of.

So, depending on available bandwidth different codecs might be chosen. [FL] Skype also does the same thing. So, if the bandwidth is different, Skype dynamically changes the codec, and that is why the quality of voice is almost is not degrading much, but it degrades if the bandwidth goes down, okay.

Student: Most of software is hard ones?

Most of the time these are embedded devices. See even if you buy a mobile phone, it contains I think most of them are built either with some processor. So, our must be in kind of become a defector standard, but of course, now atom is also being used extensively for smart phones. So, in India we have got I think only one manufacture providing a phone with atom; most of the other smart phones are all with RM series, okay, but these are pretty much very powerful processors. So, it is like a computer; your android phone is a computer technically.

We will be actually buying again a smart phone kind of a stub. We have already brought them, but it is a desktop versions. It is a pretty must like a computer, but there is no keyboard. Keyboard is like a telephone keyboard. The box will look like a telephone.

Yes yes, right right; it is exactly the same thing. It will be connecting to Ethernet port. But conventional phones also can be used by putting a small media gateway box. So, go to Cisco dot com site; there are many manufacturer which are making this small box a small [Fl]. It also has a small processor which is sufficient enough to do encoding, packetization, transmission, SIP signaling, your RTP; there is no MRGACO required in this case; this media gateway is not controlled by anybody. This is uncontrolled media gateway; this is only for one port, right.

So, it is a different name given, but technically it is a media gateway, but it does not require a MEGACO protocol. In fact, you can do away with MEGACO; you need not actually have it. You can still do SIP based control. But then you require lot of power here which needs to be done, but if you are connecting to a PSTN, in our case SIP actually was fine, because we are not having actually any switch. Our condition was there is a media gateway and then there are 5000 phones analog phones connected on this side and then connecting over IP network and then we have actually a twin redundant indexing servers; we call it SIP proxies, okay.

And then people are going to have their own VoIP phones. They might actually put a conventional phone with their media gateway; they can actually use a laptop or a machine a soft phone running on that; all three possibilities will exist. Depending on the capability of devices, you can set up even video calls, you can set up conference calls, you can set up chat sessions; actually email Id’s technically can be used for doing all communication. So, when you want to make a phone call to an email Id. So, it will go to again the indexing server; it will find out this email I d, what is the corresponding number and then this will be controlling. In our case, it is not controlling; it is a SIP based system

So, it also is just visible as a 6000 wi phones connected through one single port, but if the PSTN’s which would have been there if there would have been a exchanges network. So, you require a separate signaling between this. So, you require media gateway control in this case, because there is something more additional thing which are required.

That life is over; IT exchange will no more be there. We are just connecting existing 5000 lines onto this media gateway. The problem with this media gateway is that again they are not coming very large sizes; they are mostly 24 port which are there. So, we will have lot of them stagged. So, 3000 will be here and 2000 will be there near the hall one, and they will be connected through IP network. Earlier we were actually having a separate fiber connecting to exchange

Over the time; currently not because the people still have an issue if the power fails see problem at a house if I put a wi phone, I will not be having ADSL. ADSL is being thrown out now. So, we have Ethernet at every house, and it is connected through optical fiber. Optical fiber cannot carry power; we still do not have I think any matured technology by which I can also transfer power through optical fiber. I think we need to even have that kind of stuff. So, where actually we transmit signal as well as power through optical fiber optically and then we trap it out and use it for the switches on the other end.

So, unless the fiber cut actually happens, the power can still be transported. Currently, if transport is required, you require either a separate power cable which has not been done in our case, okay. So, it is a locally power disturb but with some kind of battery backup. But suppose a power fails in campus for more than eight hours, then there is a problem; UPS will also age. So, some sides there will be disconnection because of that. So, those VoIP phones at houses will not work. So, you cannot even make a complaint to the IWT that my power is not coming. So, this is a very sensitive issue. [FL] This we have gone through.

Ultimately, we figured out we cannot, we have to actually provide powering mechanism from the exchange side. And that is why this media gateway has come into picture with conventional phones. So, every house in a campus will actually have two phones, and technically, it will have the same number, and this will be an example of what I will talk about in SIP. When a phone will come, both the phones will ring, and you can pick up any one of them, and they are not parallel phones. They are technically two separate phones. An indexing server will now be sending message one to the gateway and one to the actual VoIP phone which are associated with this same number.

See interesting thing is with the same IP address and port will have one possibility for the same user Id; I can register multiple devices. So, when number will come, this can be routed to all of them parallel can be probed; any one of them is picked it is done. This is known as forking, okay, in this case of SIP or one by one. If you try this thing, this does not respond within a time, go to the next one, go to the next one. Ultimately, none of them are been picked up, go to the voice mailbox which is again maintained as a separate server, and SIP will be routing the call through that; that is one possibility. All of them can be done in parallelly; he will be actually using that now.

There is another interesting which can be done; you are going to your friend's house. You can go and actually type in a number and register even your phone number on that port. Now both the numbers are on the same port; it is like dual sim system dual sim phone. So, multiple identities can be there on single port; that is possible.

Ordinary phone ordinary analog phone which is already there as of now; the big gateway we are already purchasing 6000 lines. That is already there; that is a multiple of twenty four ports actually they will put in the rack. That is the issue of reliability, reliability and power. See when I am putting everything in the exchange; I have my good power backup source. If I do it distributively at twenty sides there has to be, for example, which have been built; they have got a small UPS. Now maintaining the number of the UPS if they are more, the more will be problem for you for maintenance.

I think the only problem with this system is this is not for the countries where power situation is bad. Now that is the only thing which I feel is a problem; I cannot transfer power. I can I think over ADSL also it can be done, but usually ADSL ports are powered up locally all ADSL modems; they are not powered through the telephone lines. But if you can do that if you can actually make very low power ADSL modems which can take up a power only from the telephone line itself, then actually this is possible. You should also now appreciate one more thing; we call it loop unbundling.

I can change my operator. I got a wire late from telephone exchange of BSNL; telephone exchange is no more required you are going to for voice over IP only ADSL. So, that media copper media is only used for data transport. So, voice is transported over ADSL over IP now. So, if at that exchange itself it is there; I can actually now simply change my SIP provider, my phone is still the same, my port physical port is same, IP address is same, port number is same. I can keep on changing my service provider at any point of time.

And I only pay to BSNL only for the loop and data transport charges and remaining charges I pay to the SIP service provider. Now here also issue of how you will do the billing will also come in to the picture. Currently, what you do? You get a telephone; for every telephone, you make the call, make the payment, one rupee per minute, okay, standard call rates. Now what you will do? You are actually going to have a line. You will pay for how many bytes which have been transferred, okay, and are you going to pay additionally to the SIP service provider that one rupee per minute. His role is only when the call is setup.

Once the call is setup, you are directly talking peer to peer. You no more require any more services. So, whether you talk for one hour, whether you talk for two hours, it does not matter. So, it is how many times call will be setup; per call setup the charges will be there by the SIP service provider. And for the duration it means call will be free? No, it is not free. If you talk for two hours, you are transporting more bytes. And if you pay on per byte basis, you are still paying to BSNL in that case, but you keep on changing your SIP service provider the way you want. The way Vonage gateway people are actually using and it is a US number, you are using SIP service provider of US. It is I think fixed annual fee which is charged by in US actually by the companies, but here you pay to BSNL for your data transport charge. ADSL is only a media link layer mechanism which provides a data transport.

Student: But voice is not going as IP sir in ASDSL.

It can when you are using Vonage, it is going actually as voice over I p; conventionally, it is not. Because what happens from your phone? You have an ADSL modem; you have your machine or whatever is your IP phone or wifi, whatever it is. This is going over a single line; you have what we call a filter here. There is a filter even before this modem; actually there is a filter here. So, this is the way it goes actually. So, in this direction it is band pass only lower band goes; for this side, it is the higher band which is not used by voice that will be used for data transport high pass filter, okay. This is a low pass filter on this side, high pass on this side.

So, similarly there is going to be a filtering. So, all high pass stuff will come, low pass will go, and this will be going for the exchange part wherever is the line card. So, your voice can be routed through these conventional phones. As far as this phone is concerned, it does not see actually your digital part; it does not see the ADSL thing. It is just from this point onward to this point, we using common media, but technically there are two separate networks. Physically they may be one, because this line is maintained by the same guy.

And on top of it, you are using voice over I p, you are using your cable TV what is your IP TV we call it; you are using your data connection all three. So, data, voice and t v, but I call it now actually multimedia. So, data, voice, TV, everything we actually fasted up and its all kind of will mix into each other over time. Currently, that is the way the BSNL actually sells rest of all three together; triple play they call it no. So, triple play is for these three things. So, you have to buy a separate setup box IP setup box here for connecting, but I do not know whether it is the same ADSL modem plus setup box is common or its separate. I have never seen this. Okay, then logically it must be separate, because it is optional thing IP TV stuff, okay. So, these four protocols will be actually used.

This simply will give you textual description of that; for example, I am sending a SDP message to you, it will be sent through SIP mechanism. And once I send a message, I will say I want to actually have a phone call, and I am going to use this particular codec rate. This is my capability, because I am going to transport video; there will be I am transporting audio, no video. I do not need any video call, okay. My bit rate maximum is this, and I have these these codecs available with me for example. This message goes to you; you will read this message, and you have your own capabilities.

You will say, yes, I can make you audio call; I have all that hardware with me. You will confirm back, and you will then find out what all audio codecs actually I have written; what is my bit rate. You know your capability. Whatever is common that common thing is which what you are going to send it back to me. So, the common subset of the two will be send back to me again in another SDP reply; reverse also s d p comes, and based on that I know, okay, we both are using the same rule. So, this is the codec which I have to use; I will trigger that codec. You will also tell me I have to connect to your IP address is this; your port number is this on which I have to connect for my media.

Student: One is sending set of files, other is sending another one.

Yes, and once your thing come, I will do acknowledgement and then we will setup a media stream and we will start talking and then we will disengage. These guys the SIP service provider will not be aware of what I am doing actually; except if he is routing the call in this fashion. If there is intermediary gateway or then only he will know it; media gateway also he might know it actually.

RTP is for signaling; in real time, for example, I want to switch over, RTP is again end to end. It is nothing to do with SIP. Once I know my peers address, I have a signaling thing. I can a start a RTP session. So, I can do a signaling on that. So, we will switch over the audio codec from higher to lower bandwidth depending on if my bandwidths are changing. So, it is basically real time you will keep on telling you what is the performance of the reception, whether something has to be dynamically changed.

Streaming is I will be just maintaining what is called buffer control. For example, we do not use in our Bhraspathi Singh, we are not using RTSP. Of course, it is a bad design, but that is the best which I could have done with whatever is existing manpower which I have in the project. So, we use buffer management; a buffer over actually crosses a certain flow, we start dropping packets, okay. Every third packet will be dropped. If there are again two much happen every alternate packet will be dropped or every ten packets. We start with tenth actually; every tenth packet will be dropped. So, we keep the buffer within the bounds.

If the buffer actually empties out, it means you are consuming at faster rate, and buffers incoming rate is lower. So, I have to now wait for some time. So, it is not that when the first packet comes, I will start playing immediately; I will wait. I will wait for ten packets to accumulate and then I will start my player. An audio and video need to be synchronized, remember; audio and video need to be synchronized, time stamping is important there. So, when I am writing here something and I am speaking. So, all actions have to be synchronized. So, that is why what actually this streaming protocol actually does.

Sir, what we get in terms of feedback; in real time we are playing, timing changes are there. So we need a feedback, sir.

This is what is that. Feedback comes through RTP. RTSP does not bother about feedback; it does the synchronization part.

How many packets were dropped? Every packet will be numbered. See, the hundred packets were transmitted; I got only these many, these many media packets were dropped. So, you will try to make a guess based on that. You remember you are doing UDP transport; it is not TCP. There is no retransmission. Even when you are doing signaling and its signaling has to be done in real time; can you do TCP? You cannot; you have to still do UDP. Something is missed out crucial or not crucial that has to be seen.

All of you have actually for example, a suffering from a situation. We are actually currently in Bhrahaspati Singh not using any one of these. Well, this is not required there, because it is a proprietary system. These two could have been used; this could have been used. This we are not using; we are using our own XML schema description proprietary. This we are not using; we are using buffer control here. We do not bother if slight voice mismatch something happens. You will always find the lip-sync is a problem in is Bhrahaspati Singh. In Skype lip sync usually you would not actually figure it out; they use again their proprietary codec equivalent of which actually does provide synchronization of the d s streams.

Synchronization I think is the one of the major issues. So, it is always done through time is tamping and creating separate buffers for each media stream and dropping and heading depending on the requirement. A real time I think what it means is another connotation of this is the clocks which I am going to use. Everybody’s clock if you have actually done GPS sync may be giving a GPS clock. So, everybody’s clock will be uniform. So, today I have matched my clock with yours is 9.40, and tomorrow if I come at the same time, your clock says 9.35; mine says 9.40. So, five minutes gap has been created; this takes care of even that scenario.

When play back rate and recording rates are different, they are bound to be different because all clocks are not synchronized over internet. This sometimes what we call buffer overflow or buffer underflow; both problems can actually happen, but amazingly this is not perceived by us, because when you play MP3 music, for example; on every computer, the clock rate is different. So, you buy two gigahertz, but actually it is not two gigahertz, two plus minus something which is there. So, when you are playing the music, you may not be actually listening to the original music. It is always slightly disturbed version of that unless the clock rates are perfectly synced.

But humans should not be able to peruse only that much tolerance is there. Zither is going to be there; you cannot move that. So far you can communicate, it is fine. Okay, now there is one more issue [FL]; I would like to explain that and then come to this SIP. Sometimes, the problem of behind a proxy or behind a net that actually comes into picture; in IIT, Kanpur Skype does work actually without any issues even if you are behind a proxy. So, a Skype if you go to tools options settings whatever it is, then you will find there is a proxy setting there; proxy username and password is also there, okay.

So, it is a STTP based transport; what is STTP based transport? STTP is a very unique kind of protocol; it is I send a request, you give me a response; you forget about me after that. So, it does not remember what was my earlier request and what is my state. So, it is a stateless protocol; there are no states maintained, okay. Web servers of course, do it actually, maintain it through a different mechanism. These are known as cookies session cookies basically through that which are sent with the every request. So, it can figure out I am talking about which particular; state information actually is transacted with every transaction. So, then it can remember that a state on that basis using those pointers. So, it is a stateless protocol.

So, important thing that you want to send something to me, how; what happens in IIT K? Suppose, this is an IIT k network; this is your machine. This is the proxy; this is the outside world. I cannot send any information to you. This is not a web server. I am using private IP address. So, this guy cannot connect here. So, if I connect a Skype running here, this Skype cannot send audio packets or video packets onto this side; it is not possible. It is always this is going to start the connection, request the proxy for getting some information. It will go to server and then information will come back; this guy cannot initiate this stuff.

Now that is the bottle neck here. So far I have been using symmetric system; both sides can independently push the information, it will flow and it will reach. In this case they cannot reach; it is not possible. So, what you will do is we have actually faced this problem. So, we have used a technique here. We used what we call timeout mechanism when the periodic fetch. So, whenever you might have seen in your certain web pages, when you download, after some time it times out, and it again refreshes the page. So, there is a refresh interval.

So, usually what happens whenever the STTP request is sent, then the STTP response is sent. In STTP response I actually can set a refresh interval. So, this guy is going to set the refresh interval, and it is periodically keep on fetching. Now there again two methods in STTP; one is get, another one is post. Get method is you will send only the URL; there is nothing is there in the payload part. So, you only can send everything in the URL itself, and then only you retrieve the information in response to the get, and there is a post method. In the post method, I will have a URL. I also can add the payload part, and there is a response to that post method, I will get something in the payload.

So, what actually usually Skype does or for that matter Bhrahaspati Singh does client is periodically doing post method to a URL which responds to this, and whatever packets which need to be transported on this direction will be sent as the payload of the post. And when the response of that will be coming, whatever were the packets which are cured up here will be sent in this. Now it does not if it still stops sending post method that STTP request. All cured up packets will remain cured up here; they cannot be retrieved. This is in no way can send the packets; only this has to push the packets and pull from there, push the packets and pull from there. This has to happen periodically at faster rate.

Now that is the problem with STT, and STTP has its own over additions; it has to setup a TCP connection all the way here. Fortunately for every request, we do not now setup a TCP connection again and again. Usually, you setup one single connection; keep on sending a request on top of it. And this will again setup a TCP connection all the way here and then STTP will be separate text based messaging on this TCP system. So, this is something which sometimes is required if you are behind a proxy, and nothing else can be done.

Even if you are behind a netting router, then also it is not possible that this guy can send the information back here, unless a staggered entry gets created on this stating router. So, in IIT, Kanpur it is not possible that somebody from outside can come into and send information to ourselves; we have to keep on pulling it out. So, your Skype which you are running in your machine is periodically pulling from some other client which is running outside, because your friend with whom you are talking might also be there behind a proxy. This poor guy also can only push and periodically fetch. These two guys when they need to communicate, they require something outside; without this, this system cannot work

It is encryption; it is purely encryption. I am going to come to how the authentication will be done in this case, because I think Skype that is my guess; I have not read the documents. But the way it is operating I could actually figure out; most likely this could be the one possible reason, and I think the design should work. So, that is what I am also planning for our new Bhrahaspati four system as well as for Bharaspathi same clients. Ultimately, we will actually move to that; we will not be using no more login passwords. Login passwords will be used to get initial certificates only, okay.

So, Skype for example, everybody starts doing login when the machine boots up is going to be a big problem on the server. So, I think they are using security certificates. When you do first time login, it takes some time. Once the certificates comes and gets installed, everybody has authenticates by presentation of those certificates. So, you can authenticate with every your super peer with that; you need not go to the central server. So, even if central server is down for some time, services will be not off. People we can still authenticate each other; that is the good thing about that system, and it is actually server failure proof system. Even if server fails, it is okay.

Yeah, I will come to that. Now this is what is known as reflector. Reflector in we call it VRVS. This is again a multimedia system which was built I think by lot of open source guys; this view also cause call this same thing as a reflector. In Bhrahaspati Singh system, this is known as super peer or super node in case of Skype, and this actually means you are running a Skype. You are connecting through your ADSL, and you are using a public IP; that technically means if you are not using STTP proxy, you are behind a netting router also it is fine, because it is usually in the static net, or if you are in a public IP; you run a Skype. You are not making any call, but still data is being transacted. And remember your operator might be charging on per byte basis; you are going to still pay without making any call. So, I think we close here, and I will now come to the security architecture in the next class and then SIP after that.