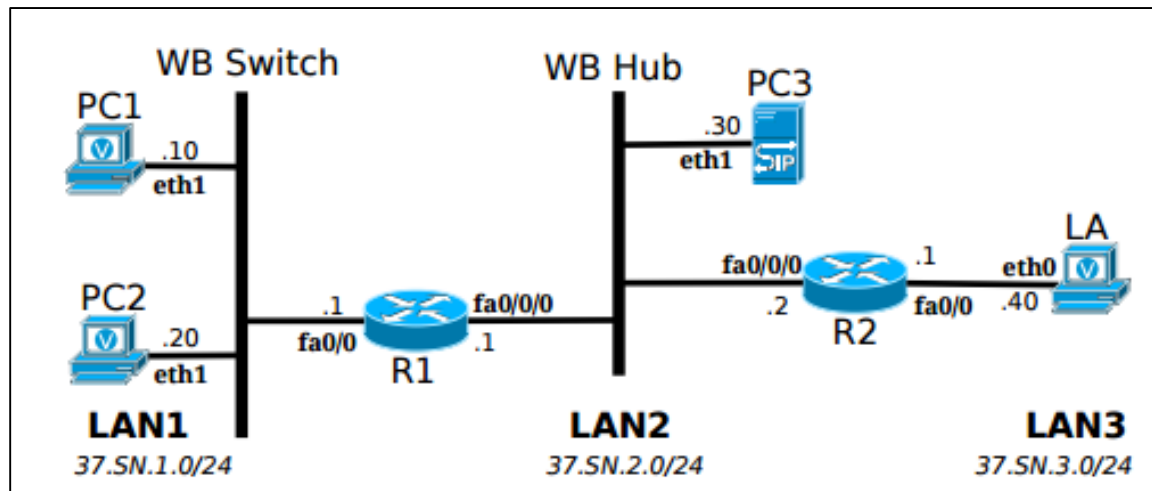


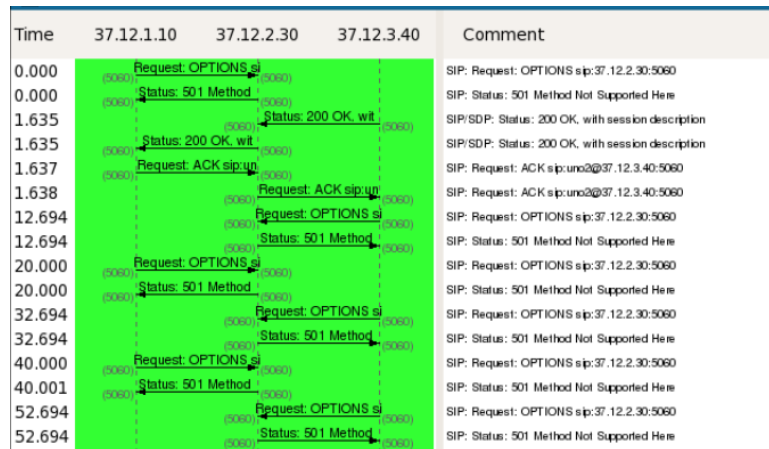
## Experiment 4: Topology



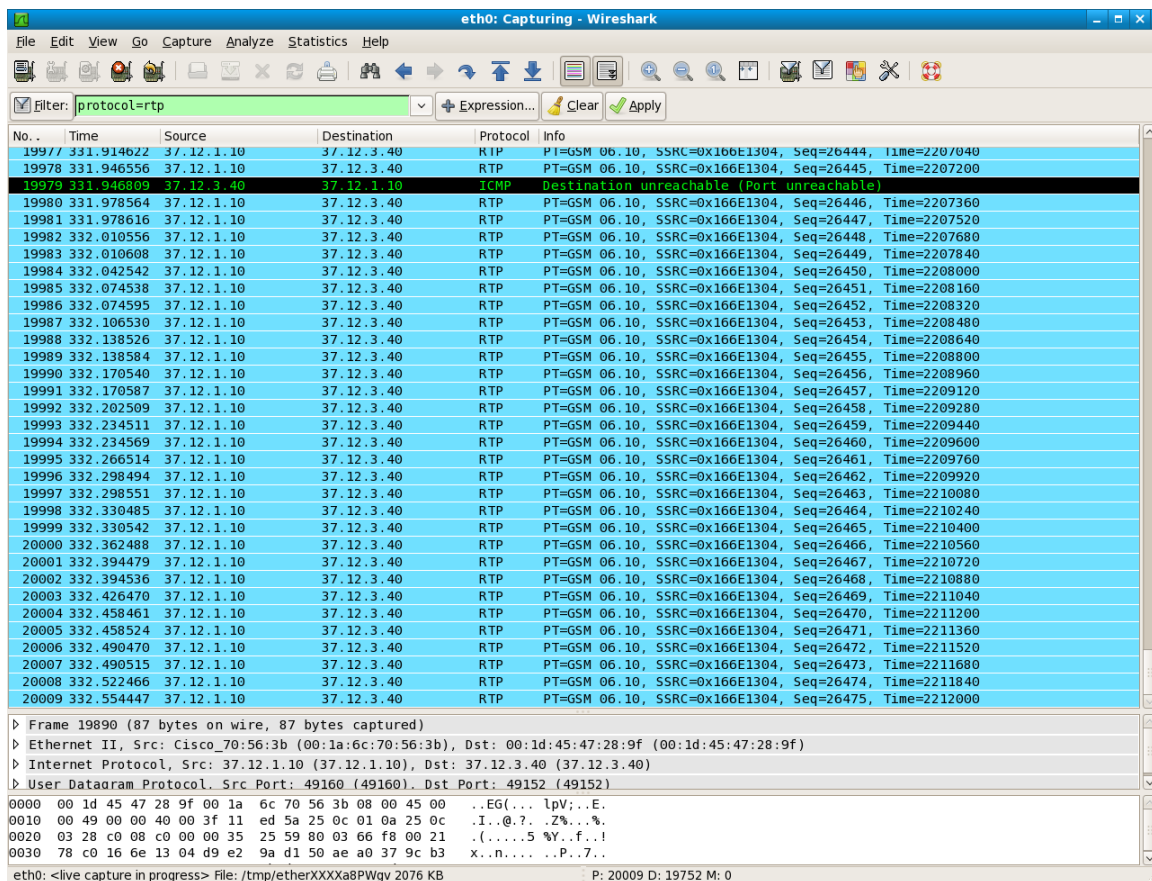
- The devices were connected as shown in the topology above.
- Interface eth0 of PC3 was connected to hub and Wireshark was started to capture the traffic between the LANs.
- Static routes were added on PC3 and the routers to achieve complete connectivity and using ping the connectivity was confirmed.

### Experiment 4.1: Direct communication between user agents

- The SJphone user agent was launched on the PC1 and laptop by executing the command `sjphone.sh` on the Linux command line.
- The name was set by filling the name field.
- The phone was configured to make a PC-PC call using the SIP signaling protocol.
- In the address box of the phone interface of PC1, the complete SIP address of the user to be called was entered in the form `<username>@<user agent IP address>`, and the call was started by clicking dial.
- We were able to hear the ringtone on the laptop.
- The call connection setup and the teardown can be seen in the Wireshark capture below.
- As we can see, the PC phone @ 37.12.3.40 was called from the PC phone @ 37.12.1.10. And we see the initial call trying, and the ringing phase. Finally, once the call is completed, and the hang-up button is pressed we see the BYE messages as well. The flow graph of the same transaction is also shown below.



- From the captures we can see that the port number for the signaling flow is **5060** which is the SIP protocol.
- The voice packets i.e. RTP packets are also shown below.



- The captures show us the voice transactions occurring between the 2 PC phones.
- The similar transactions are shown in the flow graph below.

Graph Analysis			
Time	37.12.1.10	37.12.3.40	Comment
1.643	PT=GSM 06.10, SSRC=0x490AA08C, Seq=45595, Time=0, Mask		RTP: PT=GSM 06.10, SSRC=0x490AA08C, Seq=45595, Time=0, Mask
1.675	PT=GSM 06.10, SSRC=0x490AA08C, Seq=45596, Time=160		RTP: PT=GSM 06.10, SSRC=0x490AA08C, Seq=45596, Time=160
1.675	PT=GSM 06.10, SSRC=0x490AA08C, Seq=45597, Time=320		RTP: PT=GSM 06.10, SSRC=0x490AA08C, Seq=45597, Time=320
1.695	PT=GSM 06.10, SSRC=0x82F0280C, Seq=31025, Time=0, Mask		RTP: PT=GSM 06.10, SSRC=0x82F0280C, Seq=31025, Time=0, Mask
1.696	PT=GSM 06.10, SSRC=0x82F0280C, Seq=31027, Time=160		RTP: PT=GSM 06.10, SSRC=0x82F0280C, Seq=31027, Time=160
1.696	PT=GSM 06.10, SSRC=0x82F0280C, Seq=31028, Time=320		RTP: PT=GSM 06.10, SSRC=0x82F0280C, Seq=31028, Time=320
1.707	PT=GSM 06.10, SSRC=0x490AA08C, Seq=45598, Time=480		RTP: PT=GSM 06.10, SSRC=0x490AA08C, Seq=45598, Time=480
1.738	PT=GSM 06.10, SSRC=0x490AA08C, Seq=45599, Time=640		RTP: PT=GSM 06.10, SSRC=0x490AA08C, Seq=45599, Time=640
1.739	PT=GSM 06.10, SSRC=0x490AA08C, Seq=45600, Time=800		RTP: PT=GSM 06.10, SSRC=0x490AA08C, Seq=45600, Time=800
1.759	PT=GSM 06.10, SSRC=0x82F0280C, Seq=31029, Time=480		RTP: PT=GSM 06.10, SSRC=0x82F0280C, Seq=31029, Time=480
1.760	PT=GSM 06.10, SSRC=0x82F0280C, Seq=31030, Time=640		RTP: PT=GSM 06.10, SSRC=0x82F0280C, Seq=31030, Time=640
1.760	PT=GSM 06.10, SSRC=0x82F0280C, Seq=31031, Time=800		RTP: PT=GSM 06.10, SSRC=0x82F0280C, Seq=31031, Time=800
1.771	PT=GSM 06.10, SSRC=0x490AA08C, Seq=45601, Time=960		RTP: PT=GSM 06.10, SSRC=0x490AA08C, Seq=45601, Time=960
1.771	PT=GSM 06.10, SSRC=0x490AA08C, Seq=45602, Time=1120		RTP: PT=GSM 06.10, SSRC=0x490AA08C, Seq=45602, Time=1120
1.803	PT=GSM 06.10, SSRC=0x490AA08C, Seq=45603, Time=1280		RTP: PT=GSM 06.10, SSRC=0x490AA08C, Seq=45603, Time=1280
1.823	PT=GSM 06.10, SSRC=0x82F0280C, Seq=31032, Time=960		RTP: PT=GSM 06.10, SSRC=0x82F0280C, Seq=31032, Time=960
1.823	PT=GSM 06.10, SSRC=0x82F0280C, Seq=31033, Time=1120		RTP: PT=GSM 06.10, SSRC=0x82F0280C, Seq=31033, Time=1120
1.824	PT=GSM 06.10, SSRC=0x82F0280C, Seq=31034, Time=1280		RTP: PT=GSM 06.10, SSRC=0x82F0280C, Seq=31034, Time=1280
1.834	PT=GSM 06.10, SSRC=0x490AA08C, Seq=45604, Time=1440		RTP: PT=GSM 06.10, SSRC=0x490AA08C, Seq=45604, Time=1440
1.835	PT=GSM 06.10, SSRC=0x490AA08C, Seq=45605, Time=1600		RTP: PT=GSM 06.10, SSRC=0x490AA08C, Seq=45605, Time=1600
1.867	PT=GSM 06.10, SSRC=0x490AA08C, Seq=45606, Time=1760		RTP: PT=GSM 06.10, SSRC=0x490AA08C, Seq=45606, Time=1760
1.887	PT=GSM 06.10, SSRC=0x82F0280C, Seq=31035, Time=1440		RTP: PT=GSM 06.10, SSRC=0x82F0280C, Seq=31035, Time=1440
1.887	PT=GSM 06.10, SSRC=0x82F0280C, Seq=31036, Time=1600		RTP: PT=GSM 06.10, SSRC=0x82F0280C, Seq=31036, Time=1600
1.888	PT=GSM 06.10, SSRC=0x82F0280C, Seq=31037, Time=1760		RTP: PT=GSM 06.10, SSRC=0x82F0280C, Seq=31037, Time=1760
1.898	PT=GSM 06.10, SSRC=0x490AA08C, Seq=45607, Time=1920		RTP: PT=GSM 06.10, SSRC=0x490AA08C, Seq=45607, Time=1920
1.899	PT=GSM 06.10, SSRC=0x490AA08C, Seq=45608, Time=2080		RTP: PT=GSM 06.10, SSRC=0x490AA08C, Seq=45608, Time=2080
1.919	PT=GSM 06.10, SSRC=0x82F0280C, Seq=31038, Time=1920		RTP: PT=GSM 06.10, SSRC=0x82F0280C, Seq=31038, Time=1920
1.919	PT=GSM 06.10, SSRC=0x82F0280C, Seq=31039, Time=2080		RTP: PT=GSM 06.10, SSRC=0x82F0280C, Seq=31039, Time=2080
1.930	PT=GSM 06.10, SSRC=0x490AA08C, Seq=45609, Time=2240		RTP: PT=GSM 06.10, SSRC=0x490AA08C, Seq=45609, Time=2240
1.931	PT=GSM 06.10, SSRC=0x490AA08C, Seq=45610, Time=2400		RTP: PT=GSM 06.10, SSRC=0x490AA08C, Seq=45610, Time=2400
1.962	PT=GSM 06.10, SSRC=0x490AA08C, Seq=45611, Time=2560		RTP: PT=GSM 06.10, SSRC=0x490AA08C, Seq=45611, Time=2560
1.983	PT=GSM 06.10, SSRC=0x82F0280C, Seq=31040, Time=2240		RTP: PT=GSM 06.10, SSRC=0x82F0280C, Seq=31040, Time=2240
1.983	PT=GSM 06.10, SSRC=0x82F0280C, Seq=31041, Time=2400		RTP: PT=GSM 06.10, SSRC=0x82F0280C, Seq=31041, Time=2400
1.984	PT=GSM 06.10, SSRC=0x82F0280C, Seq=31042, Time=2560		RTP: PT=GSM 06.10, SSRC=0x82F0280C, Seq=31042, Time=2560
1.994	PT=GSM 06.10, SSRC=0x490AA08C, Seq=45612, Time=2720		RTP: PT=GSM 06.10, SSRC=0x490AA08C, Seq=45612, Time=2720
1.995	PT=GSM 06.10, SSRC=0x490AA08C, Seq=45613, Time=2880		RTP: PT=GSM 06.10, SSRC=0x490AA08C, Seq=45613, Time=2880
2.026	PT=GSM 06.10, SSRC=0x490AA08C, Seq=45614, Time=3040		RTP: PT=GSM 06.10, SSRC=0x490AA08C, Seq=45614, Time=3040
2.047	PT=GSM 06.10, SSRC=0x82F0280C, Seq=31043, Time=2720		RTP: PT=GSM 06.10, SSRC=0x82F0280C, Seq=31043, Time=2720
2.047	PT=GSM 06.10, SSRC=0x82F0280C, Seq=31044, Time=2880		RTP: PT=GSM 06.10, SSRC=0x82F0280C, Seq=31044, Time=2880
2.048	PT=GSM 06.10, SSRC=0x82F0280C, Seq=31045, Time=3040		RTP: PT=GSM 06.10, SSRC=0x82F0280C, Seq=31045, Time=3040
2.058	PT=GSM 06.10, SSRC=0x490AA08C, Seq=45615, Time=3200		RTP: PT=GSM 06.10, SSRC=0x490AA08C, Seq=45615, Time=3200
2.058	PT=GSM 06.10, SSRC=0x490AA08C, Seq=45616, Time=3360		RTP: PT=GSM 06.10, SSRC=0x490AA08C, Seq=45616, Time=3360

- From the captures, we can see that the port on which the RTP transaction occurs is **49152**.
- All the SJPhone instances were shutdown.

#### Experiment 4.2: Communication via a SIP proxy server

- Using the following steps, the SIP Proxy server was configured.

```
sudo /etc/init.d/mysqld start
sudo openserdbctl drop
sudo openserdbctl create
cp /etc/openser/openser-nonat.cfg /etc/openser/openser.cfg
```

- The configuration file at /etc/openser/openser.cfg was opened in a text editor. At the listen directive, the part between the colons was changed to the IP address of the SIP proxy server i.e. **37.12.2.30**.
- The openser daemon was started. Ensured that it is working.
- The SIP domain was created using the following steps and SIP users were added using the steps below.

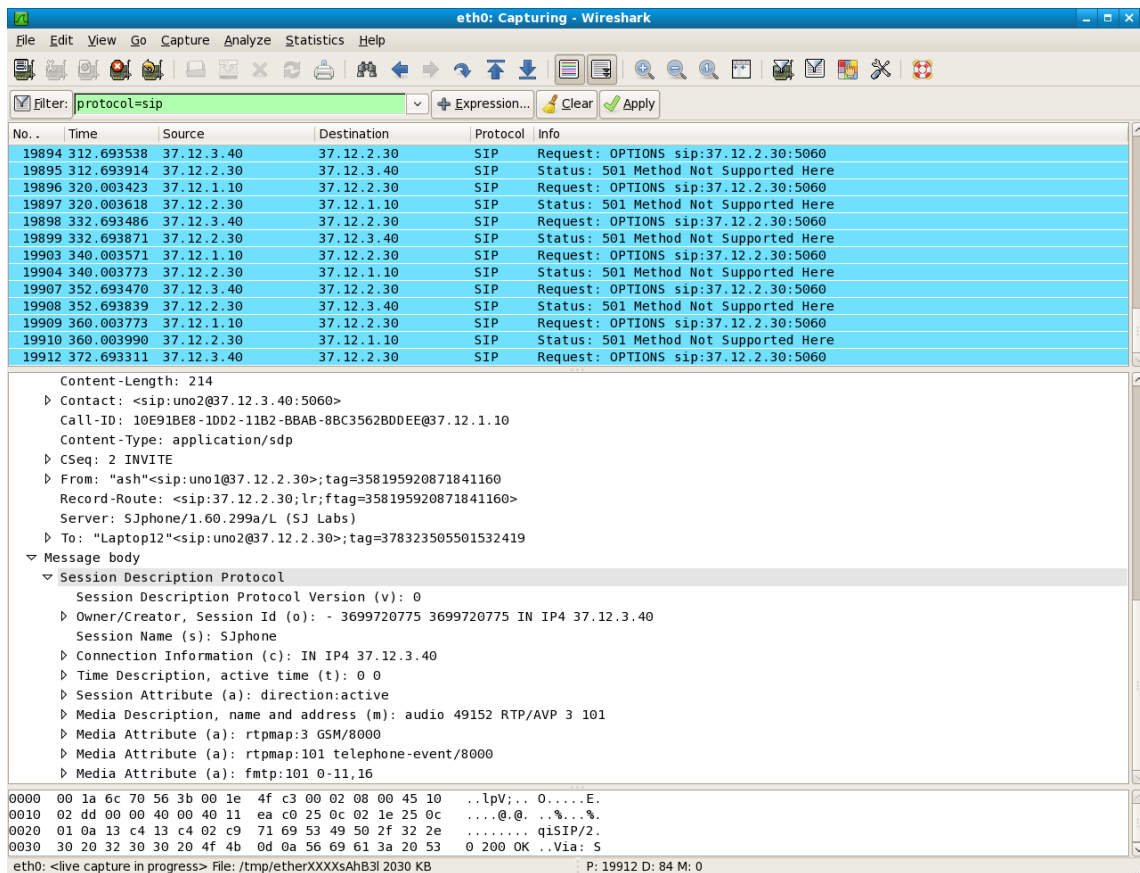
```
sudo openserctl domain add 37.12.2.30
sudo openserctl domain reload
sudo openserctl domain show
sudo openserctl add uno1@37.12.2.30 openserrw uno1@pitt.edu
```

```
sudo openserctl add uno2@37.12.2.30 openserrw uno2@pitt.edu
```

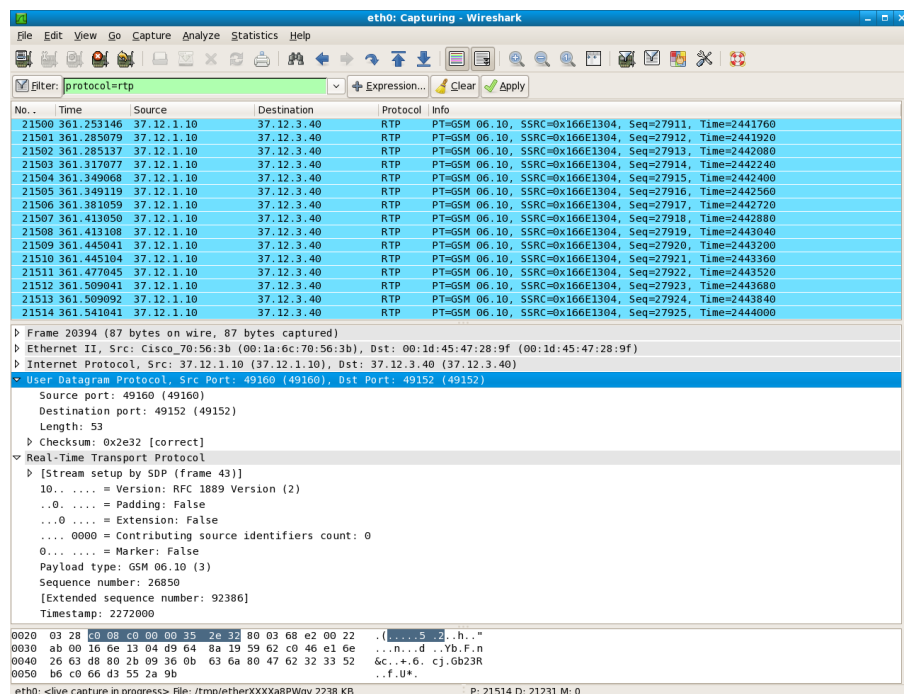
- SJPhone was launched on PC1, and configured to register with SIP proxy server. After the configuration, we were able to see the message “ready to call” which indicates that the user is registered with the SIP proxy server.
- Similar steps were carried out on the SJPhone application on the laptop and it was registered with the SIP proxy server.

```
[team12@netlab-wb2pc3 ~]$ sudo openserctl ul show
Domain:: location table=512 records=2 max_slot=1
AOR:: uno2
    Contact:: sip:uno2@37.12.3.40:5060 Q=0
        Expires:: 3557
        Callid:: B6BA4848-1DD1-11B2-AAB7-BD0D33A5B951@37.12.3.40
        Cseq:: 2
        User-agent:: SJphone/1.60.299a/L (SJ Labs)
        State:: CS_SYNC
        Flags:: 0
        Cflag:: 0
        Socket:: udp:37.12.2.30:5060
        Methods:: 4294967295
AOR:: uno1
    Contact:: sip:uno1@37.12.1.10:5060 Q=0
        Expires:: 3324
        Callid:: 4F13C162-1DD2-11B2-BBAA-8BC3562BDDEE@37.12.1.10
        Cseq:: 2
        User-agent:: SJphone/1.60.299a/L (SJ Labs)
        State:: CS_SYNC
        Flags:: 0
        Cflag:: 0
        Socket:: udp:37.12.2.30:5060
        Methods:: 4294967295
```

- Now the call was placed between the 2 user agents registered with the SIP proxy using just the username.
- The connection setup captures are shown. As it can be the calling party is checking with the SIP server **37.12.2.30**. The SIP server in turn places the call to the called party at **37.12.3.40**.
- Also it can be seen in the packet details below, at the end of the call initialization, the SIP server is sending the IP information of the calling party to the called party. And the RTP port **49152** is also shared.
- The RTP transaction occurs directly between the user agents on the IP and port, highlighted in the snapshot above. RTP captures are shown below.



- The RTP transaction occurs directly between the user agents on the IP and port, highlighted in the snapshot above. RTP captures are shown below.



- All the SJPhone instances were closed.

### **Experiment 4.3: NAT issues with VOIP Traffic – Router based solution**

- The IP addresses on LAN3 were changed to the subnet 10.31.0.0/24.
- On router R2, NAT with overload was configured so that hosts in LAN3 can access the outside world using the public IP of the interface of fa0/0/0 on router R2.
- Confirmed connectivity to LAN1 and LAN2 from LAN3 using ping.
- Wireshark was started on the laptop to monitor eth0.
- SJPhone application was launched on PC1 and Laptop.
- Registered users on the SIP server are shown below.

```
team12@netlab-wb2pc3:~$ sudo openseectl domain reload
[team12@netlab-wb2pc3 ~]$ sudo openseectl domain show
37.12.2.30
[team12@netlab-wb2pc3 ~]$ sudo openseectl add uno1@37.12.2.30 openserrw uno1@pitt.edu
new user 'uno1@37.12.2.30' added
[team12@netlab-wb2pc3 ~]$ sudo openseectl add uno2@37.12.2.30 openserrw uno2@pitt.edu
new user 'uno2@37.12.2.30' added
[team12@netlab-wb2pc3 ~]$ sudo openseectl db show subscriber
-----
| id | username | domain | password | first_name | last_name | email_address |
|-----|-----|-----|-----|-----|-----|-----|
| 1 | uno1 | 37.12.2.30 | openserrw | | | uno1@pitt.edu |
| 2 | uno2 | 37.12.2.30 | openserrw | | | uno2@pitt.edu |
-----
[team12@netlab-wb2pc3 ~]$ sudo openseectl ul show
Domain:: location table=512 records=2 max_slot=1
AOR:: uno2
Contact:: sip:uno2@37.12.3.40:5060 Q=0
Expires:: 3557
Callid:: 868A4848-10D1-11B2-AAB7-BD0033A5B951@37.12.3.40
Cseq:: 2
User-agent:: SJphone/1.60.299a/L (SJ Labs)
State:: CS_SYNC
Flags:: 0
Cflag:: 0
Socket:: udp:37.12.2.30:5060
Methods:: 4294967295
AOR:: uno1
Contact:: sip:uno1@37.12.1.10:5060 Q=0
Expires:: 3324
Callid:: 4F13C162-10D2-11B2-BBAA-8BC3562B0DDE@37.12.1.10
Cseq:: 2
User-agent:: SJphone/1.60.299a/L (SJ Labs)
State:: CS_SYNC
Flags:: 0
Cflag:: 0
Socket:: udp:37.12.2.30:5060
Methods:: 4294967295
[team12@netlab-wb2pc3 ~]$
```

- Call was started from PC1 to the Laptop. The call was successful as the Cisco 2811 routers are equipped with Application-level Gateway for SIP traffic and they are capable of parsing the SIP registration and apply the necessary NAT-traversal measures.
- The call setup and teardown traffic was captured. Captures are shown below. We can see the INVITE, ACK and the BYE messages.



eth0: Capturing - Wireshark

Filter: protocol==sip

No.	Time	Source	Destination	Protocol	Info
3630	174.001440	37.12.2.30	37.12.2.10	SIP	Status: 501 Method Not Supported Here
3641	174.138883	37.12.2.2	37.12.2.30	SIP	Request: OPTIONS sip:37.12.2.30:5060
3642	174.139277	37.12.2.30	37.12.2.2	SIP	Status: 501 Method Not Supported Here
3646	182.081221	37.12.1.10	37.12.2.30	SIP	Request: OPTIONS sip:37.12.2.30:5060
3647	182.081640	37.12.2.30	37.12.1.10	SIP	Status: 501 Method Not Supported Here
3648	194.137686	37.12.2.2	37.12.2.30	SIP	Request: OPTIONS sip:37.12.2.30:5060
3649	194.138054	37.12.2.30	37.12.2.2	SIP	Status: 501 Method Not Supported Here
3650	202.081462	37.12.1.10	37.12.2.30	SIP	Request: OPTIONS sip:37.12.2.30:5060
3651	202.081877	37.12.2.30	37.12.1.10	SIP	Status: 501 Method Not Supported Here
3655	214.136614	37.12.2.2	37.12.2.30	SIP	Request: OPTIONS sip:37.12.2.30:5060
3656	214.136949	37.12.2.30	37.12.2.2	SIP	Status: 501 Method Not Supported Here
3659	222.081666	37.12.1.10	37.12.2.30	SIP	Request: OPTIONS sip:37.12.2.30:5060
3660	222.082075	37.12.2.30	37.12.1.10	SIP	Status: 501 Method Not Supported Here
3662	234.135573	37.12.2.2	37.12.2.30	SIP	Request: OPTIONS sip:37.12.2.30:5060
3663	234.135944	37.12.2.30	37.12.2.2	SIP	Status: 501 Method Not Supported Here
3671	242.081882	37.12.1.10	37.12.2.30	SIP	Request: OPTIONS sip:37.12.2.30:5060
3672	242.082249	37.12.2.30	37.12.1.10	SIP	Status: 501 Method Not Supported Here
3675	254.134492	37.12.2.2	37.12.2.30	SIP	Request: OPTIONS sip:37.12.2.30:5060
3676	254.134901	37.12.2.30	37.12.2.2	SIP	Status: 501 Method Not Supported Here
3679	260.368482	37.12.1.10	37.12.2.30	SIP/SDP	Request: INVITE sip:uno2@37.12.2.30, with session description
3680	260.368701	37.12.2.30	37.12.1.10	SIP	Status: 100 Trying
3681	260.368748	37.12.2.30	37.12.1.10	SIP	Status: 407 Proxy Authentication Required
3682	260.372633	37.12.1.10	37.12.2.30	SIP	Request: ACK sip:uno2@37.12.2.30
3683	260.373275	37.12.1.10	37.12.2.30	SIP/SDP	Request: INVITE sip:uno2@37.12.2.30, with session description
3684	260.375569	37.12.2.30	37.12.1.10	SIP	Status: 100 Trying
3685	260.376005	37.12.2.30	37.12.2.2	SIP/SDP	Request: INVITE sip:uno2@37.12.2.2:1025, with session description
3686	260.385617	37.12.2.2	37.12.2.30	SIP	Status: 100 Trying
3687	260.685307	37.12.2.2	37.12.2.30	SIP	Status: 180 Ringing
3688	260.685519	37.12.2.30	37.12.1.10	SIP	Status: 180 Ringing
3689	262.082384	37.12.1.10	37.12.2.30	SIP	Request: OPTIONS sip:37.12.2.30:5060
3690	262.082771	37.12.2.30	37.12.1.10	SIP	Status: 501 Method Not Supported Here
3692	274.133421	37.12.2.2	37.12.2.30	SIP	Request: OPTIONS sip:37.12.2.30:5060
3693	274.133804	37.12.2.30	37.12.2.2	SIP	Status: 501 Method Not Supported Here
3694	274.205783	37.12.2.2	37.12.2.30	SIP/SDP	Status: 200 OK, with session description
3695	274.206106	37.12.2.30	37.12.1.10	SIP/SDP	Status: 200 OK, with session description
3696	274.208335	37.12.1.10	37.12.2.30	SIP	Request: ACK sip:uno2@37.12.2.2:1025
3697	274.208691	37.12.2.30	37.12.2.2	SIP	Request: ACK sip:uno2@37.12.2.2:1025
4341	280.641335	37.12.2.2	37.12.2.30	SIP	Request: BYE sip:uno1@37.12.1.10:5060
4342	280.641799	37.12.2.30	37.12.1.10	SIP	Request: BYE sip:uno1@37.12.1.10:5060
4343	280.643436	37.12.1.10	37.12.2.30	SIP	Status: 200 OK
4344	280.643728	37.12.2.30	37.12.2.2	SIP	Status: 200 OK
4345	282.083275	37.12.1.10	37.12.2.30	SIP	Request: OPTIONS sip:37.12.2.30:5060

eth0: <live capture in progress> File: /tmp/etherXXXXH0Mife 470 KB P: 4358 D: 88 M: 0

- The SDP messages exchanged and the SDP flow on both PC1 and the Laptop are captured and shown below.

eth0: Capturing - Wireshark

Filter: protocol==sip

No.	Time	Source	Destination	Protocol	Info
3684	260.375569	37.12.2.30	37.12.1.10	SIP	Status: 100 Trying
3685	260.376005	37.12.2.30	37.12.2.2	SIP/SDP	Request: INVITE sip:uno2@37.12.2.2:1025, with session description
3686	260.385617	37.12.2.2	37.12.2.30	SIP	Status: 100 Trying
3687	260.685307	37.12.2.2	37.12.2.30	SIP	Status: 180 Ringing
3688	260.685519	37.12.2.30	37.12.1.10	SIP	Status: 180 Ringing
3689	262.082384	37.12.1.10	37.12.2.30	SIP	Request: OPTIONS sip:37.12.2.30:5060
3690	262.082771	37.12.2.30	37.12.1.10	SIP	Status: 501 Method Not supported Here
3692	274.133421	37.12.2.2	37.12.2.30	SIP	Request: OPTIONS sip:37.12.2.30:5060
3693	274.133804	37.12.2.30	37.12.2.2	SIP	Status: 501 Method Not Supported Here
3694	274.205783	37.12.2.2	37.12.2.30	SIP/SDP	Status: 200 OK, with session description
3695	274.206106	37.12.2.30	37.12.1.10	SIP/SDP	Status: 200 OK, with session description
3696	274.208335	37.12.1.10	37.12.2.30	SIP	Request: ACK sip:uno2@37.12.2.2:1025
3697	274.208691	37.12.2.30	37.12.2.2	SIP	Request: ACK sip:uno2@37.12.2.2:1025

▶ CSeq: 1 INVITE  
 ▶ From: "ash"<sip:uno1@37.12.2.30>;tag=3610540841768477263  
 Max-Forwards: 70  
 ▶ To: <sip:uno2@37.12.2.30>  
 User-Agent: SJphone/1.60.299a/L (SJ Labs)

▼ Message body

Session Description Protocol

Session Description Protocol Version (v): 0

Owner/Creator, Session Id (o): - 3699703505 3699703505 IN IP4 37.12.1.10

Session Name (s): SJphone

Connection Information (c): IN IP4 37.12.1.10

Time Description, active time (t): 0 0

Session Attribute (a): direction:active

Media Description, name and address (m): audio 49156 RTP/AVP 3 97 98 110 8 0 101

Media Attribute (a): rtpmap:3 GSM/8000

Media Attribute (a): rtpmap:97 iLBC/8000

Media Attribute (a): rtpmap:98 iLBC/8000

Media Attribute (a): fmp:98 mode=20

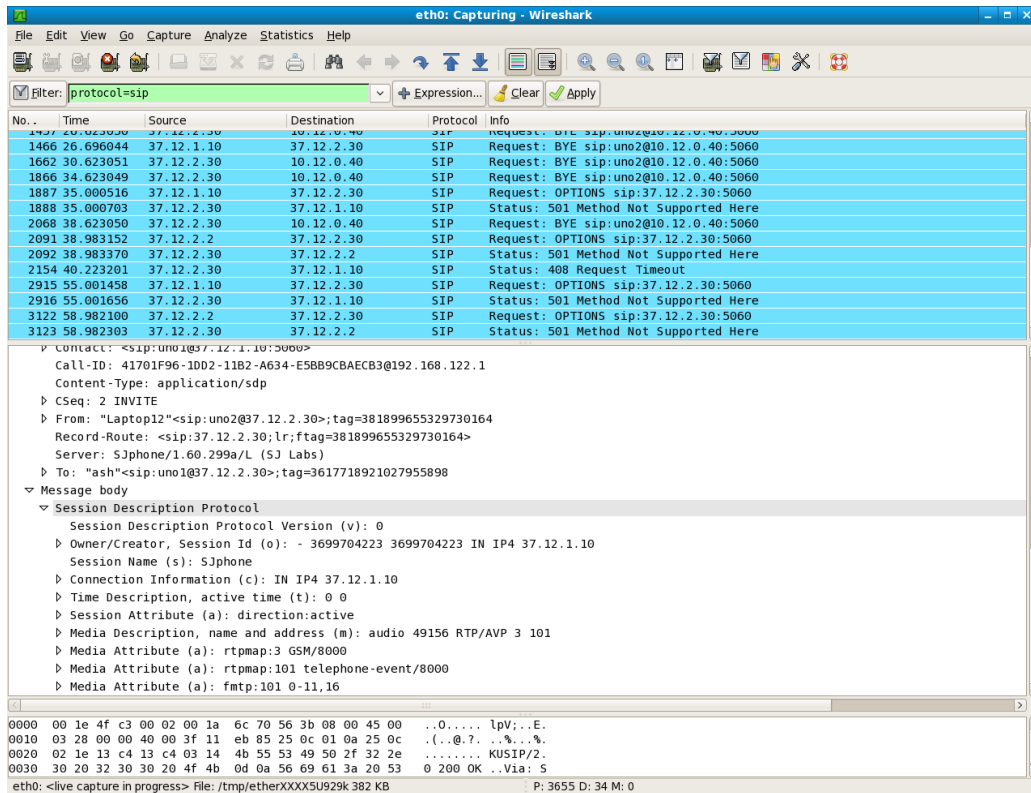
Media Attribute (a): rtpmap:110 speex/8000

Media Attribute (a): rtpmap:8 PCMA/8000

Media Attribute (a): rtpmap:0 PCMU/8000

0000 00 1e 4f c3 00 02 00 1a 6c 70 56 3b 08 00 45 00 ..0....lpV;..E.  
 0010 03 3b 00 00 40 00 3f 11 eb 72 25 0c 01 0a 25 0c ...@?..r%...%  
 0020 02 1e 13 c4 13 c4 03 27 27 80 49 4e 56 49 54 45 .....'.INVITE  
 0030 20 73 69 70 3a 75 6e 6f 32 40 33 37 2e 31 32 2e sip:uno 2@37.12.

eth0: <live capture in progress> File: /tmp/etherXXXXH0Mife 430 KB P: 4002 D: 74 M: 0



- As it can be seen from the laptop point of view, for SIP messages, it is always communicating with the SIP server and we see the private IP address of the Laptop. Whereas from the PC3 point of view, it observes that the request is coming from the PC1 to the SIP server and the SIP server is forwarding the requests to the public IP of the router R2. It does not know the private IP address of the Laptop. The router takes care of forwarding the messages to the Laptop received from the SIP server.
- All SJPhone instances were closed.

#### **Experiment 4.4: NAT issues with VOIP traffic – without routers support**

- Ensured that on PC3 there are 2 static routes and there is no default gateway set.
- On the router R2, SIP NAT was disabled using the commands given below.

```
no ip nat service sip tcp port 5060
no ip nat service sip udp port 5060
```

- The NAT translations were cleared on the router.
- The SJPhone instances were launched again and the call was repeated from the PC1 to Laptop.
- This call failed. As it can be seen in the captures below, the PC1 is not able to reach the Laptop now since the SIP NAT was disabled on the router. An ICMP unreachable message is sent back to the PC1.



eth0: Capturing - Wireshark

File Edit View Go Capture Analyze Statistics Help

Filter: protocol=sip

No.	Time	Source	Destination	Protocol	Info
21068	352.906356	37.12.3.40	37.12.2.30	SIP	Request: REGISTER sip:37.12.2.30:5060
21069	352.906488	37.12.2.30	37.12.3.40	ICMP	Destination unreachable (Port unreachable)
21070	352.907206	37.12.2.30	37.12.3.40	ICMP	Destination unreachable (Port unreachable)
21275	356.906258	37.12.3.40	37.12.2.30	SIP	Request: REGISTER sip:37.12.2.30:5060
21276	356.906392	37.12.2.30	37.12.3.40	ICMP	Destination unreachable (Port unreachable)
21277	356.907110	37.12.2.30	37.12.3.40	ICMP	Destination unreachable (Port unreachable)
21724	365.656075	37.12.3.40	37.12.2.30	SIP	Request: OPTIONS sip:37.12.2.30:5060
21725	365.656361	37.12.3.40	37.12.4.40	ICMP	Destination unreachable (Port unreachable)

Frame 43 (708 bytes on wire, 708 bytes captured)

Ethernet II, Src: 00:1d:45:47:28:9f (00:1d:45:47:28:9f), Dst: Cisco\_70:56:3b (00:1a:6c:70:56:3b)

Internet Protocol, Src: 37.12.3.40 (37.12.3.40), Dst: 37.12.1.10 (37.12.1.10)

User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)

Source port: sip (5060)  
Destination port: sip (5060)  
Length: 674  
Checksum: 0x7660 [correct]

Session Initiation Protocol

Status-Line: SIP/2.0 200 OK

Message Header

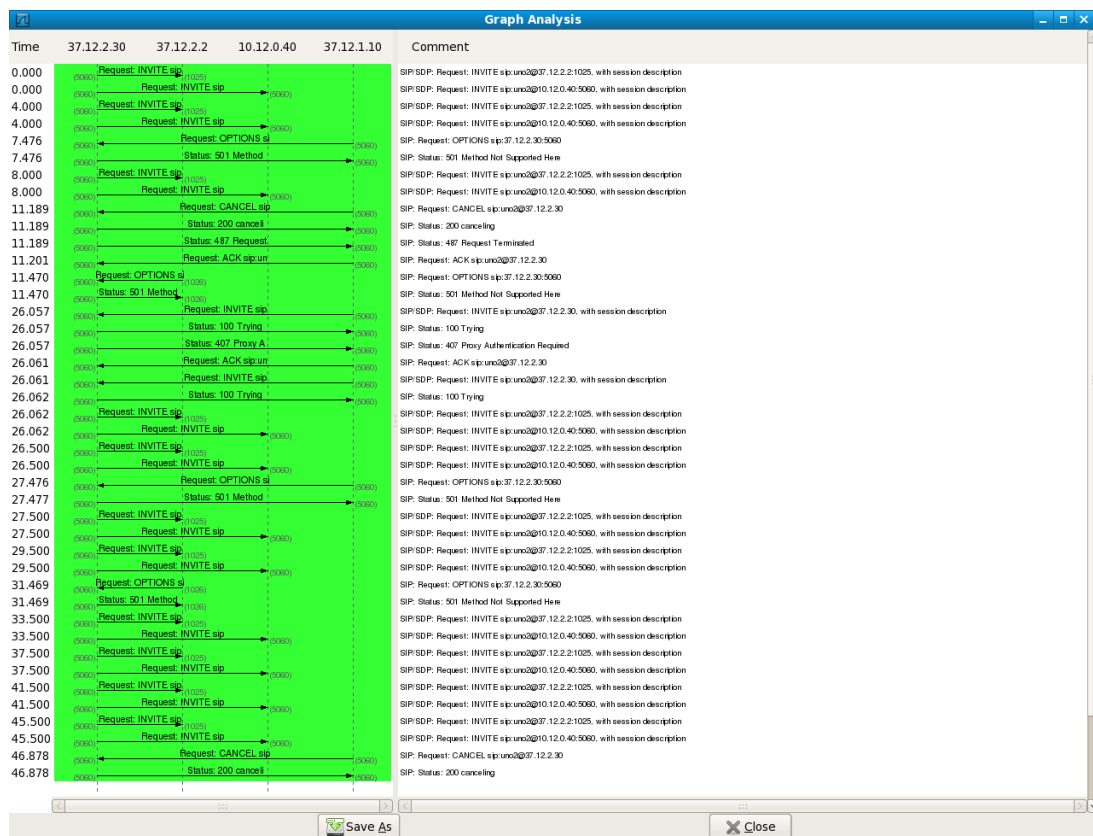
Via: SIP/2.0/UDP 37.12.1.10;rport=5060;received=37.12.1.10;branch=z9hG4bK250c010a0000002358da6dad3ebcf47d00000005  
Content-Length: 214  
Contact: <sip:team2@37.12.3.40:5060>  
Call-ID: 9BD2464E-1DD1-11B2-9F38-E2D374BFED8B@37.12.1.10  
Content-Type: application/sdp  
CSeq: 1 INVITE  
From: "ash"<sip:37.12.1.10>;tag=3562814901645483102  
Server: SJphone/1.60.299a/L (SJ Labs)  
To: "Laptop12"<sip:team12@37.12.3.40>;tag=3764304171474556021

Message body

Session Description Protocol

Session Description Protocol Version (v): 0  
Owner/Creator, Session Id (o): - 3699718882 3699718882 IN IP4 37.12.3.40  
Session Name (s): SJphone

eth0: <live capture in progress> File: /tmp/etherXXXa8PWgv 2521 KB P: 24284 D: 196 M: 0



- Now we tried a call from Laptop to PC1. This call was successful because a NAT overload was configured earlier on router R2 and since this connection is originating from inside LAN3 the router will overload the connection to its outside interface IP address and add a NAT translation entry.

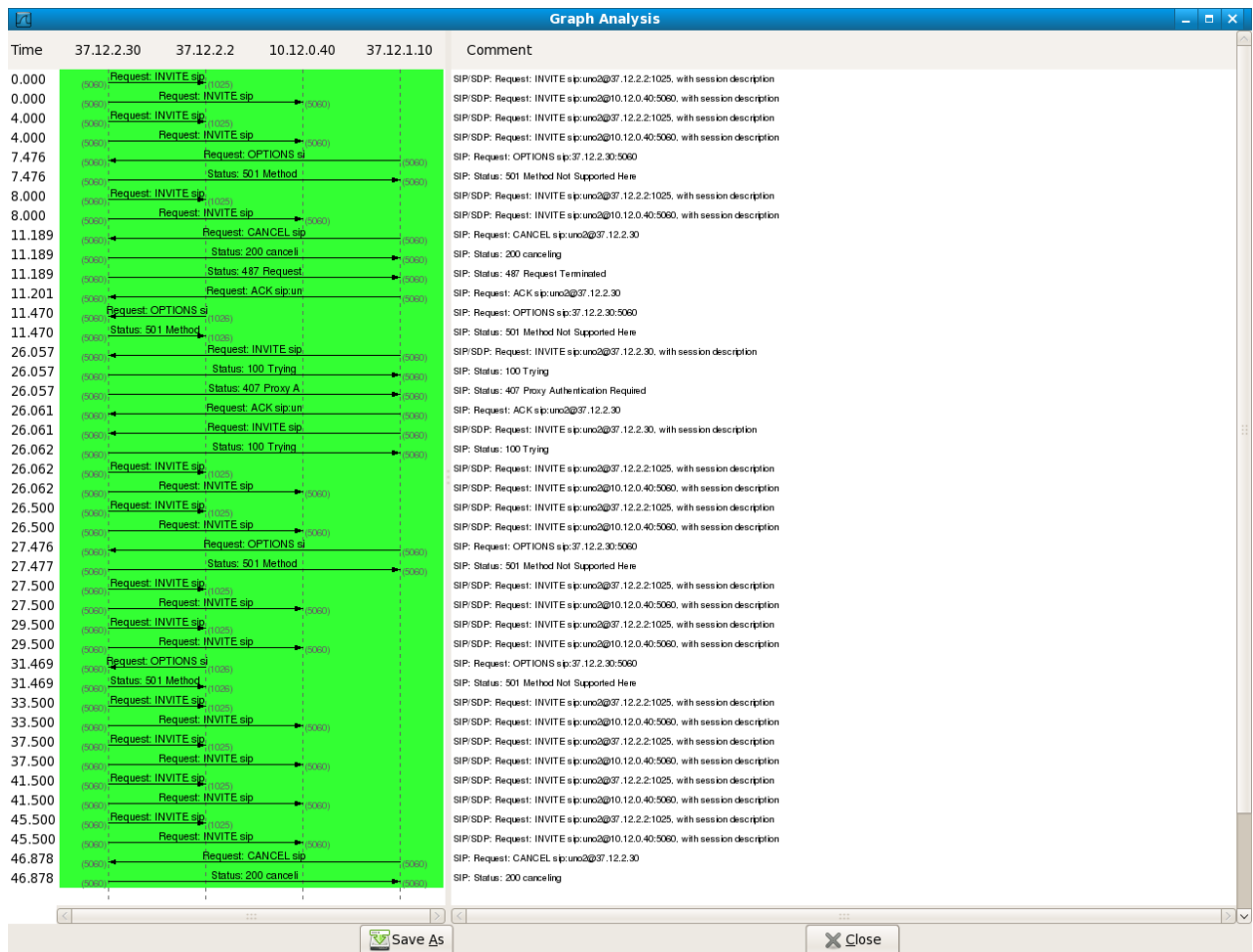
eth0: Capturing - Wireshark

File Edit View Go Capture Analyze Statistics Help

Filter: protocol=sip

No.	Time	Source	Destination	Protocol	Info
21	11.183440	37.12.2.30	37.12.1.10	SIP	Status: 487 Request Terminated
22	11.200767	37.12.1.10	37.12.2.30	SIP	Request: ACK sip:uno2@37.12.2.30
23	11.470165	37.12.2.2	37.12.2.30	SIP	Request: OPTIONS sip:37.12.2.30:5060
24	11.470328	37.12.2.30	37.12.2.2	SIP	Status: 501 Method Not Supported Here
25	26.056871	37.12.1.10	37.12.2.30	SIP/SDP	Request: INVITE sip:uno2@37.12.2.30, with session description
26	26.056992	37.12.2.30	37.12.1.10	SIP	Status: 100 Trying
27	26.057037	37.12.2.30	37.12.1.10	SIP	Status: 407 Proxy Authentication Required
28	26.060813	37.12.1.10	37.12.2.30	SIP	Request: ACK sip:uno2@37.12.2.30
29	26.061448	37.12.1.10	37.12.2.30	SIP/SDP	Request: INVITE sip:uno2@37.12.2.30, with session description
30	26.061690	37.12.2.30	37.12.1.10	SIP	Status: 100 Trying
31	26.062263	37.12.2.30	37.12.2.2	SIP/SDP	Request: INVITE sip:uno2@37.12.2.2:1025, with session description
32	26.062336	37.12.2.30	10.12.0.40	SIP/SDP	Request: INVITE sip:uno2@10.12.0.40:5060, with session description
35	26.499988	37.12.2.30	37.12.2.2	SIP/SDP	Request: INVITE sip:uno2@37.12.2.2:1025, with session description
36	26.500065	37.12.2.30	10.12.0.40	SIP/SDP	Request: INVITE sip:uno2@10.12.0.40:5060, with session description
37	27.476432	37.12.1.10	37.12.2.30	SIP	Request: OPTIONS sip:37.12.2.30:5060
38	27.476789	37.12.2.30	37.12.1.10	SIP	Status: 501 Method Not Supported Here
39	27.499990	37.12.2.30	37.12.2.2	SIP/SDP	Request: INVITE sip:uno2@37.12.2.2:1025, with session description
40	27.500064	37.12.2.30	10.12.0.40	SIP/SDP	Request: INVITE sip:uno2@10.12.0.40:5060, with session description
43	29.499987	37.12.2.30	37.12.2.2	SIP/SDP	Request: INVITE sip:uno2@37.12.2.2:1025, with session description
44	29.500066	37.12.2.30	10.12.0.40	SIP/SDP	Request: INVITE sip:uno2@10.12.0.40:5060, with session description
48	31.469233	37.12.2.2	37.12.2.30	SIP	Request: OPTIONS sip:37.12.2.30:5060
49	31.469443	37.12.2.30	37.12.2.2	SIP	Status: 501 Method Not Supported Here
50	33.499989	37.12.2.30	37.12.2.2	SIP/SDP	Request: INVITE sip:uno2@37.12.2.2:1025, with session description
51	33.500067	37.12.2.30	10.12.0.40	SIP/SDP	Request: INVITE sip:uno2@10.12.0.40:5060, with session description
54	37.499992	37.12.2.30	37.12.2.2	SIP/SDP	Request: INVITE sip:uno2@37.12.2.2:1025, with session description
55	37.500070	37.12.2.30	10.12.0.40	SIP/SDP	Request: INVITE sip:uno2@10.12.0.40:5060, with session description
61	41.500016	37.12.2.30	37.12.2.2	SIP/SDP	Request: INVITE sip:uno2@37.12.2.2:1025, with session description
62	41.500094	37.12.2.30	10.12.0.40	SIP/SDP	Request: INVITE sip:uno2@10.12.0.40:5060, with session description
67	45.499996	37.12.2.30	37.12.2.2	SIP/SDP	Request: INVITE sip:uno2@37.12.2.2:1025, with session description
68	45.500074	37.12.2.30	10.12.0.40	SIP/SDP	Request: INVITE sip:uno2@10.12.0.40:5060, with session description
71	46.878151	37.12.1.10	37.12.2.30	SIP	Request: CANCEL sip:uno2@37.12.2.30
72	46.878458	37.12.2.30	37.12.1.10	SIP	Status: 200 canceling
73	46.878497	37.12.2.30	37.12.1.10	SIP	Status: 487 Request Terminated
74	46.889395	37.12.1.10	37.12.2.30	SIP	Request: ACK sip:uno2@37.12.2.30
75	47.476614	37.12.1.10	37.12.2.30	SIP	Request: OPTIONS sip:37.12.2.30:5060
76	47.476870	37.12.2.30	37.12.1.10	SIP	Status: 501 Method Not Supported Here
77	51.468061	37.12.2.2	37.12.2.30	SIP	Request: OPTIONS sip:37.12.2.30:5060
78	51.468295	37.12.2.30	37.12.2.2	SIP	Status: 501 Method Not Supported Here
80	67.476884	37.12.1.10	37.12.2.30	SIP	Request: OPTIONS sip:37.12.2.30:5060
81	67.477126	37.12.2.30	37.12.1.10	SIP	Status: 501 Method Not Supported Here
82	71.466911	37.12.2.2	37.12.2.30	SIP	Request: OPTIONS sip:37.12.2.30:5060
83	71.467168	37.12.2.30	37.12.2.2	SIP	Status: 501 Method Not Supported Here

eth0: <live capture in progress> File: /tmp/etherXXXXL71egf 38 KB P: 86 D: 52 M: 0



- From the captures and the flow graph we can see that the call is successful and there is no ICMP unreachable message sent.
- After a few seconds, the call was ended from the PC1 side. But this call was not terminated on the laptop side because the SIP message is generated from PC1 and sent to SIP server. The SIP server now tries to send this message to the Laptop but since there is no NAT translation entry on the router for the SIP server and since SIP NAT is disabled on the router, it does not allow this message to pass through and hence the Laptop still thinks that the call is still going on.
- In the captures given above we can see once the PC1 sends a BYE message to the SIP server, the SIP server tries to forward that message to the Laptop. But since it is not successful it sends a message back to the PC1 saying that the sending has failed and we see a “send failed” packet.

## Exit Procedures

- Ensured there is one cable connecting each PCx21 port to the DELL switch (ports 1-20).
- Ensured there is one cable connecting port LA111 to the DELL switch (ports 1-20).
- The LNET port is connected to port 19 of the DELL switch.
- The MGT port is connected to port 23 of the DELL switch.
- Nothing else is connected to the switch, hub or router ports.
- Unused cables were put on the cabling rack and all computers were shut down.