Digital Switching
Electronics and Communication Engineering
Prof. Y. N. Singh
Department of Electrical Engineering
Indian Institute of Technology, Kanpur

Lecture – 33

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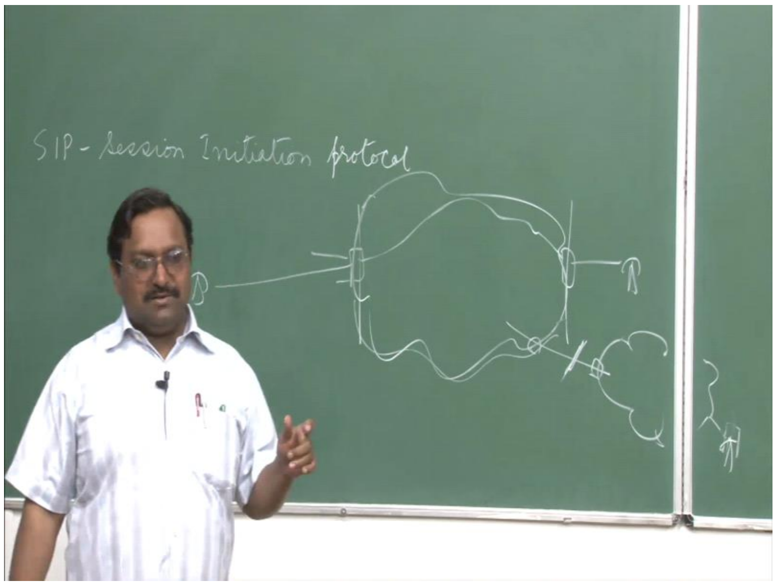

It is today we will start with SIP, SIP is stands for session initiation protocol. So, some of the smallest of actually I had been telling about this protocol already. So, what has happened is as peer to peer computing actually has evolved over time. And it become the basis for voice over I p telephoning. So, people actually thought of building up a protocol a stack. So, this was 1 of the thing them actually then there was another 1 remaining four actually I have told in my earlier lectures.

So, those kind of if they played together then they if actually form a complete multimedia architecture functionality essentially. And in very limited sense you can build up simple voip telephony system. In fact, the all existing telephony infrastructure can be implemented with this. So. In fact, there is all ready most of the telecom operators slowly will be moving in this direction. So, one of the first once in think was British telecom which made a complete change to voice IP base telephony.

They are no more using circuit switching in their network they are actually using completely voip systems and even for telecom operator it does make sense to actually build up this kind of thing. But, as of now india I think it is not legally permitted to use IP base telephony unless until this PC to PC PC to PST anything is permitted here. So, only exception is call centers were over a IP network actually all the calls will come and will be received here and over a ordinary telephone line usually people will be receiving. So, that is the only exception.

As of now operators actually do a I think that is my feeling I have not verified this fact. But it is possible that from your home the phone is their it goes all the way over same copper cable and in the exchange they just put a media gate way. And then they can within their network can route this as voice over I p call goes to other end point and then again going as a ordinary copper cable actually.

So, this is technically possible. So, you actually do not get a feel you still do all dialup everything all information signaling which happens in the conventional structure. But, intermediate in the core it make sensor to convert it to. Voice over IP because your cost will be very, very low here only you going to use bandwidth and your actually the person is going to talk otherwise, there is no packet being generated there is a silence being detected here nobody is talking. So, do not generate any packet.

Now, that is a very specific feature in any recording system in any digital recording system this actually happens. So, when for example, I am not speaking here there system will figure out that I am not speaking there is a gap. So, even if there is some noise which is going perturbation whatever it is system should able to identify through the threshold detection. Because, when I am speaking actually signal level is greater than a certain threshold noise is below the threshold actually.

So, when it is below the threshold do not do detect digitization you can actually do digitization you will get some number some PCM code for that, but that is immaterial you simply do not discard. It actually. So, that will bring by all the systems this actually means for telecom operator even in the core if they start using this infrastructure. It make sense because they will be route of push through large number of calls. And of course, they did not maintain 2 separate overlay networks over same infrastructure currently they have to maintain a SDH voice actually is being transported to that thing without any compression, because it is like a 64 kbps channel which is fixed for every voice.

So, he is going to consume 64 kbps per voice channel plus overheads and everything and then he has to actually overlay a data network, because a most of these guys are also providing IP connectivity. If they go for this kind of infrastructure there is no 64 kbps limit there is a silence detection and for tremendous amount of compression which can happen and no separate data in voice network is required.

Only thing is that for voice thing you require quality of service support. So, that can be handled using MPLS in this case. So, that will be 1 way but, introducer will never figure it out this within an operator. And when an operator is going to talk another operator for example, if I make this as cloud of an operator he need to talk to another operator because the destination phone is here.

This peering is not IP based this peering is like conventional peer I lines kind of thing traditional circuits. So, this critically happens in various different cities between the switch is this is done, but this is an option, but I am not sure in india this is being used as of now or not. This is a mostly it will be circuit switch link it will be mostly PRI lines-Primary Rate Interface ISDN. Is a 2 mbps circuit over which 30 voice circuits can be set up at any point of time.

I say even PRI even PRI is ISDN PRI. So, in fact, that even is of 2 kinds even R 2 MFC and even PRI, but even R 2 MFC is no more even used is mostly PRI which are used only difference is the signaling channels. How the signaling is actually happening for various voice circuits, that is a only major difference. So, I think now R 2 MFC are no more in use no body uses them I have not , we discard them almost seven or eight years back at IIT K. Now coming to SIP, so how will SIP actually will work.

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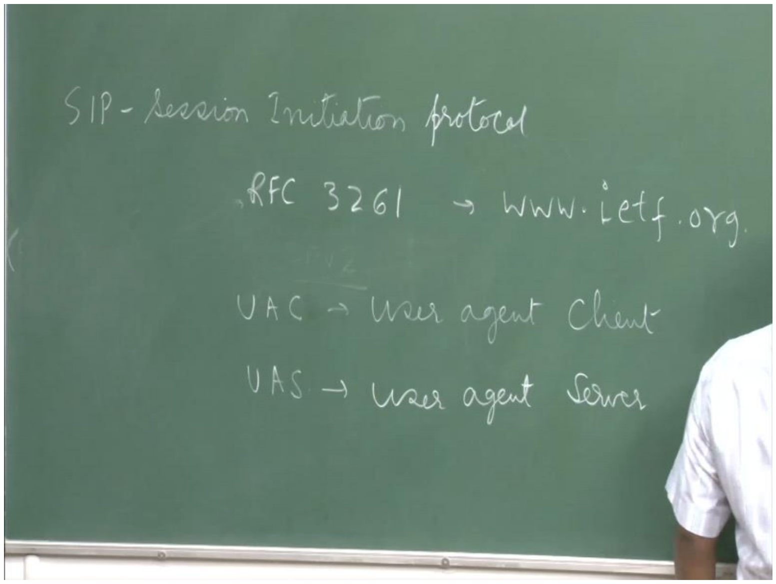

So, far I have given in the name now lets me give the number of the RFC. So, RFC number will be in this case which defines RFC 3 2 6 1 inch defines the core of this. This RFC is available at and currently it actually absolutes an earlier RFC, RFC which was 2 5 4 3, these was the older 1 this has been scrapped. Now, R systems actually follow this SIP and its not explicitly mention, but nowhere in the text in RFC we call it SIP version 2 except, when you will look at the headers fields which are been transported between 2 endpoints.

That is why SIP slash 2 points 0 will actually come the way http. We have http slash 1.0 http slash 1.1 same way will specification here, but the this standard itself does not tell then it is SIP version 2. This was SIP version 1. So, currently we always look for most of the cube mates are build with this. So, there has been some changes in between. So, I am not going to cover this particular part.

So, we will be working only with this 1 now the entity is which are defined first thing are a 2 basic entities. So, 1 is known as u a c u a s this is typically known as user agent client . And this is known as user agent server and important both of them are actually user agent the way why this definition actually has come is there is 1 peer.

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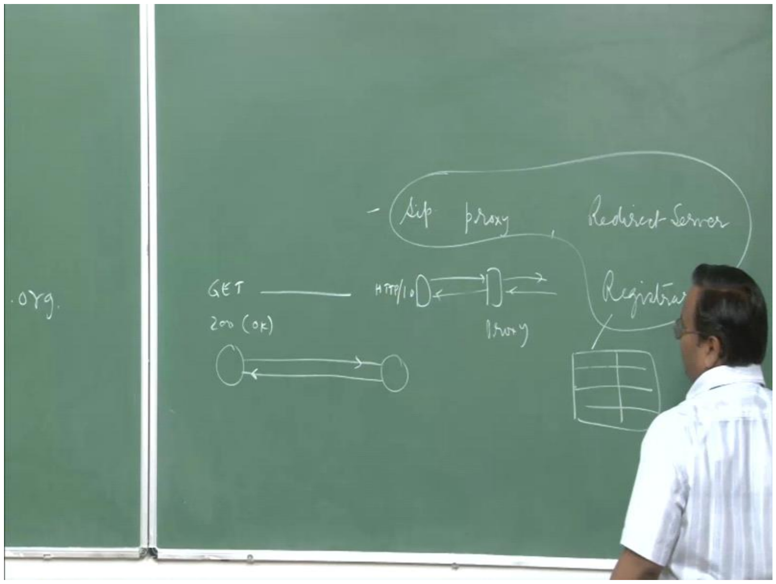

There is another peer if these persons sends a request it is a client it is a user agent only is your soft phone. And, but it is acting as a client because it is sending the request and this guy is now going to respond. So, this is known as user agent server, but both of them are user agents. So, it will respond back to the client. Similarly, who is initiating it depends on that both of them are user agents. So, 1 is known as caller other 1 is calling. So, this guy is making staring the starting the call this is receiving the call this responds.

It can be other way around this can work as user agent client and this can come user agent server. So, usually s r is the software design is concerned there will be 2 separate portions 1 which depends on the which takes over the request. And whenever, a request comes what is the corresponding response which has to be generated is handled by user user agent server. So, when this guy is initiating there is separate software component which will be participating, but each client will contain both user agent client as well as user agent server both will present.

Now, there are many other things these are the 2 entities most of the SIP phones will actually will provide this or if you get a phone on your machine. So, 1 of the very popular SIP phone is software is Ekiga. So, this is available for your windows. So, you can download and install it even can actually make a phone call through this, this is technically possible. And this is a pure simple client similarly this also comes for windows linux and window both those of you actually have a android phones and would like to communicate over a wi fi.

So, there is also SIP clients which are available even for android phones, but they use either by wi fi or a 3G data connectivity they do not talk to your GSM and other kind of things which are built inside the phone. They are they are many SIP clients. So, what I have been using is c SIP client actually that is a light weight 1, but there are many high way high end ones also are which are some of them are paid versions. Also I am using a free 1 that can connect to multiple proxies that is a advantage it can register with multiple of them.

So, actually my c SIP client and android also connects to my ekiga account. So, I have an account here. So, you can give me a call even on that. So, my phone number there will be actually dash nine that is my then my phone number: 9, 4, 5, 2, 0, 4, 8, 4, 5, 1 at the rate Ekiga dot net. So, that is my SIP URI, but it was not a phone through Ekiga that 1 that 1 will be same of course. If you call it no body as so far called me I have got this account for more than 6 months.

I have not received any call on this account I am still waiting for the first one. So, user clients now you will have more elements. So, 1 of them is known as proxy. Typically this is also known as SIP server this is technically nothing but it is not an indexing server remember see in earlier case I have not talked about proxies were not there. So, equivalent of proxy was a reflector node or a super peer super peer in your skype or in case of it is a reflector that was equivalent to proxy.

But we call them SIP proxies that is the third entity which we will user agent client is one. So, 1 will be server 1 will be always a client then SIP proxy. This usually is a intermediary program you want to make a call to somebody you cannot make a direct connection. So, you tell somebody who is on your behalf will try to set up the connection will find out talk will register we will find out somebody will set up a connection on your behalf and we will inform you back. To whom you have to connect on which port and what all parameters which are accepted and based on that you will make the call.

So, you does everything on your behalf. So, this will typically will be requirement because of multiple domains, because I might actually get a my own SIP proxy my own SIP proxy will only permit the phones will only knows the numbers which are existing within IIT Kanpur. So, whenever you are trying to talk you always request that proxy 2 set up the connection. Proxy will also be having a indexing server we call them register actually in this case equivalent of that.

So, whenever my device is active connected to internet port powered on it will register with the register and of course, MAC address binding will be done usually. So, it knows that this IP address corresponds to this particular phone number, but this mac address also corresponds to this phone number. So, usually what happens is somebody else can spoof your ID, but cannot spoof your mac address.

So, in between somebody cannot take it unless you login even from that mac address. So, always that thing is also additionally done has a security thing this is evolved over time and IIT Kanpur you want to make a call somebody out side. There are 2 ways: 1 is there is a media gate way and I have even PRI lines. So, that is a connecting to PSG network. So, I can route the call through that through even PRI or 1 way is suppose if BSNL is going to run its own SIP proxy.

So, we will have an agreement we will know each other and we will authenticate each other that is a way, because even now of my connect my even PRIwire comes from BSNL and technically we authenticate using those physical port. Somebody takes that physical port out connects to his own machine in between. He can start making call like IIT Kanpur actually does not matter. So, there is technically no security even for except it is a physical port connectivity here there is no physical port I am connecting over a internet line.

So, 2 SIP servers will be talking to each other. So, whenever you want to talk to somebody who is 1 BSNL line and I have only then tie up with BSNL now you want to make a information call how this will be done. So, most likely I will tell my SIP sever that I want to make a call my SIP server should talk to BSNL SIP server its its like an exchange. And you will then talk to somebody else and ultimately the other guy will inform his SIP phone at the destination.

Once he accepts all the message will come all the way back to me and once we both are I will know whose are who is my destination what is the IP address. He will know what is my IP address and port number and then will make a peer to peer connection and call is set up and we are keep on going to work with that, but they are complications in this. For example, I cannot do a midway negotiation is not through SIP proxy it can be done directly also. But sometimes it is not required it is just not to be done, because there can be a problem; for example, you have done this kind of thing you are making a call to your friend.

In between that third phone actually comes in and you want to add him to the call now adding third party to a call is a complicated process is known as mid call future this cannot be implemented is a peer to peer relationship is done. So, signaling usually is always has to be through always proxies it will never be direct even if the direct connection is set up. But that is an optional thing I will tell how this option gets implemented in say. So, SIP proxy will do that it will basically a client will be their connect to the SIP proxy for signaling purpose not for transport; it will then make a request response comes it tells the response.

So, this will be proxy the way we have http proxy similarly we will have t c p proxy and our h t t p proxy also checks the login and password otherwise does not permit. Same way it will also check the validity of all users here and we will have something called redirect servers is another entity forwarding agents technically. So, you can register yourself on to the redirect server and based on that all calls will be actually terminating for you, we come to redirect server most likely it will be a proxy.

So, far whether a domain I can actually have a SIP proxy for that domain I can have a redirect server. So, SIP proxy can give it to a and this will maintain where the currently the person is moving he can be mobile he is moving somewhere else. So, I can give a redirection URL’s. So, you are change the organization gone from IIT Kanpur to IIT bombay, but you want to still to remain the actually maintain this number. So, every time the call will come will come at whatever your address at the rate iitk.ac.in.

So, once the phone call comes here rewrite server will find out where the call have to be forward it where the signaling has to be forward it. So, remember call is a still not voice call is not going to come to Kanpur its only peer to peer directly between end users this request can be sent to multiple places simultaneously through redirect server. That is a beauty of it and then whichever places you will actually give a reply it will come back. So, this done through mapping remember it is not the same address 1 address is been map to another address.

Through a mapping table that is only what has been done the way the mail is forwarded for example, you can forward your mail coming to iitk.c.in to gmail.com. It is a technically doing that particular work only for signaling this is special kind of proxy technically. These all are logical entities which I am telling as of now you can put them in a same box running as a separate procedure x I, physical is only a box executing a program operating system can done multiple instances separate instances of separate programs.

But physically it is a same box roaming is slightly different implementation roaming what happens is, your name your, your destination address does not change, your phone number remains the same redirect is you have got some number you go to some other place you got another number. I am redirecting the call to that it can be even conditional redirectional. So, you are trying trying to dial you your number you are not picking up after some time the call will be forwarded to another number is a call forwarding actually. Equivalent to that in signaling, but they call it redirect server in SIP terminology.

SIP proxy it is a known as a register sir.

No register is separate logical entity and this is not indexing server sir SIP proxy is not an indexing server.

SIP proxy in not indexing server who will maintain the directory sir who will tell the user that this is a just connected over their because directory functionality is done by.

There is no directory you have to get somebody has to tell what is the I have to tell you what is my phone number then only you can call me know directory you have to buy from BSNL. If BSNL is maintaining mobile directory or I have to give you a number or you have to have my e mail or website from where you will get the number. Unless I give you my number you cannot call me and there is no directory server in PSGN. We cannot say give me the phone number or its maintain over a separate website separate booklet.

For example I dial that SIP number sir.

See otherwise what will happen you can start calling me you anybody who does not want to receive your call.

If I dial a SIP number sir.

Where does this request go to sir, because I will not.

Request usually I am coming to that particular sequence of events. I am going to come to that of trapezoid of communication. The next 1 is your register the final entity. So, here the moment you will have an ID this is known as a your SIP URL that is what it actually known as which will identify unit. So, for me it is YNS dash 9 4 5 2 0 4 8 4 5 1 at the rate giga dot net. So, that is my phone number. So, there is no numeric thing that is a important thing that is a alpha numeric establish comes into picture now.

But, I would have kept 9 4 5 2 0 4 8 5 1 at the rate giga.net also would have been fine. But putting YNS make sense. So, mobile phone also now you dial you do not dial by putting numbers you always dial you store them and dial with the name. And androids of course, this become a stemley powerful because they are synchronizing directory from the lot of servers which are available over the internet.

Linkedin face book and everything. So, you actually dial names e mail id's technically and there is a display name correspondingly, you will dial that or you search with an organization my organization name is also there. So, google does provide that service. So, in the phone itself I can find out I can push the number it will knows all the numbers from that face book all other linkedin surf. So, far there is a common e mail Id across all the accounts all of them are join together that is a functionality of android.

In fact, I think almost now blackberry is a as well as this 1 your apple they are also providing very similar services otherwise they would not sell in the market. Honestly speaking a this is extremely, because you can through your phone buy a new phone. The old directories can be restored back which is not possible in the conventional mobile phones. So, ultimately they will move to SIP telephony they also do not have much option and with Y max now coming up or four g. So, Y max telephony is going to come by other Ambani brother who is not in telephony currently.

So, he will be providing this site same SIP telephony systems over Y max. So, by that time he is also hoping it will become legal I think this year and it should be there as already we have know technically, but they have to only change the revolution. So, there is mostly all these will be nothing but the separate software process is running in the same machine mostly. When you will you just run them what this register will contain is only have the access to data base there. When you will attach your phone or whenever you change your IP address you will reregister with them it maintains your SIP URL. And what is the corresponding attachment point as of now the attachment point is technically IP address mac address port number.

There will be a signaling thing and there is signaling port and there will be then data transfer port see voice for media. For example, which can be dynamically change for security purpose this is not a new technique this has been used in FPT protocol. FTP is not that you just tell your machine on the other side and the machine sends you the file on the same port number.

This does not happen this way you tell make an FTP request over a signaling ports which are identified FTP actually happens on I think 21 FTP is 21 or 20 21 of them is s s I 23 s s I think. So, FTP port when you connect that is signaling you tell that I want this file to be transfer to me to put or whatever.

So, the other machine actually tells on which port or from which port you should transport on which port I am going to set up a connection. So, that side server actually sets up a connection in the reverse direction. But, TCP connection is not set by client for data transfer it is set by the done by the server servers actually sends the connection back on that port where you are suppose to receive it and then the transaction of actual data file actually happens.

So, again this was all done for security reasons this was a kind of a hack which was figured out or implemented in FTP protocol and this was earliest 1 earliest design of this kind. So, register will maintain this redirect server also will get most of the information from here technically it will be nothing but proxy, but I will have access to all these data alternate parts. So, this register is currently where you are connected or what all active machines are their here you will be telling where the call has to be forwarded in what case in what scenarios.

So, this will be a separate data base attached to this this will only maintain what we call the current active signaling paths only information about those. So, typically now what happens the whole structure is very similar to http R hypertext transport protocol. So, whatever you response you will send the first line will contain the message, which was sending in the signaling mechanism. So, far example in http you always do that post. So, usually you will say get and then the URI and then you will have http 1.0 or 1.1 whatever it is you want. So, that is a kind of thing.

So, same thing will be used here in the response you will always say 200, it means or something like these numeric coding actually is used for giving the responses whatever other fields will come estimate. But, these actually tells the machine that what the state next states of the FSM, which is used for implementing the protocol FSM is finally, state machine. So, going from 1 state to another end so on.

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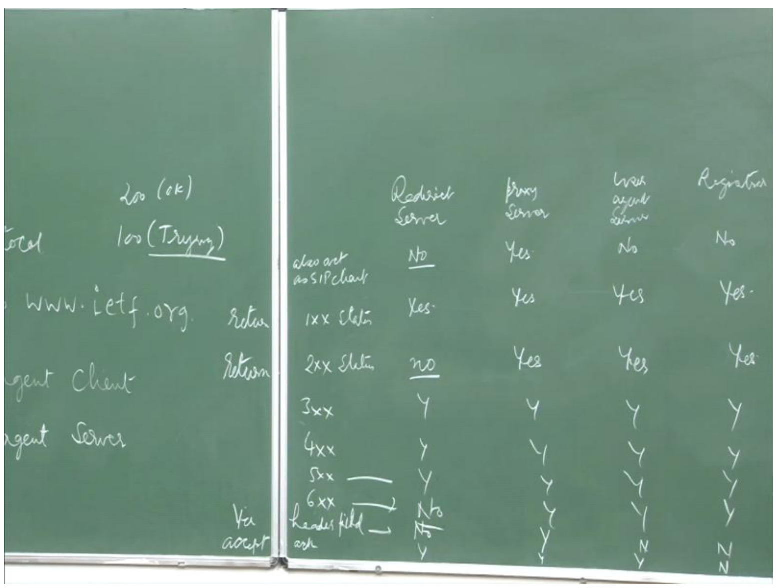

So, there is a list again which is been provided for various capabilities of these logical entities. So, whether now I can actually put them yes four actually here you will have redirect server is 1 entity; you will have a proxy you will have a user agent server. And you will have a registrar. So, whether this can act as a SIP client. So, what about redirect server it cannot act as a SIP client it is only passing the signaling information.

It cannot act as a SIP client. So, it is known for actually in this case. So, it is actually sending back the replying back it is not forwarding the request like a SIP client. So, you request actually goes to him it sends you back that go now. You should send the information here I cannot send the information I cannot forward on your behalf. But I am giving alternate entries. So, this cannot proxy server yes it can user agent server is the last end point actually you say user agent and this is server part where the call signaling path terminates.

So, this cannot act as a SIP client this will be acting as only as SIP server last server. So, this will also know register is no only SIP proxy can do this job because this is doing signaling on the behalf of a client. So, that is a first capability then 1 x x this usually is is a status code. So, 1 0 0 1 1 0 1 whatever number, but the number will start with 1. So, this status code is retuned actually this status code redirect server actually can return this usually is that you kindly I am still trying to keep on waiting. It is not final response it is a intermediary response.

so the transaction is still not complete is basically for that purpose redirect server may say kindly where I am still trying I am still trying to search my data base I will give you a redirection URL. Proxy server yes this each 1 of them can tell this thing status 2 we will come to this numbers actually what for this purpose actually these are used as we go along. They are used exactly whatever the http responses are used for these codes are exactly same used in many other protocols.

Then 1 dot series which is successful completion kind of thing 200 is for example, 100 is for trying I am still trying for making the connection who excessive status is ... So, this how these things get differentiated. So, this will not send ... So, 2 x, is because signaling will not be terminating. So, 2 x x cannot be send by redirect server other guys can send, because they can terminate the connection you are registering your thing in register he is sending I have registered everything cleared transaction over. So, it can send.

So, 200 can be send by them, but not this guy transaction will not be completes giving a redirection. And, because transition complete has not come this guy has to keep on working based on the response. Similarly, you will have 3 x x all 4 can do this 4 x x 5 and 6; there only 6 up to the 6 actually. I will explain all these later on not now as we keep on using I will just for example, 100 if for trying because that will come when I will start explaining all the call set up. 200 is when the call actually gets transition gets completed it is usually.

So, that is what happen whenever I will write hundred in bracket I will write trying. So, protocol machine does not understand this system it only understands this. But, the meaning will be this actually that is the way to be return in the description. 200 similarly, will be written for ... So, 4 x x is also yes 5 x x is also yes actually for all of them 6 x x is not used here these are the only exception rest everything is yes.

Now, there is something called wire adder field in the adder which is sent there is something called wire. Wire always tells that when the response is going to be send back do not send a directly it has to be send wire in. See for example, he asking me you set me call for me, I set the message signaling message to you. But, I set the wire field that wire; it is me. So, you know that who is the guy who is for whom actually the call is been setup you will not send the response back to him you will send to me.

You can send to at another person, but you do not had a wire free. So, that person when he will actually respond back. He will see the first wire which is for me he will send the signal to me I will send it back. So, anybody who remains in the path signaling path was the everything has been setup they have to keep on adding this wire in field now, which all servers actually can do this. So, next 1 is whether wire adder field can be added or not; obviously, redirect server cannot add the wire adder field, because it is not suppose to be the intermediary for signaling proxy server. Yes, this is not a intermediary that is also not a intermediary.

So, they cannot add wire adder field. So, this know for everybody except the proxy acknowledgement who will actually send the acknowledgment acceptance, actually this is excepts acknowledgement who will accept the acknowledgment. So, it is yes for everybody except the last 1 in rest are typically uses the software base system you have to periodically keep on updating if it times out the entry will remove. If your registration entry, because sometimes you have registered your number with the certain IP address binding to the register. And certainly your power goes off you never add a chance of removing the entry from the register.

So, it is a soft history it is not a hard state time out it will expire you have to periodically keep on refreshing that entry. The same thing is done with even GSM for example, every 6 hours usually a registration has to be sent back and tell that you are in the particular area. And suddenly you take the battery out he will still keep on trying that area your registration do not be half and then he will figure out that you are off. But, if you properly shutdown your mobile phone it will be deregister from their and then it will not be sending any information back. It will be telling form their itself the phone is switched off.

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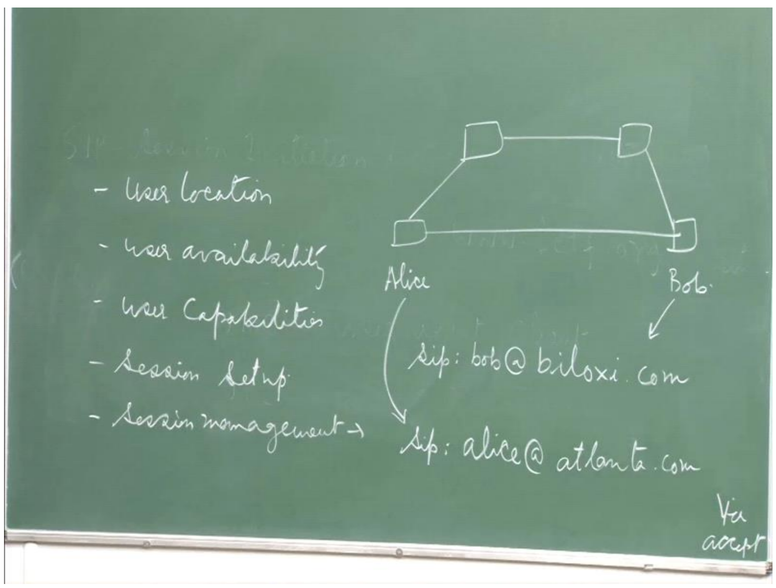

So, now let us go to how the procedure ... So, you will have now these thing which will be done by SIP user location this I have also already mentioned actually I think. But not have I have not written it actually user availability you have user capabilities actually. I am just making my more explicit, but most of it is already understood session set up. This basically, it means ringing and setting of session parameters that is what the session set up actually means and then, there is a session management.

Here, you include transfer and termination of the session modifying the session parameter and working services there are three things which will come here. So, that is what the SIP will do. So, usually you will actually have a trapezoid; we call it a trapezoid actually I am just taking a same example given in RFC. So, like in most of the examples alice and bob are again present here going to use the same name it is there the RFC as it is.

So, does it come from alice most likely. So, alice would like to call bob that is a problem here and this has to be solved. So, bob will be identified by a URI which will be something like this. Since, it is been done in USO, they have chosen those names I am sticking to them I hope this is these are all prestigious name they do not exist actual operators have non there with this name.

So, alice is their very particular about it no branding. So, this will give the URI and this is for the gentleman name bob. So, alice somehow will get to know of these thing the way I have told you that my URI for making a phone call is SIP colon YSN dash my phone number at the rate at giga.net for see might have also figure it out from somewhere. Once this is their she has to make a call the URI for this alice is . So, that is a URI that is a domain and that is a name within a domain same name can be given to cannot be given to 2 different way naming as to be unique within a domain. But, same name can be used across 2 different domains, but those will be counted as 2 separate persons.

So, this is like your you have name what they are actually lot of students with the same name. So, usually in your register duos register it is always son of or daughter of for other name is also attached, that is counted as domain name unfortunately fathers name is also they can also get the ... So, we do not have a concept of domain name, but here this as been taking care of 2 domains cannot exist to cannot have same names. So, when a new domain name actually is been requested it always search, it is already existing it is never given actually.

So, that is for example, you cannot have iitk.c.in been given as domain to somebody else its only for IIT K you can change something else or you can become a sub domain with an IIT. For example, e.iit.kc.in is fine or you have to change IIT K either you have to change either a c you have to change in you have to change. So, domains are always unique. So, that is the way the naming will be there. Now, how the message what is the fundamental thing which will be done. So, most fundamental thing is the invite that is a most basic thing.

So, when the message or a phone call has to be made. So, it is like when you dial a number a message has to go to that I want to set up a call call is set up. So, when this is a SIP phone she is going to we call it a soft phone if it is on a PC or it can be hardware box containing all the SIP software setting inside it and a invite has to be sent to box. So, that call can be made, but she does not know she cannot find out where the bob is or bob may not except it. Actually even you directly because there is no way you can authenticate that is a problem.

So, she does not know where the bob is she does not know where the proxy server for the bob a proxy has to be connected because the IP address of these guys will keep on changing dynamically you switch of your machine. Next day you tomorrow you come up it takes another new I p address from DSP it might be different entry every time every day. And may be different port every time which is available on which your server can be activated.

So, she has to find out a way. So, only way is she has to look at SIP server within her own domain then I told that within IIT Kanpur domain. It will have over own SIP server running may be after 2 months and then all the phones will have to be configured with that SIP server. So, may be when the phone was provided to alice her phone was configured to use SIP server which corresponds to atlanta.com.

Or when the machine boots up in the morning there is a d s p dynamic host configuration protocol it keeps on looking for the d s p broadcast once it finds out a d s p server it says requires and then get all the information. So, it may get which is a current time which is the domain name, but domain name usually will be configured because identity is there is a key which is involved in every SIP registration, because it has to connect to even to register. But remember register is logically different than a proxy only for connecting to register and telling what is your IP address you require a authentication. But not for connecting to proxy, but usually proxy will authenticate you by challenge response and you will also has suppose to do which is the current proxy. So, you are going to at basically have faith on your DSAP that it will give you correct proxy the way it gives you correct routers here.

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So, now there is another variant of this switch exist not only SIP you can also have SIP SI can add an s here. So, this is very similar to you have http and http s what is the difference between the 2 it is a secure option. So, similarly for SIP also there is secure option it works exactly same way they way http works method is exactly same. So, alice will find out this server and she will send a invite request.

This is written in capital to her own proxy which is configured here this invite message will tell that what a server is suppose to do when you see gate some URI. You are telling the http server this is what you suppose to execute. So, you are telling the server what the method is suppose to do and response has to come back to me. So, invite is the method being told which is to be ultimately executed here.

So, actually this currently it is been sent here. So, this will execute and give the response it is a proxy it will understand and will again sent invite to this that is a different matter it got it can do something else also for example, if she has not paired has used. So, the invite will not be accepted and it will be refused the call cannot be made through the ... You might have to pay monthly rental for taking the service of a proxy server because there is cost involved cost has to be borned by the users.

So, invite know usually will contain header fields invite message typically how the invite will look is this. So, I am just writing how it will look like in the text it is a purely a text message h t t p is also a pure text. And remember the all header fields are named attributes. So, there will be a name and there is a there is a attribute name and the corresponding value of that attribute. So, for example, 2 will be the named attribute and whatever is their bob at the rate a biloxi dot com will be the value of that attribute and colon will be the separate tree usually that is a standard.

So, invite will the first line there is no named attribute that is important thing first line only tells what is the method which is been invoke. And then after that you will have all the header fields then there is 1 blank line and then whatever is been sent header actually ends there with the blank line. And then whatever is the pay load part of that. So, payload for example, invite might contain s d p s d p will be in the payload part. So, invite and what is the argument of the invite is this you are trying to invite this bob at the rate biloxi.com.

Now, this is something which is this is 2.0 SIP slash 2.0 if you have been using 1.0 you would drive SIP slash 1.0. And depending on whether it is 1.0 2.0 server is suppose to interpret the fields accordingly the interpretation of fields actually changes across the versions. The version has to be their both versions 1 and version 2 actually provide for invite. So, type of header fields here or type of request is version 2. And then after that you will have the first field which is first letter is capital then a small via why is basically through a certain path or I think it is a street in italian via probably stands for that.

So, you will say it is SIP 2.0 comparitible over a UDP transport. So, this actually tells the response is suppose to come here back at this place. So, any signaling response to this should be sent here I collect more wire fields also and they will be in order if a new wire filed it has to be here. It will keep on pushing now this thing. So, PC 33 is I am again taking the same thing, but that thing does not matter now this is the domain name of the machine. This is the name of the machine on which the soft phone is installed or this can be name of the phone itself phone is like a computer and of course, this line is still not finished remember.

So, I am writing I should not put a black slash actually this should be long boats, but the remaining text I am just putting indent in remember, this whole thing is 1 line in 1 line you will have a branch equal to some random mystery. So, random mystery also again copying from IIT f draft, but this is actually random I can put anything. I will literally which I wish does not matter no relevance to us as for is there is no meaning the meaning is session depending.

So, they out some values. So, you put whatever it said a b c 1 2 3 4 5 a b c again f g h whatever you want. So, semicolon is the separator within this. So, the 2 1 is the attribute this tells the version of the SIP which has to be used the transport the machine to whom the response has to be send. And response has to come for a SIP 2.0 claim it has to come by UDP that is what it has been explicitly told and this is a branch parameter.

So, branch is going to be unique branch is a random parameter which will be used to identified the session uniqueness basically there are many random fields. So, no 2 phone calls even a between the same persons alice bob can set up another call and the every call and the every call can be uniquely identified this can also be locked.

Sir PC 32 what is?

PC 33 some number given to that machine for example, my machine in my office is actually having a name: yamini.ee.iit.k in ac.in that is a name of my daughter. So, I kept my machines name on that. So, you have a machine in electrical agent you want to have a name for that you have to send a mail to me and I can put that mail and then whatever is the name e.iit.k in ac.in will resolve to an IP address which will be for that machine.

So, this similarly a name, if you resolve in the DNS for atlanta.com domain given IP address where the soft phone is installed. That is what it means names are easier to remember by human beings otherwise IP address also. I have been put it is what I will do is I will close here, because recording is finishing and we will I will continue with this description tomorrow. So, this is whole trapezoid. We have to complete we have to have more fields we have to study their interpretation and the reverse direction also there.