SAN JOSÉ STATE UNIVERSITY

EE 284

PROJECT 1:

Voice over Wireless Ad-Hoc Network, A Hands-on SIP-based VoIP Experiments on: Call

Establishment, Busy Lines, Call on Hold, and Conference Calling

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1.PART-I:

For this part we used the virtual machine approach as creation of Ad-Hoc networks was not suppoted.

The below is the network settings available for VMWare fusion we used:

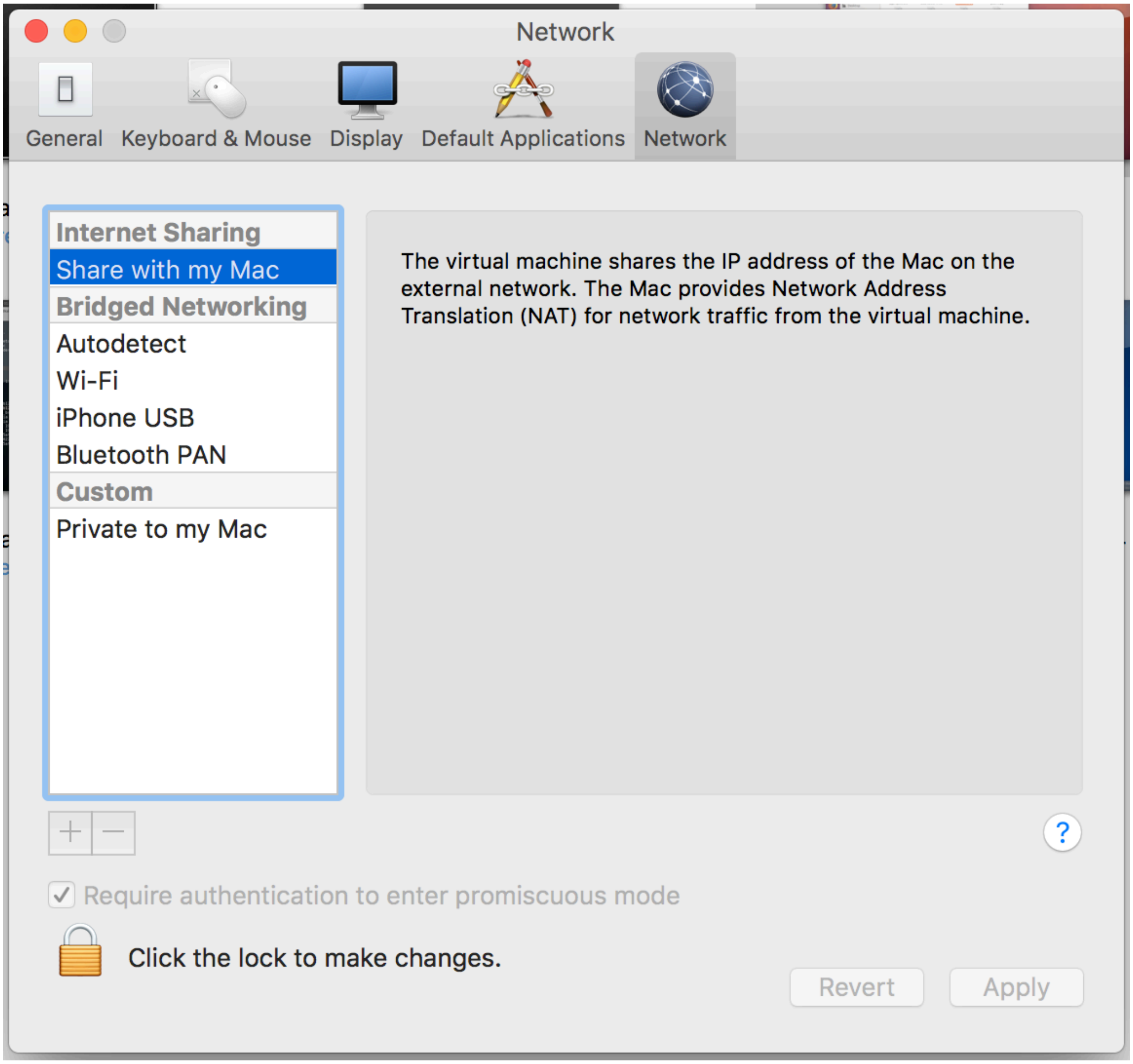


Figure : 1.0.1

The asterisk server is installed on Kali Linux the machine details of which are as under:

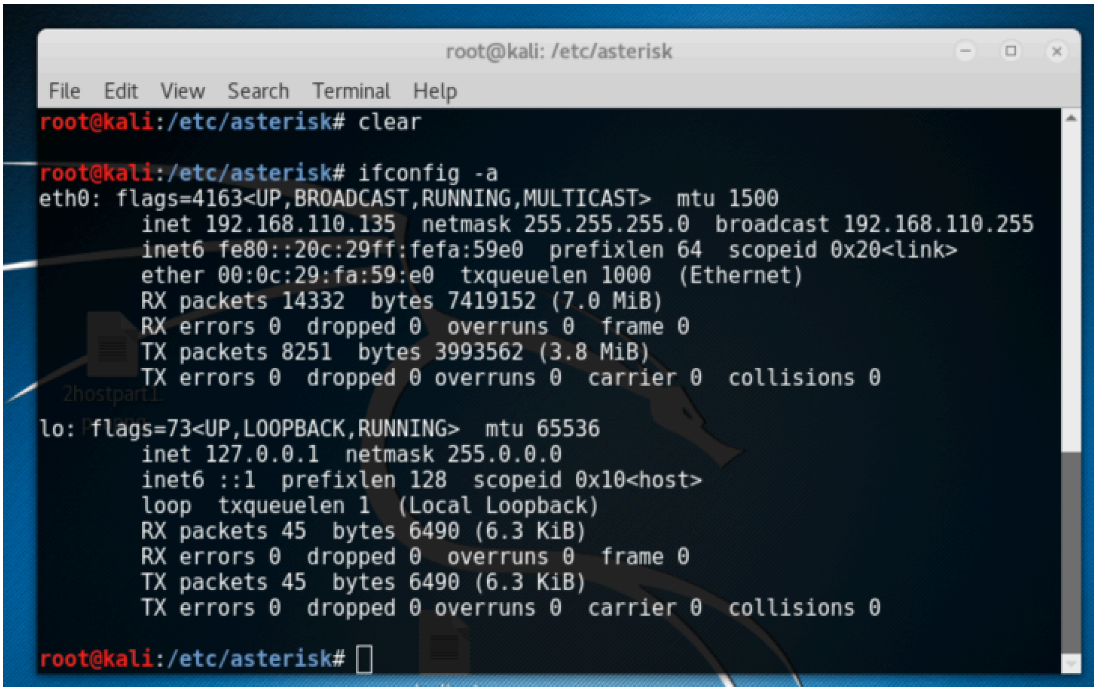


Figure : 1.0.2

Three clients as per requirement of project were created in VMWare Fusion with Windows 7 ultimate OS with usernames :2000, 2010 and 2009 respectively whose configs are as under:

2000:

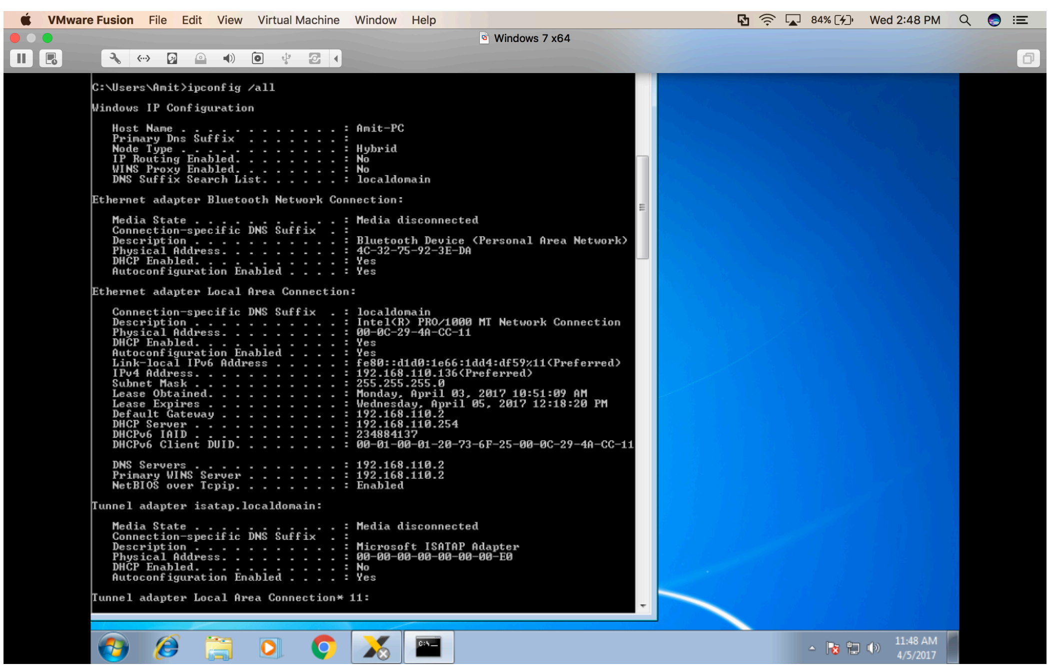


Figure : 1.0.3

2010:

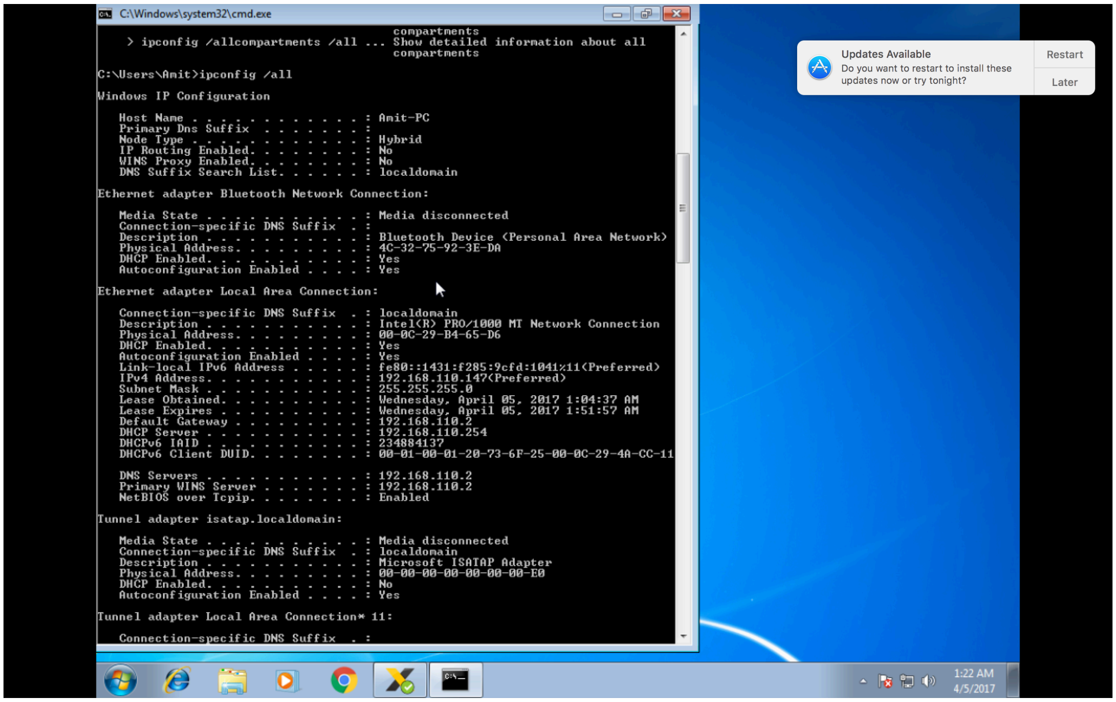


Figure : 1.0.4

2009:

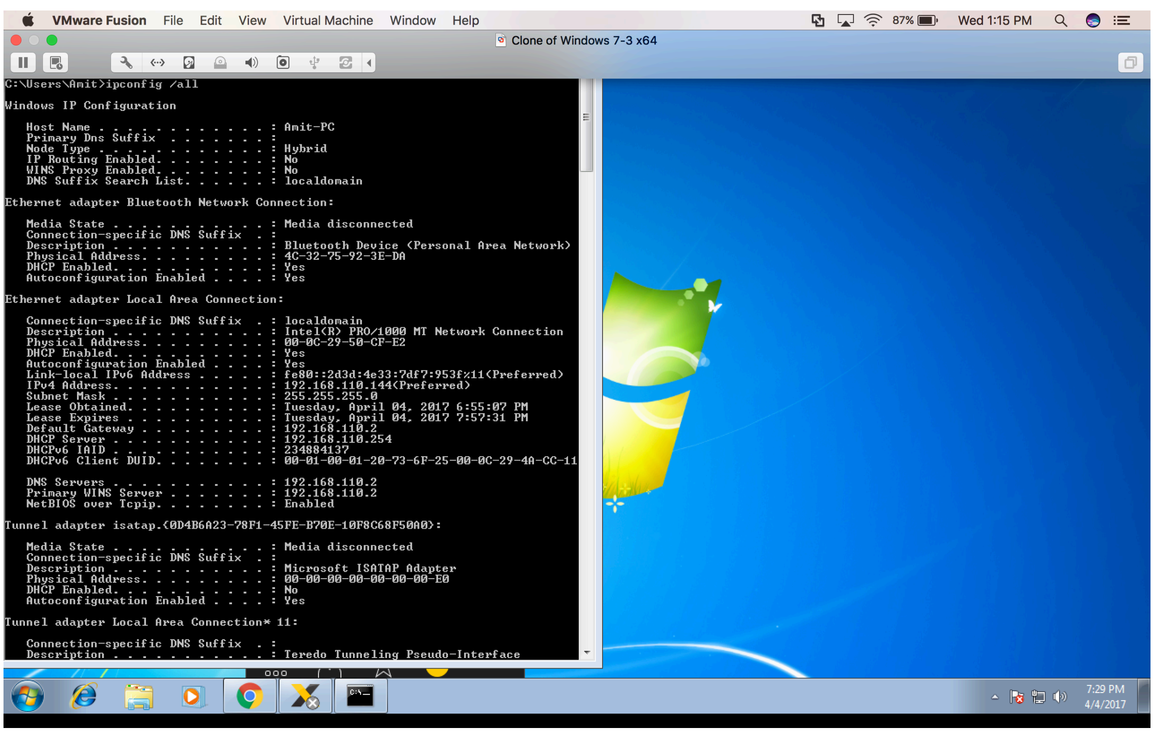
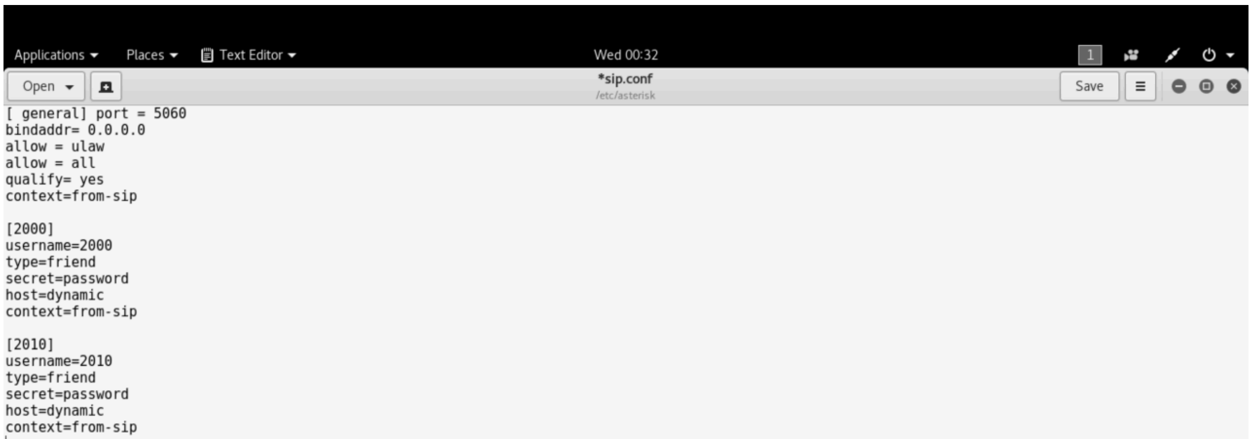


Figure : 1.0.5

PHASE 1: **<u>Establish and Analyze a Successful Call Between 2 SIP Clients.</u>**

Here we use two sip clients 2000 and 2010 and call from 2000 to 2010 in order to establish call.

Below figure 6 and 7 are sip.conf and extensions.conf files respectively.

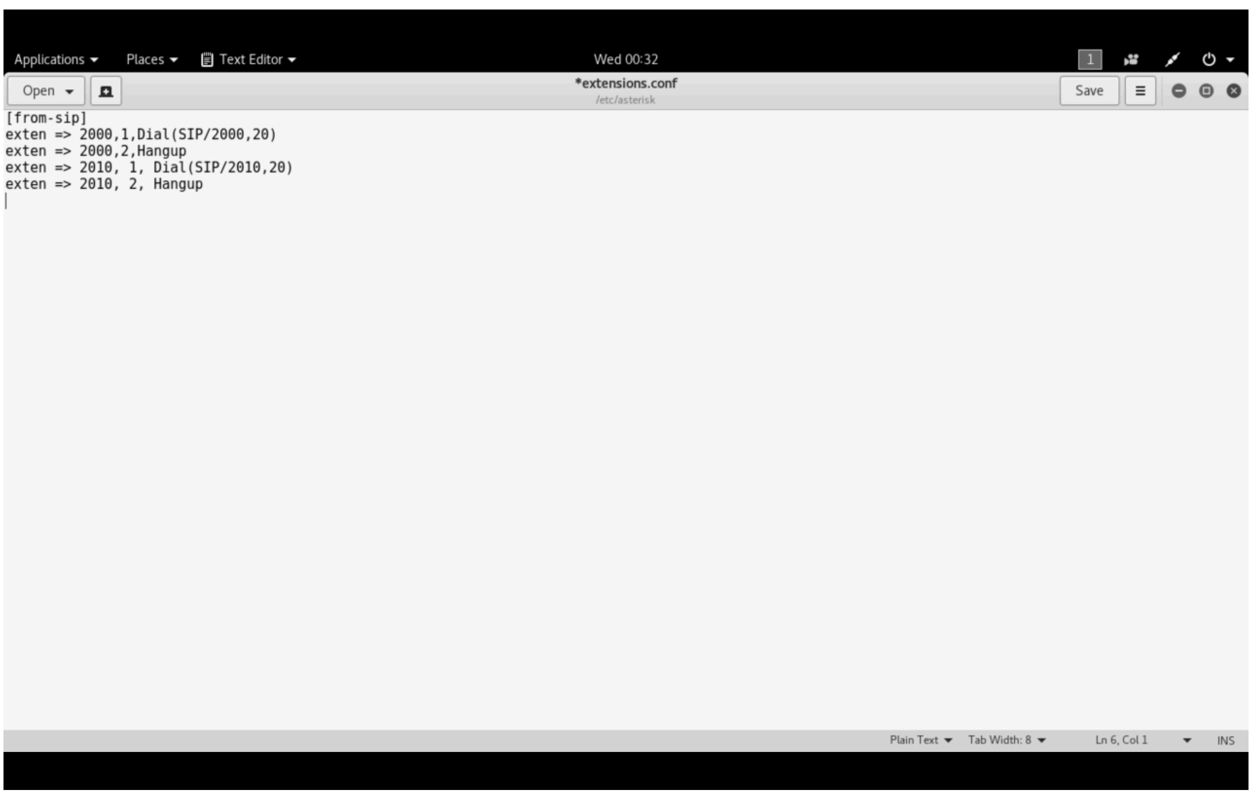


```
[ general] port = 5060
bindaddr= 0.0.0.0
allow = ulaw
allow = all
qualify= yes
context=from-sip

[2000]
username=2000
type=friend
secret=password
host=dynamic
context=from-sip

[2010]
username=2010
type=friend
secret=password
host=dynamic
context=from-sip
```

Figure : 1.1.1



```
[from-sip]
exten => 2000,1,Dial(SIP/2000,20)
exten => 2000,2,Hangup
exten => 2010, 1, Dial(SIP/2010,20)
exten => 2010, 2, Hangup
```

Figure : 1.1.2

The below figure shows registration details of a Exlite SIP account for client 2010

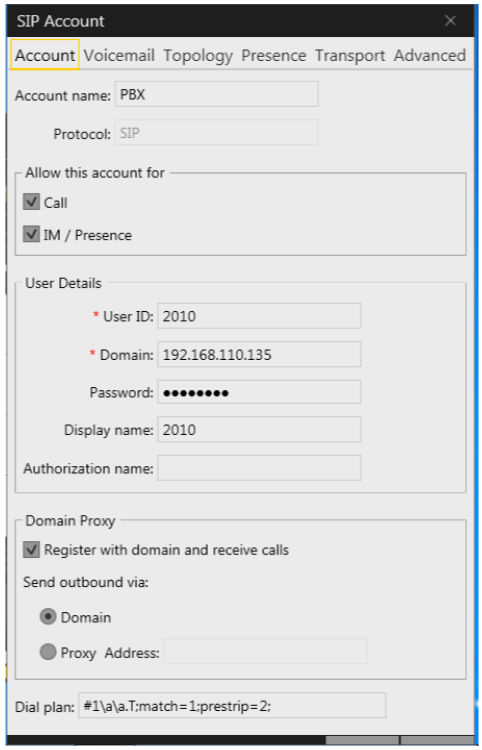


Figure : 1.1.3

This screenshot shows caller 2000 calling 2010

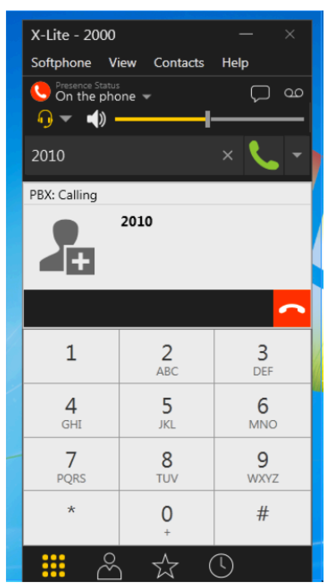


Figure : 1.1.4

2010 online:

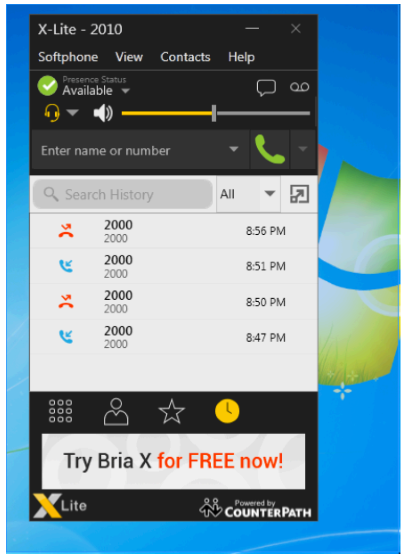


Figure : 1.1.5

Incoming call at 2010:

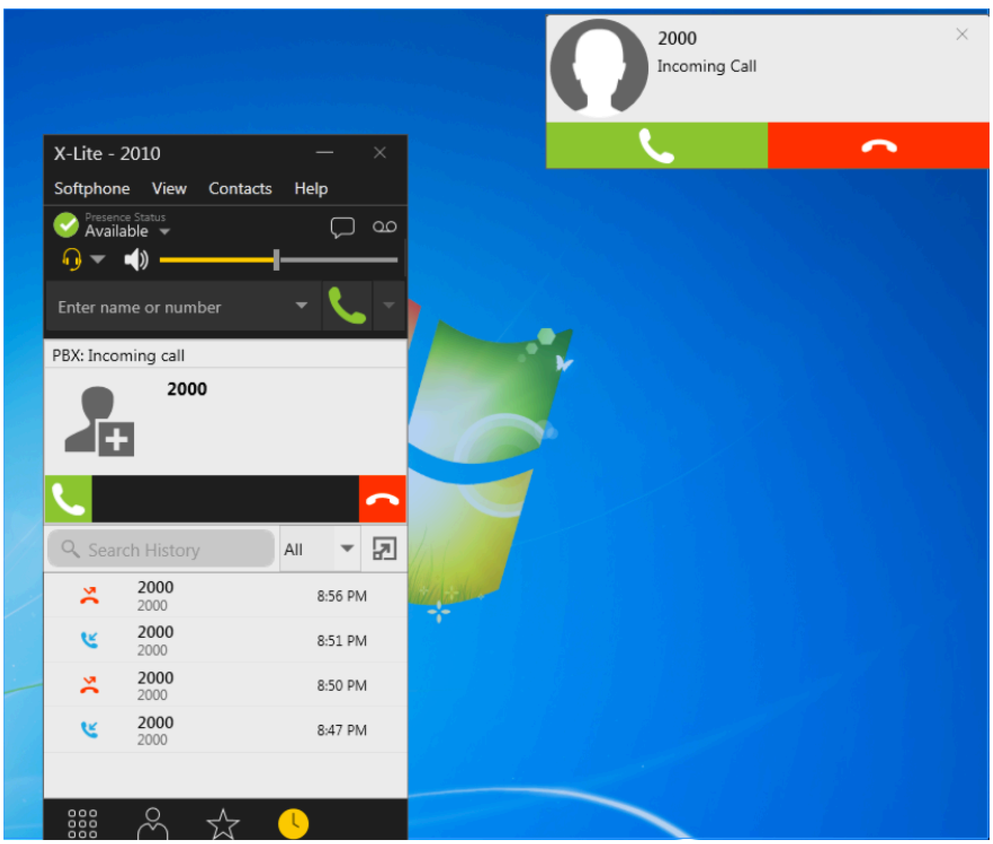


Figure : 1.1.6

Call established between 2000 and 2010:

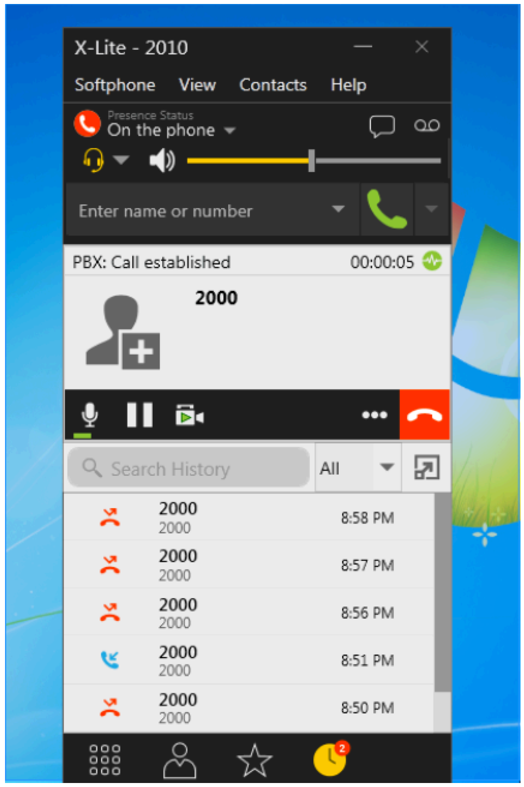


Figure : 1.1.7

Various events at asterisk server:

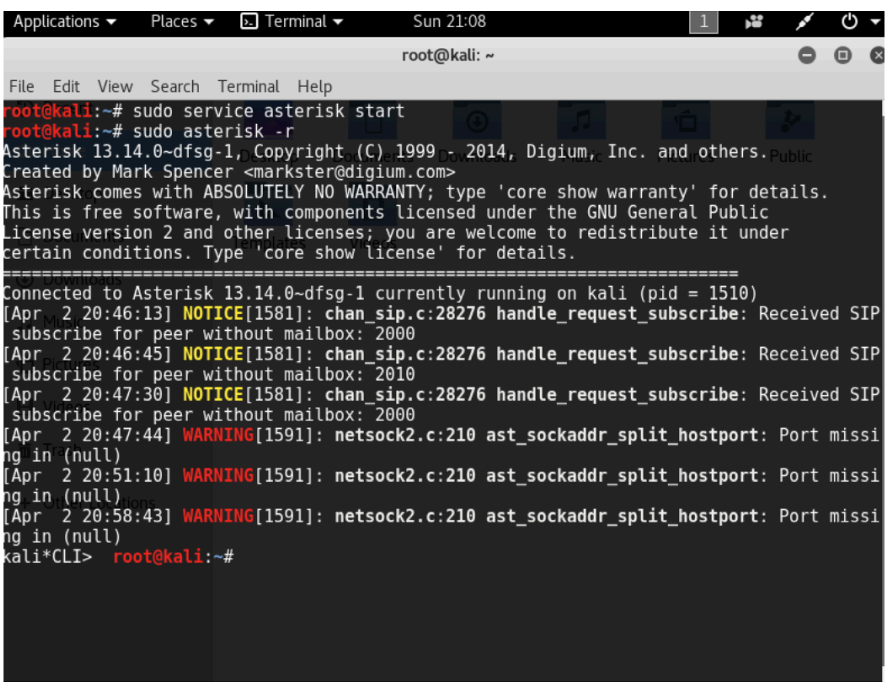


Figure : 1.1.8

Wireshark Capture during phase 1 with SIP filter applied:

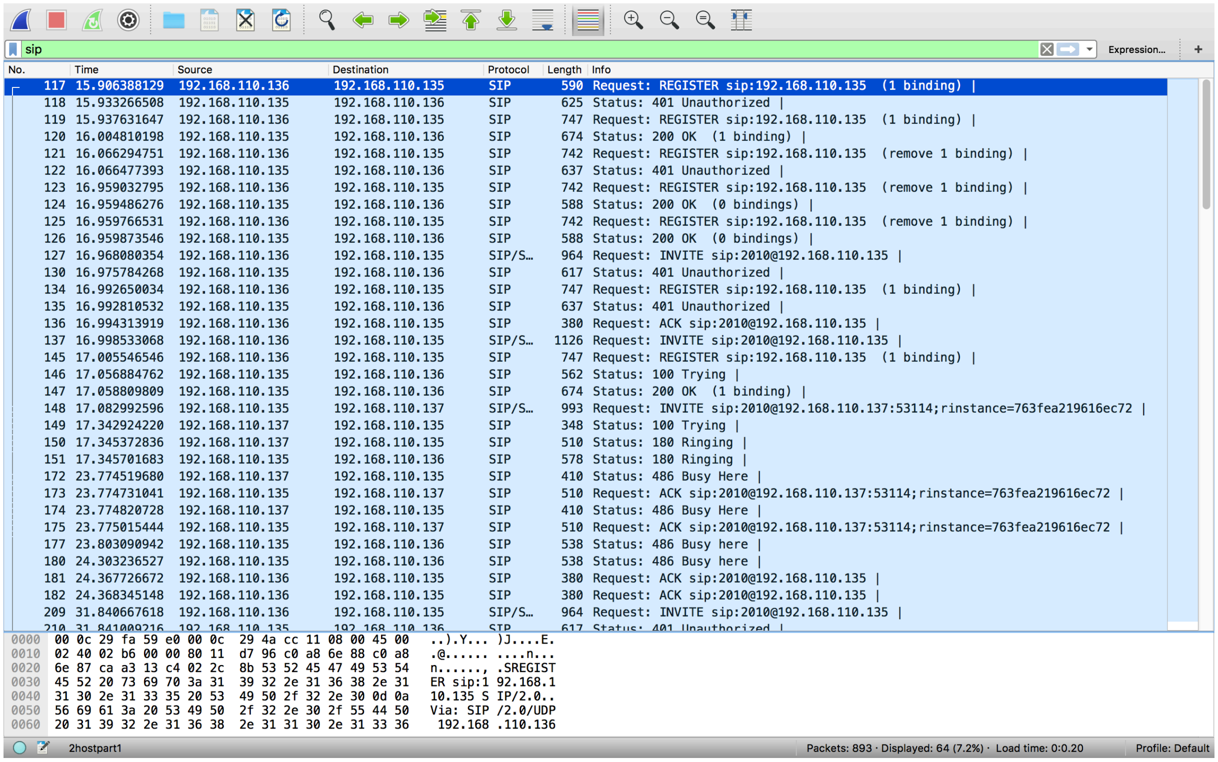


Figure : 1.1.9

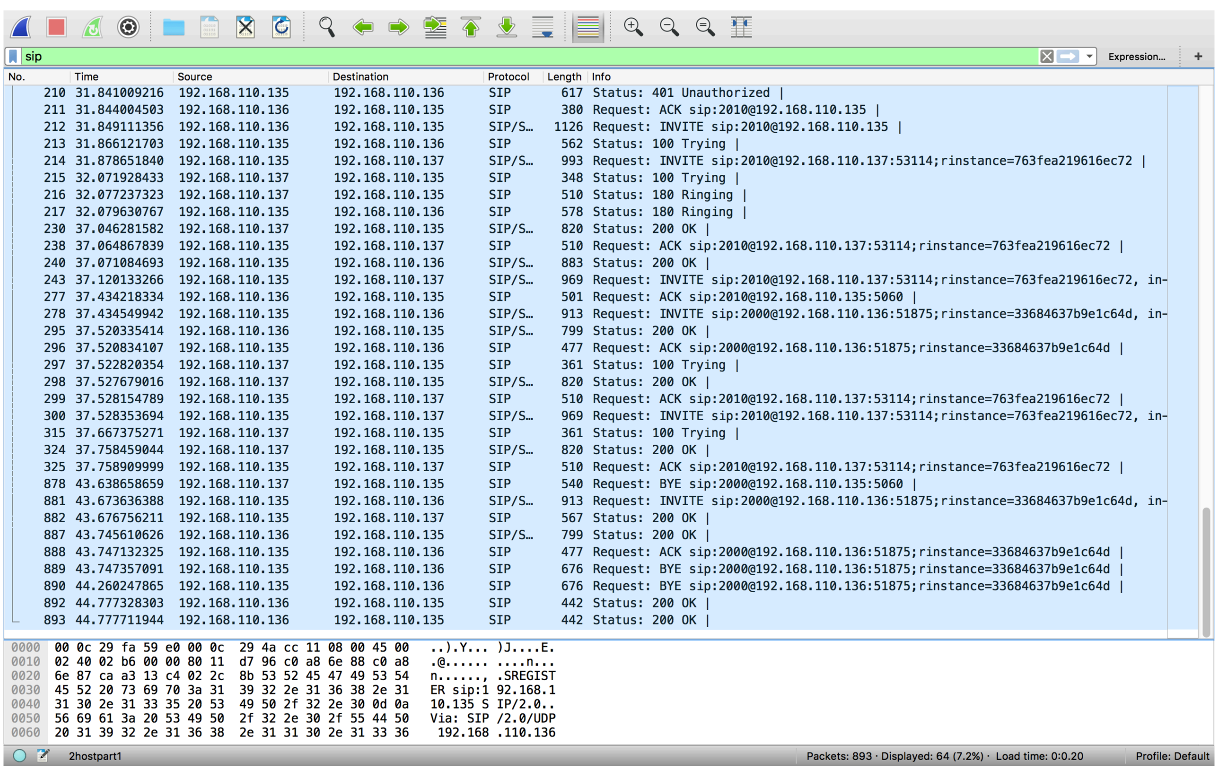


Figure : 1.1.10

Timing Diagrams of SIP packets:

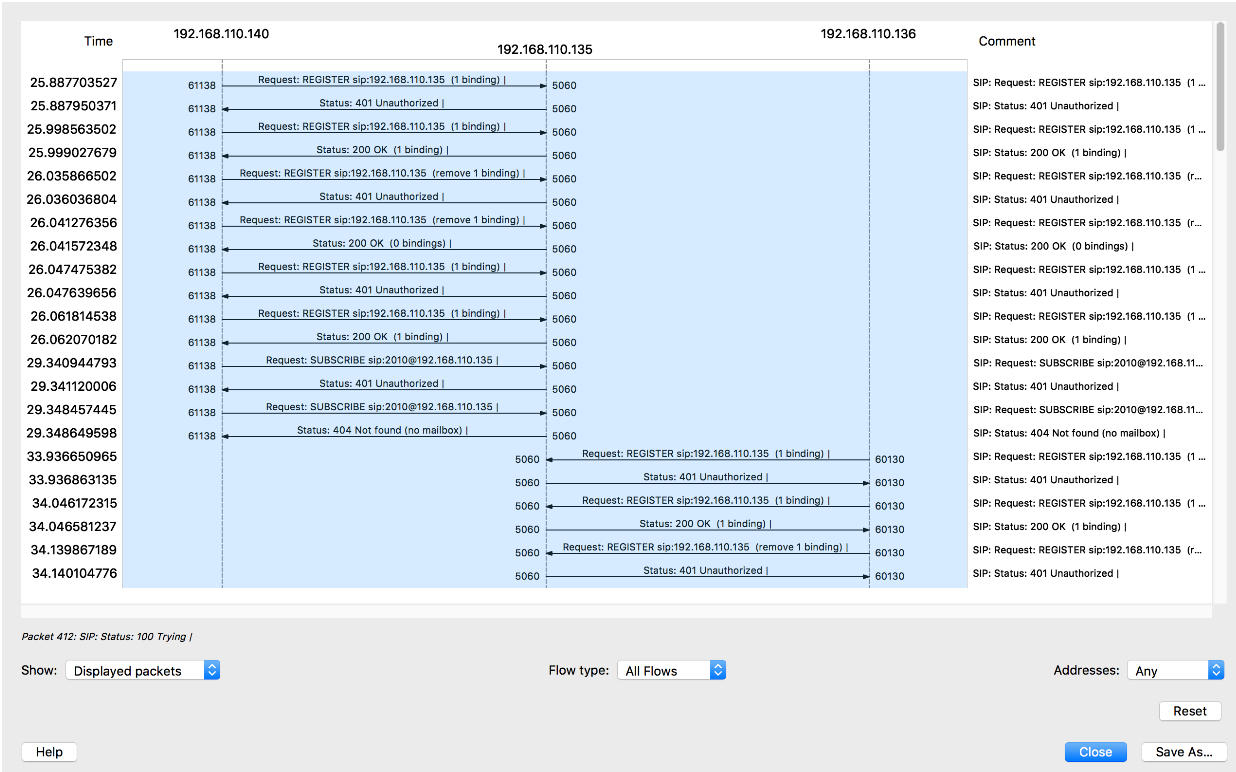


Figure : 1.1.11

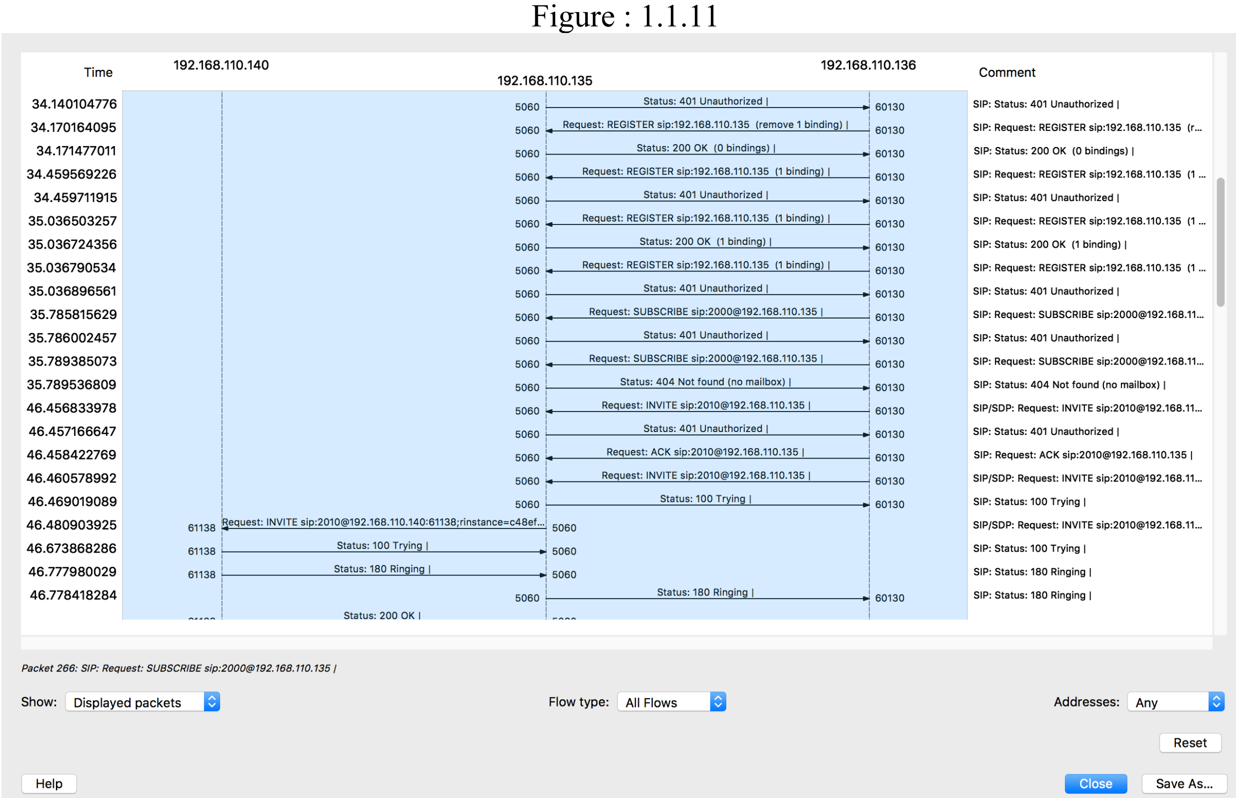


Figure : 1.1.12

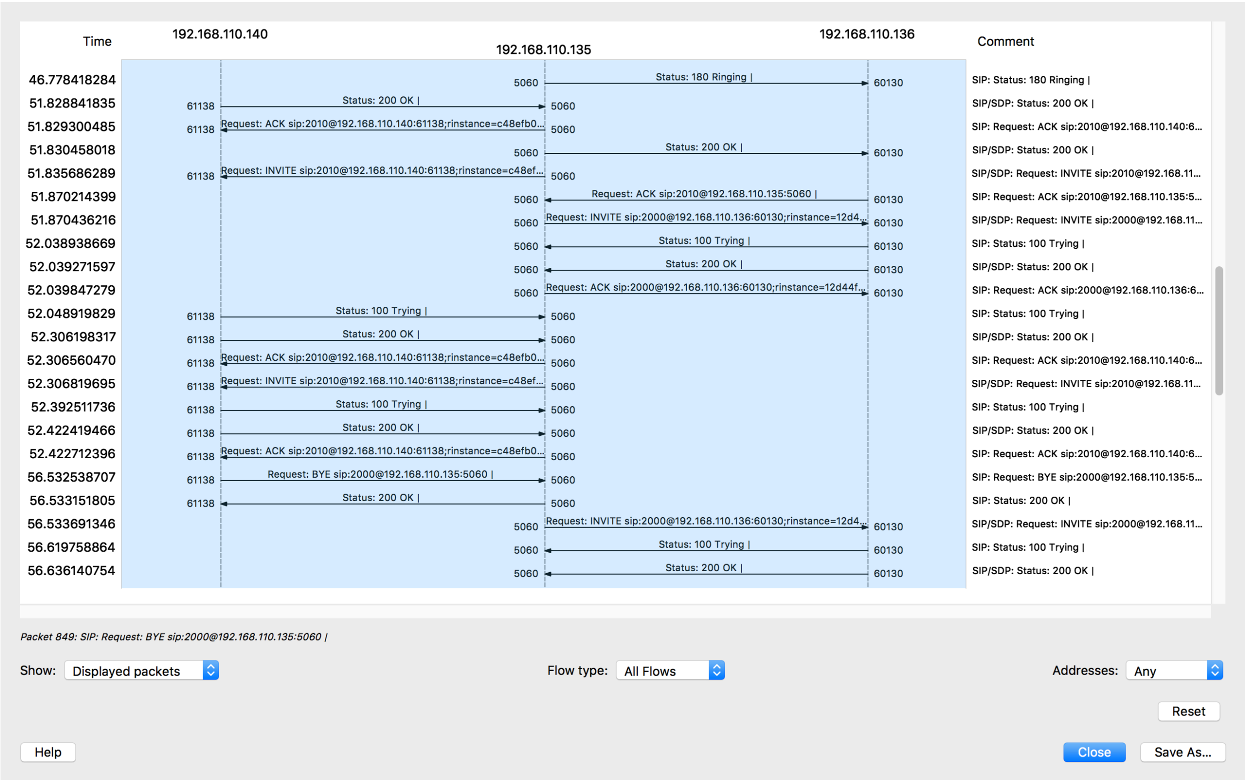


Figure : 1.1.13

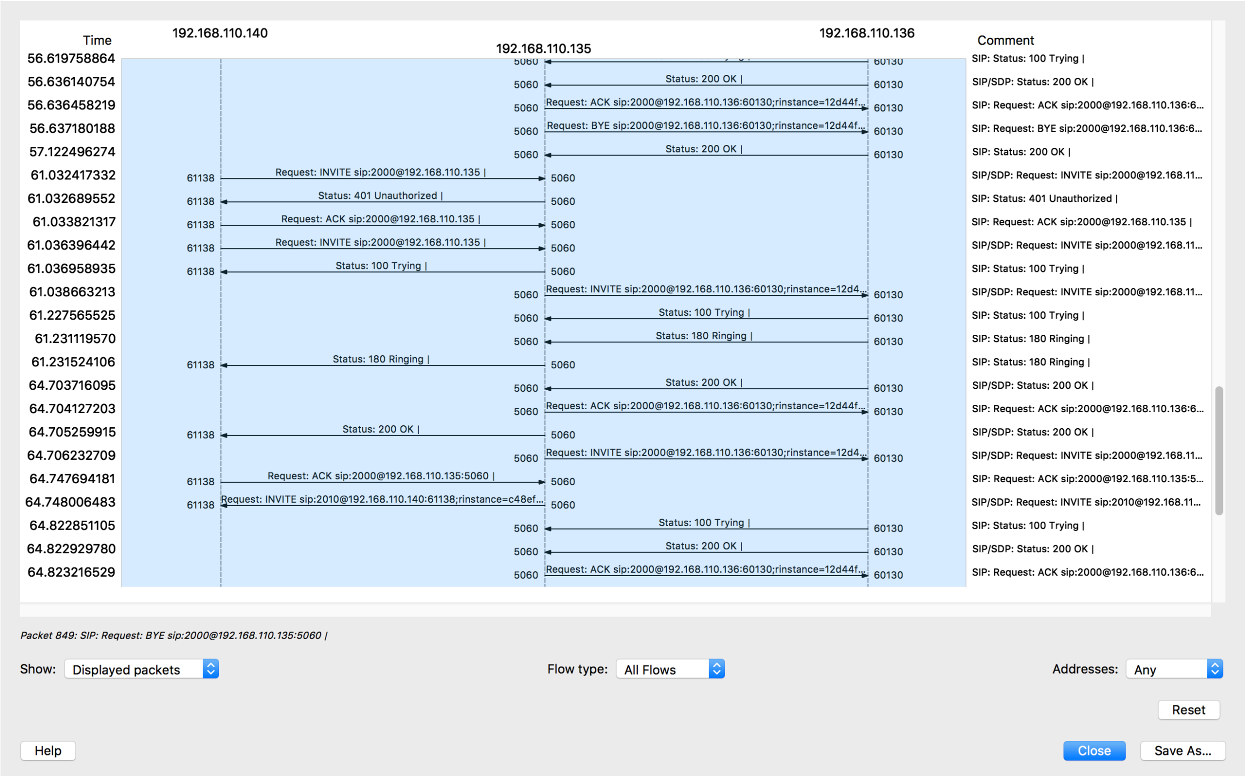


Figure : 1.1.14

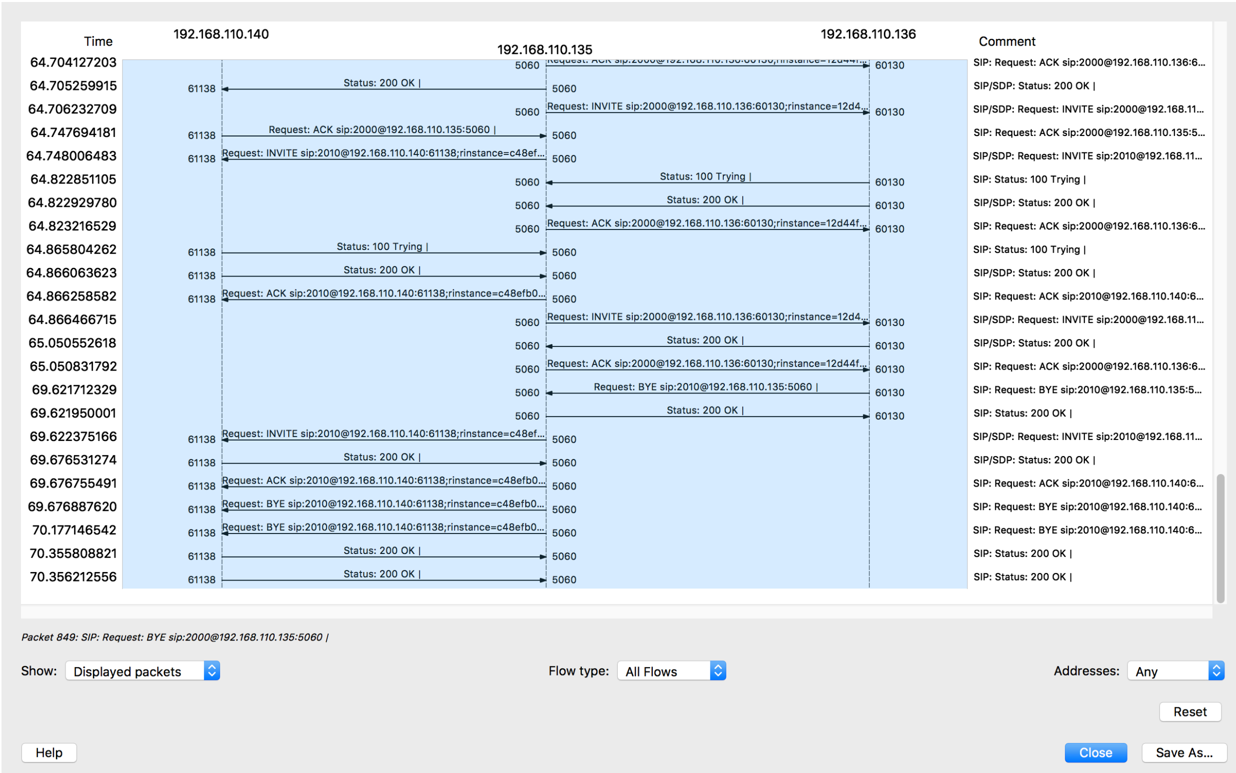
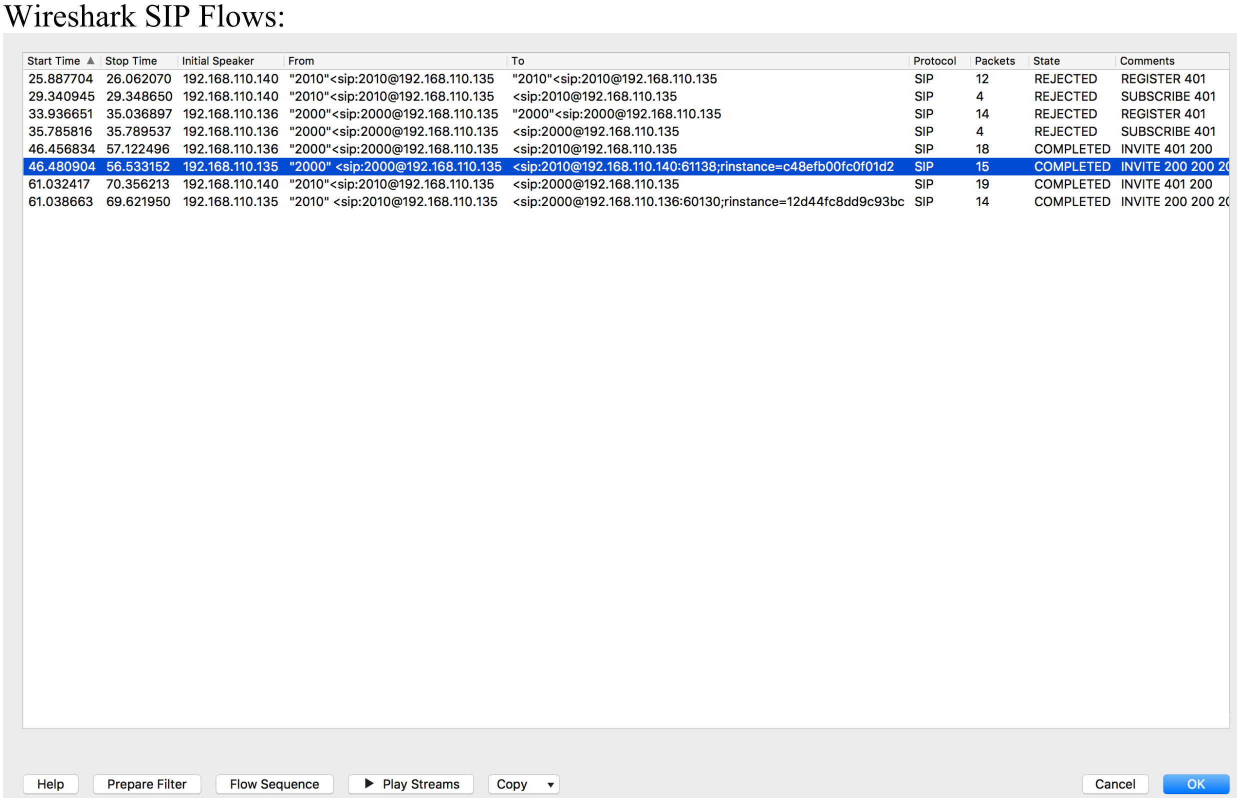


Figure : 1.1.15

Wireshark SIP Flows:



Start Time ▲	Stop Time	Initial Speaker	From	To	Protocol	Packets	State	Comments
25.887704	26.062070	192.168.110.140	"2010"<sip:2010@192.168.110.135	"2010"<sip:2010@192.168.110.135	SIP	12	REJECTED	REGISTER 401
29.340945	29.348650	192.168.110.140	"2010"<sip:2010@192.168.110.135	<sip:2010@192.168.110.135	SIP	4	REJECTED	SUBSCRIBE 401
33.936651	35.036897	192.168.110.136	"2000"<sip:2000@192.168.110.135	"2000"<sip:2000@192.168.110.135	SIP	14	REJECTED	REGISTER 401
35.785816	35.789537	192.168.110.136	"2000"<sip:2000@192.168.110.135	<sip:2000@192.168.110.135	SIP	4	REJECTED	SUBSCRIBE 401
46.456834	57.122496	192.168.110.136	"2000"<sip:2000@192.168.110.135	<sip:2010@192.168.110.135	SIP	18	COMPLETED	INVITE 401 200
46.480904	56.533152	192.168.110.135	"2000" <sip:2000@192.168.110.135	<sip:2010@192.168.110.140:61138;rinstance=c48efb00fc0f01d2	SIP	15	COMPLETED	INVITE 200 200 20
61.032417	70.356213	192.168.110.140	"2010"<sip:2010@192.168.110.135	<sip:2000@192.168.110.135	SIP	19	COMPLETED	INVITE 401 200
61.038663	69.621950	192.168.110.135	"2010" <sip:2010@192.168.110.135	<sip:2000@192.168.110.136:60130;rinstance=12d44fc8dd9c93bc	SIP	14	COMPLETED	INVITE 200 200 20

Figure : 1.1.16

The below are logical flow charts of our interest:

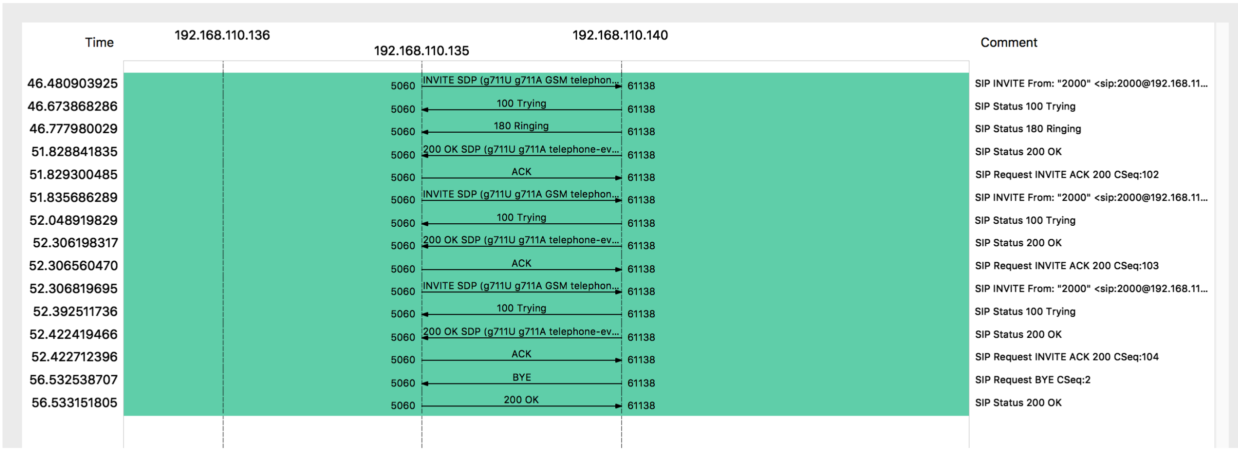


Figure : 1.1.17

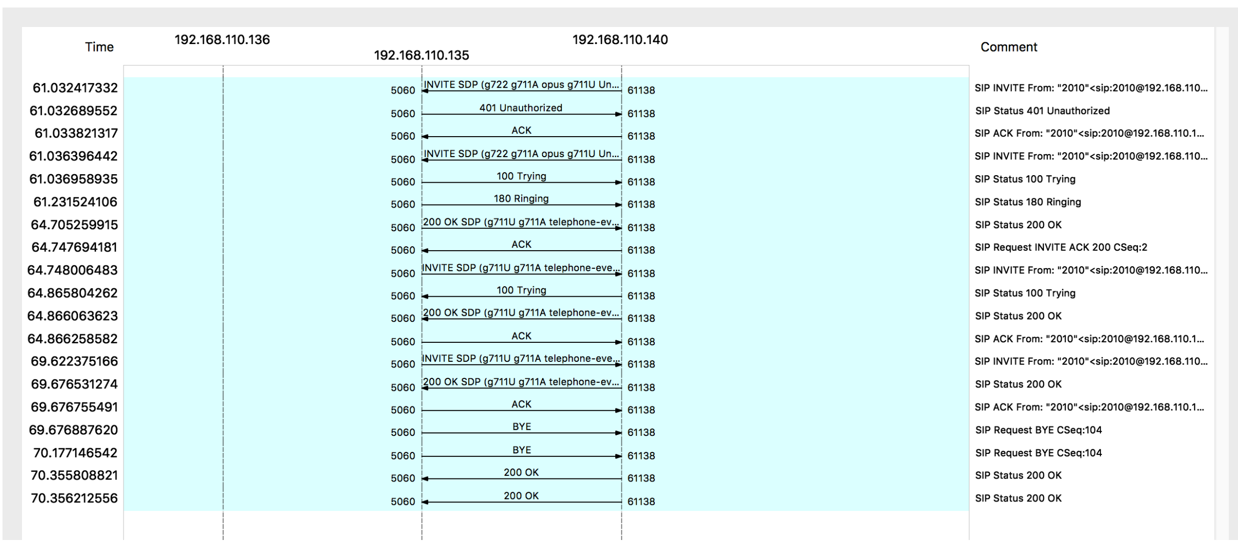


Figure : 1.1.18

The logical flow shows asterisk server notifying user 2000 about call request in the very first line of figure 1.1.17. Then 2010 sends Trying Ringing and OK to which the server acknowledges completing the SIP 3 way handshake. It is also shown that it uses G.711 codec and figure:1.1.18 shows the user 2010 joining the call by sending invite.

PHASE 2: Busy User:

The below is modified extensions.conf file to make busy user with delay of 10

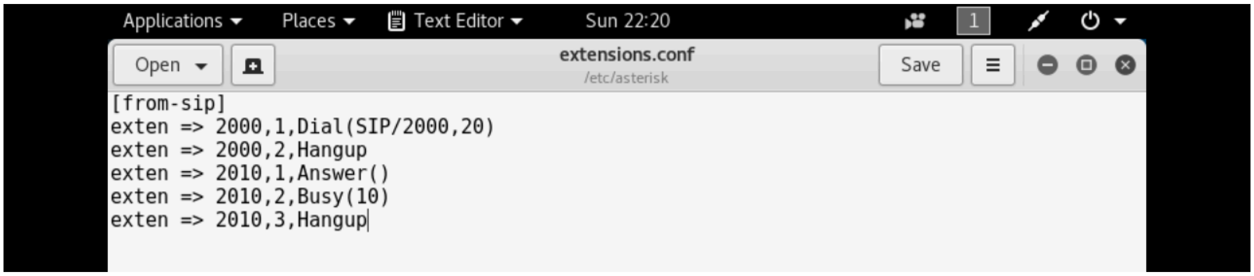


Figure : 1.2.1

The below screenshot shows 2000 calling 2010 which is a busy user and the call ends after giving the busy tone instead of connecting the call

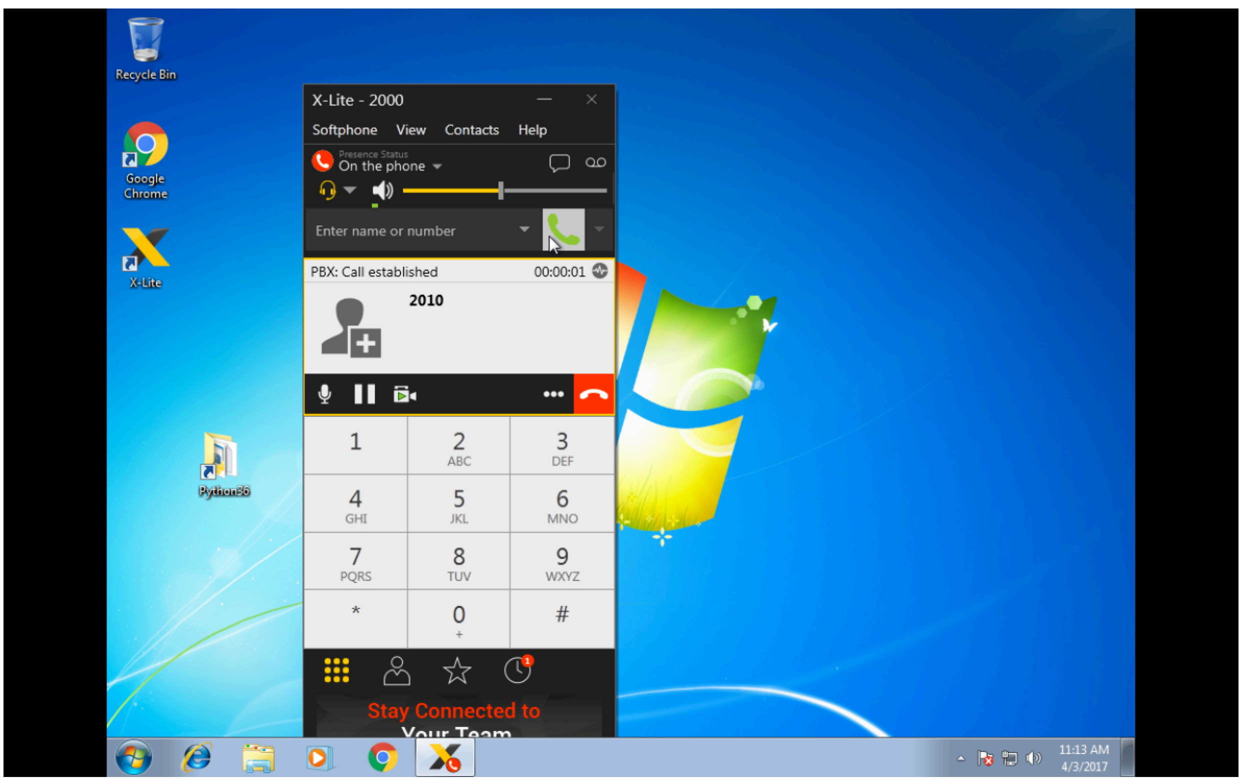


Figure : 1.2.2

The below are wireshark captures for busy user phase:

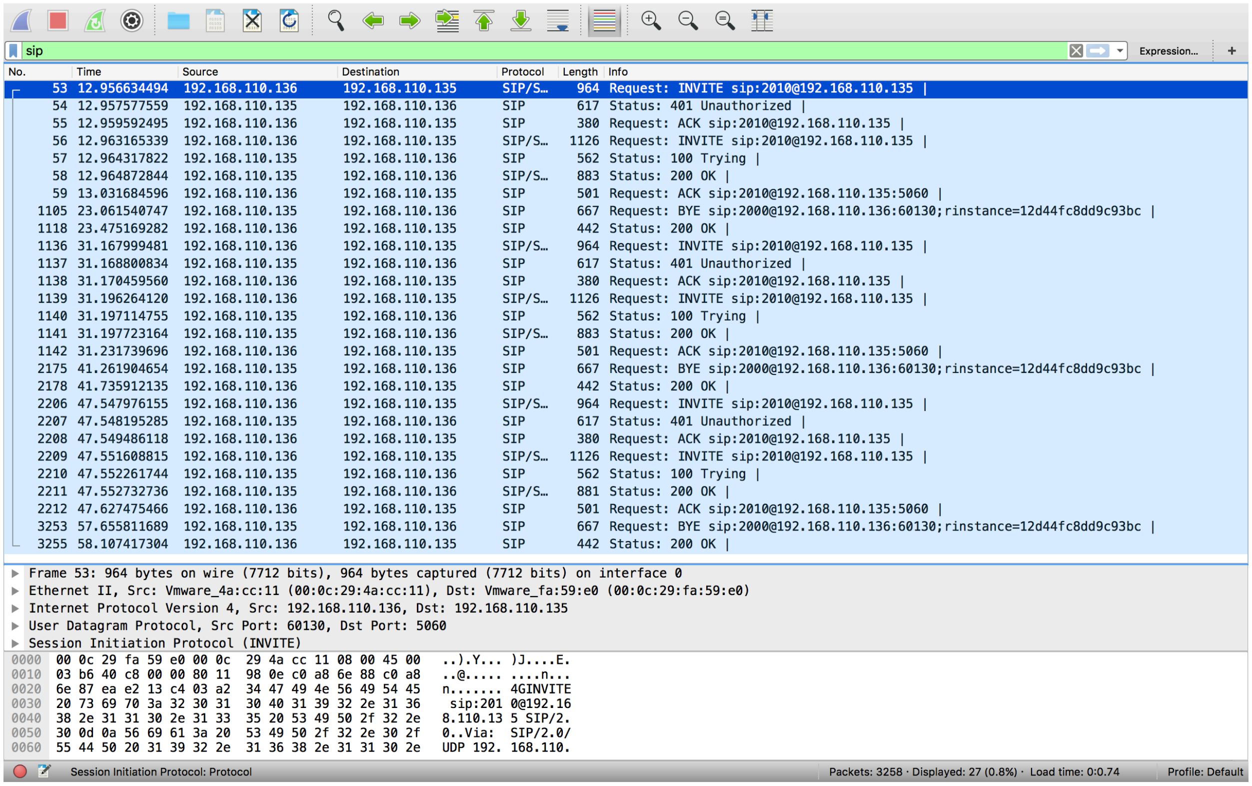


Figure : 1.2.3

Timing Diagram:

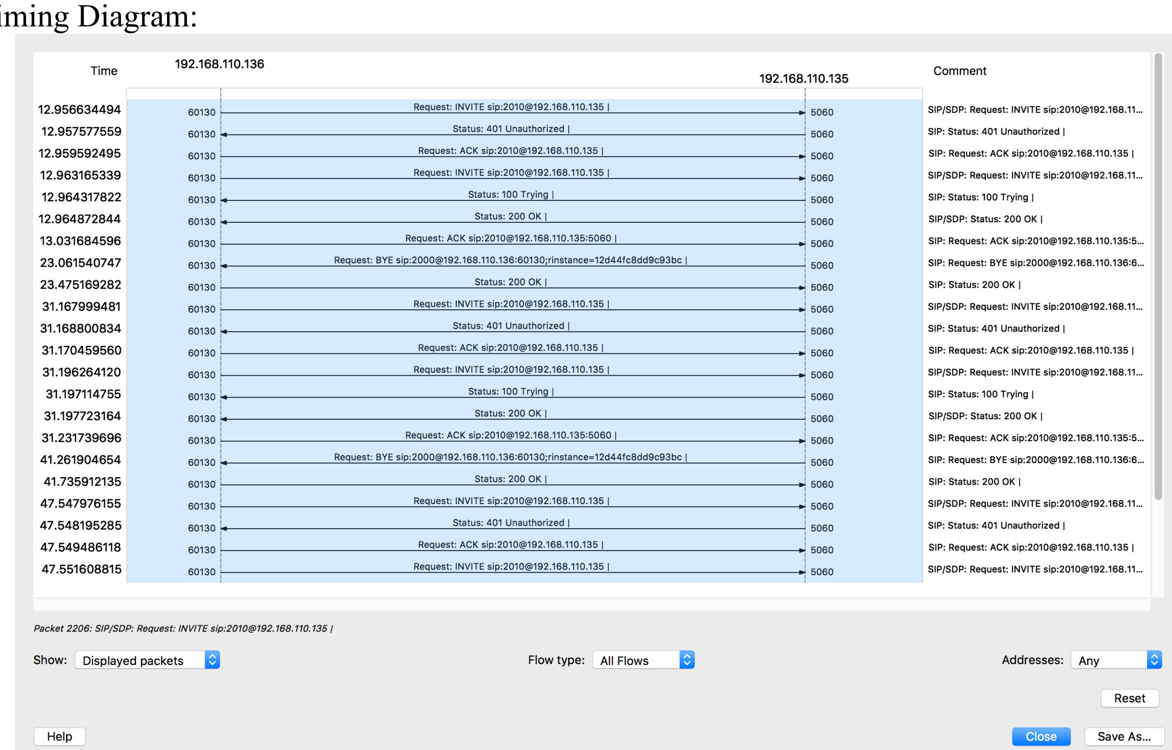


Figure : 1.2.4

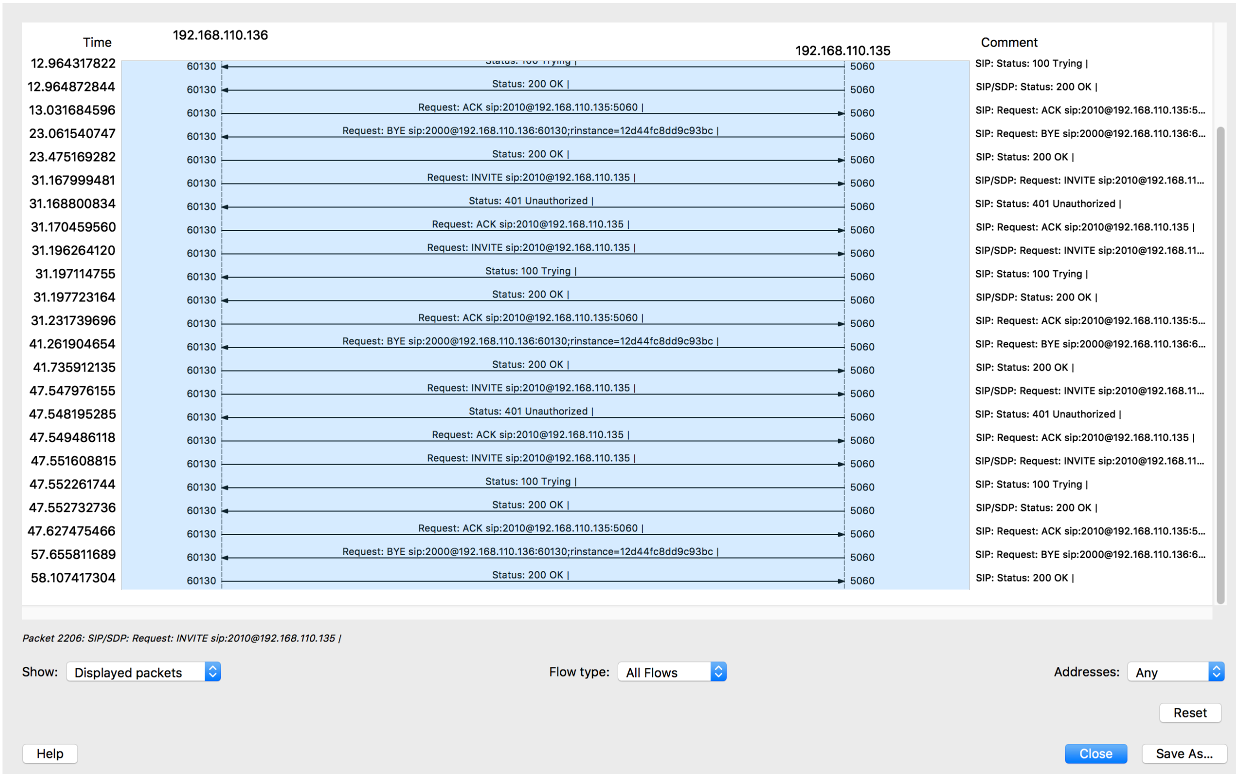


Figure : 1.2.5

SIP flows:

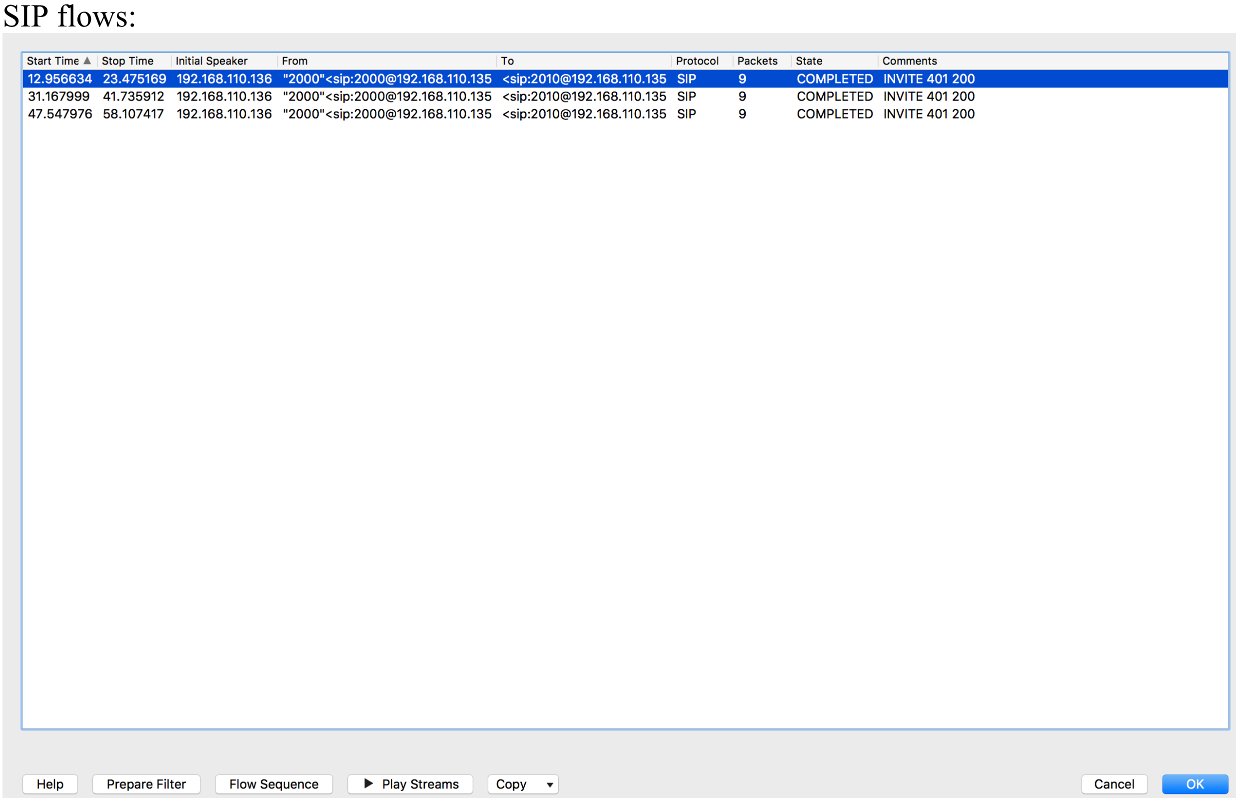


Figure : 1.2.6

User to asterisk:

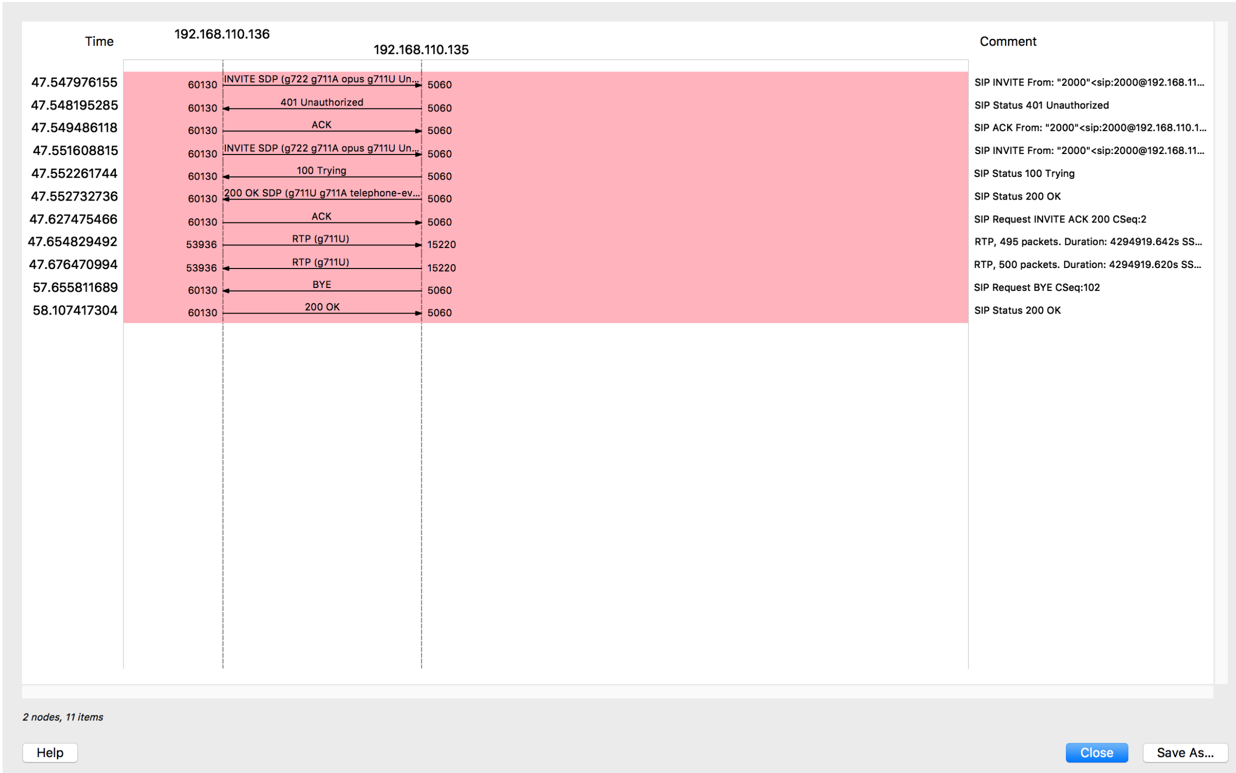


Figure : 1.2.7

It can be clearly seen in figure:1.2.7 that when user 2000 tries to call 2010 the server sends busy tone for 10secs from 47.65 and a BYE at the end at 57.65 to terminate the call according to the delay parameter specified.

PHASE 3 - Call on Hold:

Below is the modified extensions file to add third user 2009:

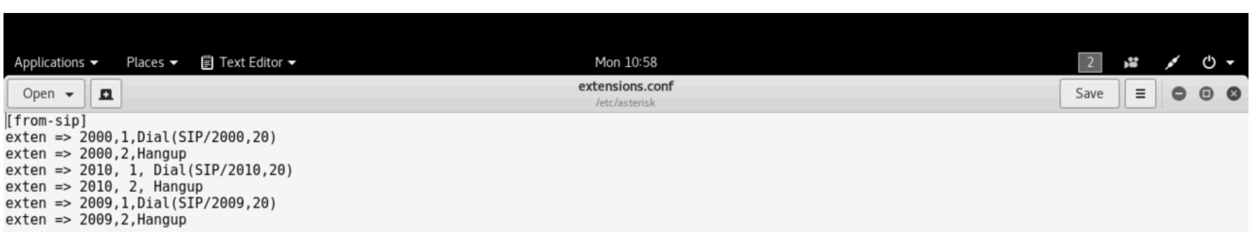


Figure : 1.3.1

The SIP.conf file with three users 2000,2010 and 2009

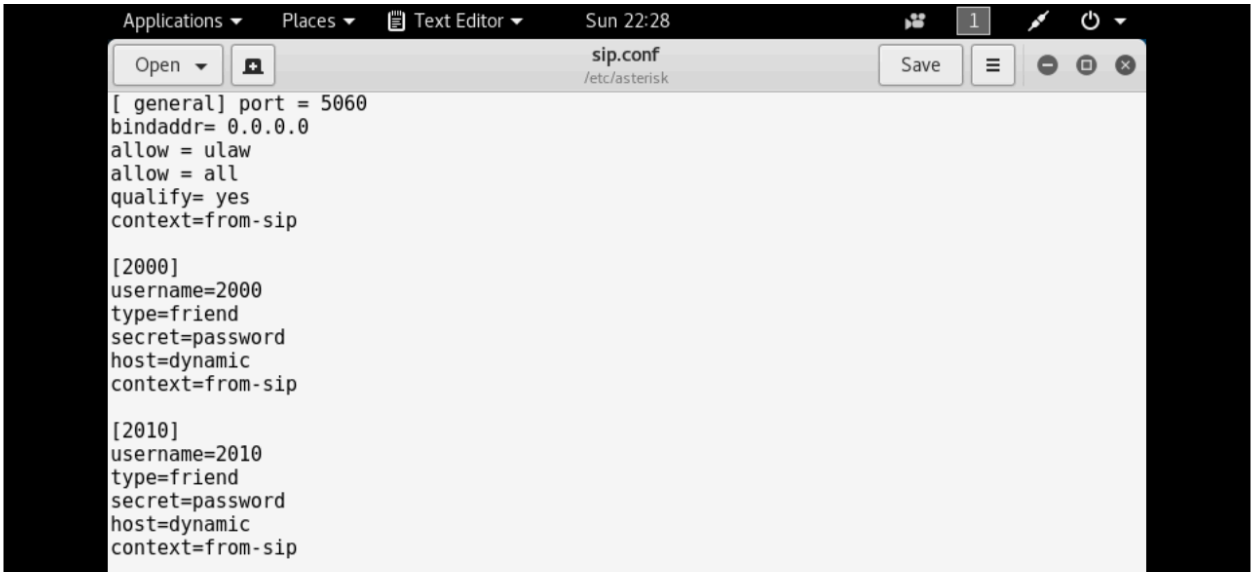


Figure : 1.3.2

The figure shows ongoing all between 2010 and 2000 at 2000's side:

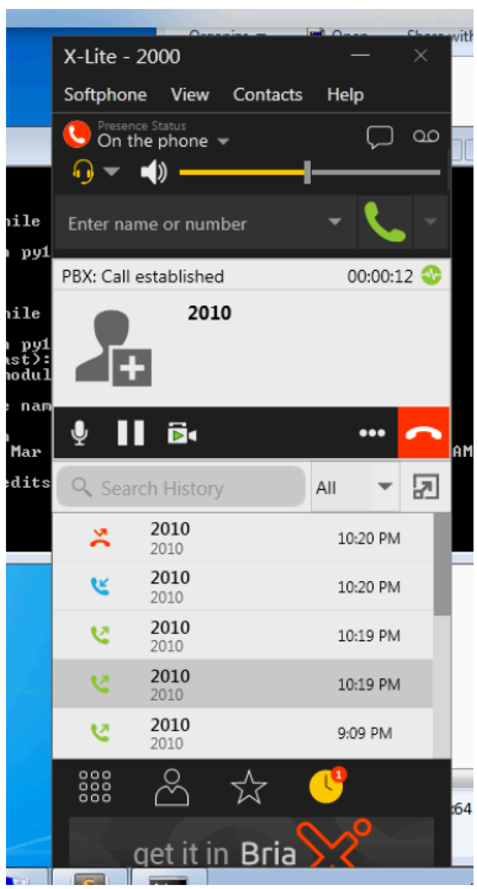


Figure : 1.3.3

2009 online:

Figure : 1.3.4

2009 calling 2000:

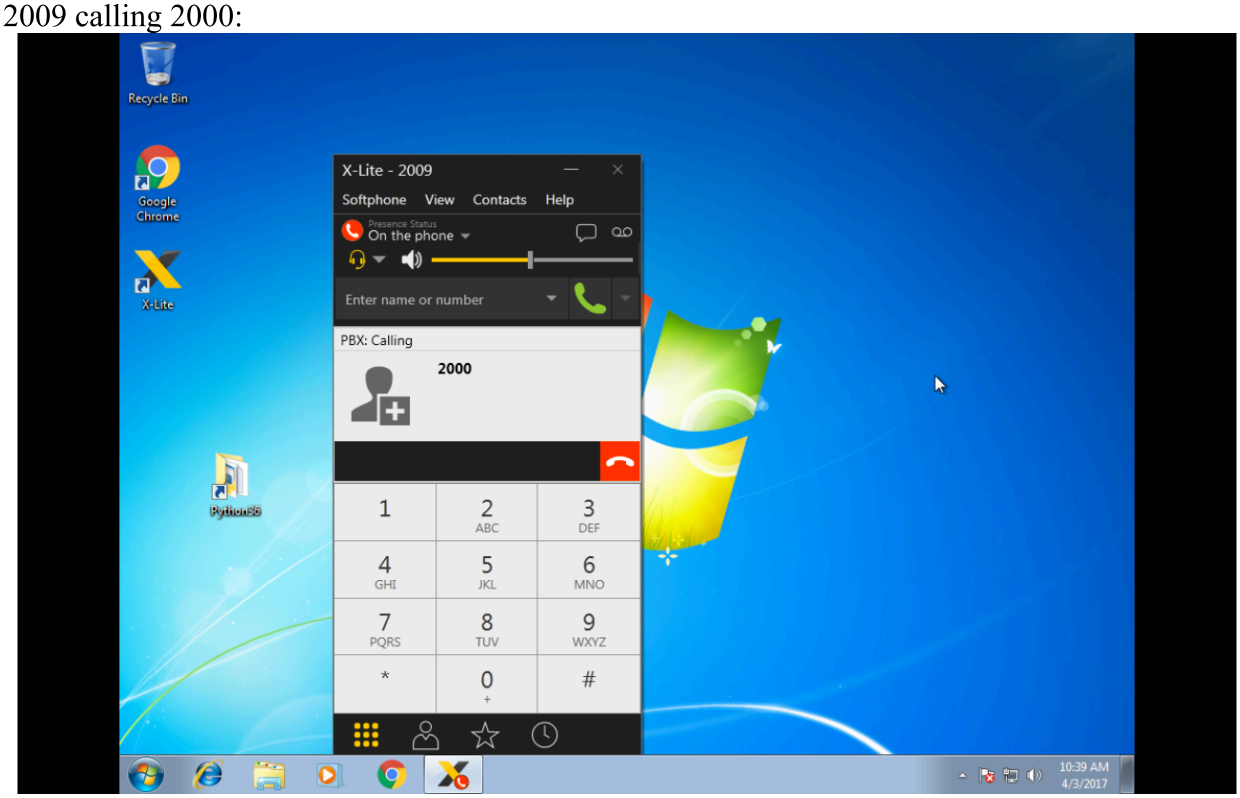


Figure : 1.3.5

2000 accepts call from 2009 while keeping 2010 on hold:

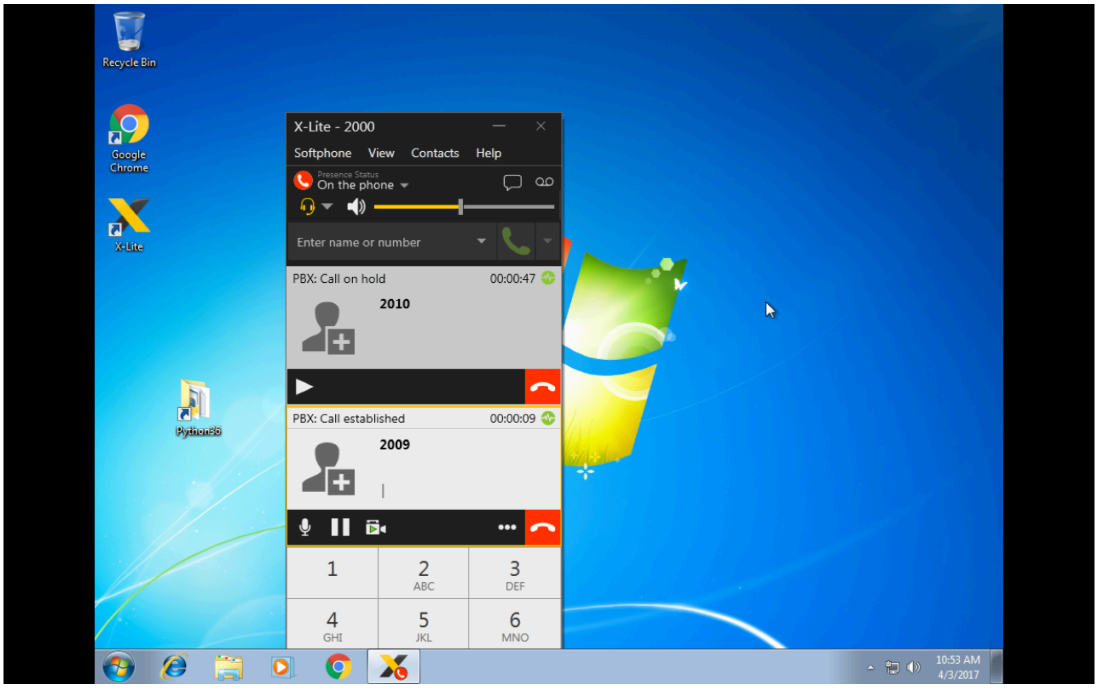


Figure : 1.3.6

2000 ends call with 2009 and 2010 still on hold:

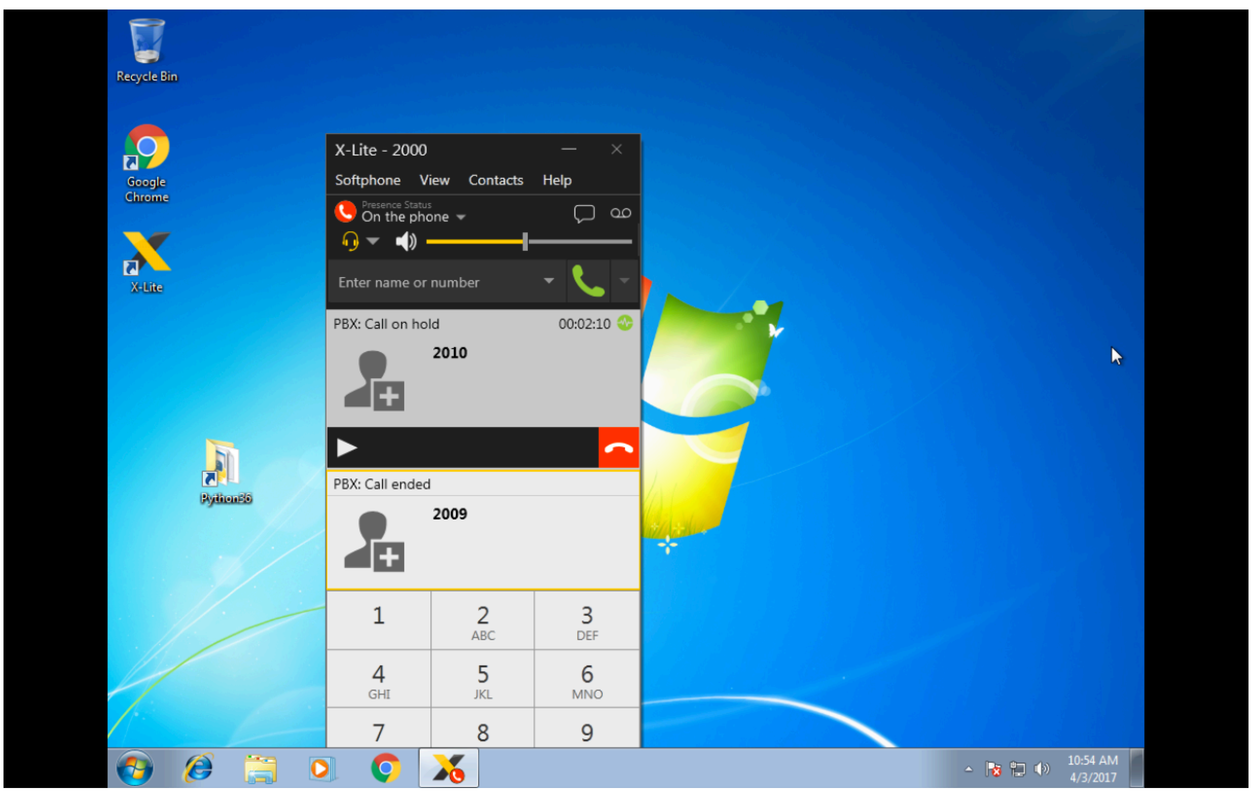


Figure : 1.3.7

2000 only has one call from 2010 on hold:

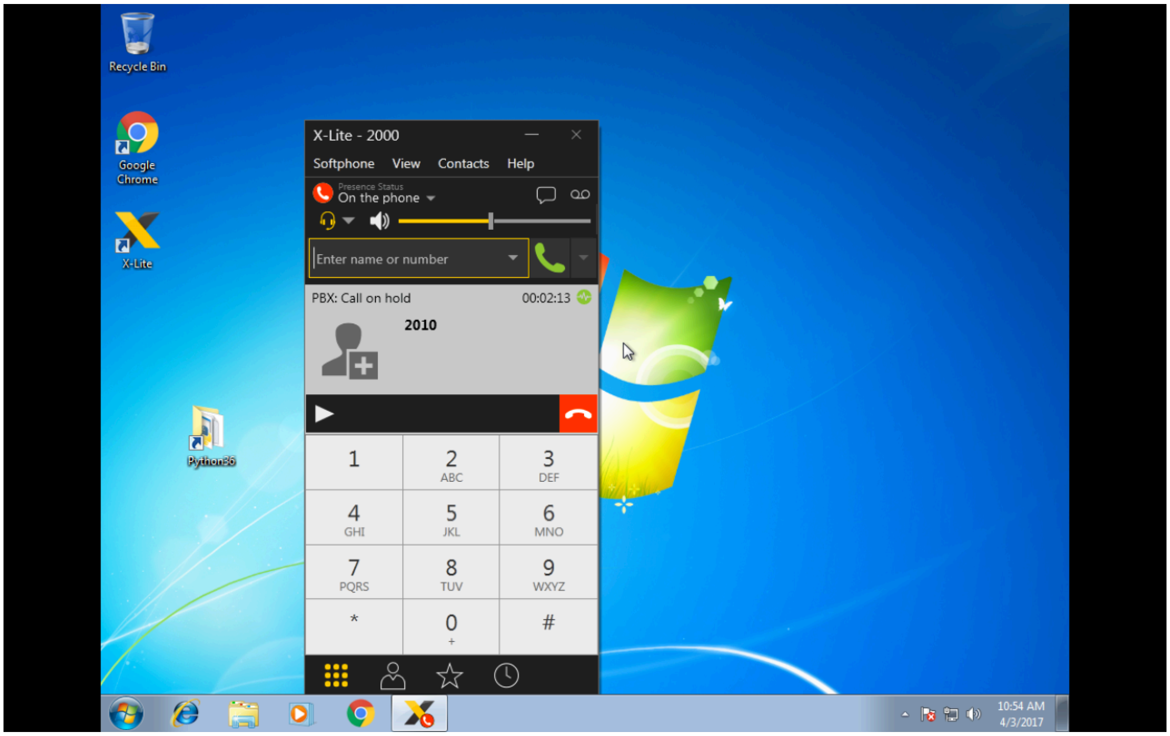


Figure : 1.3.8

2000 removes 2010 from hold:

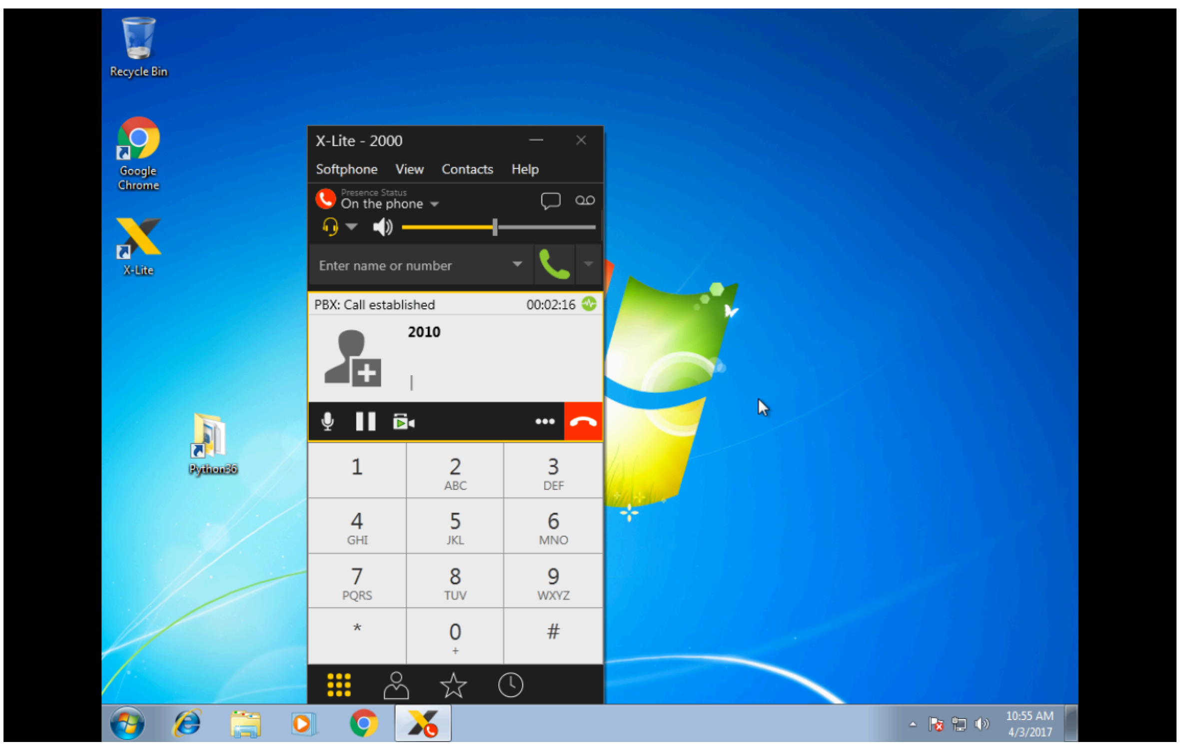


Figure : 1.3.9

Wireshark Captures:

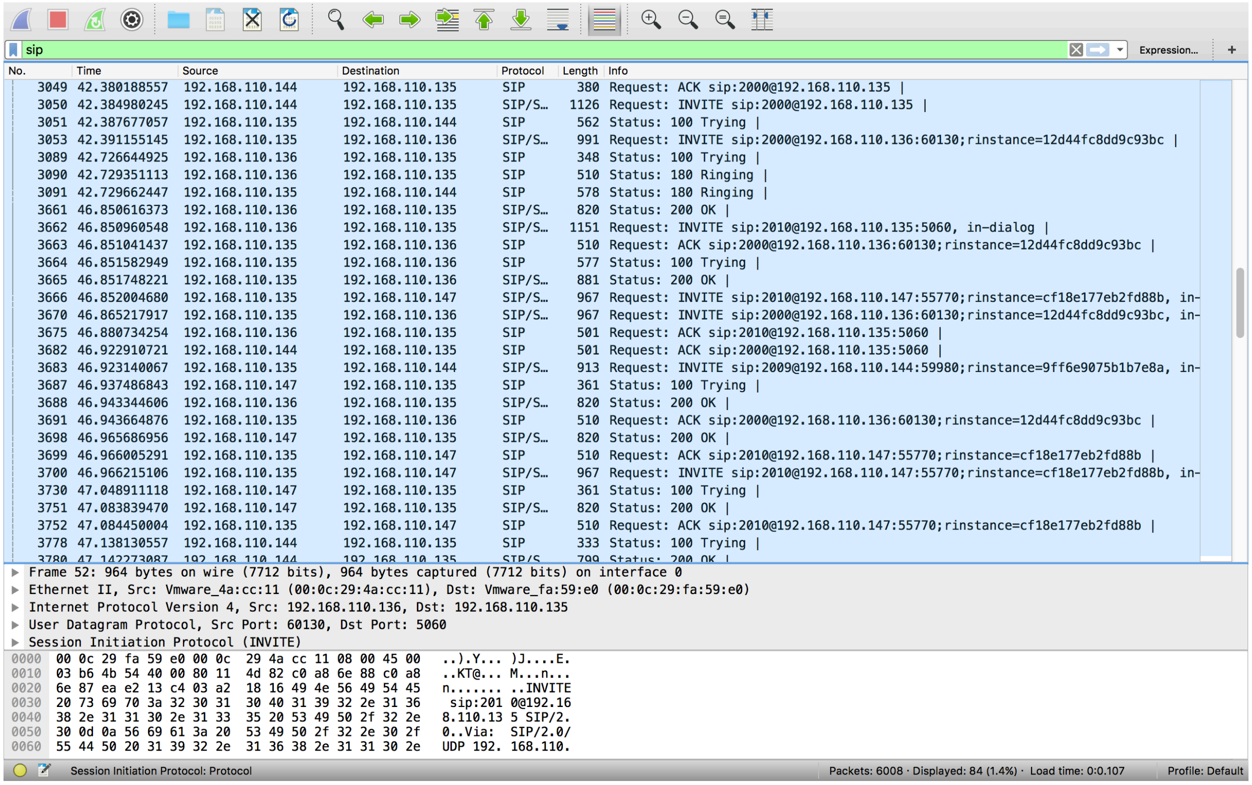


Figure : 1.3.4

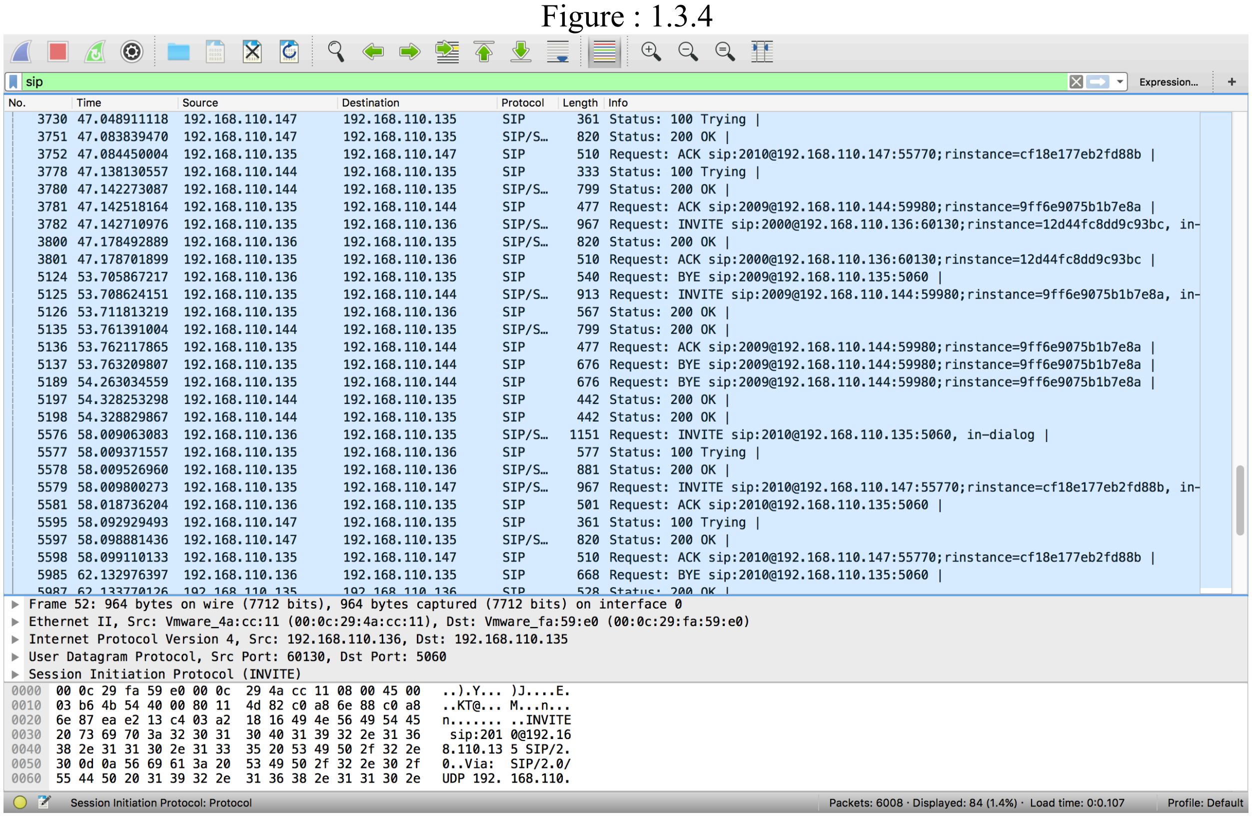


Figure : 1.3.5

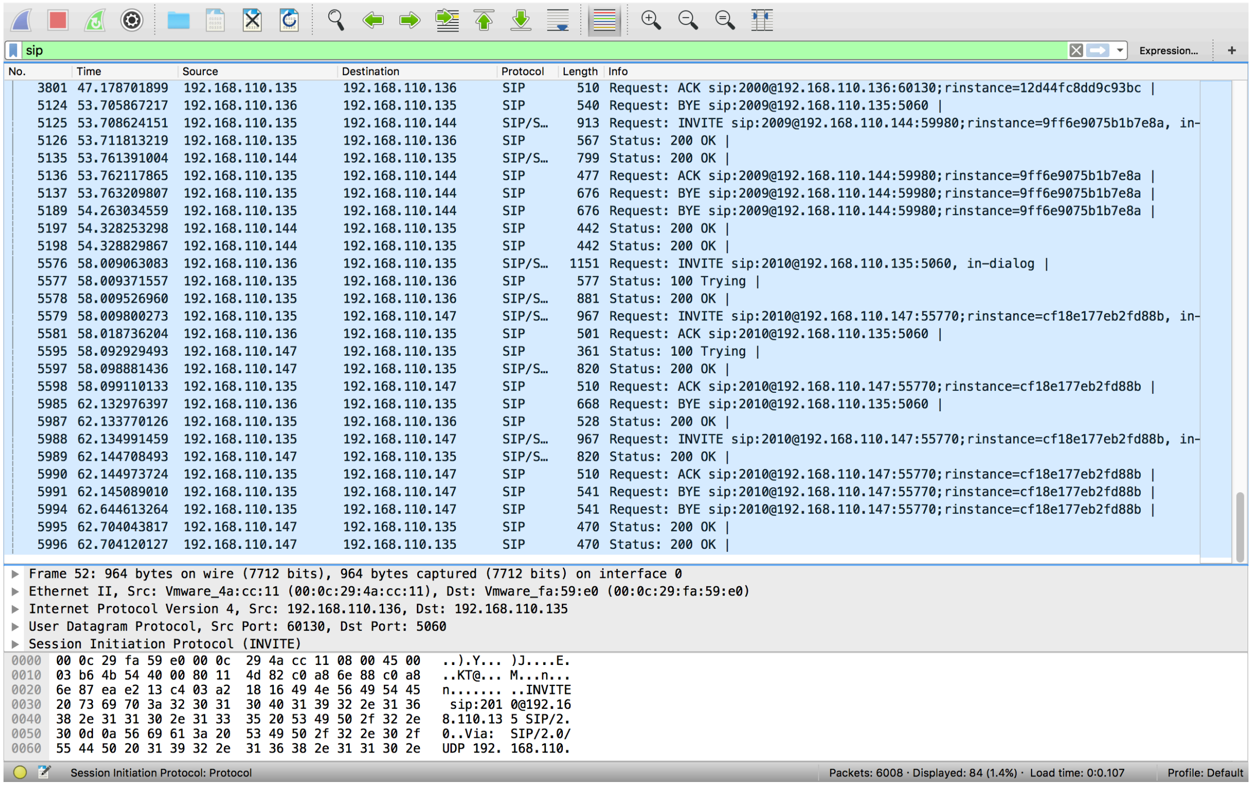


Figure : 1.3.6

Timing Diagrams:

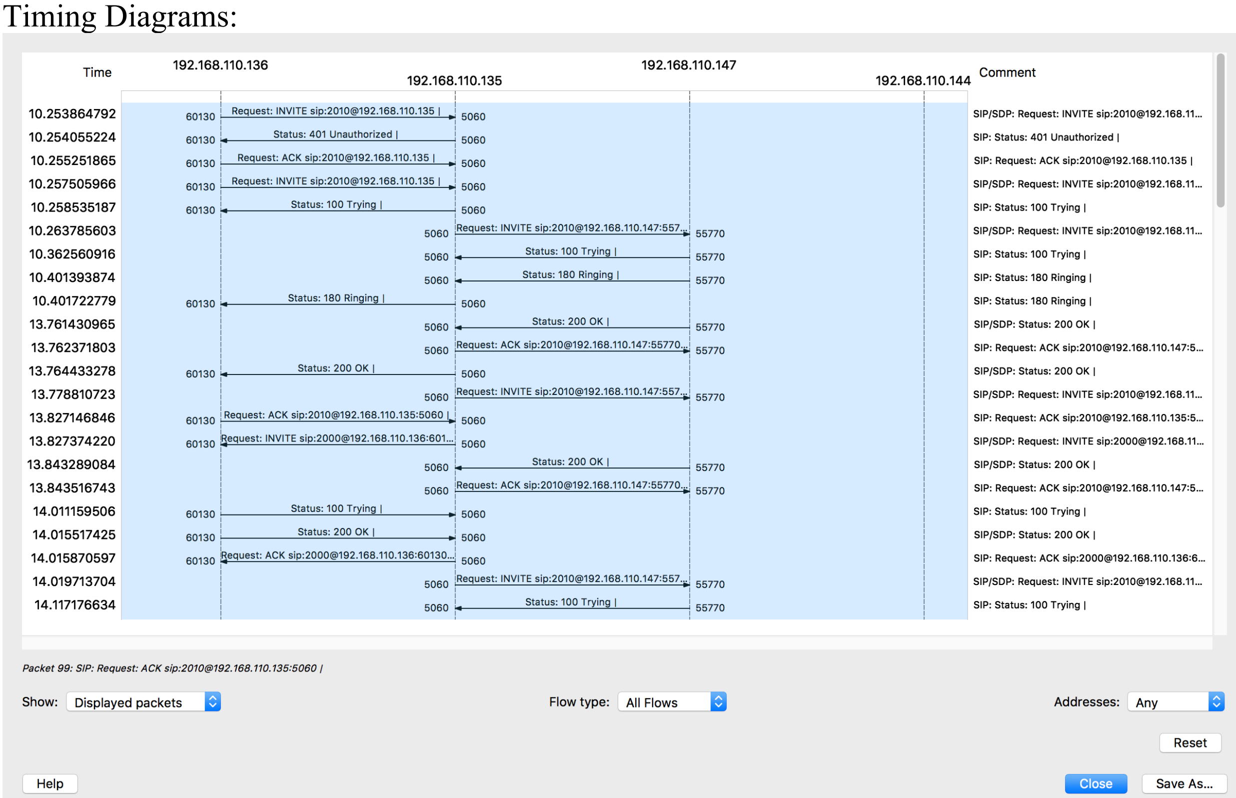


Figure : 1.3.7

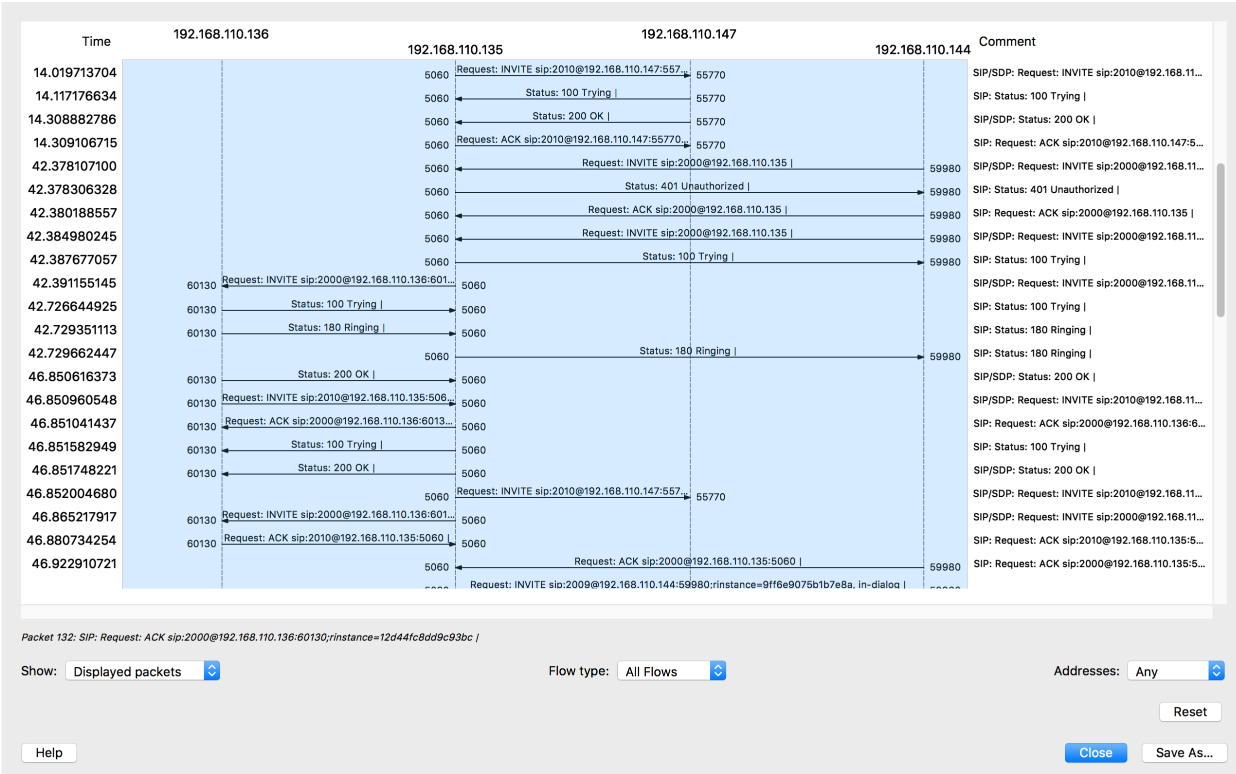


Figure : 1.3.8

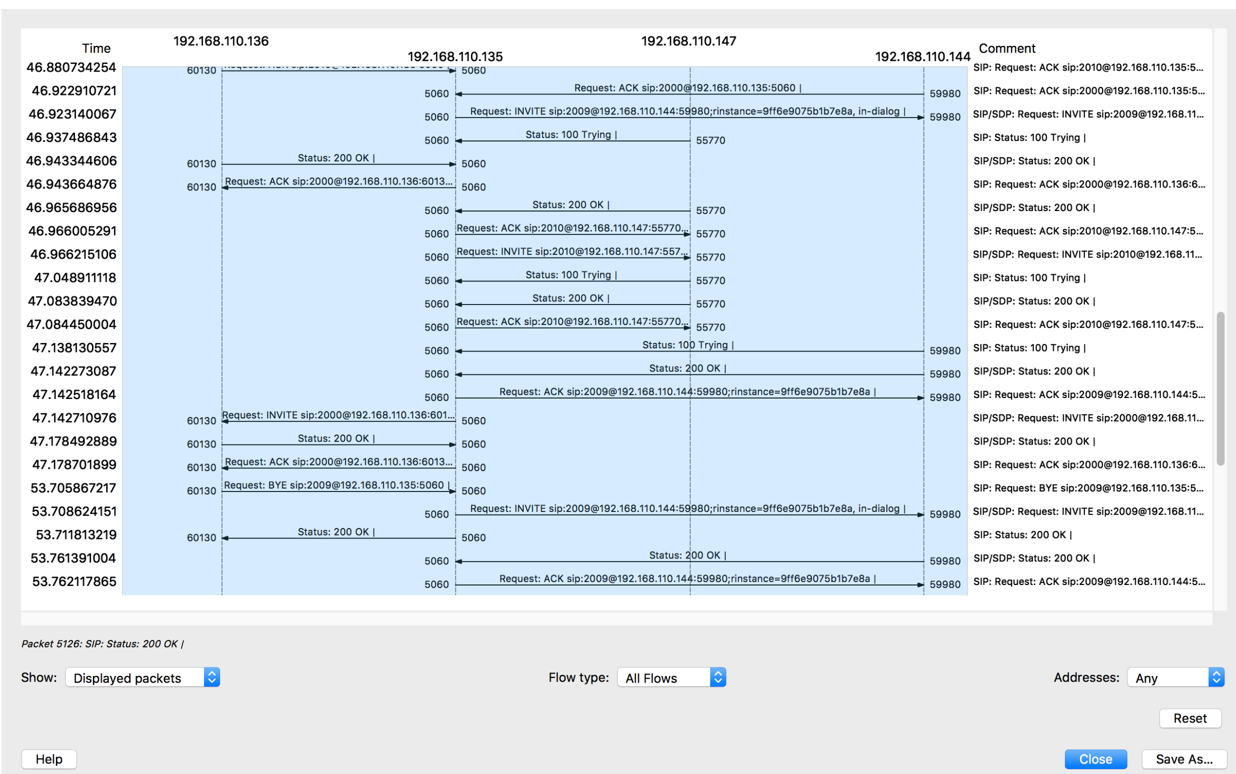


Figure : 1.3.9

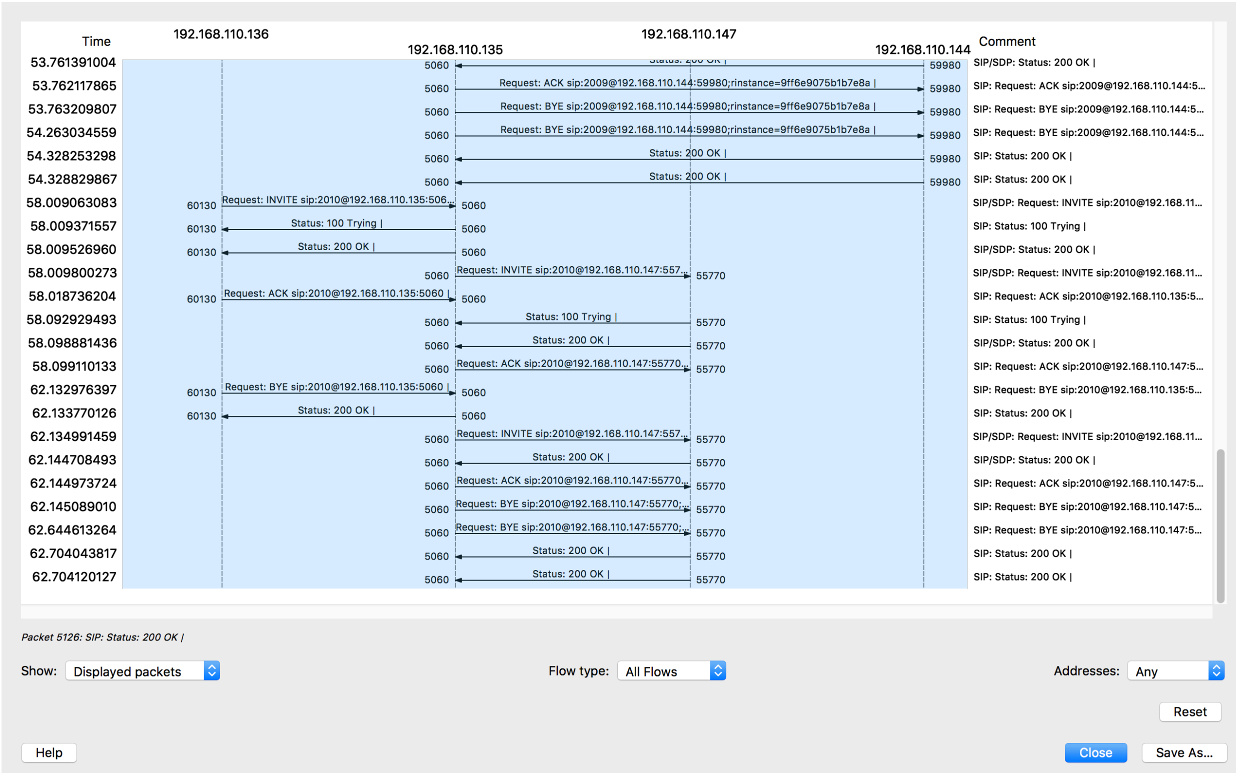
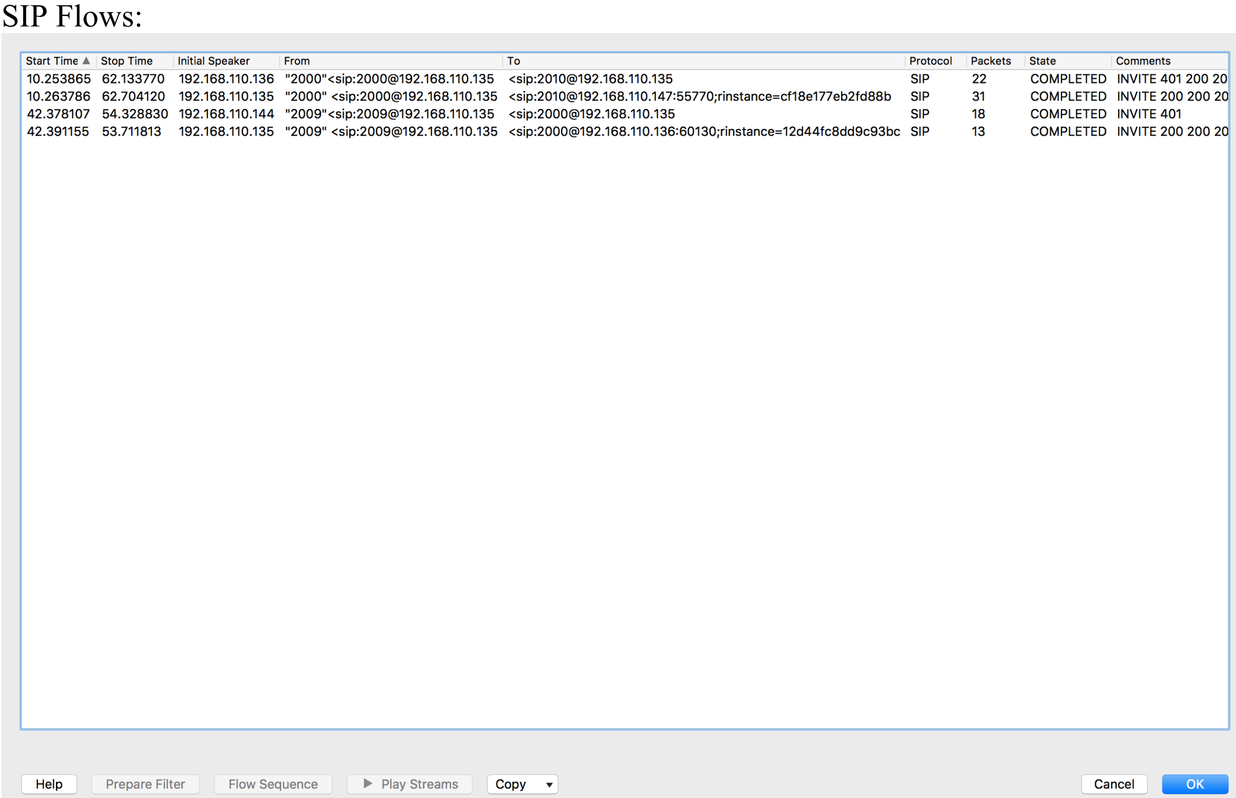


Figure : 1.3.10

SIP Flows:



Start Time ▲	Stop Time	Initial Speaker	From	To	Protocol	Packets	State	Comments
10.253865	62.133770	192.168.110.136	"2000"<sip:2000@192.168.110.135	<sip:2010@192.168.110.135	SIP	22	COMPLETED	INVITE 401 200 20
10.263786	62.704120	192.168.110.135	"2000" <sip:2000@192.168.110.135	<sip:2010@192.168.110.147:55770;rinstance=cf18e177eb2fd88b	SIP	31	COMPLETED	INVITE 200 200 20
42.378107	54.328830	192.168.110.144	"2009"<sip:2009@192.168.110.135	<sip:2000@192.168.110.135	SIP	18	COMPLETED	INVITE 401
42.391155	53.711813	192.168.110.135	"2009" <sip:2009@192.168.110.135	<sip:2000@192.168.110.136:60130;rinstance=12d44fc8dd9c93bc	SIP	13	COMPLETED	INVITE 200 200 20

Figure : 1.3.11

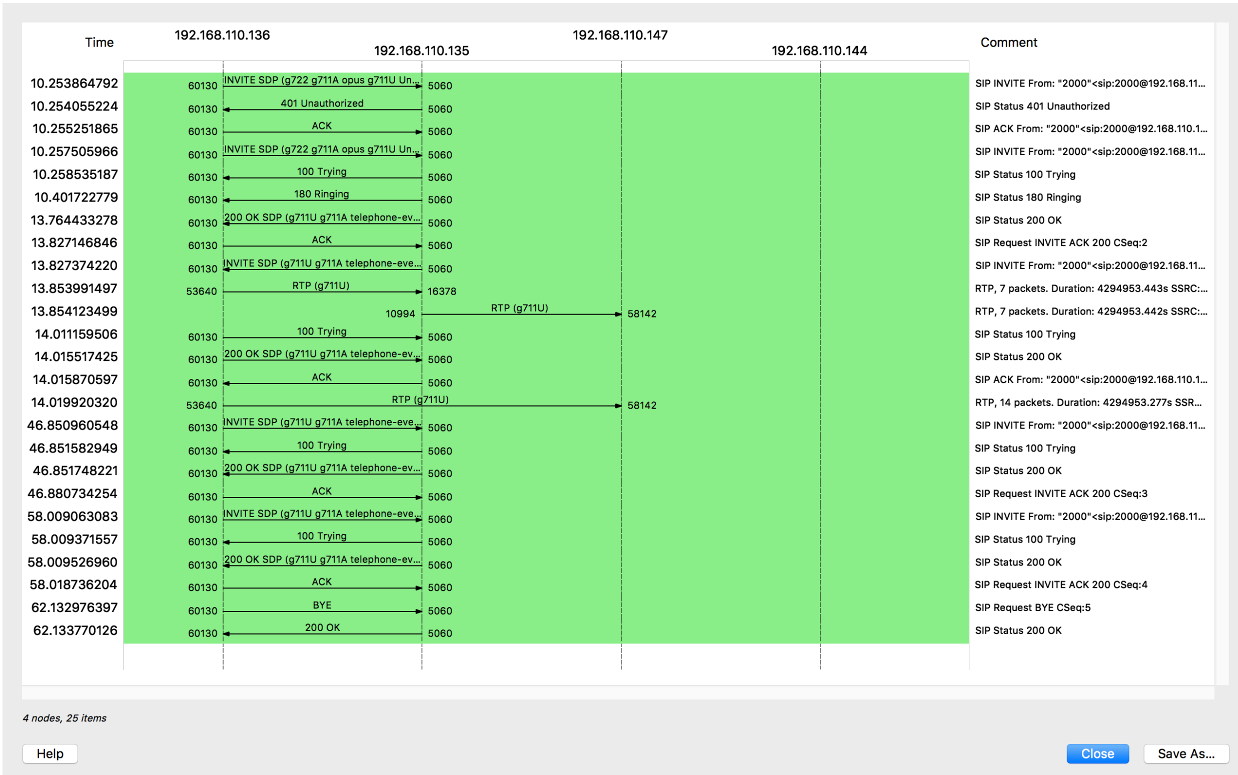


Figure : 1.3.12

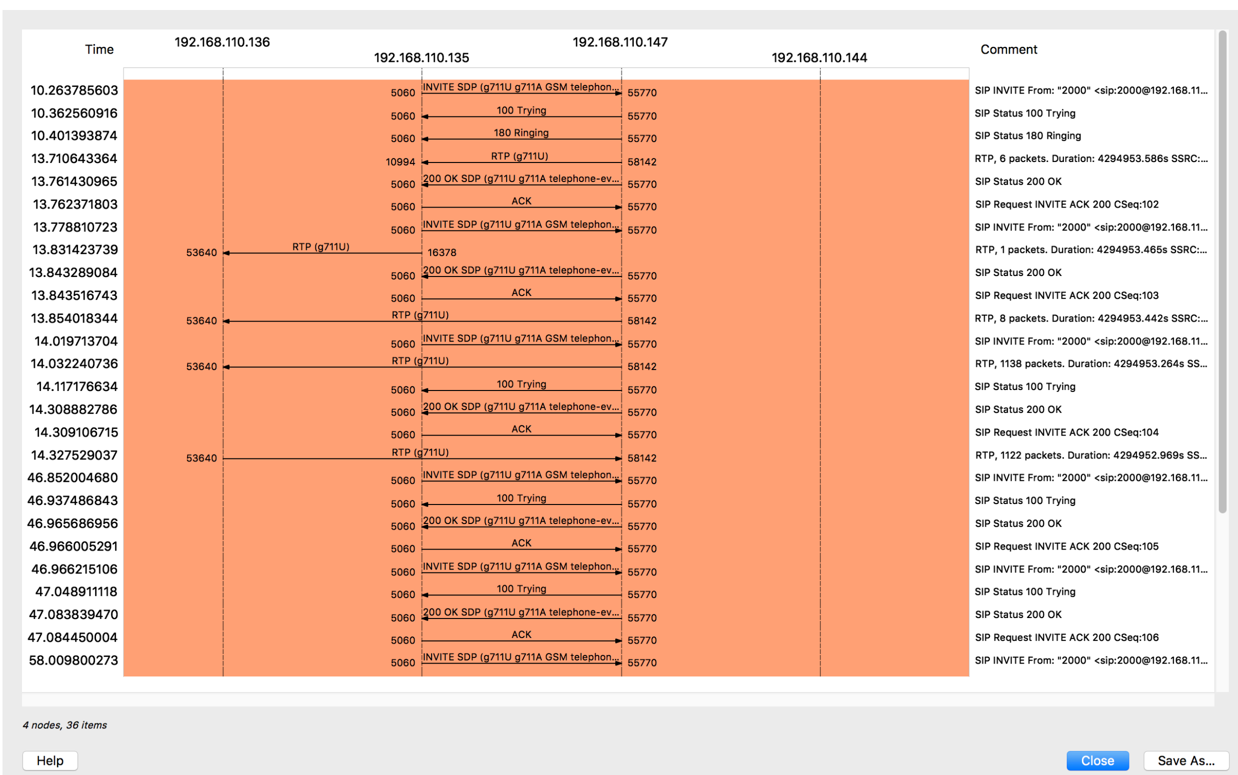


Figure : 1.3.13

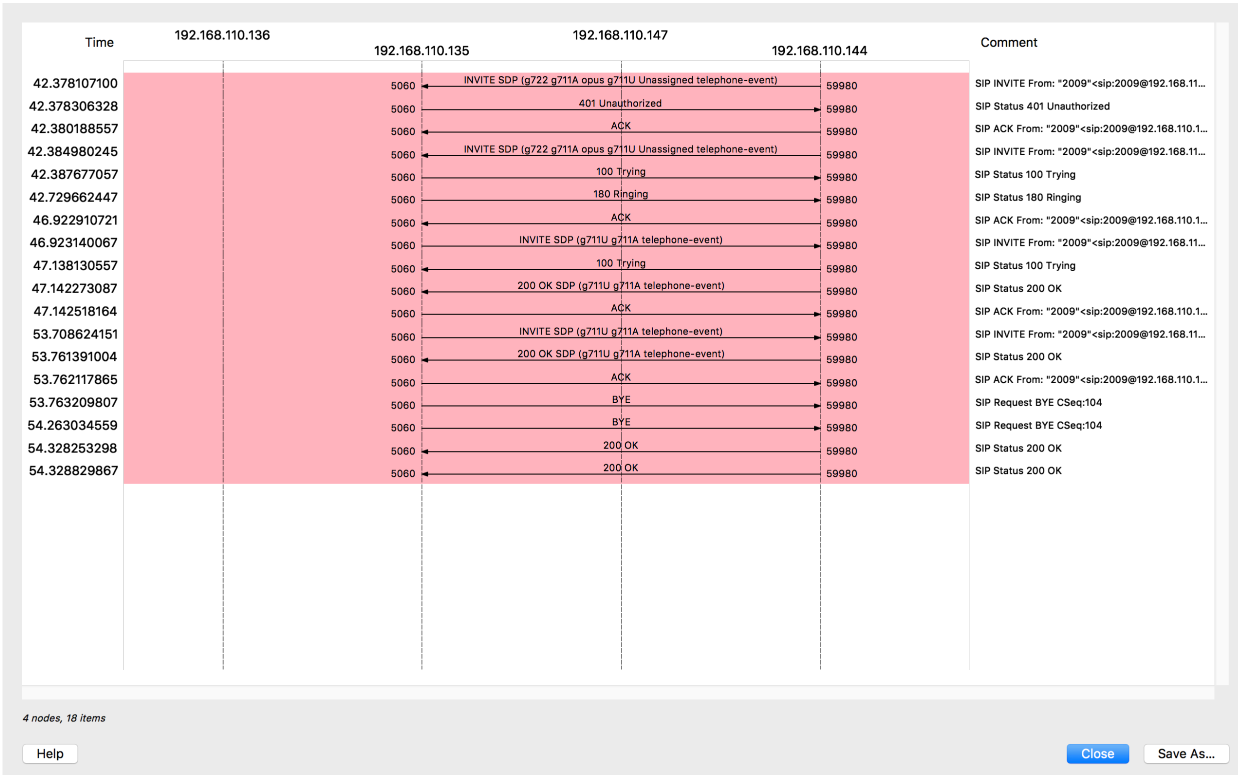


Figure : 1.3.14

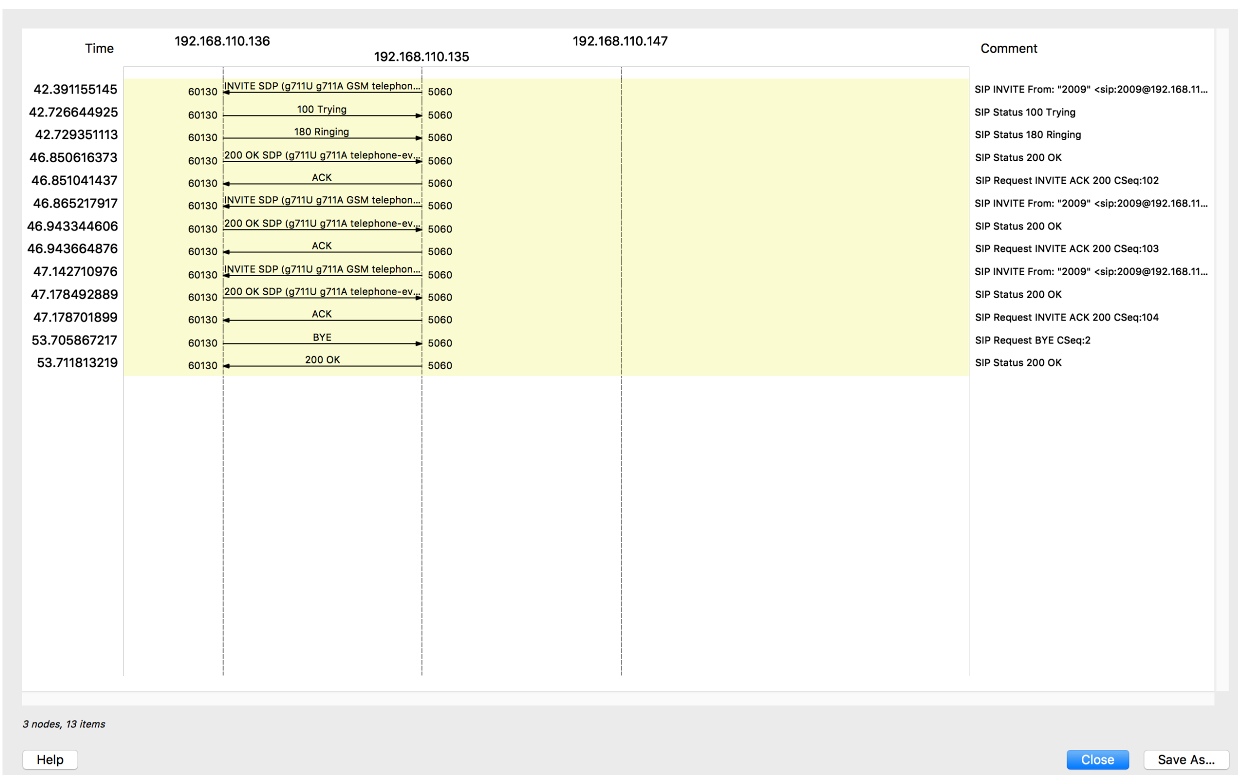


Figure : 1.3.15

We can see in fig:1.3.12 and 1.3.13 the call between 2000 and 2010 in progress from 2000 and 2010's side respectively. Fig 1.3.14 shows the call from 2009 is received by 2000 while keeping 2010 on hold and after completing call with 2009 again resuming call with 2010 as seen in fig: 1.3.15.

PHASE 4 – Call Conferencing:

The picture shows call in progress between 2000 and 2010 with 2009 calling 2000

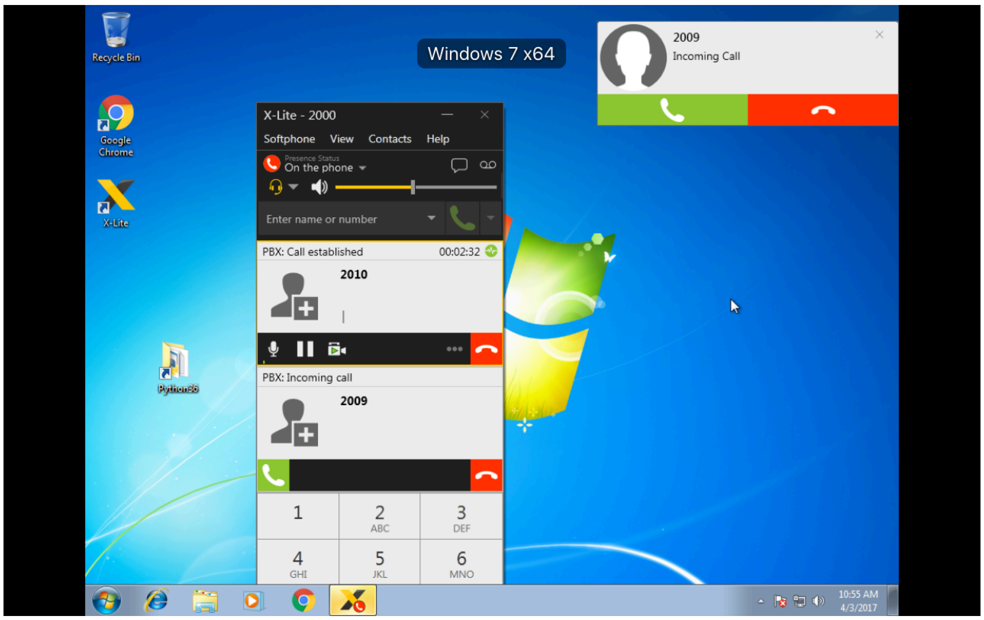


Figure : 1.4.1

Here when 2000 first accepts call 2010 goes on hold automatically as in real time cellular conferencing

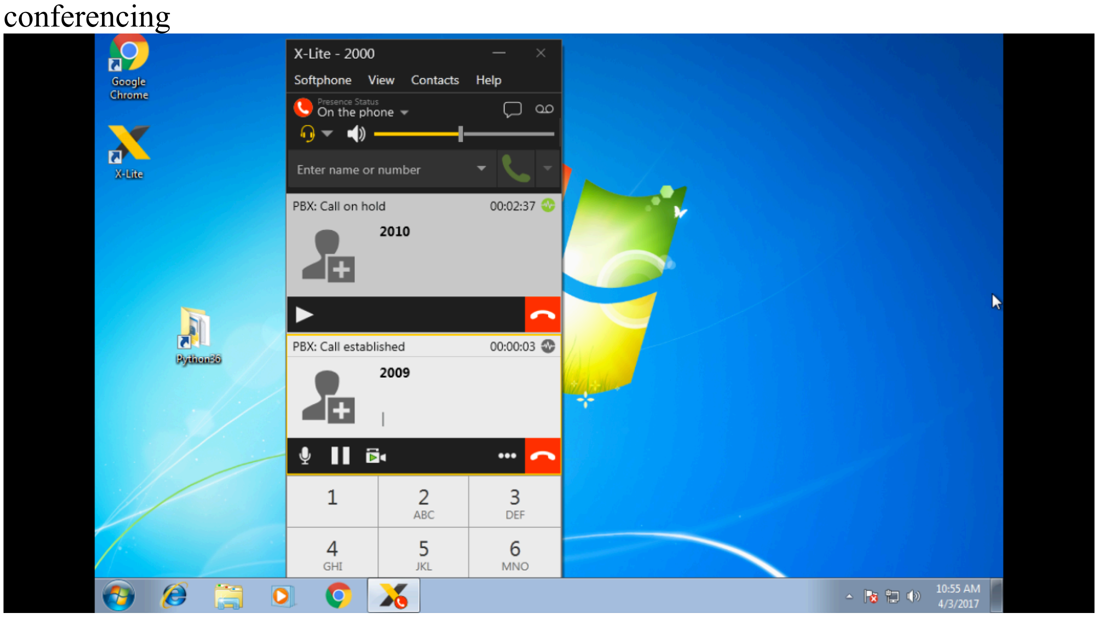


Figure : 1.4.2

This screenshot shows 2000 merging the two calls togather:

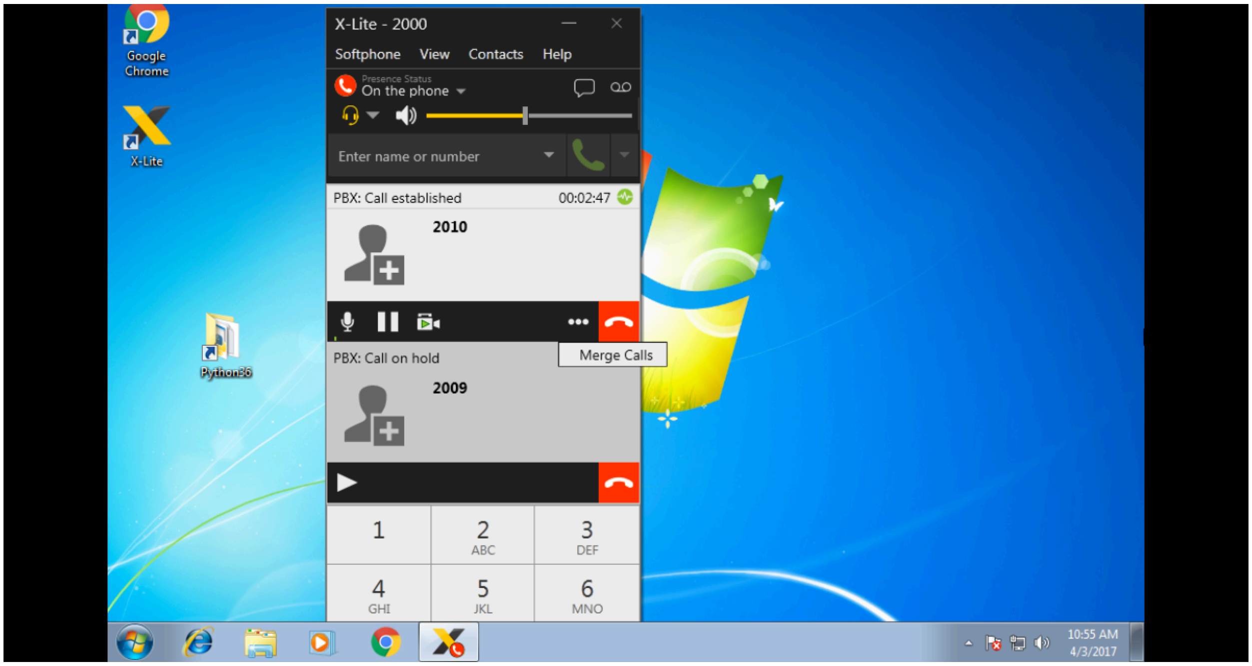


Figure : 1.4.3

This figure signify successful merger of calls with call conferencing:

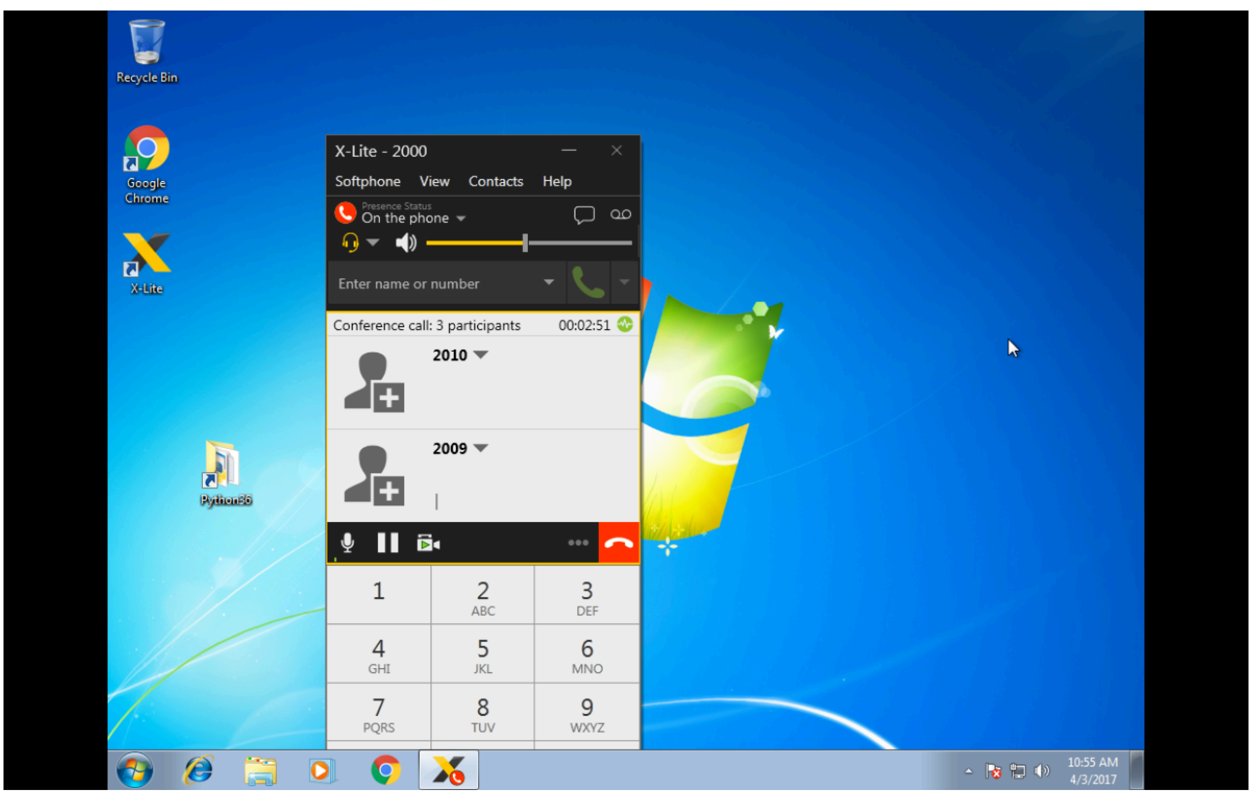


Figure : 1.4.4

This screenshot shows 2000 ending conference and both the calls simultaneously:

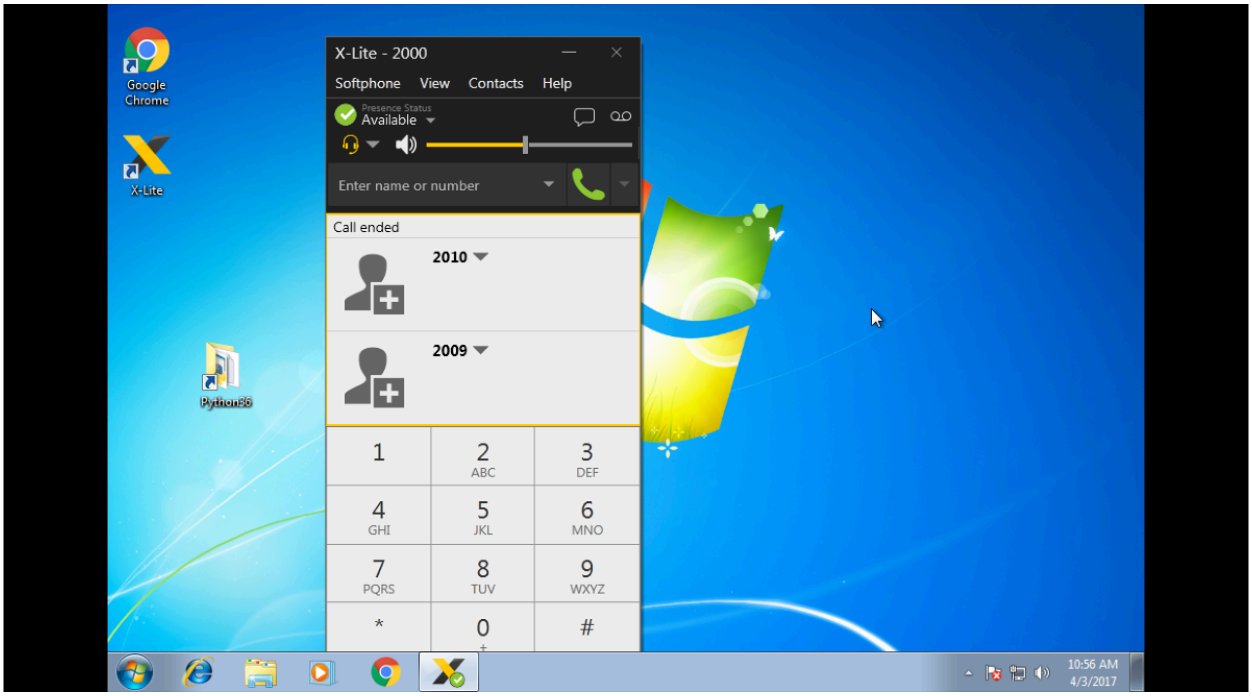


Figure : 1.4.5

Wireshark Captures:

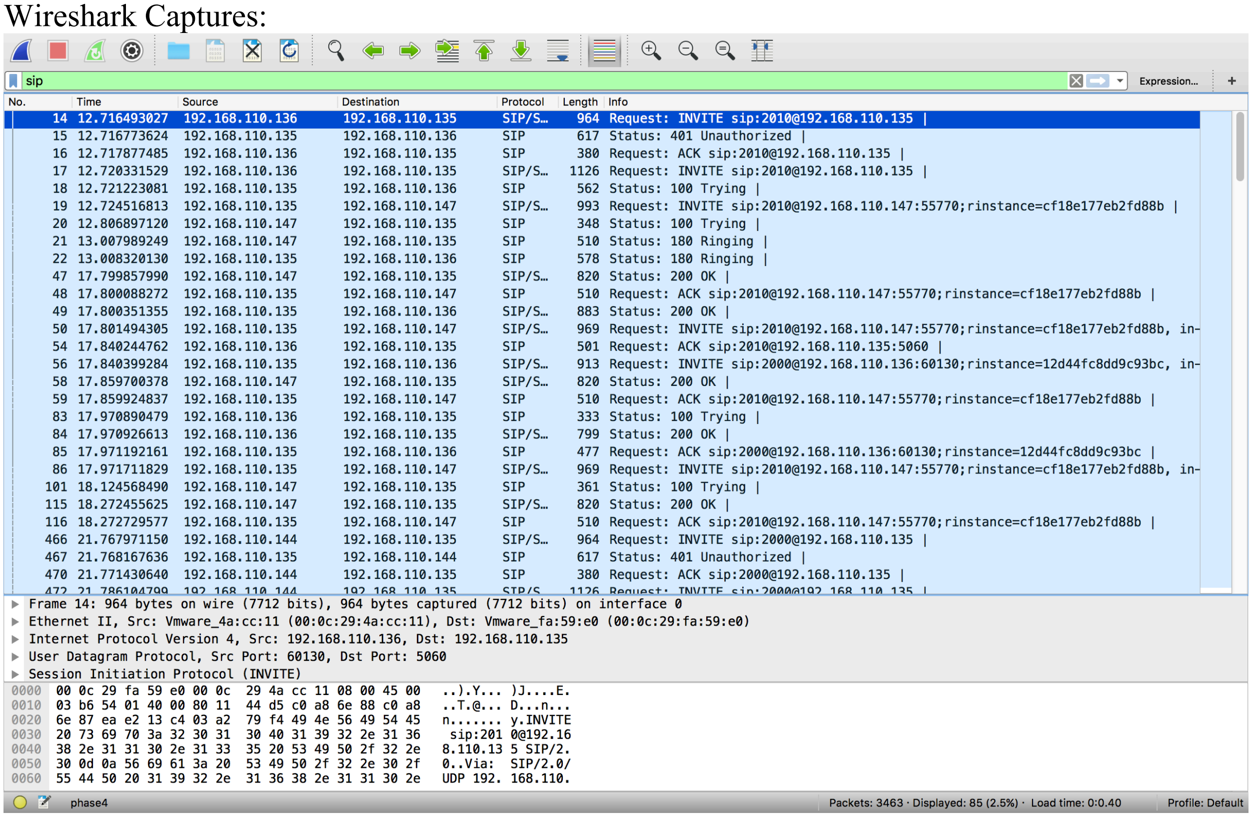


Figure : 1.4.6

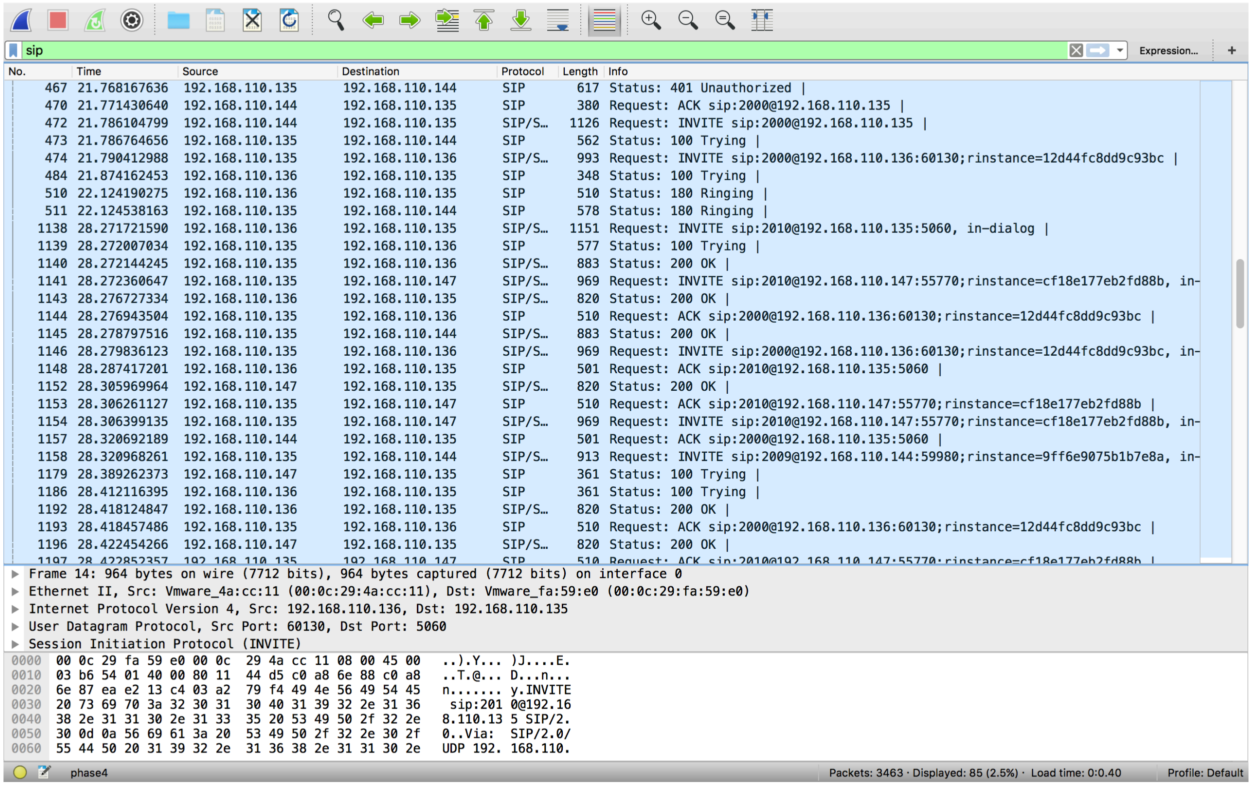


Figure : 1.4.7

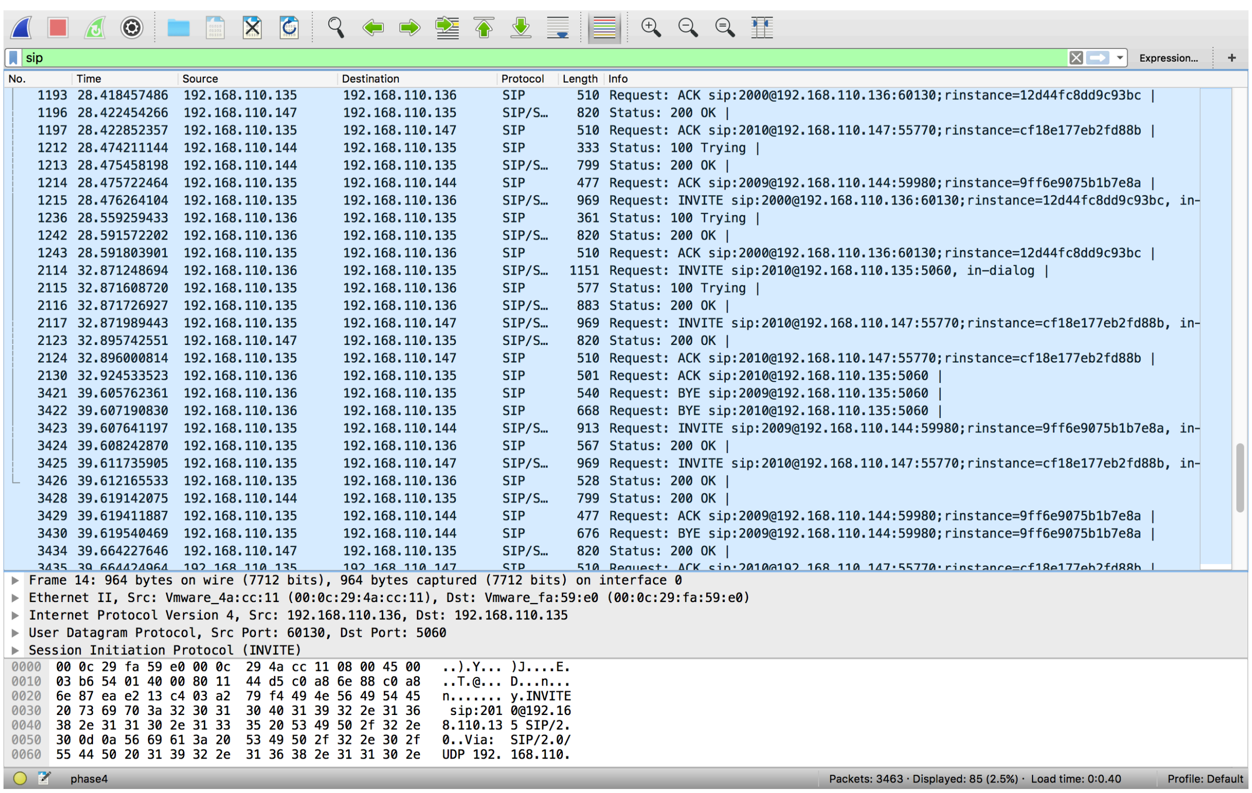


Figure : 1.4.8

Timing Diagram:

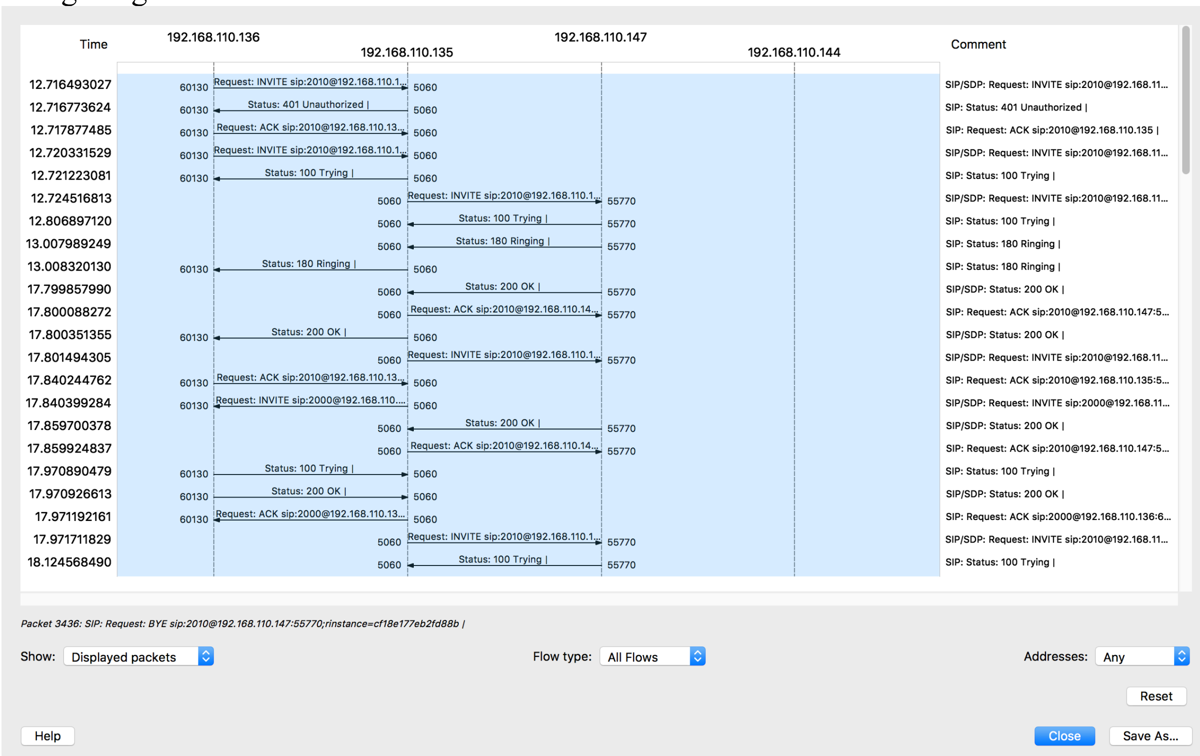


Figure : 1.4.9

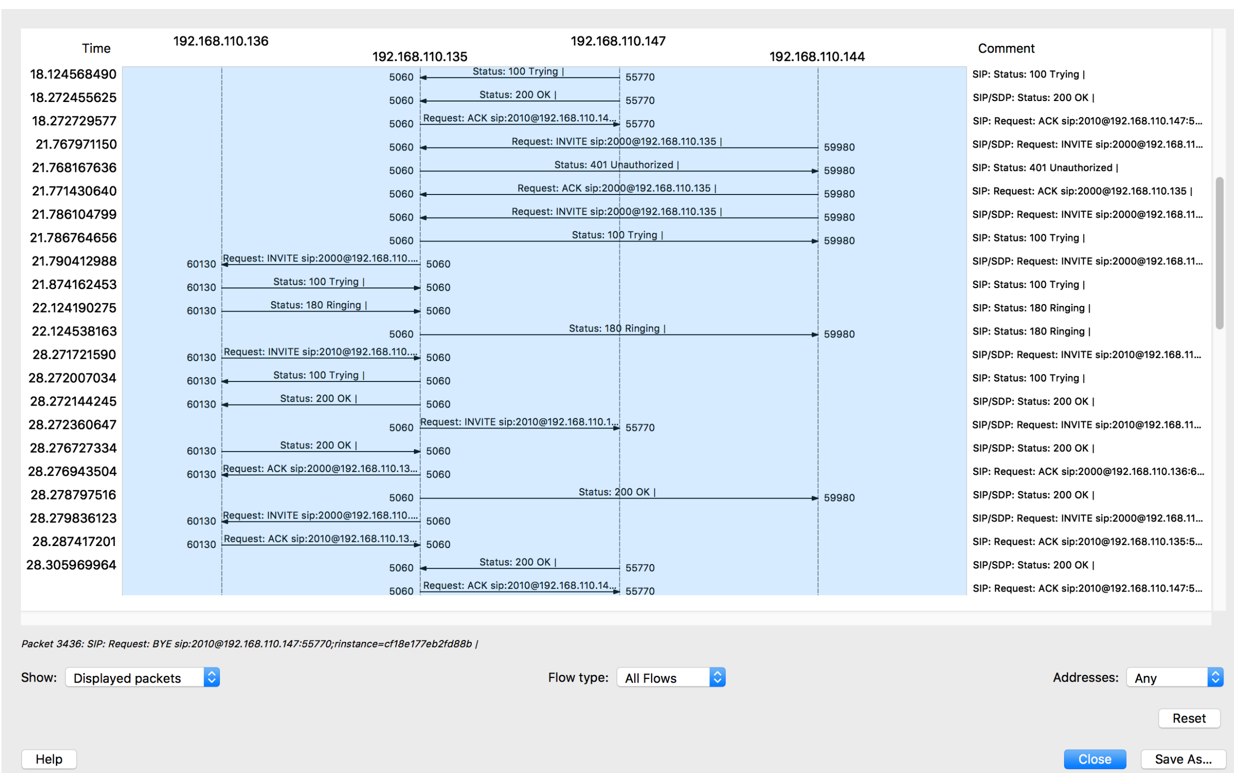


Figure : 1.4.10

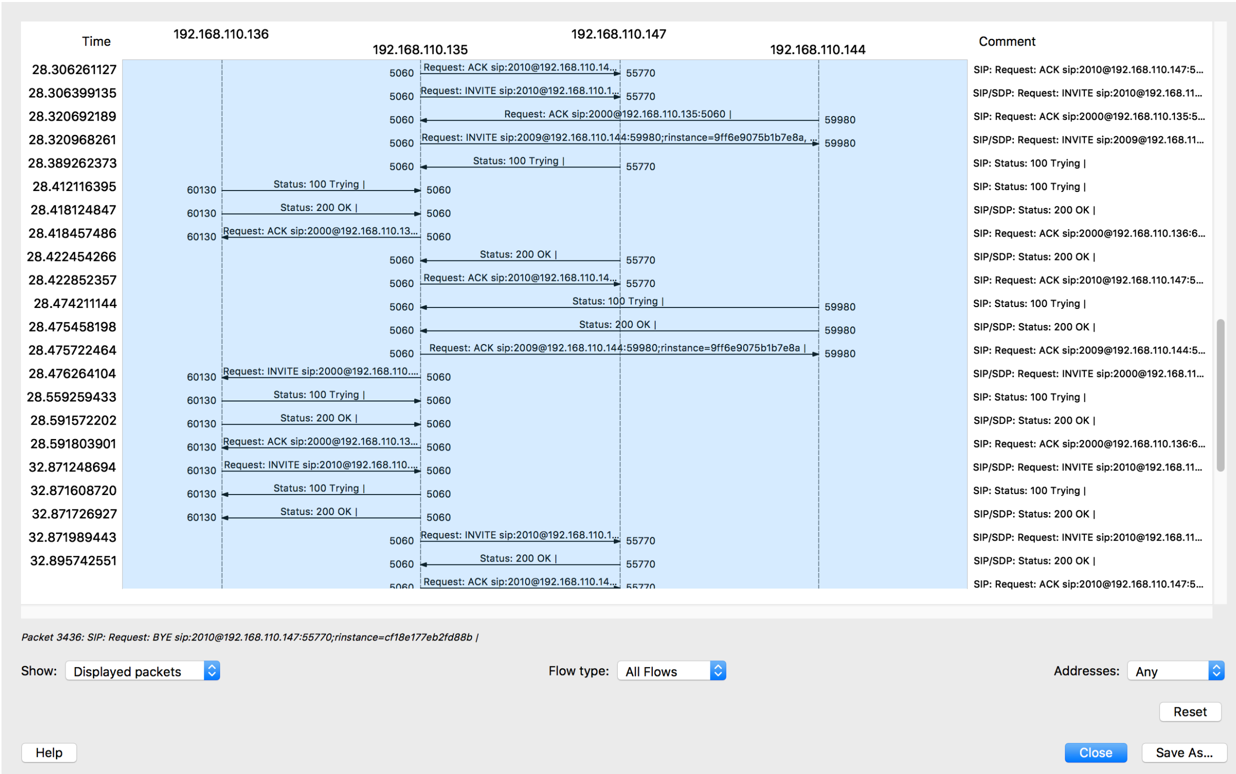


Figure : 1.4.11

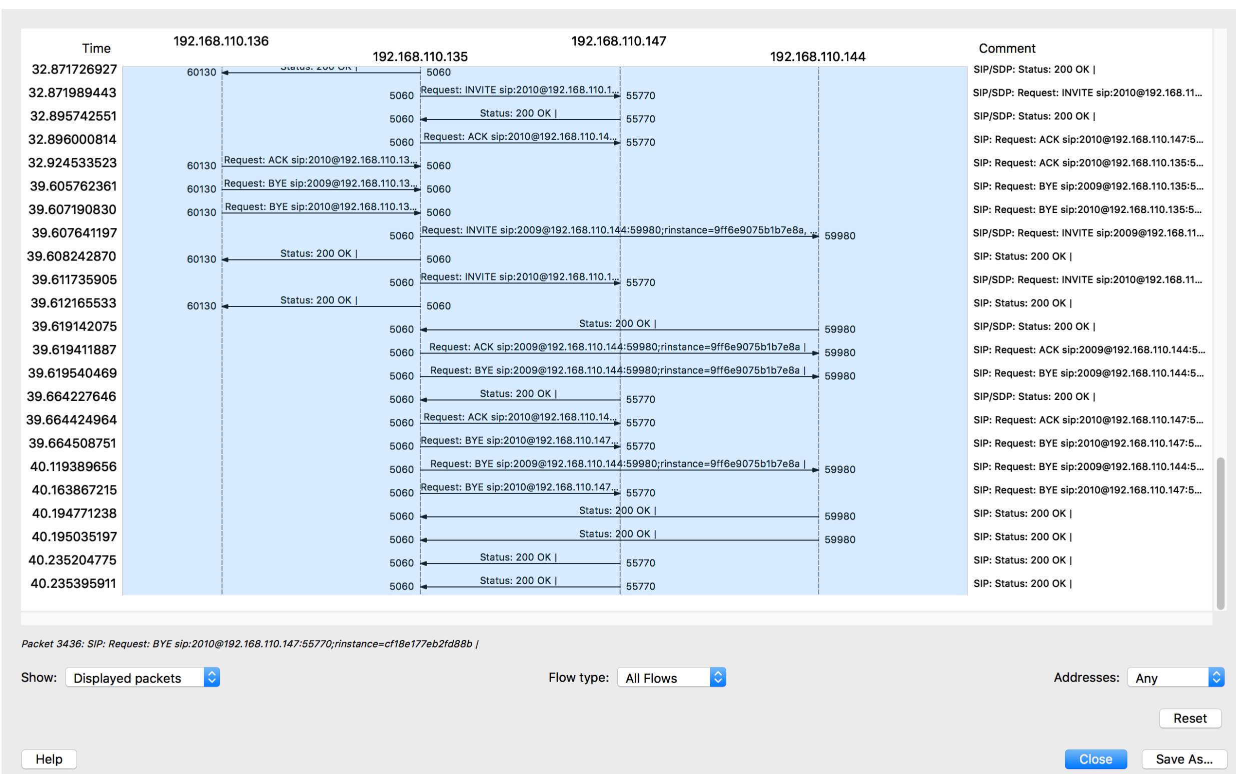
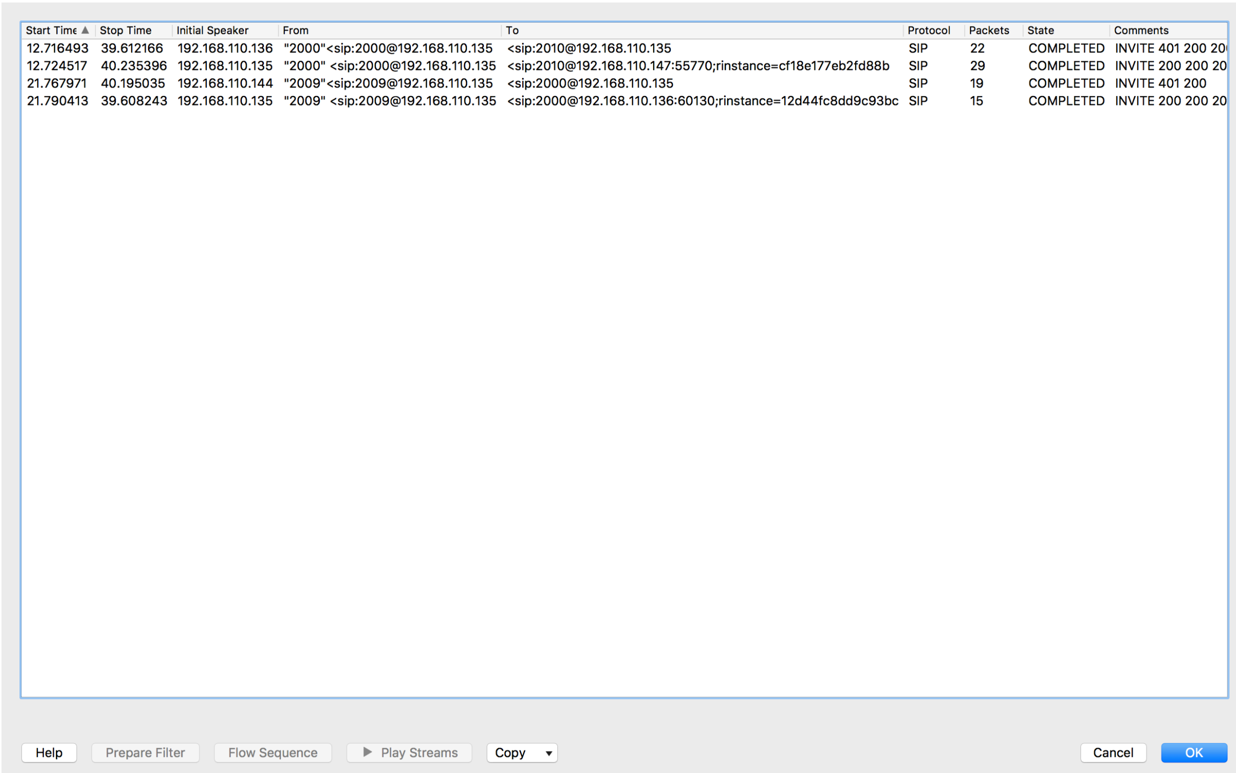


Figure : 1.4.12

SIP Flows:



Start Time	Stop Time	Initial Speaker	From	To	Protocol	Packets	State	Comments
12.716493	39.612166	192.168.110.136	"2000"<sip:2000@192.168.110.135	<sip:2010@192.168.110.135	SIP	22	COMPLETED	INVITE 401 200 20
12.724517	40.235396	192.168.110.135	"2000" <sip:2000@192.168.110.135	<sip:2010@192.168.110.147:55770;rinstance=cf18e177eb2fd88b	SIP	29	COMPLETED	INVITE 200 200 20
21.767971	40.195035	192.168.110.144	"2009"<sip:2009@192.168.110.135	<sip:2000@192.168.110.135	SIP	19	COMPLETED	INVITE 401 200
21.790413	39.608243	192.168.110.135	"2009" <sip:2009@192.168.110.135	<sip:2000@192.168.110.136:60130;rinstance=12d44fc8dd9c93bc	SIP	15	COMPLETED	INVITE 200 200 20

Figure : 1.4.13

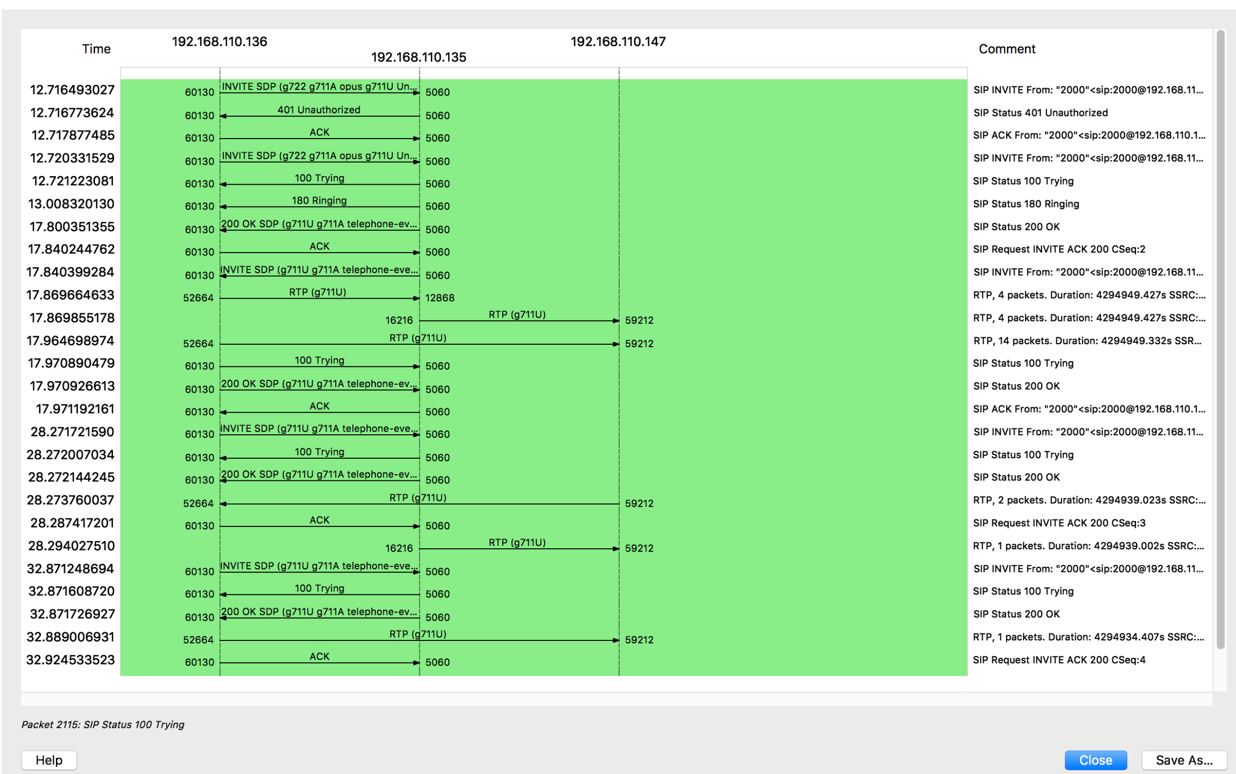


Figure : 1.4.14

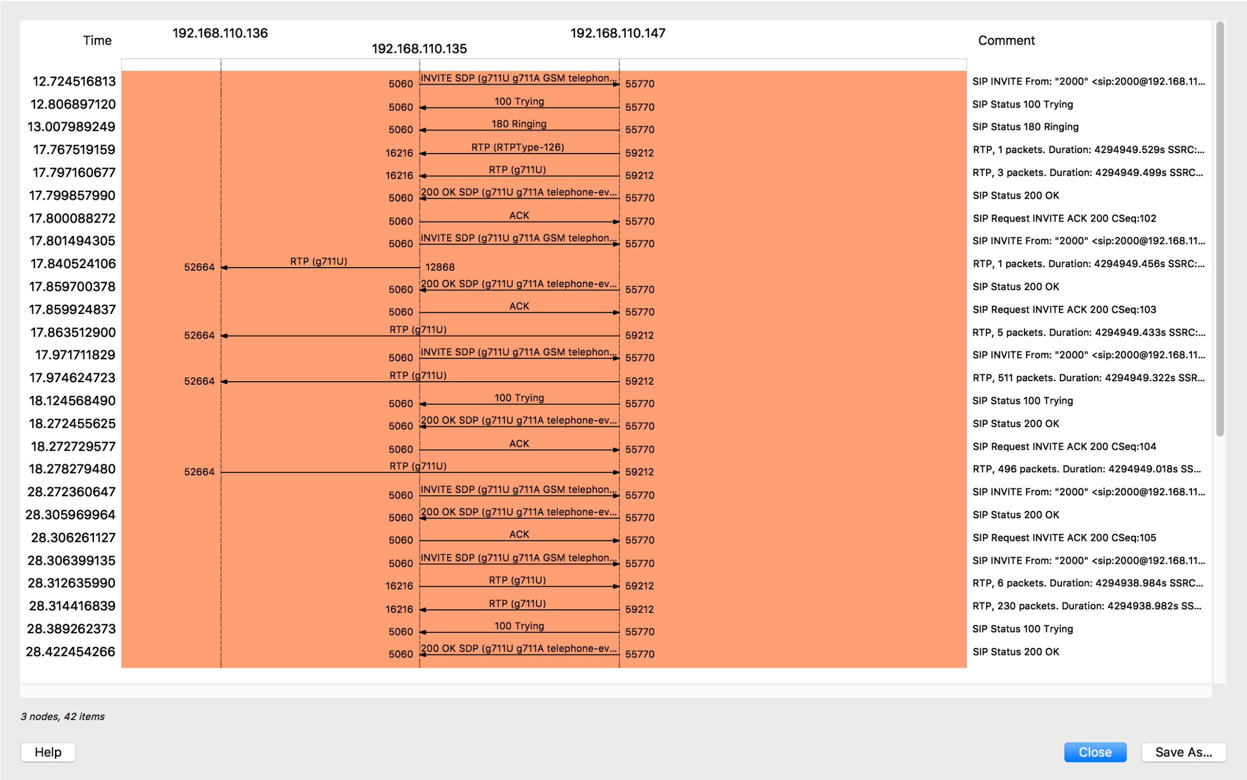


Figure : 1.4.15

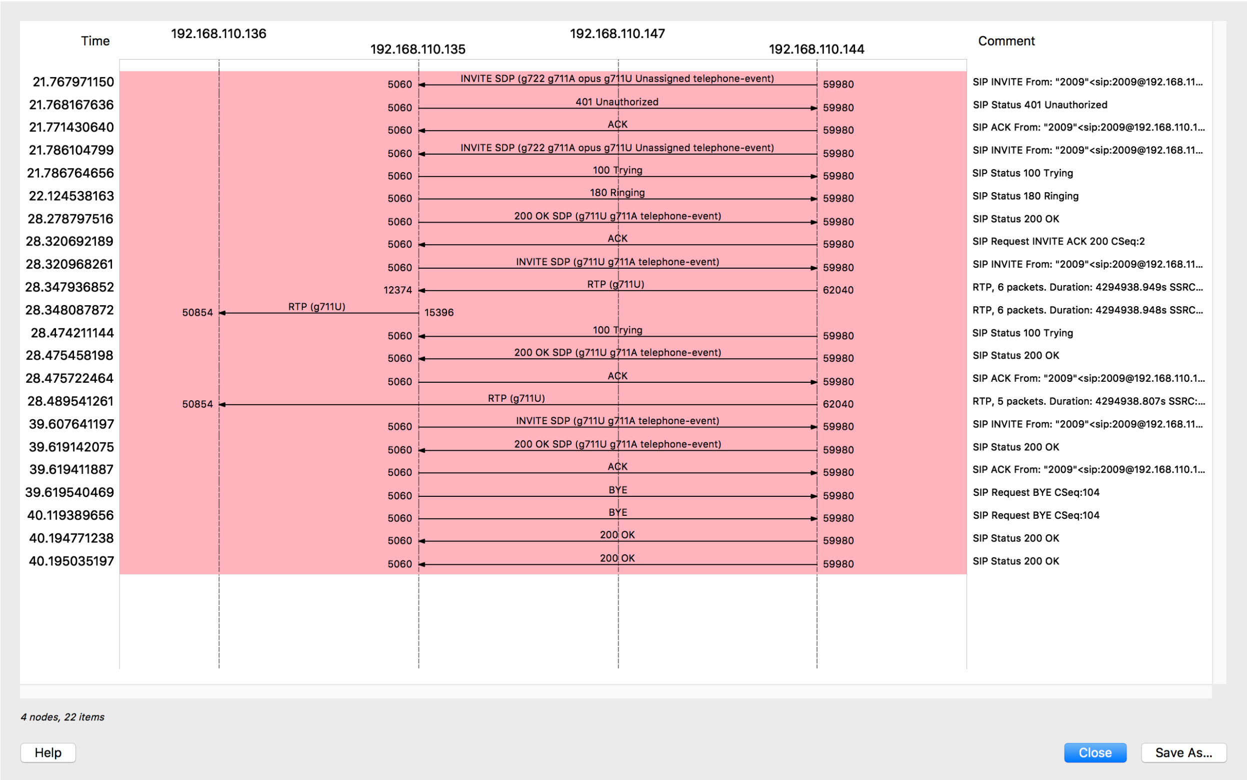


Figure : 1.4.16

Conference on:

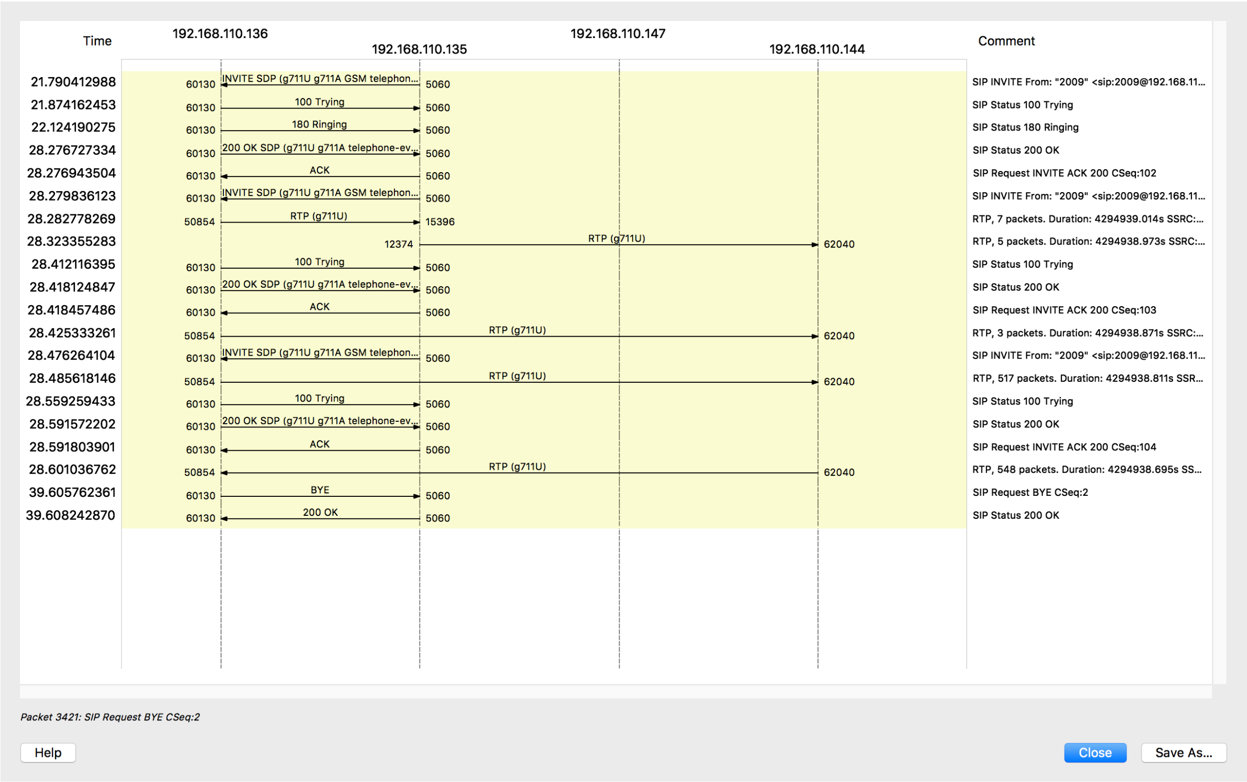


Figure : 1.4.17

Figure 1.4.14 shows establishment of calls between 2000 and 2010 and the messages exchanged at 2000's side figure:1.4.15 shows same on 2010's side. The figure:1.4.16 shows request from 2009 to 2000 to establish call when 2000 is already on call with 2010. The yellow flow graph shows commencement of conference as after accepting call from 2009 2000 invites 2010 for a conference which 2010 accepts and conference is established. The media flow can be seen in the RTP g.711 flow at 28.6 seconds in figure:1.4.17.

2.Part-II Basic SIP Client

For this part asterisk was installed on Kali Linux with one X-Lite client was loaded on Windows 7 and custom SIP client on Ubuntu their ip and MAC are as follows.

Asterisk:

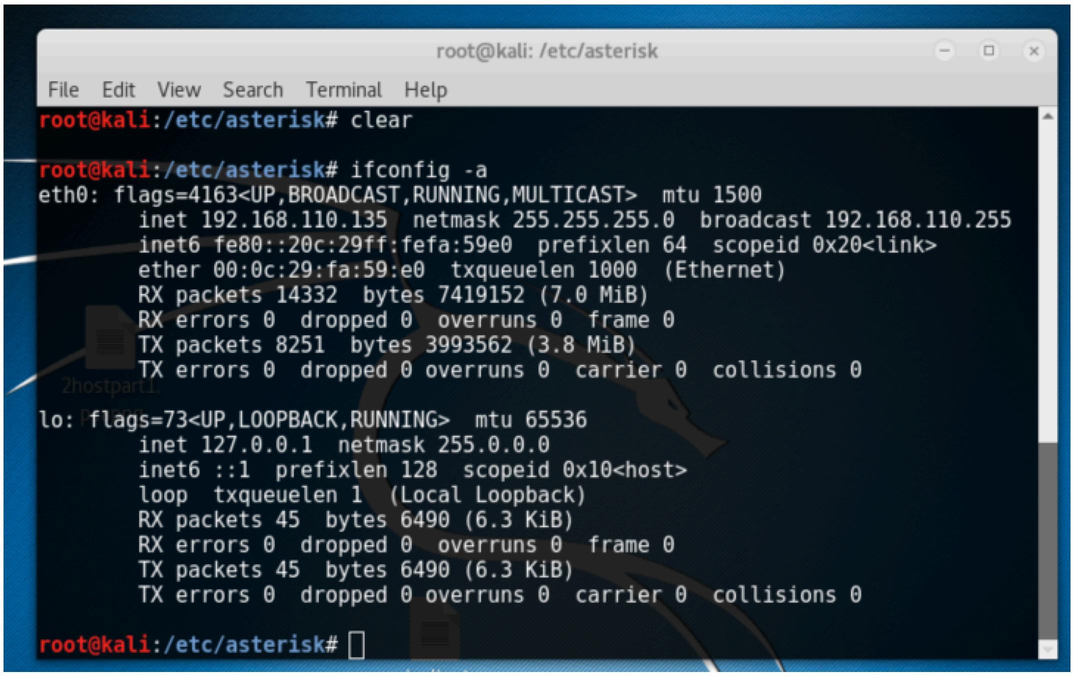


Figure : 2.1

User on Windows

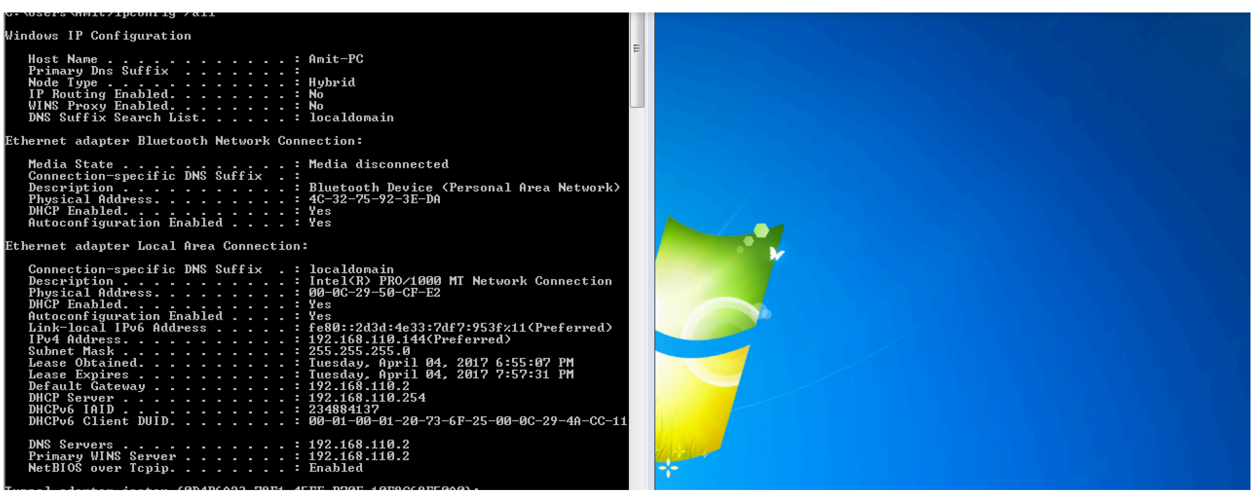


Figure : 2.2

Personalised client on Ubuntu:

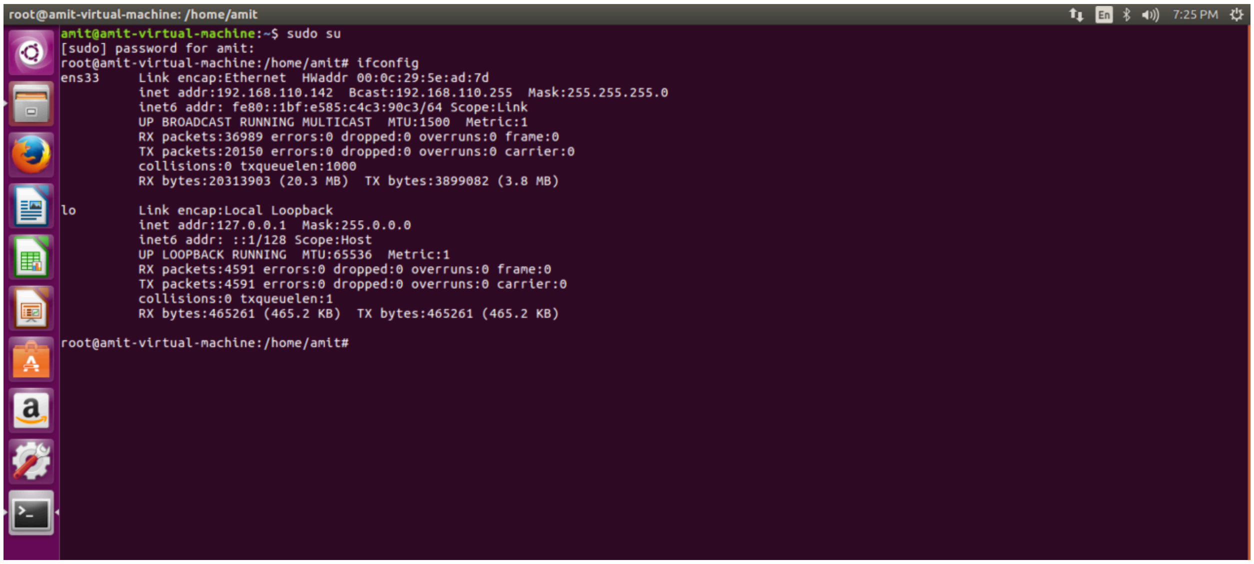


Figure : 2.3

The below screenshot shows execution of custom code 284final.py file asking user to enter his IP

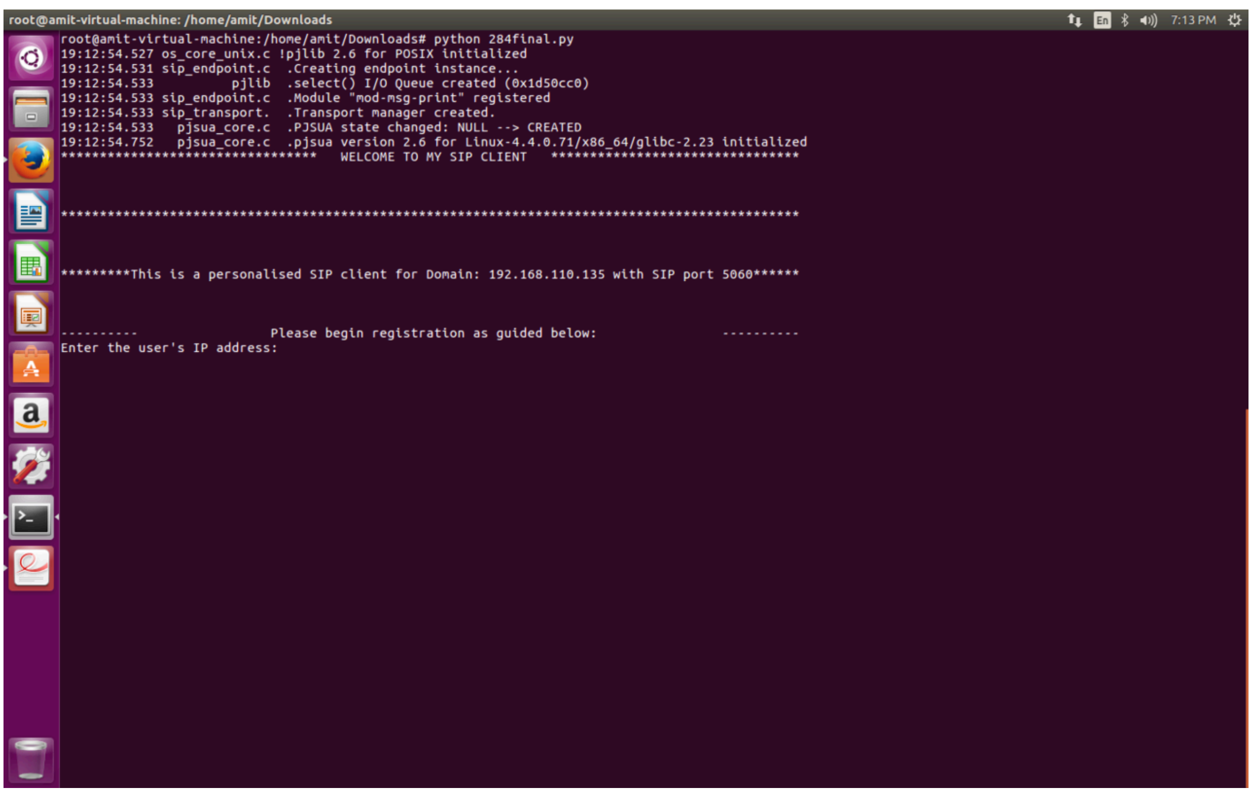


Figure : 2.4

Once the IP i.e 192.168.110.142 for client is entered authentication starts by asking the username

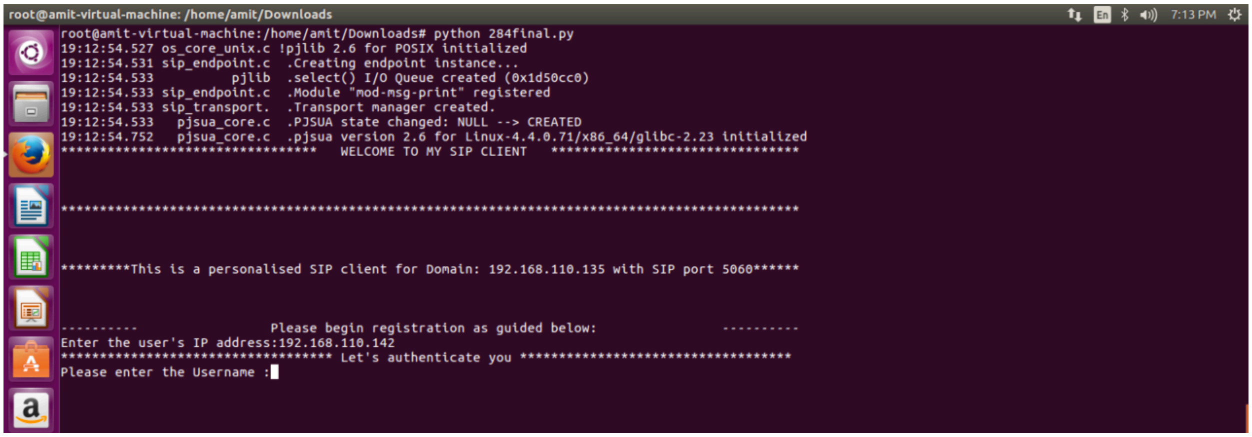


Figure : 2.5

we enter the username 2000 which we earlier included in sip and extensions configuration files and the password and using displayname Amit instead of 2000.After this the client registers himself with the asterisk server and asks if the user wants to call.

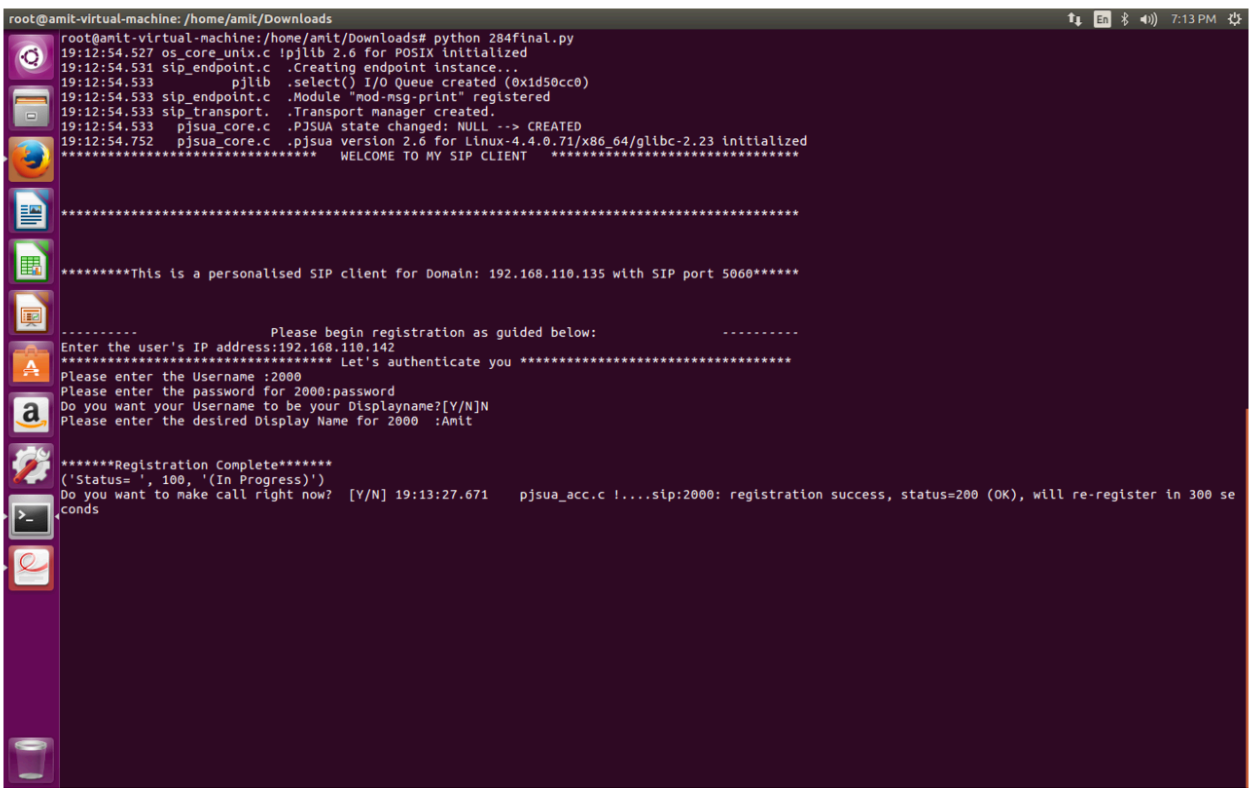


Figure : 2.6

The screenshot of asterisk server showing 2000 registered :

Figure : 2.7

On entering y client asks for username to be called at the asterisk server running on 192.168.110.135

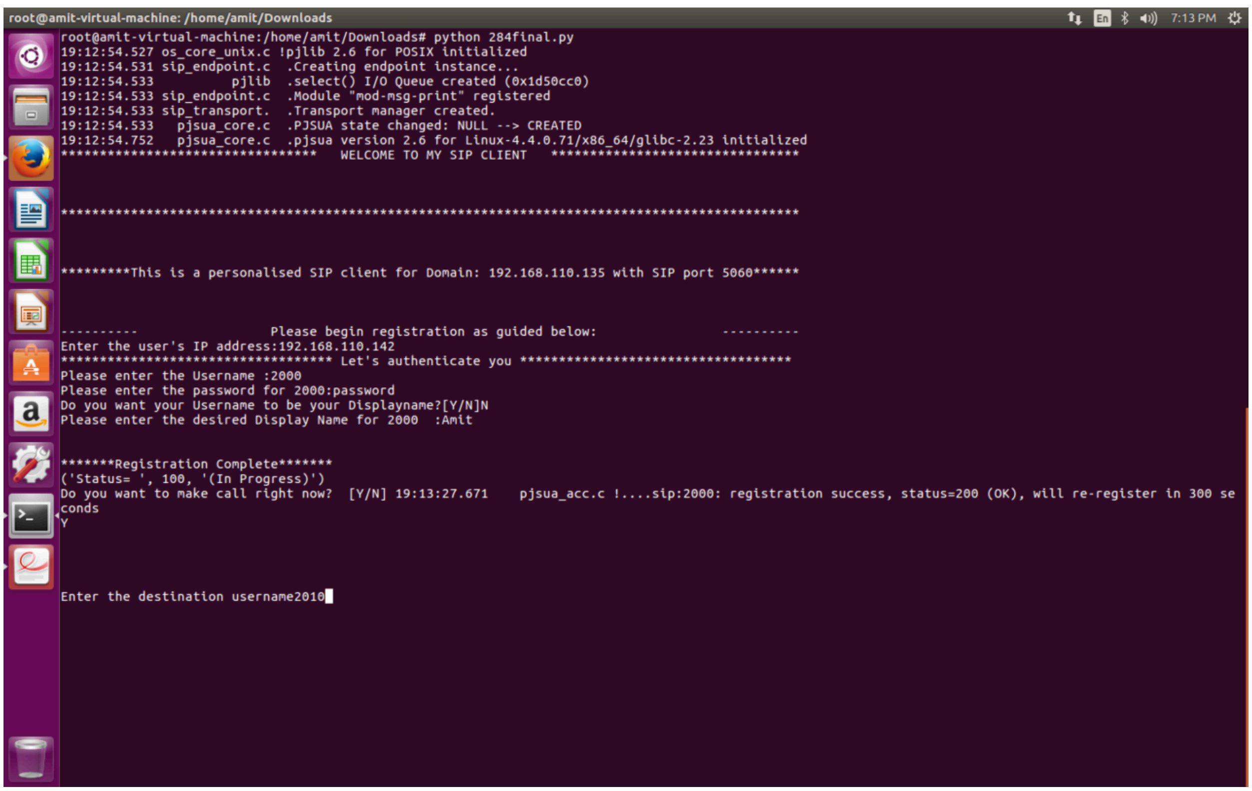


Figure : 2.8

We call 2010 from our client and the screenshot shows client calling 2010 and prompts to press enter to finish call:

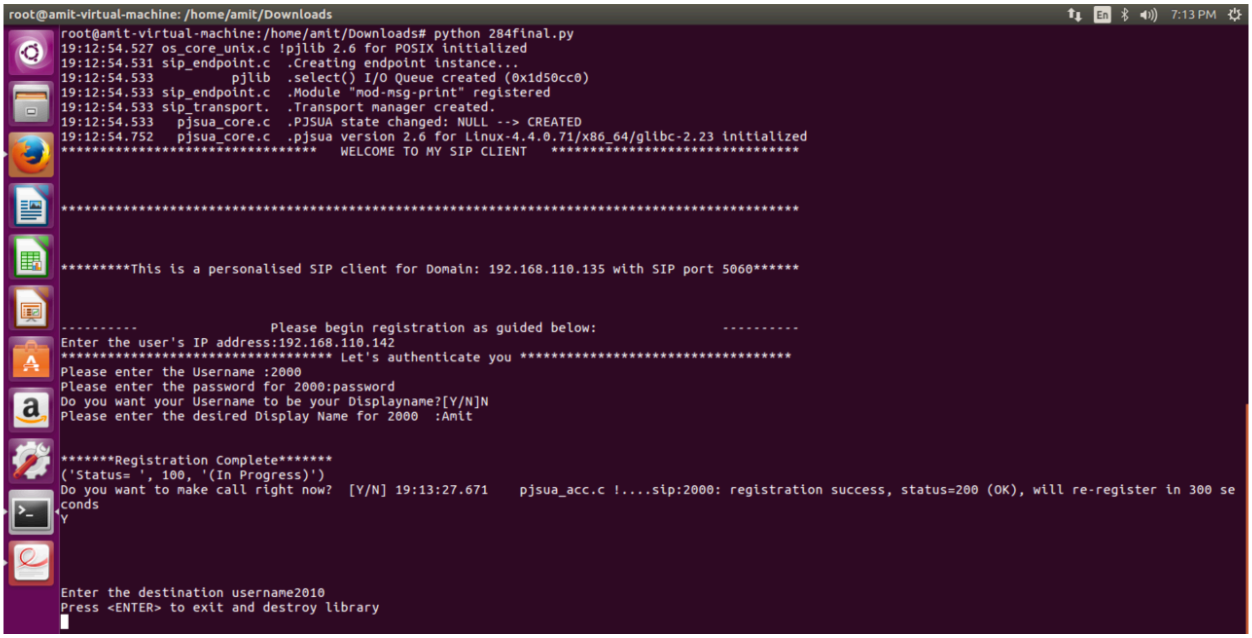


Figure : 2.9

The below figure shows asterisk server events where call from 2000 to 2010 has been initiated:

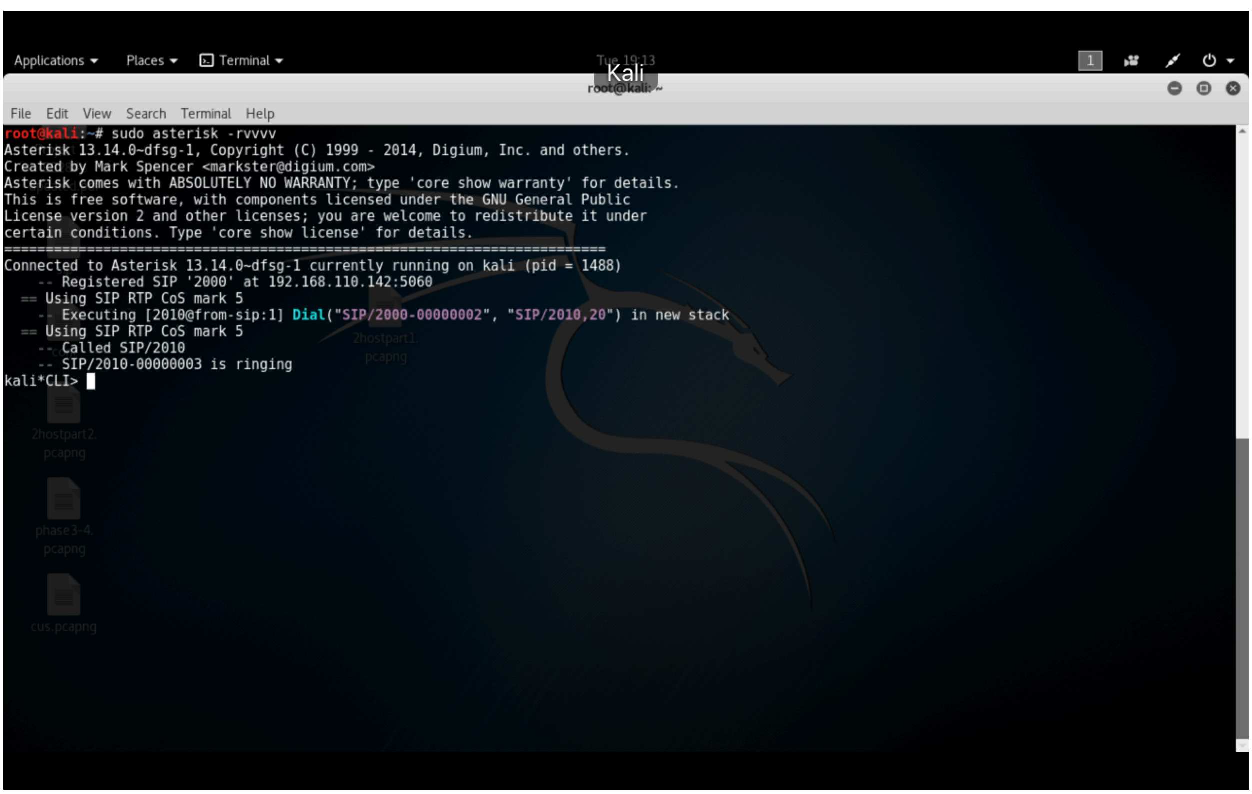


Figure : 2.10

Call from server at 2010

Figure : 2.11

2010 accepts call

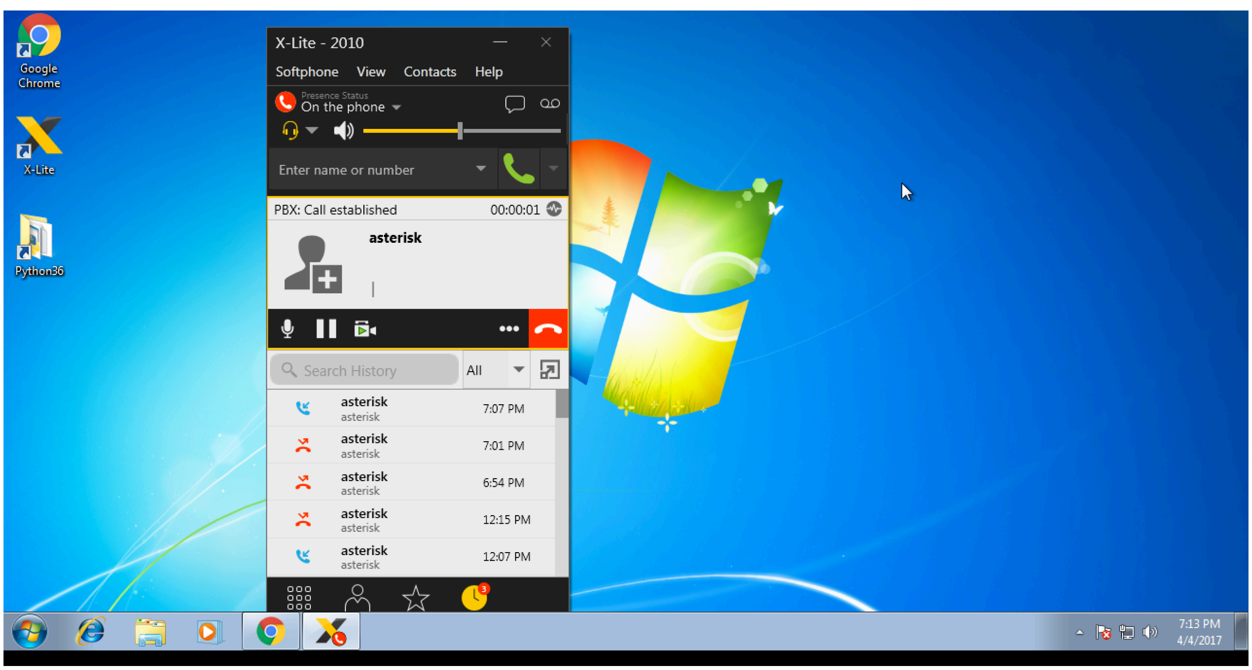


Figure : 2.12

The below is the events at asterisk server showing call established

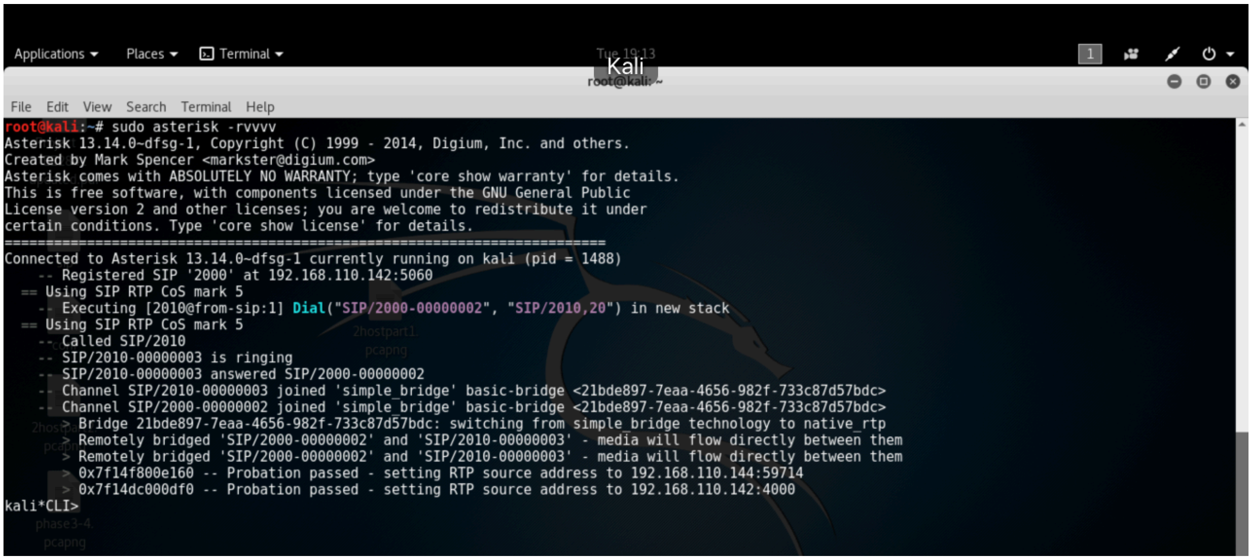


Figure : 2.13

The figure shows call ended from 2000's side and the libraries being destroyed.

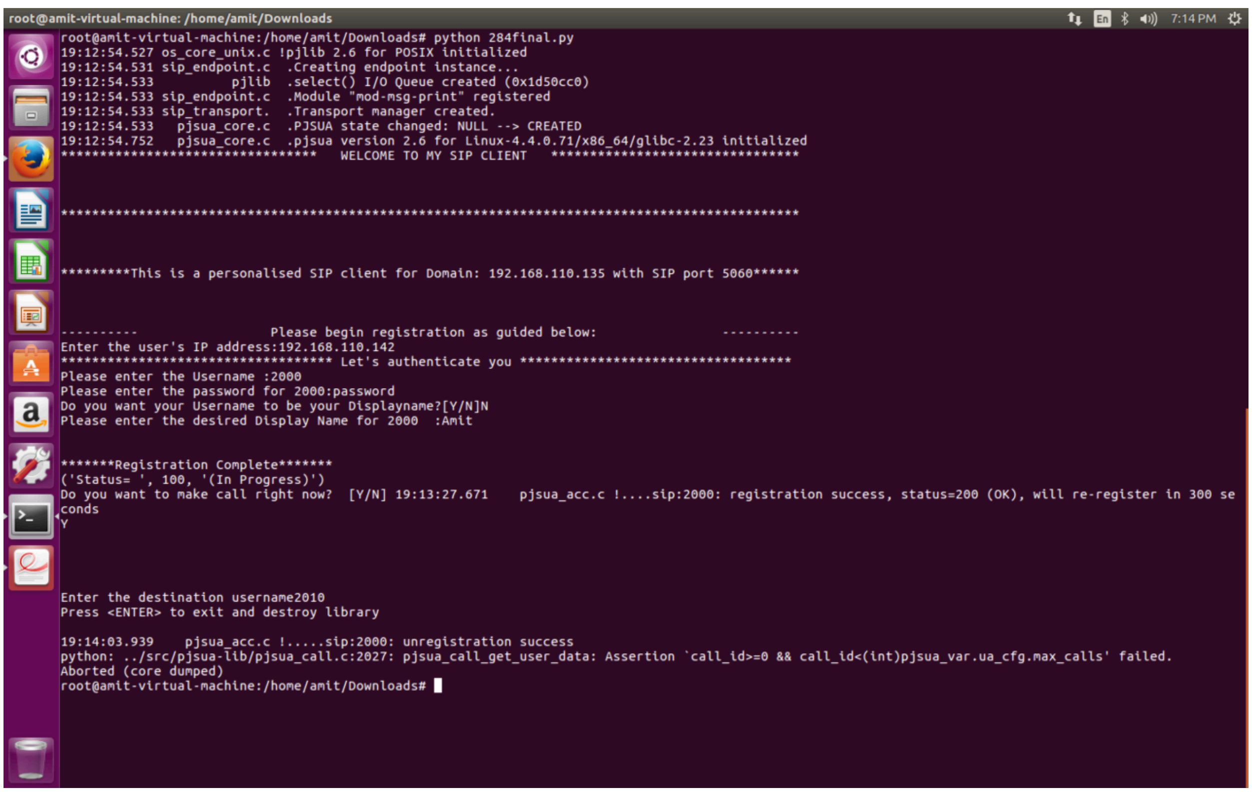


Figure : 2.14

The asterisk server’s output showing the whole process of registration ringing call and unregistration of 2000

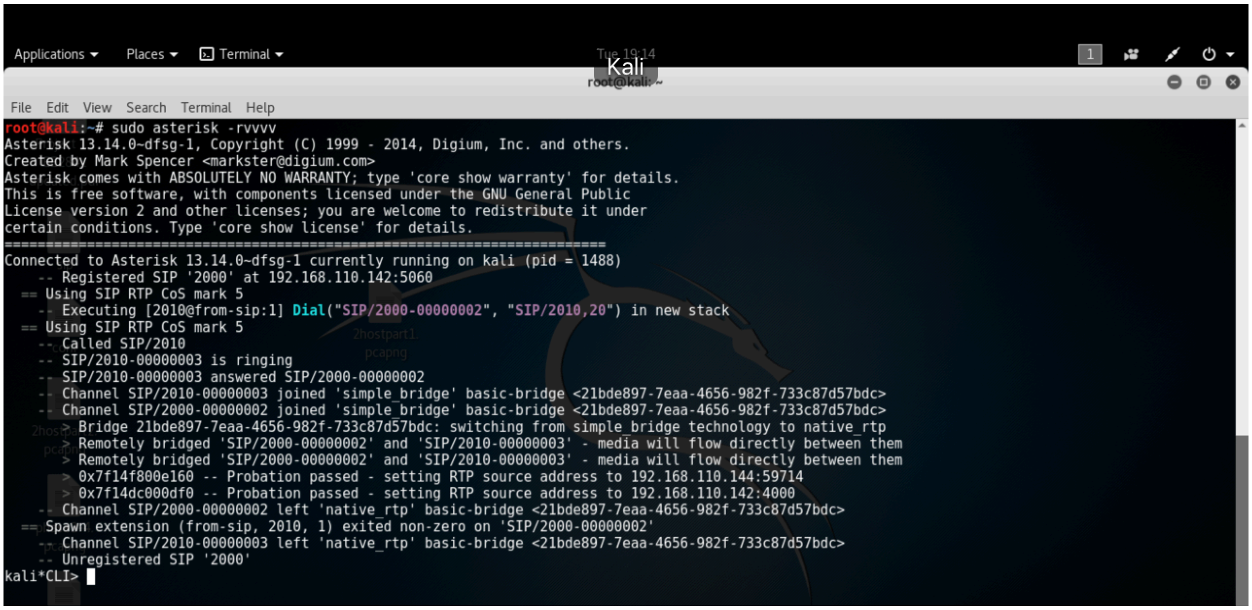


Figure : 2.15

Wireshark Captures:

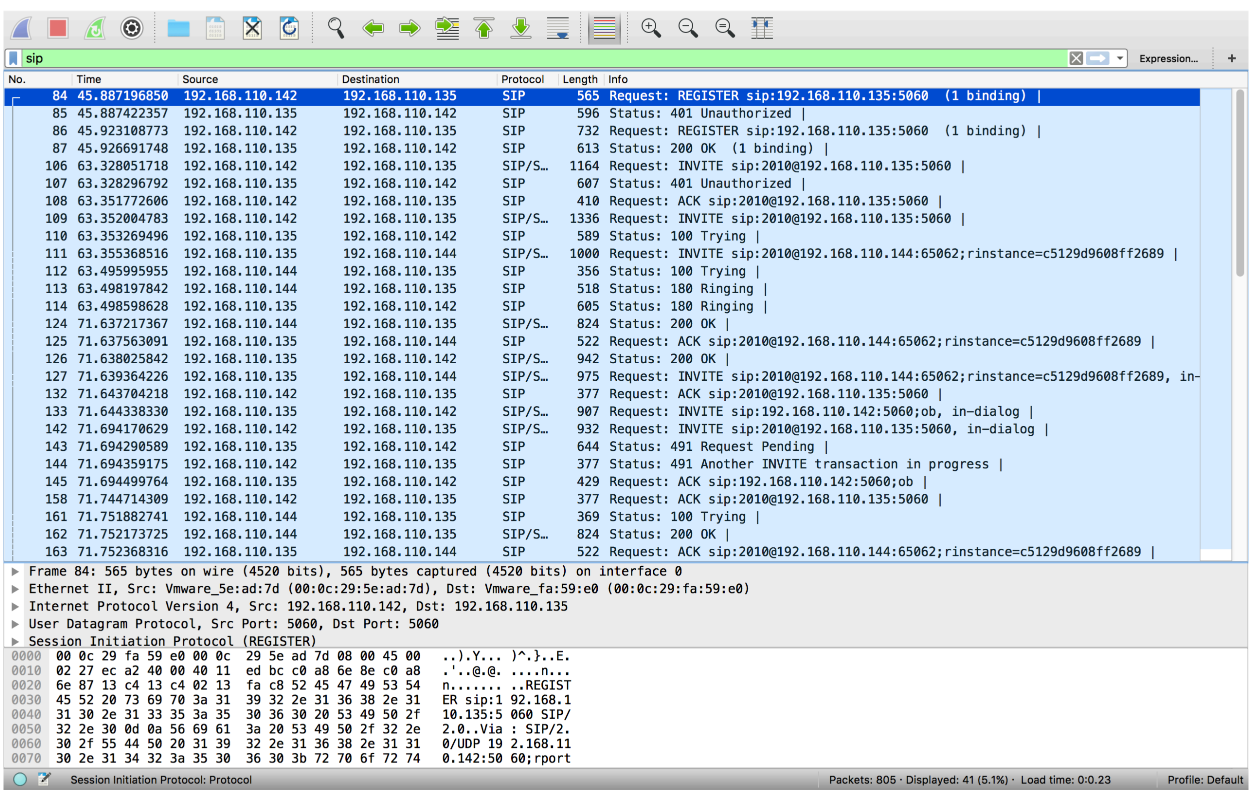


Figure : 2.16

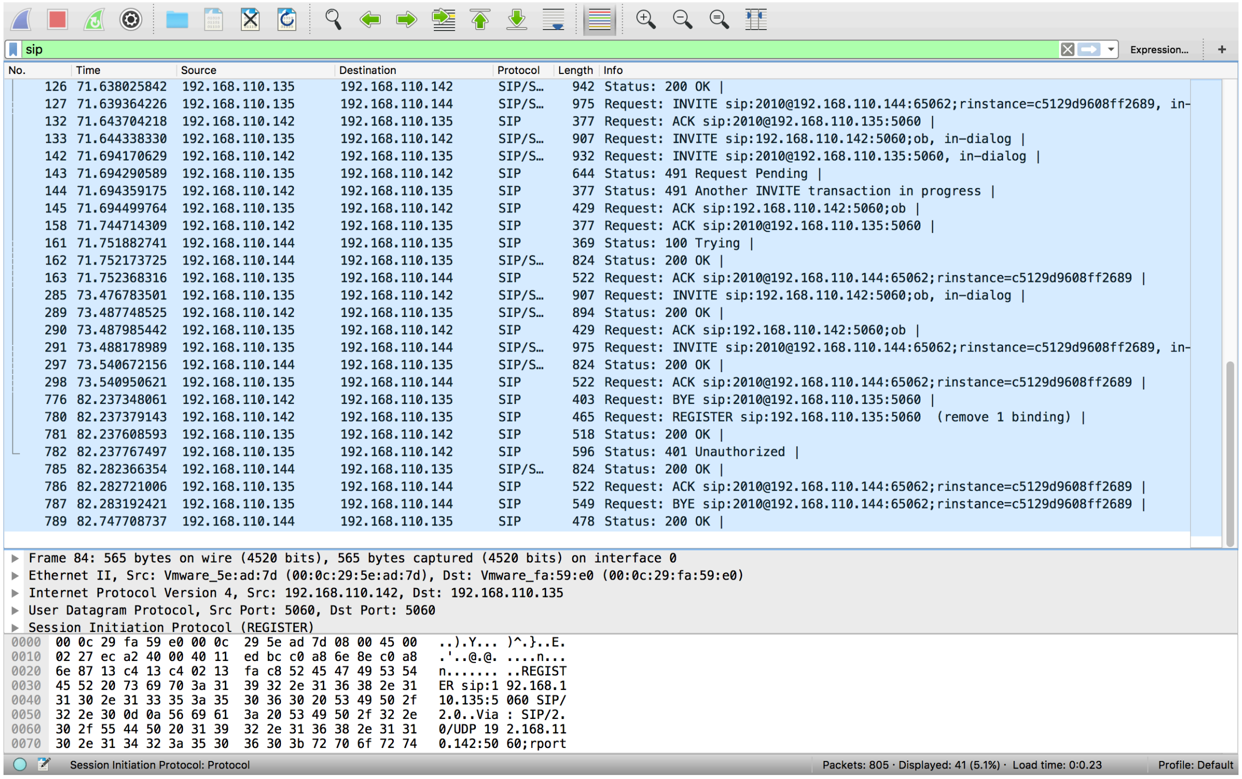


Figure : 2.17

Timing Diagram:

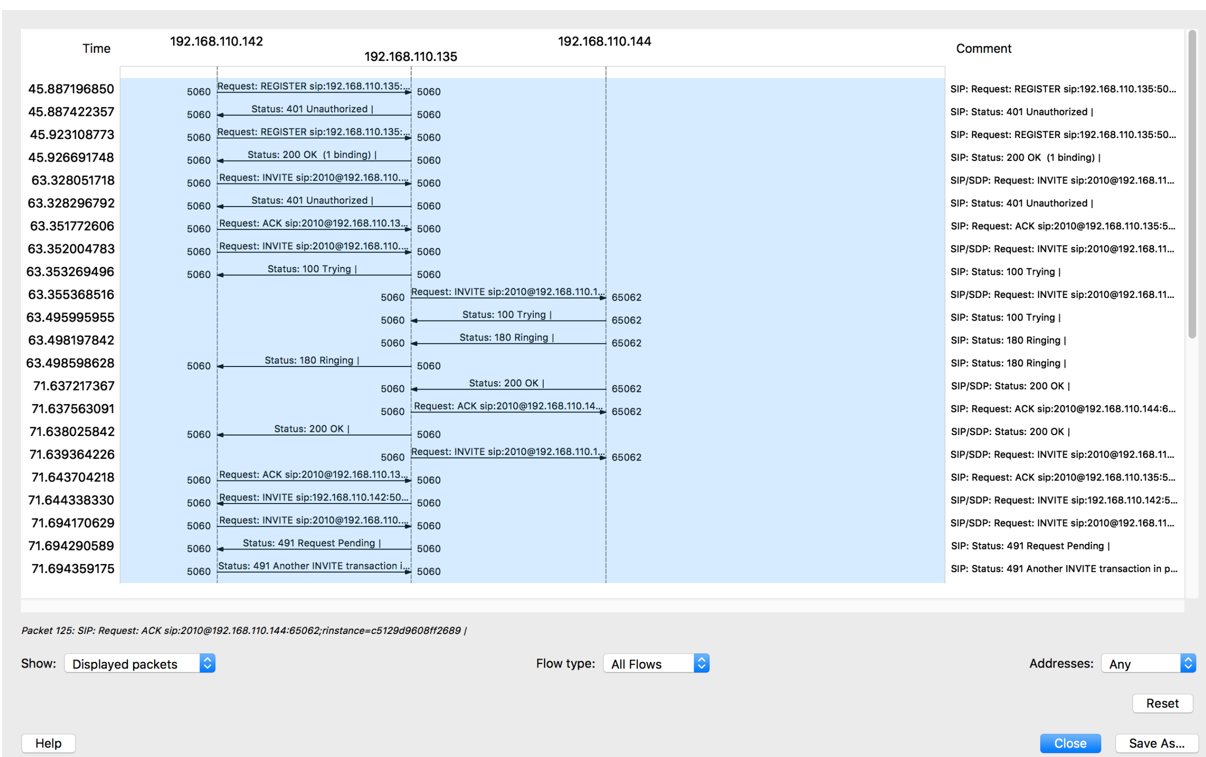


Figure : 2.18

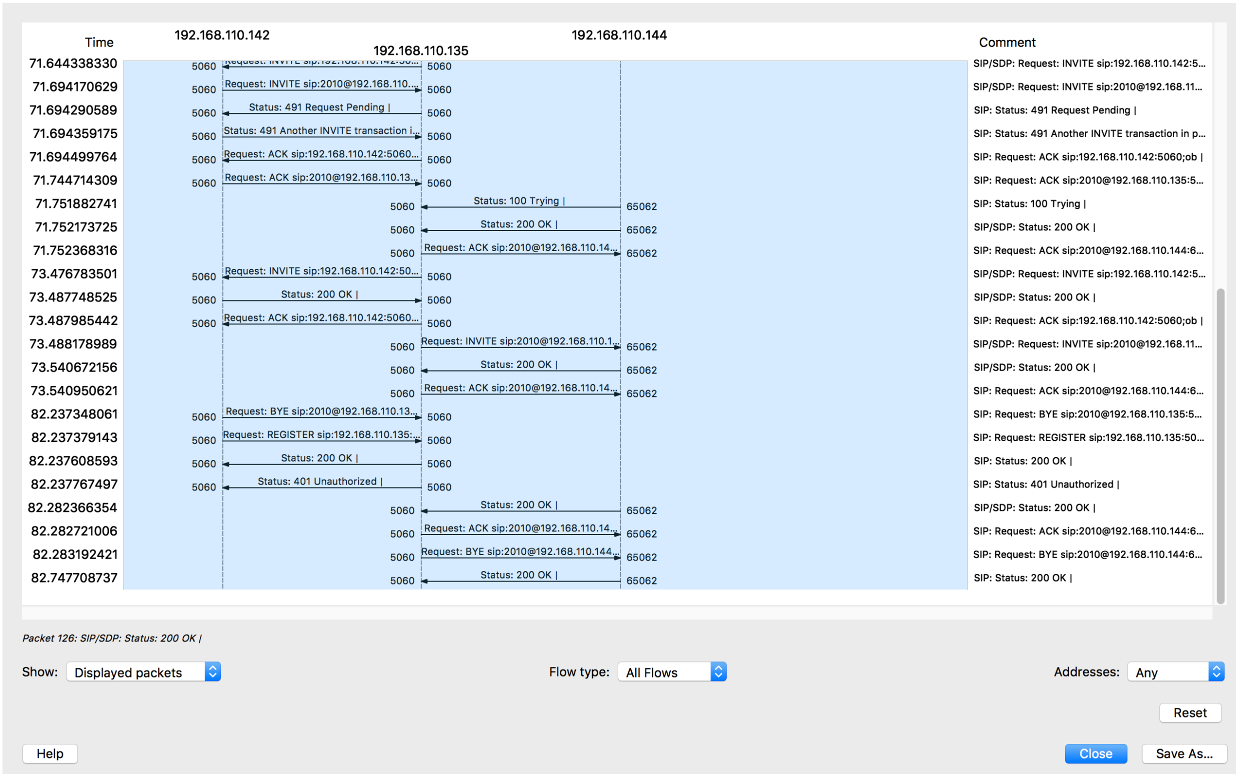


Figure : 2.19

SIP Flows:

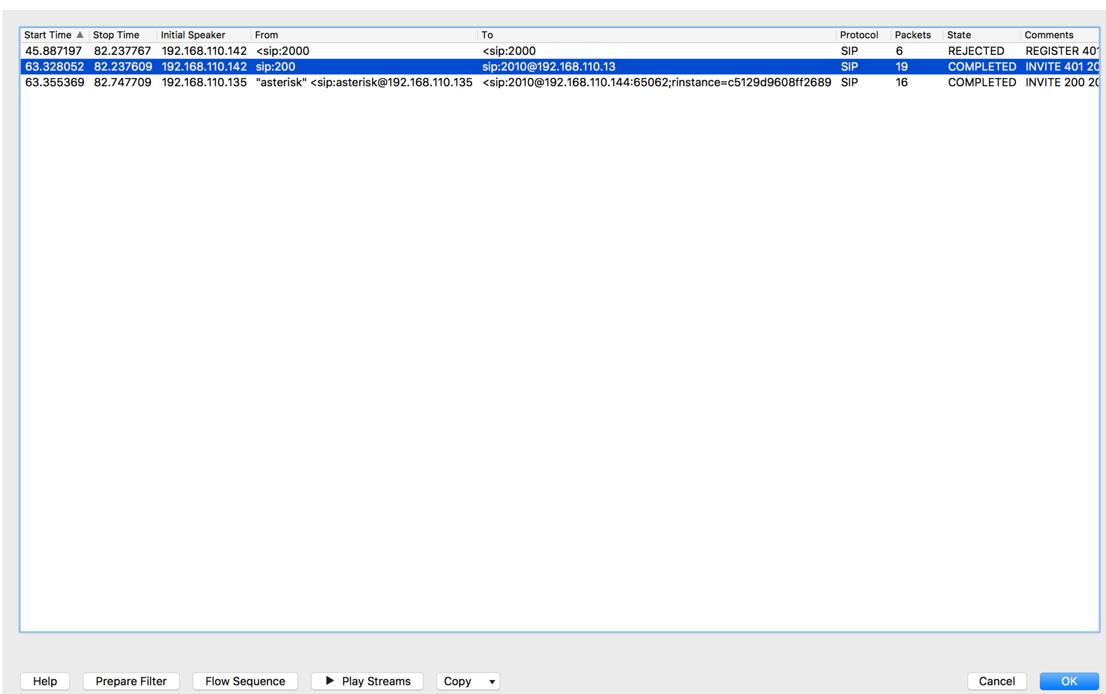


Figure : 2.20

The figure shows registration process of user 2000 via SIP Register and 200 OK

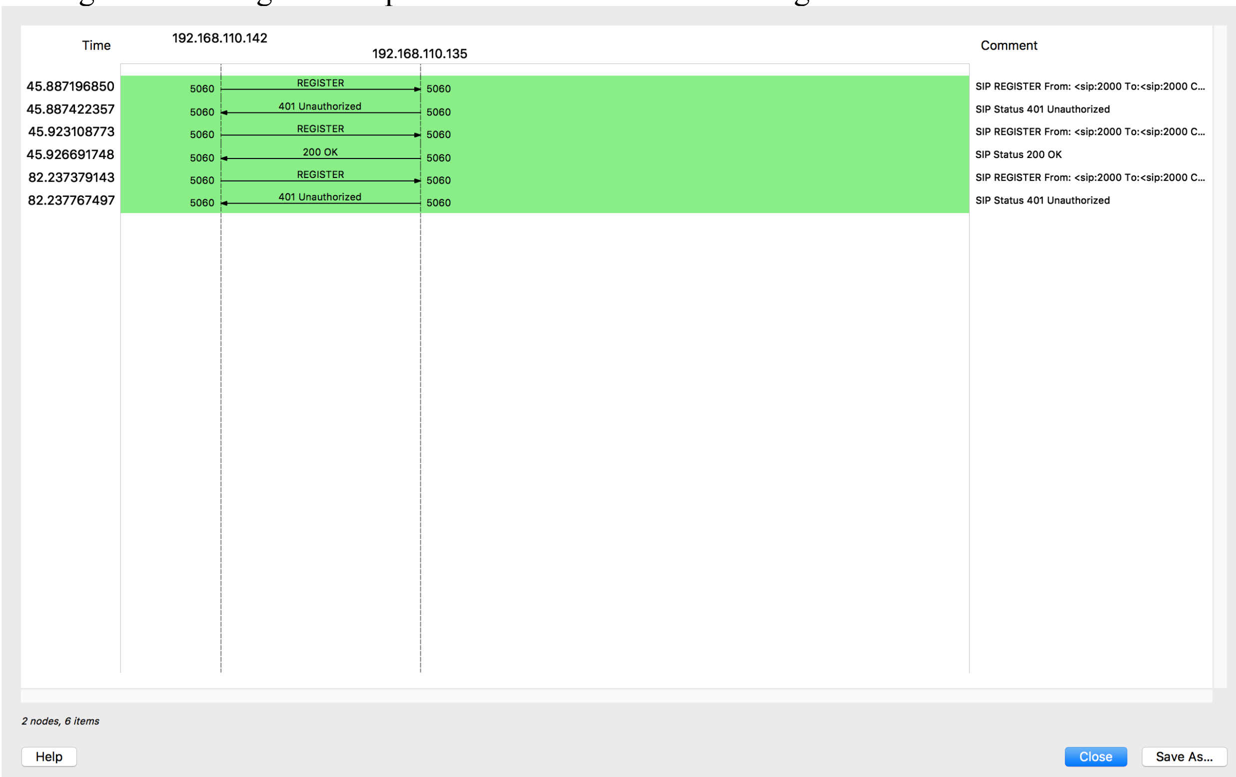


Figure : 2.21

The flow diagram shows INVITE from 2000 to 2010 for a call following the SIP 3 way handshake and establishment of call

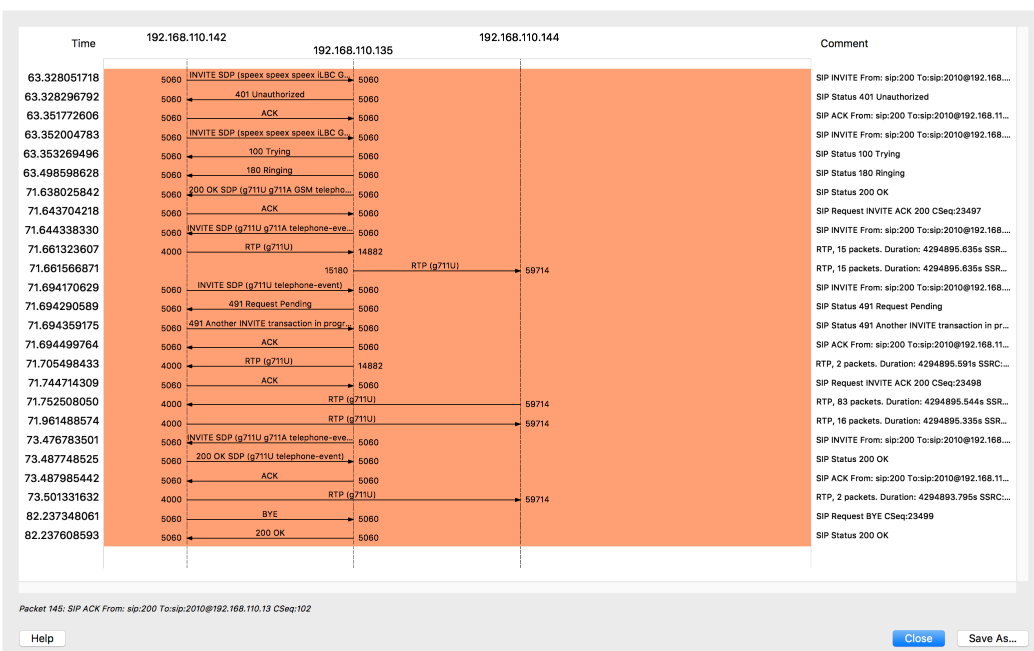


Figure : 2.22

This flow shows communication between asterisk server and SIP client 2010 for establishing a call with 2000 at 2010's side. The call is established at 73.54 seconds with the establishment of an RTP stream.

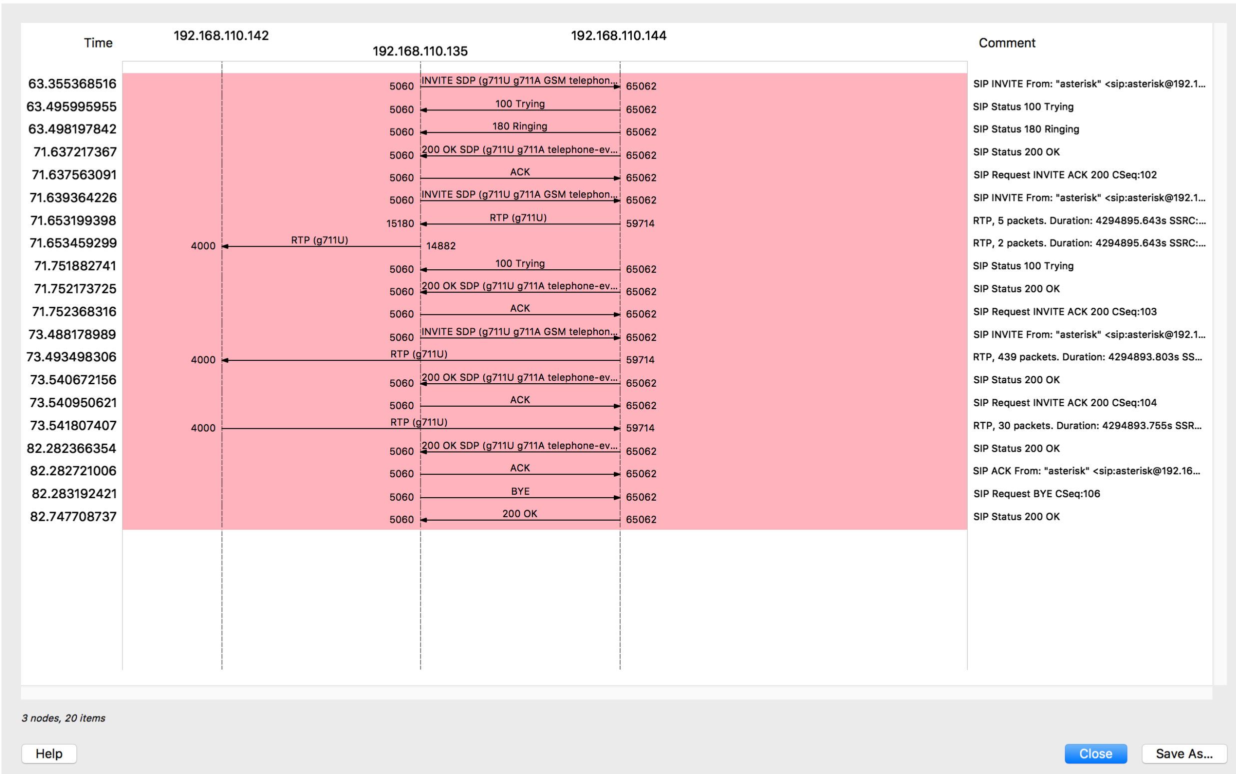


Figure : 2.23

3.SIP Packet Fields:

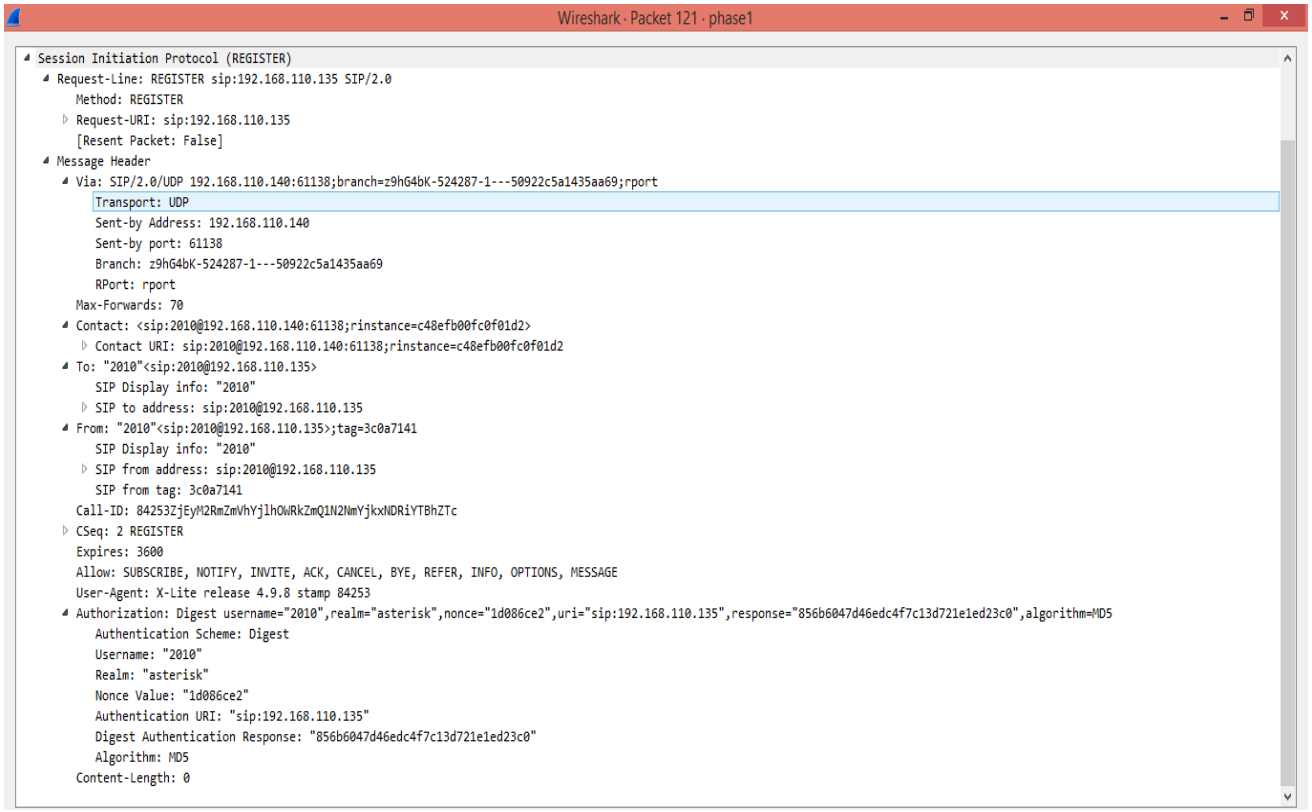


Figure : 3.1

Request Line

It shows the method of the SIP packet which is REGISTER in this case and gives the destination or the server address on which it wants to register. That is 192.168.110.135 Asterisk server address in our case.

Via

It shows the source IP address and port number of the SIP packet, transport protocol to be used and the version of SIP which is used.

Contact

This field contains the source or sender's SIP URI that can be used to contact the sender.

TO

It contains the address of the recipient of the request. It may or may not contain the address of the final destination. In our case display name 2010 is registering to Asterisk server thus it contains IP address of the asterisk server which is 192.168.110.135.

FROM

It consists of identity of the initiator of the request from the point of view of the server which of the form "user@domain" that is "2010@192.168.110.135" in our case

ALLOW

This field gives the information about the SIP Methods the caller can support which is SUBSCRIBE, BYE, ACK etc. in our case.

Figure : 3.2

For the above packet all the SIP Packet fields stays the same as it was for SIP Register packet but SIP Method used is INVITE instead of REGISTER.
Here source also shows the user agent it uses under the user agent field which is X-Lite in our case.

4.SIP 3 way handshake:

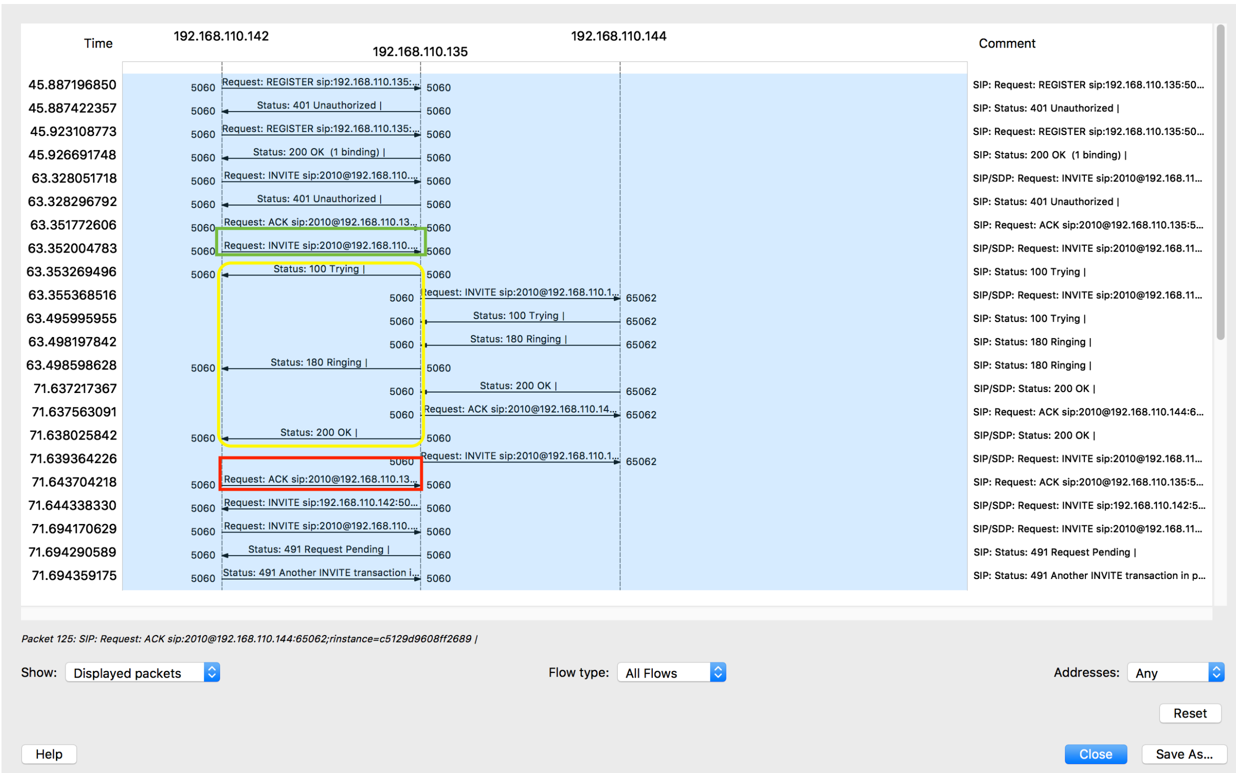


Figure : 4.1

The figure: 4.1 shows a typical SIP 3 way handshake observed during our experiment. The green box initiates the SIP 3 way handshake with the request INVITE to establish the call. The yellow box shows second part of 3 way handshake where the 100 Trying, 180Ringing and eventually 200 OK messages are sent from server to the calling client to which the calling client responds with Request_ACK as shown in the red box.

5.Ad-hoc Networks:

Ad hoc network

- Ad-hoc network is a network of devices that communicate with each other directly using Ethernet or wireless cards.
- Using this network devices can communicate without routers or access points.
- It is usually built spontaneously in Local Area Network (LAN) for a specific application or purpose i.e. it is a temporary network.
- Classification of Ad-hoc networks
    - Mobile ad-hoc networks (MANET)
    - Vehicular ad-hoc networks (VANET)
    - Internet Based mobile ad-hoc networks (iMANETS)
    - Military ad-hoc networks.
- Pros
    - Infrastructureless.
    - Low Cost.
    - Less Setup time.
    - Connects Automatically and adapts changes.