Voice Call Establishment and Forwarding over IP, a Hands-on SIP-Based Experiment

EE-284

Team Members

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SESSION INITIATION PROTOCOL

- IETF Standard defined by RFC 3261
- "The Session Initiation Protocol (SIP) is an application-layer control (signaling) protocol for creating, modifying and terminating sessions with one or more participants."
- Can be used for voice, video, instant messaging, gaming, etc.

Components of SIP

- User Agents:

Clients

Servers

- Server Types

Redirect Sever

Proxy Server

Registrar Server

Location Server

SIP Methods

Invite

Acknowledge

Options

Cancel

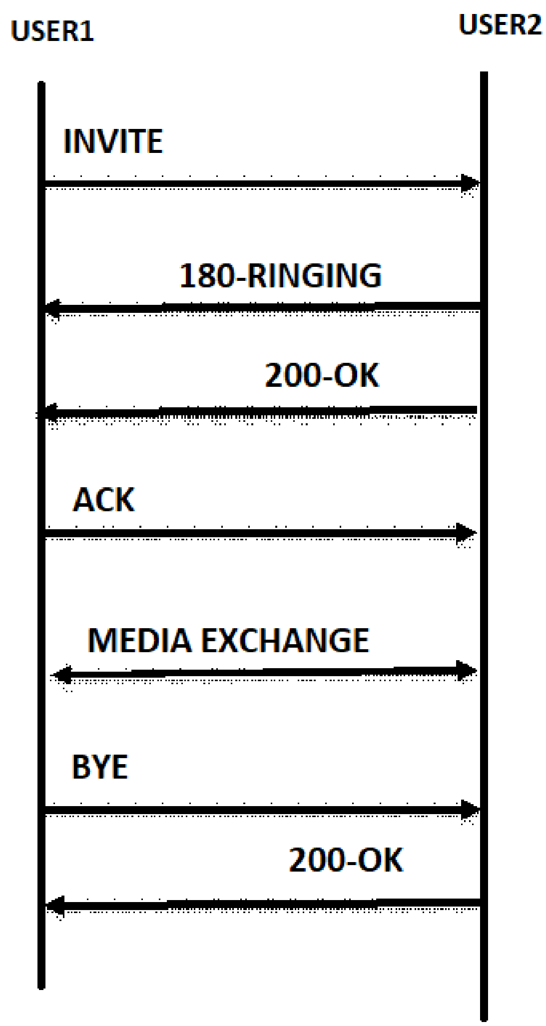
Bye

Register

SIP RESPONSES

- 1XX Provisional
- 2XX Successful
- 3XX Redirection
- 4XX Client Error
- 5XX Server Error
- 6XX Global Failure

SIP FLOW



OBJECTIVE

The objective of this project is to learn the implementation of Voice over IP (VoIP) using SIP connections and a call forwarding through a hands-on experiment over Ad-Hoc networks. The learning process is carried out by utilizing a server and a few VoIP users as clients. The server uses Asterisk software.

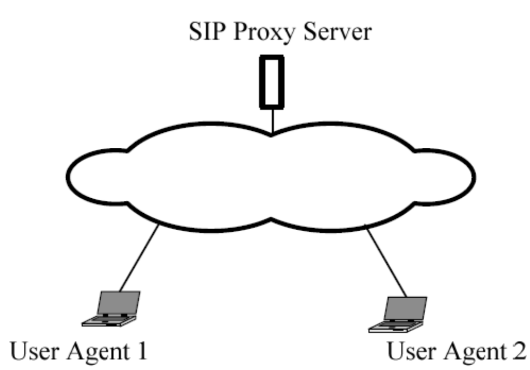
TOOLS

OS: Linux, Windows

Software: X-lite, Asterisk

SETUP:

We have used virtual machines (VMs) for the project: one acts as the SIP proxy server and two machines acts as user agents or VoIP clients as shown below



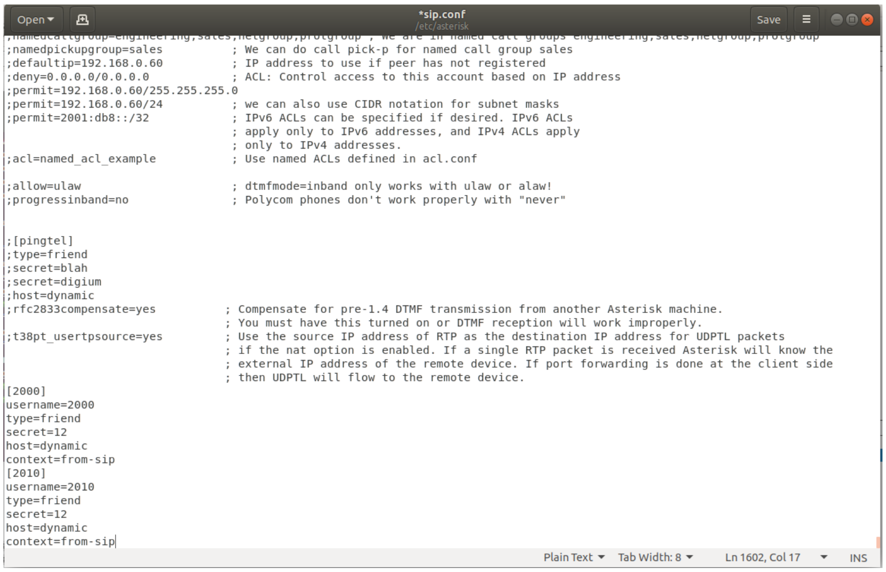
EXPERIMENT:

PART1:

Establish and Analyze a Successful Call Between Two SIP Clients.

STEP1:

In order to establish a call between two SIP clients, sip.conf file is configured as below

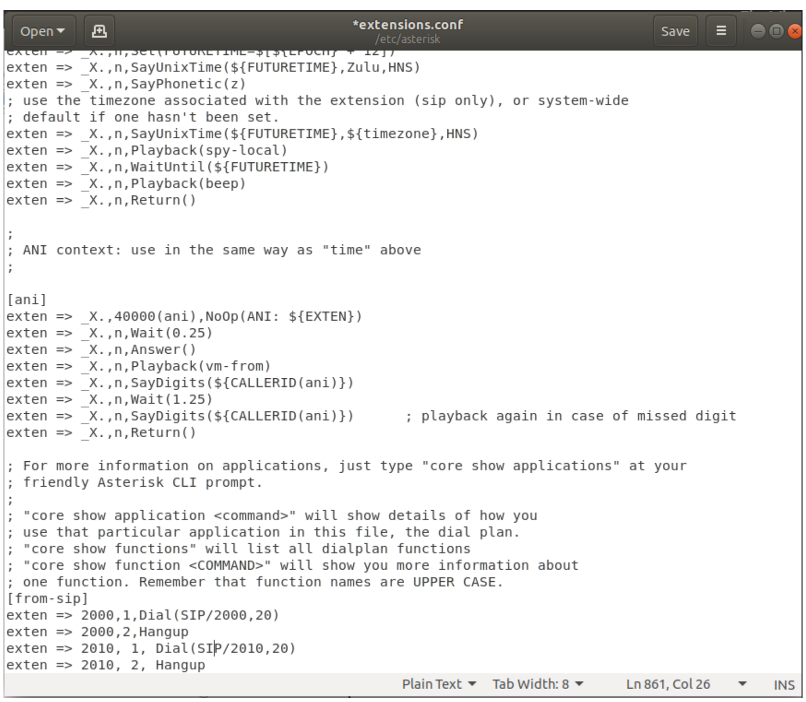


```
;namedpickupgroup=sales          ; We can do call pick-p for named call group sales
;defaultip=192.168.0.60          ; IP address to use if peer has not registered
;deny=0.0.0.0/0.0.0.0            ; ACL: Control access to this account based on IP address
;permit=192.168.0.60/255.255.255.0
;permit=192.168.0.60/24          ; we can also use CIDR notation for subnet masks
;permit=2001:db8::/32            ; IPv6 ACLs can be specified if desired. IPv6 ACLs
                                 ; apply only to IPv6 addresses, and IPv4 ACLs apply
                                 ; only to IPv4 addresses.
;acl=named_acl_example           ; Use named ACLs defined in acl.conf

;allow=ulaw                      ; dtmfmode=inband only works with ulaw or alaw!
;progressinband=no               ; Polycom phones don't work properly with "never"


;[pingtel]
;type=friend
;secret=blah
;secret=digium
;host=dynamic
;rfc2833compensate=yes          ; Compensate for pre-1.4 DTMF transmission from another Asterisk machine.
                                ; You must have this turned on or DTMF reception will work improperly.
;t38pt_usertpsource=yes         ; Use the source IP address of RTP as the destination IP address for UDPTL packets
                                ; if the nat option is enabled. If a single RTP packet is received Asterisk will know the
                                ; external IP address of the remote device. If port forwarding is done at the client side
                                ; then UDPTL will flow to the remote device.
[2000]
username=2000
type=friend
secret=12
host=dynamic
context=from-sip
[2010]
username=2010
type=friend
secret=12
host=dynamic
context=from-sip
```

STEP2: Configure extensions.conf file as below



```
exten => _X.,n,SayUnixTime(${FUTURETIME},Zulu,HNS)
exten => _X.,n,SayPhonetic(z)
; use the timezone associated with the extension (sip only), or system-wide
; default if one hasn't been set.
exten => _X.,n,SayUnixTime(${FUTURETIME},${timezone},HNS)
exten => _X.,n,Playback(spy-local)
exten => _X.,n,WaitUntil(${FUTURETIME})
exten => _X.,n,Playback(beep)
exten => _X.,n,Return()

;
; ANI context: use in the same way as "time" above
;

[ani]
exten => _X.,40000(ani),NoOp(ANI: ${EXTEN})
exten => _X.,n,Wait(0.25)
exten => _X.,n,Answer()
exten => _X.,n,Playback(vm-from)
exten => _X.,n,SayDigits(${CALLERID(ani)})
exten => _X.,n,Wait(1.25)
exten => _X.,n,SayDigits(${CALLERID(ani)})      ; playback again in case of missed digit
exten => _X.,n,Return()

; For more information on applications, just type "core show applications" at your
; friendly Asterisk CLI prompt.
;
; "core show application <command>" will show details of how you
; use that particular application in this file, the dial plan.
; "core show functions" will list all dialplan functions
; "core show function <COMMAND>" will show you more information about
; one function. Remember that function names are UPPER CASE.
[from-sip]
exten => 2000,1,Dial(SIP/2000,20)
exten => 2000,2,Hangup
exten => 2010, 1, Dial(SIP/2010,20)
exten => 2010, 2, Hangup
```

STEP3: Configure X-LITE Soft phone with the parameters such as:

Accountname

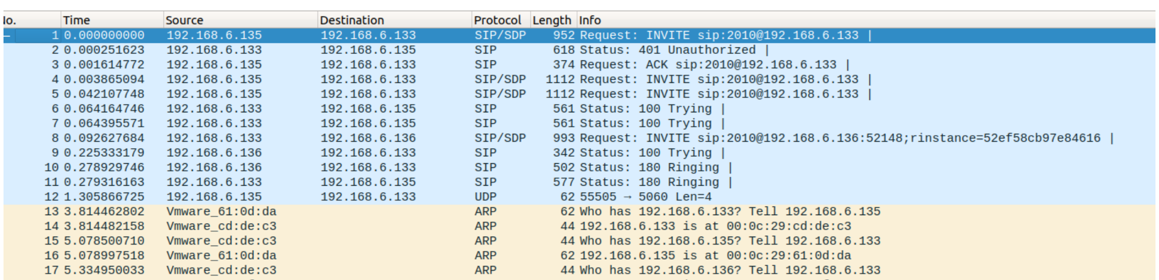
Display name

User name

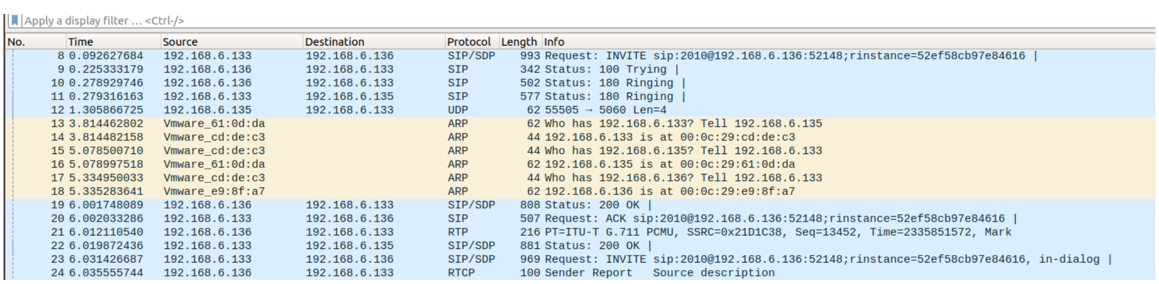
Password

Domain

When a call is placed from user 2000 (192.168.6.135) to user 2010 (192.168.6.136), detailed observation shows that user 2000 does an arp query to for IP 192.168.6.133 of sip-proxy server. Once location known then the user 2000 sends an request: INVITE sip:2010@192.168.6.133 to SIP-proxy server and sip-proxy server sends an request: INVITE to sip:2010@192.168.6.136 which is the sip id of user 2010. The user 2010 responds with RINGING to sip-proxy server and then to user 2000. This is as shown in fig1



No.	Time	Source	Destination	Protocol	Length	Info
1	0.000000000	192.168.6.135	192.168.6.133	SIP/SDP	952	Request: INVITE sip:2010@192.168.6.133 |
2	0.000251623	192.168.6.133	192.168.6.135	SIP	618	Status: 401 Unauthorized |
3	0.001614772	192.168.6.135	192.168.6.133	SIP	374	Request: ACK sip:2010@192.168.6.133 |
4	0.003865094	192.168.6.135	192.168.6.133	SIP/SDP	1112	Request: INVITE sip:2010@192.168.6.133 |
5	0.042107748	192.168.6.135	192.168.6.133	SIP/SDP	1112	Request: INVITE sip:2010@192.168.6.133 |
6	0.064164746	192.168.6.133	192.168.6.135	SIP	561	Status: 100 Trying |
7	0.064395571	192.168.6.133	192.168.6.135	SIP	561	Status: 100 Trying |
8	0.092627684	192.168.6.133	192.168.6.136	SIP/SDP	993	Request: INVITE sip:2010@192.168.6.136:52148;rinstance=52ef58cb97e84616 |
9	0.225333179	192.168.6.136	192.168.6.133	SIP	342	Status: 100 Trying |
10	0.278929746	192.168.6.136	192.168.6.133	SIP	502	Status: 180 Ringing |
11	0.279316163	192.168.6.133	192.168.6.135	SIP	577	Status: 180 Ringing |
12	1.305866725	192.168.6.135	192.168.6.133	UDP	62	55505 → 5060 Len=4
13	3.814462802	Vmware_61:0d:da		ARP	62	Who has 192.168.6.133? Tell 192.168.6.135
14	3.814482158	Vmware_cd:de:c3		ARP	44	192.168.6.133 is at 00:0c:29:cd:de:c3
15	5.078500710	Vmware_cd:de:c3		ARP	44	Who has 192.168.6.135? Tell 192.168.6.133
16	5.078997518	Vmware_61:0d:da		ARP	62	192.168.6.135 is at 00:0c:29:61:0d:da
17	5.334950033	Vmware_cd:de:c3		ARP	44	Who has 192.168.6.136? Tell 192.168.6.133
18	5.335283641	Vmware_e9:8f:a7		ARP	62	192.168.6.136 is at 00:0c:29:e9:8f:a7



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No.	Time	Source	Destination	Protocol	Length	Info
8	0.092627684	192.168.6.133	192.168.6.136	SIP/SDP	993	Request: INVITE sip:2010@192.168.6.136:52148;rinstance=52ef58cb97e84616 |
9	0.225333179	192.168.6.136	192.168.6.133	SIP	342	Status: 100 Trying |
10	0.278929746	192.168.6.136	192.168.6.133	SIP	502	Status: 180 Ringing |
11	0.279316163	192.168.6.133	192.168.6.135	SIP	577	Status: 180 Ringing |
12	1.305866725	192.168.6.135	192.168.6.133	UDP	62	55505 → 5060 Len=4
13	3.814462802	Vmware_61:0d:da		ARP	62	Who has 192.168.6.133? Tell 192.168.6.135
14	3.814482158	Vmware_cd:de:c3		ARP	44	192.168.6.133 is at 00:0c:29:cd:de:c3
15	5.078500710	Vmware_cd:de:c3		ARP	44	Who has 192.168.6.135? Tell 192.168.6.133
16	5.078997518	Vmware_61:0d:da		ARP	62	192.168.6.135 is at 00:0c:29:61:0d:da
17	5.334950033	Vmware_cd:de:c3		ARP	44	Who has 192.168.6.136? Tell 192.168.6.133
18	5.335283641	Vmware_e9:8f:a7		ARP	62	192.168.6.136 is at 00:0c:29:e9:8f:a7
19	6.001748089	192.168.6.136	192.168.6.133	SIP/SDP	808	Status: 200 OK |
20	6.002033286	192.168.6.133	192.168.6.136	SIP	507	Request: ACK sip:2010@192.168.6.136:52148;rinstance=52ef58cb97e84616 |
21	6.012110540	192.168.6.136	192.168.6.133	RTP	216	PT=ITU-T G.711 PCMU, SSRC=0x21D1C38, Seq=13452, Time=2335851572, Mark
22	6.019872436	192.168.6.133	192.168.6.135	SIP/SDP	881	Status: 200 OK |
23	6.031426687	192.168.6.133	192.168.6.136	SIP/SDP	969	Request: INVITE sip:2010@192.168.6.136:52148;rinstance=52ef58cb97e84616, in-dialog |
24	6.035555744	192.168.6.136	192.168.6.133	RTCP	100	Sender Report Source description

FIG1

The user 2010 sends ok message to sip-proxy server and sip-proxy server sends ok message to user 2000. This completes a successful connection between 2 user-agents and server as shown in FIG2

After this step, voice data is transmitted over RTP protocol which provides end-to-end transfer functions for applications transmitting real time data as shown . The data is transmitted over RTP from user 2000 to sip-proxy server to 2000

sip

No.	Time	Source	Destination	Protocol	Length	Info
10	0.278929746	192.168.6.136	192.168.6.133	SIP	502	Status: 180 Ringing |
11	0.279316163	192.168.6.133	192.168.6.135	SIP	577	Status: 180 Ringing |
19	6.001748089	192.168.6.136	192.168.6.133	SIP/SDP	808	Status: 200 OK |
20	6.002033286	192.168.6.133	192.168.6.136	SIP	507	Request: ACK sip:2010@192.168.6.136:52148;rinstance=52ef58cb97e84616 |
22	6.019872436	192.168.6.133	192.168.6.135	SIP/SDP	881	Status: 200 OK |
23	6.031426687	192.168.6.133	192.168.6.136	SIP/SDP	969	Request: INVITE sip:2010@192.168.6.136:52148;rinstance=52ef58cb97e84616, in-dialog |
27	6.046475565	192.168.6.135	192.168.6.133	SIP	493	Request: ACK sip:2010@192.168.6.133:5060 |
28	6.046645937	192.168.6.133	192.168.6.135	SIP/SDP	887	Request: INVITE sip:2000@192.168.6.135:55505;rinstance=3cd78b673e9dd0d7, in-dialog |
39	6.113578399	192.168.6.136	192.168.6.133	SIP	355	Status: 100 Trying |
57	6.205795784	192.168.6.135	192.168.6.133	SIP	329	Status: 100 Trying |
58	6.205819220	192.168.6.135	192.168.6.133	SIP/SDP	787	Status: 200 OK |
59	6.206132737	192.168.6.133	192.168.6.135	SIP	476	Request: ACK sip:2000@192.168.6.135:55505;rinstance=3cd78b673e9dd0d7 |
76	6.431087491	192.168.6.136	192.168.6.133	SIP/SDP	808	Status: 200 OK |
77	6.431357609	192.168.6.133	192.168.6.136	SIP	507	Request: ACK sip:2010@192.168.6.136:52148;rinstance=52ef58cb97e84616 |
81	20.063300848	192.168.6.136	192.168.6.133	SIP	530	Request: BYE sip:2000@192.168.6.133:5060 |
82	20.063663131	192.168.6.133	192.168.6.136	SIP	566	Status: 200 OK |
83	20.063961408	192.168.6.133	192.168.6.135	SIP/SDP	887	Request: INVITE sip:2000@192.168.6.135:55505;rinstance=3cd78b673e9dd0d7, in-dialog |
84	20.145965447	192.168.6.135	192.168.6.133	SIP	329	Status: 100 Trying |

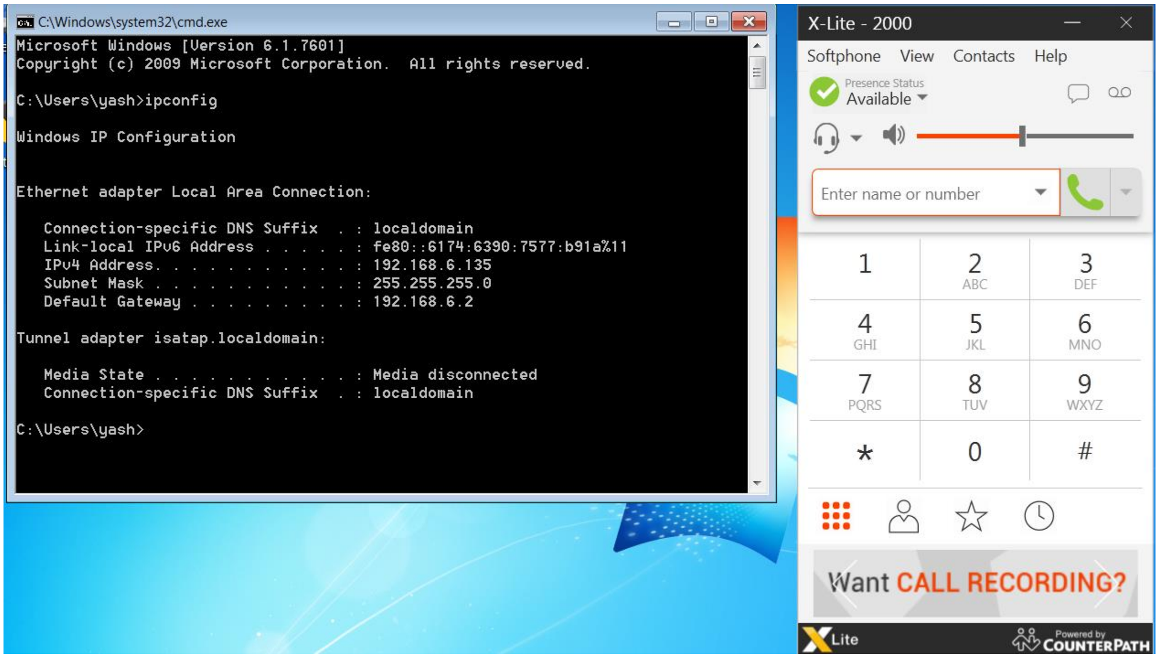
FIG2

During call termination, the terminating user (user 2010 here) requests BYE message over sip to proxy-server which forwards the request to user 2000. User 2000 and proxy server responds with 200 OK message and user 2000 sends GOODBYE REPORT to sip proxy server as shown in fig 3

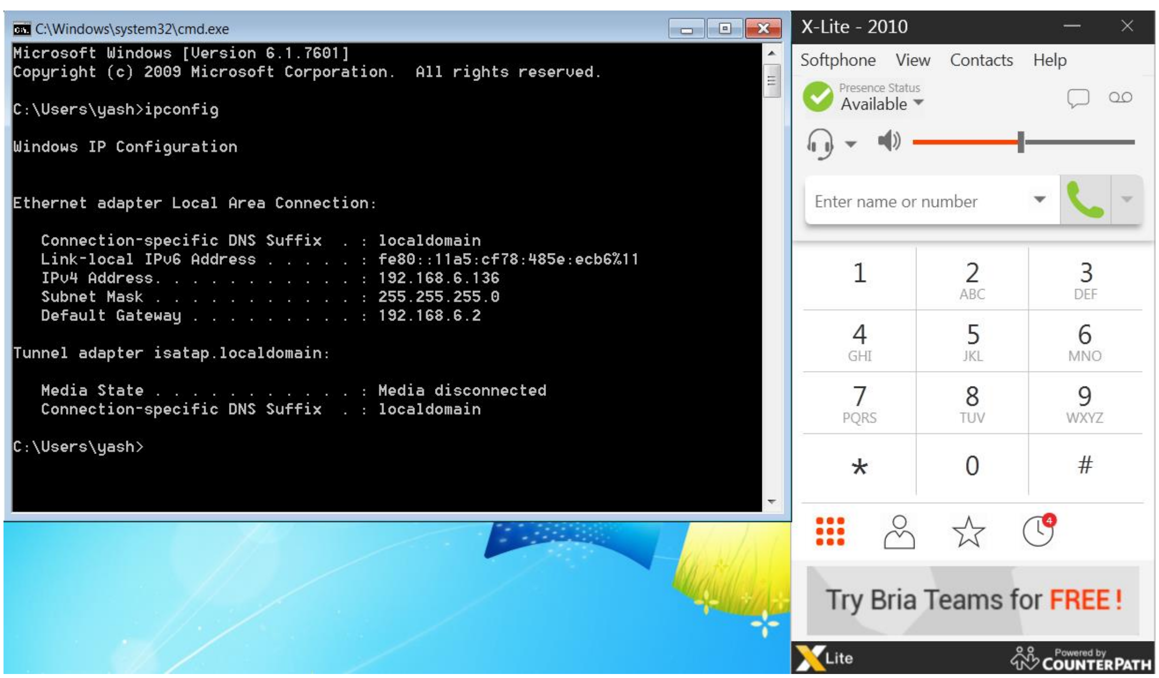
No.	Time	Source	Destination	Protocol	Length	Info
23	6.031426687	192.168.6.133	192.168.6.136	SIP/SDP	969	Request: INVITE sip:2010@192.168.6.136:52148;rinstance=52ef58cb97e84616, in-dialog |
27	6.046475565	192.168.6.135	192.168.6.133	SIP	493	Request: ACK sip:2010@192.168.6.133:5060 |
28	6.046645937	192.168.6.133	192.168.6.135	SIP/SDP	887	Request: INVITE sip:2000@192.168.6.135:55505;rinstance=3cd78b673e9dd0d7, in-dialog |
39	6.113578399	192.168.6.136	192.168.6.133	SIP	355	Status: 100 Trying |
57	6.205795784	192.168.6.135	192.168.6.133	SIP	329	Status: 100 Trying |
58	6.205819220	192.168.6.135	192.168.6.133	SIP/SDP	787	Status: 200 OK |
59	6.206132737	192.168.6.133	192.168.6.135	SIP	476	Request: ACK sip:2000@192.168.6.135:55505;rinstance=3cd78b673e9dd0d7 |
76	6.431087491	192.168.6.136	192.168.6.133	SIP/SDP	808	Status: 200 OK |
77	6.431357609	192.168.6.133	192.168.6.136	SIP	507	Request: ACK sip:2010@192.168.6.136:52148;rinstance=52ef58cb97e84616 |
81	20.063300848	192.168.6.136	192.168.6.133	SIP	530	Request: BYE sip:2000@192.168.6.133:5060 |
82	20.063663131	192.168.6.133	192.168.6.136	SIP	566	Status: 200 OK |
83	20.063961408	192.168.6.133	192.168.6.135	SIP/SDP	887	Request: INVITE sip:2000@192.168.6.135:55505;rinstance=3cd78b673e9dd0d7, in-dialog |
84	20.145965447	192.168.6.135	192.168.6.133	SIP	329	Status: 100 Trying |
85	20.171489379	192.168.6.135	192.168.6.133	SIP/SDP	787	Status: 200 OK |
86	20.171757365	192.168.6.133	192.168.6.135	SIP	476	Request: ACK sip:2000@192.168.6.135:55505;rinstance=3cd78b673e9dd0d7 |
87	20.171877221	192.168.6.133	192.168.6.135	SIP	675	Request: BYE sip:2000@192.168.6.135:55505;rinstance=3cd78b673e9dd0d7 |
88	20.243185897	192.168.6.135	192.168.6.133	SIP	436	Status: 200 OK |

FIG3

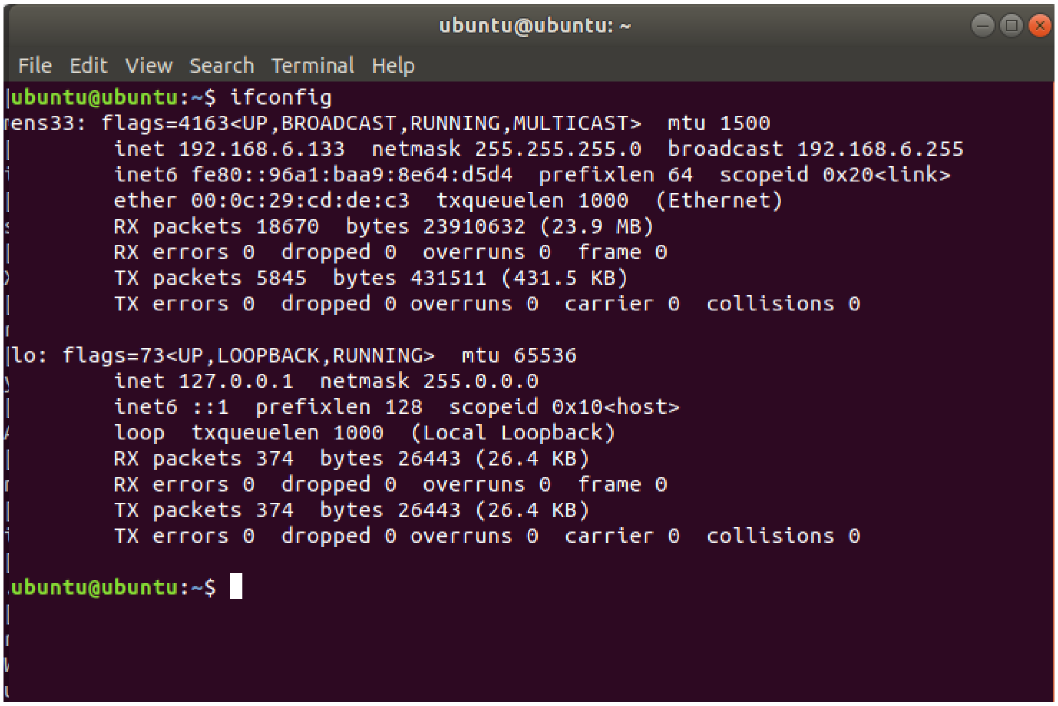
The figure below shows the IP address and the X-lite for user 2000



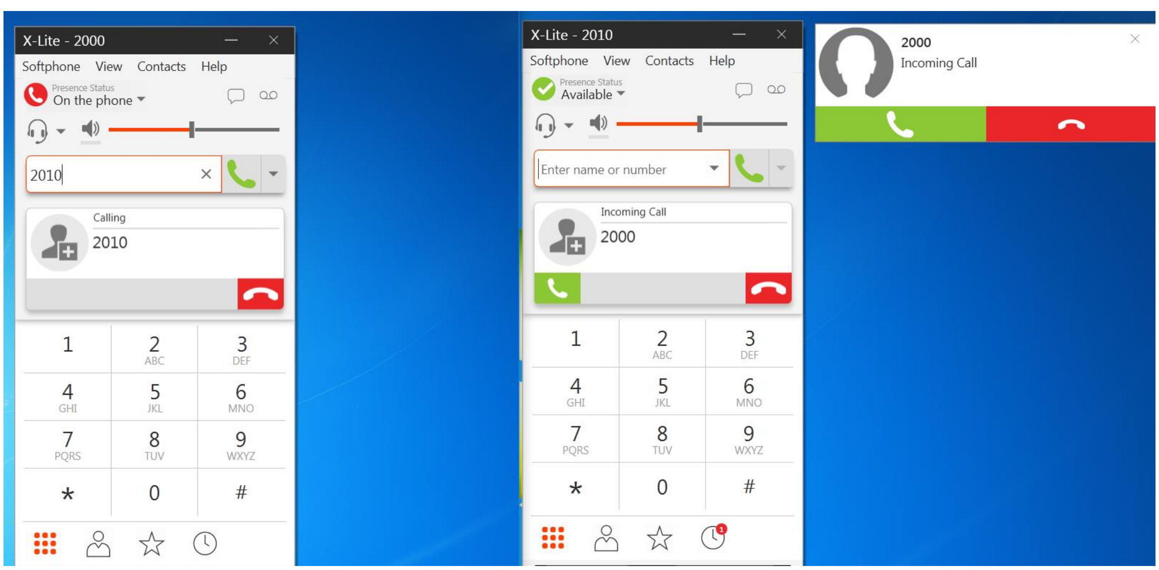
The figure below shows the IP address and the X-lite for user 2010



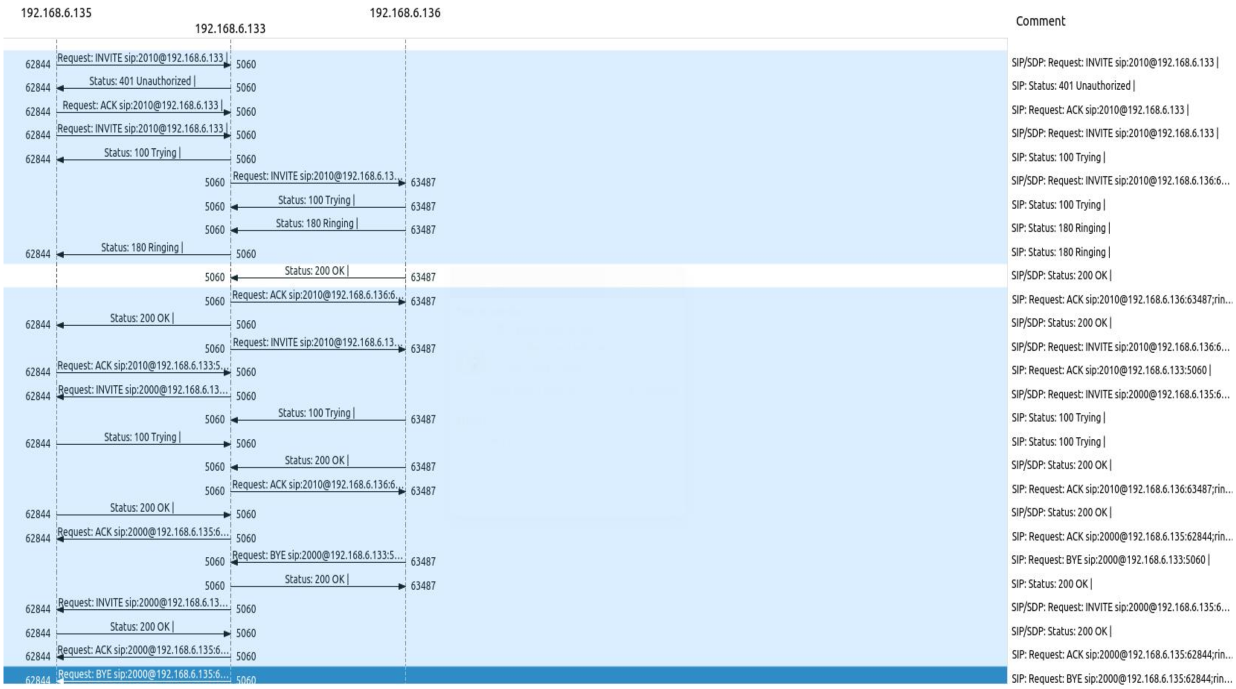
IP Configuration of SIP Proxy server



The below figure shows the GUI during the call establishment in X-lite

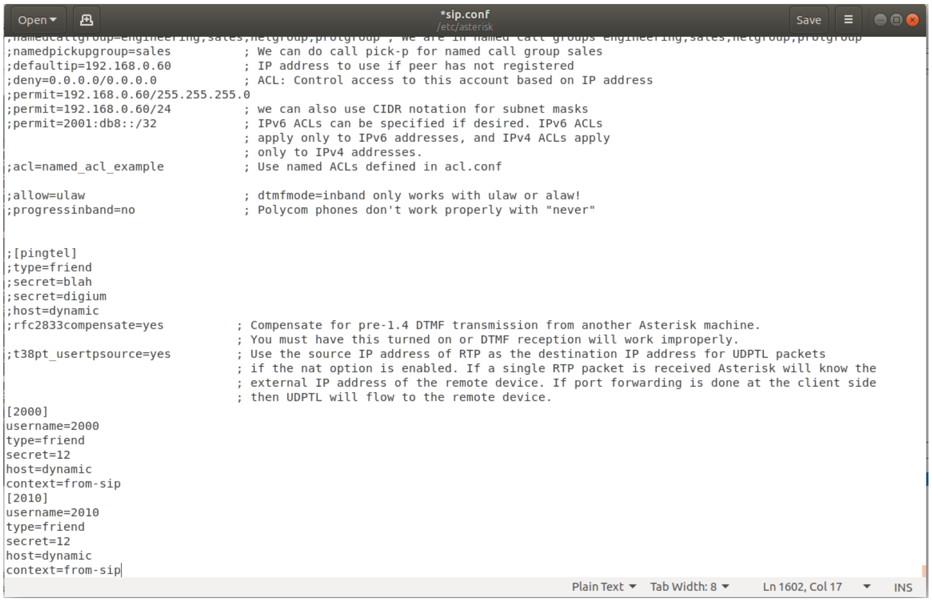


The following figure shows the Wireshark capture between user 2000 and user 2010



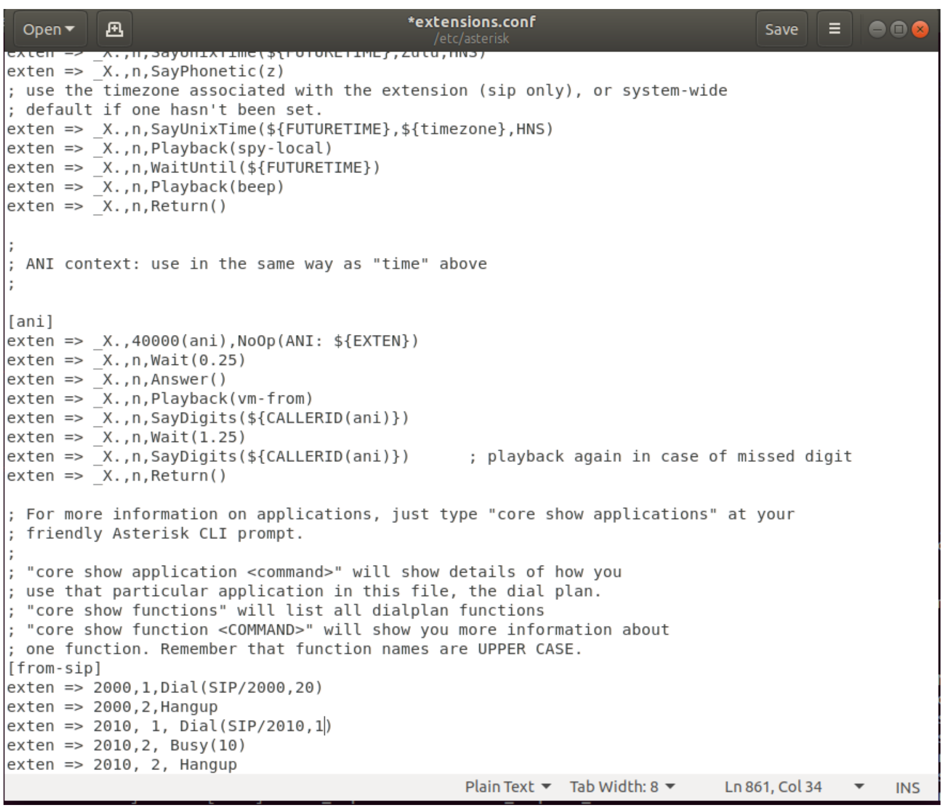
PART2: Busy User

STEP1: Configure the sip.config



```
;namedpickupgroup=sales          ; We can do call pick-p for named call group sales
;defaultip=192.168.0.60          ; IP address to use if peer has not registered
;deny=0.0.0.0/0.0.0.0            ; ACL: Control access to this account based on IP address
;permit=192.168.0.60/255.255.255.0
;permit=192.168.0.60/24          ; we can also use CIDR notation for subnet masks
;permit=2001:db8::/32            ; IPv6 ACLs can be specified if desired. IPv6 ACLs
                                 ; apply only to IPv6 addresses, and IPv4 ACLs apply
                                 ; only to IPv4 addresses.
;acl=named_acl_example           ; Use named ACLs defined in acl.conf

;allow=ulaw                      ; dtmfmode=inband only works with ulaw or alaw!
;progressinband=no               ; Polycom phones don't work properly with "never"


;[pingtel]
;type=friend
;secret=blah
;secret=digium
;host=dynamic
;rfc2833compensate=yes          ; Compensate for pre-1.4 DTMF transmission from another Asterisk machine.
                                ; You must have this turned on or DTMF reception will work improperly.
;t38pt_usertpsource=yes         ; Use the source IP address of RTP as the destination IP address for UDPTL packets
                                ; if the nat option is enabled. If a single RTP packet is received Asterisk will know the
                                ; external IP address of the remote device. If port forwarding is done at the client side
                                ; then UDPTL will flow to the remote device.
[2000]
username=2000
type=friend
secret=12
host=dynamic
context=from-sip
[2010]
username=2010
type=friend
secret=12
host=dynamic
context=from-sip
```

STEP2:Configure extensions.conf file as below

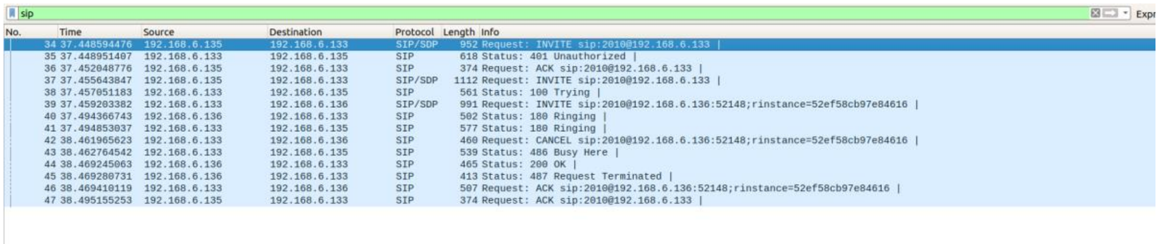


```
exten => _X.,n,SayPhonetic(z)
; use the timezone associated with the extension (sip only), or system-wide
; default if one hasn't been set.
exten => _X.,n,SayUnixTime(${FUTURETIME},${timezone},HNS)
exten => _X.,n,Playback(spy-local)
exten => _X.,n,WaitUntil(${FUTURETIME})
exten => _X.,n,Playback(beep)
exten => _X.,n,Return()

;
; ANI context: use in the same way as "time" above
;

[ani]
exten => _X.,40000(ani),NoOp(ANI: ${EXTEN})
exten => _X.,n,Wait(0.25)
exten => _X.,n,Answer()
exten => _X.,n,Playback(vm-from)
exten => _X.,n,SayDigits(${CALLERID(ani)})
exten => _X.,n,Wait(1.25)
exten => _X.,n,SayDigits(${CALLERID(ani)})      ; playback again in case of missed digit
exten => _X.,n,Return()

; For more information on applications, just type "core show applications" at your
; friendly Asterisk CLI prompt.
;
; "core show application <command>" will show details of how you
; use that particular application in this file, the dial plan.
; "core show functions" will list all dialplan functions
; "core show function <COMMAND>" will show you more information about
; one function. Remember that function names are UPPER CASE.
[from-sip]
exten => 2000,1,Dial(SIP/2000,20)
exten => 2000,2,Hangup
exten => 2010, 1, Dial(SIP/2010,1)
exten => 2010,2, Busy(10)
exten => 2010, 2, Hangup
```

STEP3: Configure X-LITE Soft phone with the parameters similar to Part1

STEP4 : Reload to update the changes

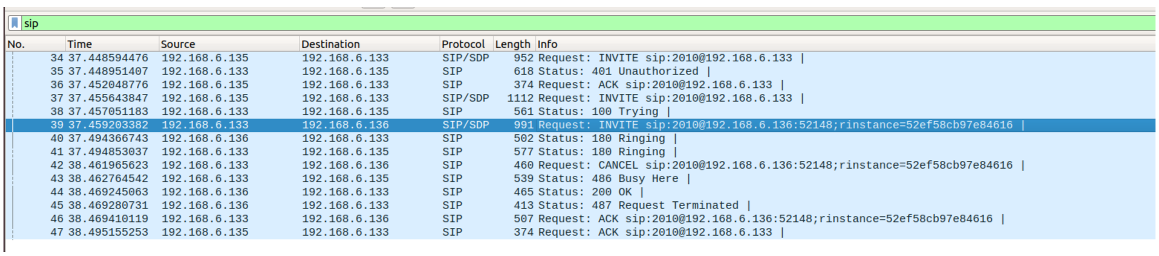
The user 2000 first requests INVITE to user2010 through the sip server using SIP protocol INVITE sip:2010@192.168.6.133 as shown below



sip

No.	Time	Source	Destination	Protocol	Length	Info
34	37.448594476	192.168.6.135	192.168.6.133	SIP/SDP	952	Request: INVITE sip:2010@192.168.6.133 |
35	37.448951407	192.168.6.133	192.168.6.135	SIP	618	Status: 401 Unauthorized |
36	37.452048776	192.168.6.135	192.168.6.133	SIP	374	Request: ACK sip:2010@192.168.6.133 |
37	37.455643847	192.168.6.135	192.168.6.133	SIP/SDP	1112	Request: INVITE sip:2010@192.168.6.133 |
38	37.457051183	192.168.6.133	192.168.6.135	SIP	561	Status: 100 Trying |
39	37.459203382	192.168.6.133	192.168.6.136	SIP/SDP	991	Request: INVITE sip:2010@192.168.6.136:52148;rinstance=52ef58cb97e84616 |
40	37.494366743	192.168.6.136	192.168.6.133	SIP	502	Status: 180 Ringing |
41	37.494853037	192.168.6.133	192.168.6.135	SIP	577	Status: 180 Ringing |
42	38.461965623	192.168.6.133	192.168.6.136	SIP	460	Request: CANCEL sip:2010@192.168.6.136:52148;rinstance=52ef58cb97e84616 |
43	38.462764542	192.168.6.133	192.168.6.135	SIP	539	Status: 486 Busy Here |
44	38.469245063	192.168.6.136	192.168.6.133	SIP	465	Status: 200 OK |
45	38.469280731	192.168.6.136	192.168.6.133	SIP	413	Status: 487 Request Terminated |
46	38.469410119	192.168.6.133	192.168.6.136	SIP	507	Request: ACK sip:2010@192.168.6.136:52148;rinstance=52ef58cb97e84616 |
47	38.495155253	192.168.6.135	192.168.6.133	SIP	374	Request: ACK sip:2010@192.168.6.133 |

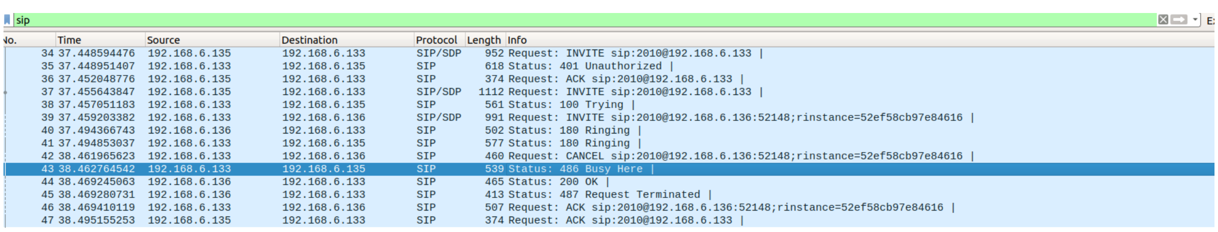
The SIP proxy server sends INVITE sip:2010@192.168.6.136 to user 2000 as shown below



sip

No.	Time	Source	Destination	Protocol	Length	Info
34	37.448594476	192.168.6.135	192.168.6.133	SIP/SDP	952	Request: INVITE sip:2010@192.168.6.133 |
35	37.448951407	192.168.6.133	192.168.6.135	SIP	618	Status: 401 Unauthorized |
36	37.452048776	192.168.6.135	192.168.6.133	SIP	374	Request: ACK sip:2010@192.168.6.133 |
37	37.455643847	192.168.6.135	192.168.6.133	SIP/SDP	1112	Request: INVITE sip:2010@192.168.6.133 |
38	37.457051183	192.168.6.133	192.168.6.135	SIP	561	Status: 100 Trying |
39	37.459203382	192.168.6.133	192.168.6.136	SIP/SDP	991	Request: INVITE sip:2010@192.168.6.136:52148;rinstance=52ef58cb97e84616 |
40	37.494366743	192.168.6.136	192.168.6.133	SIP	502	Status: 180 Ringing |
41	37.494853037	192.168.6.133	192.168.6.135	SIP	577	Status: 180 Ringing |
42	38.461965623	192.168.6.133	192.168.6.136	SIP	460	Request: CANCEL sip:2010@192.168.6.136:52148;rinstance=52ef58cb97e84616 |
43	38.462764542	192.168.6.133	192.168.6.135	SIP	539	Status: 486 Busy Here |
44	38.469245063	192.168.6.136	192.168.6.133	SIP	465	Status: 200 OK |
45	38.469280731	192.168.6.136	192.168.6.133	SIP	413	Status: 487 Request Terminated |
46	38.469410119	192.168.6.133	192.168.6.136	SIP	507	Request: ACK sip:2010@192.168.6.136:52148;rinstance=52ef58cb97e84616 |
47	38.495155253	192.168.6.135	192.168.6.133	SIP	374	Request: ACK sip:2010@192.168.6.133 |

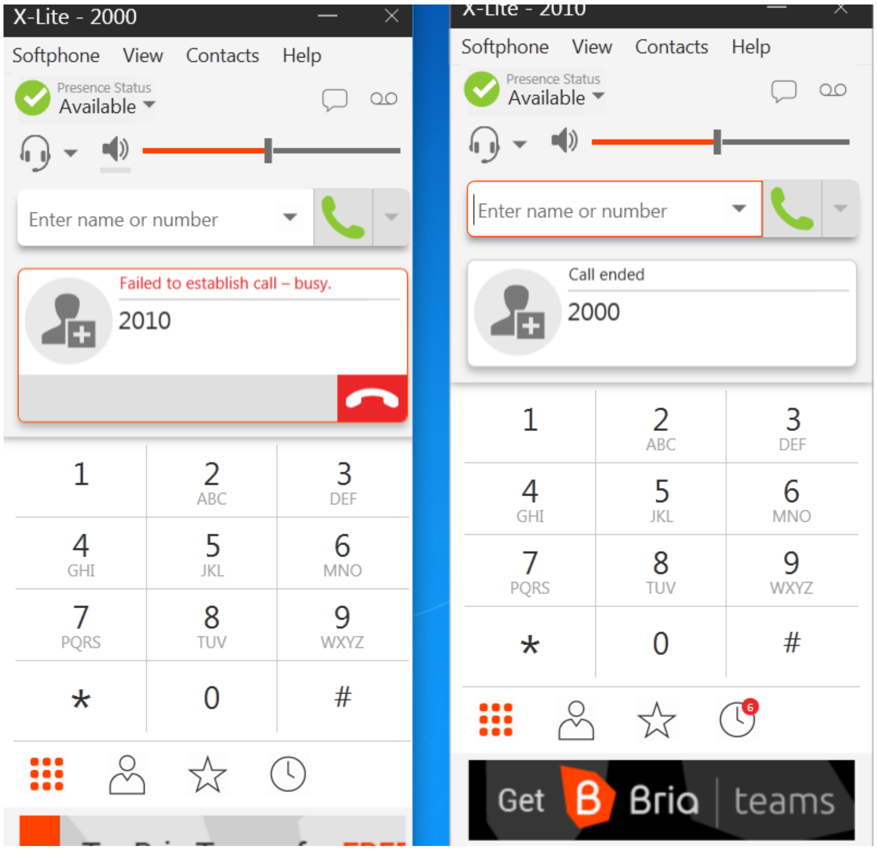
As busy tone is generated after 10 sec of delay, the call termination occurs and the SIP server sends a busy tone and the call is terminated as shown below.



sip

No.	Time	Source	Destination	Protocol	Length	Info
34	37.448594476	192.168.6.135	192.168.6.133	SIP/SDP	952	Request: INVITE sip:2010@192.168.6.133 |
35	37.448951407	192.168.6.133	192.168.6.135	SIP	618	Status: 401 Unauthorized |
36	37.452048776	192.168.6.135	192.168.6.133	SIP	374	Request: ACK sip:2010@192.168.6.133 |
37	37.455643847	192.168.6.135	192.168.6.133	SIP/SDP	1112	Request: INVITE sip:2010@192.168.6.133 |
38	37.457051183	192.168.6.133	192.168.6.135	SIP	561	Status: 100 Trying |
39	37.459203382	192.168.6.133	192.168.6.136	SIP/SDP	991	Request: INVITE sip:2010@192.168.6.136:52148;rinstance=52ef58cb97e84616 |
40	37.494366743	192.168.6.136	192.168.6.133	SIP	502	Status: 180 Ringing |
41	37.494853037	192.168.6.133	192.168.6.135	SIP	577	Status: 180 Ringing |
42	38.461965623	192.168.6.133	192.168.6.136	SIP	460	Request: CANCEL sip:2010@192.168.6.136:52148;rinstance=52ef58cb97e84616 |
43	38.462764542	192.168.6.133	192.168.6.135	SIP	539	Status: 486 Busy Here |
44	38.469245063	192.168.6.136	192.168.6.133	SIP	465	Status: 200 OK |
45	38.469280731	192.168.6.136	192.168.6.133	SIP	413	Status: 487 Request Terminated |
46	38.469410119	192.168.6.133	192.168.6.136	SIP	507	Request: ACK sip:2010@192.168.6.136:52148;rinstance=52ef58cb97e84616 |
47	38.495155253	192.168.6.135	192.168.6.133	SIP	374	Request: ACK sip:2010@192.168.6.133 |

The following shows the GUI of X-lite between two users.



PART3: CALL FORWARDING

STEP1: Configure the sip.config as shown below

```
*sip.conf
/etc/asterisk

                                        ; be played upon completion of
                                        ; an attended transfer to the
                                        ; target of the transfer.

;[pre14-asterisk]
;type=friend
;secret=digium
;host=dynamic
;rfc2833compensate=yes          ; Compensate for pre-1.4 DTMF transmission from another
Asterisk machine.
                                ; You must have this turned on or DTMF reception will work
improperly.
;t38pt_usertpsource=yes         ; Use the source IP address of RTP as the destination IP
address for UDPTL packets
                                ; if the nat option is enabled. If a single RTP packet is
received Asterisk will know the
                                ; external IP address of the remote device. If port forwarding
is done at the client side
                                ; then UDPTL will flow to the remote device.
[2000]
username=2000
type=friend
secret=12
host=dynamic
context=from-sip
[2010]
username=2010
type=friend
secret=12
host=dynamic
context=from-sip
[2010]
username=2020
type=friend
secret=12
host=dynamic
context=from-sip
```

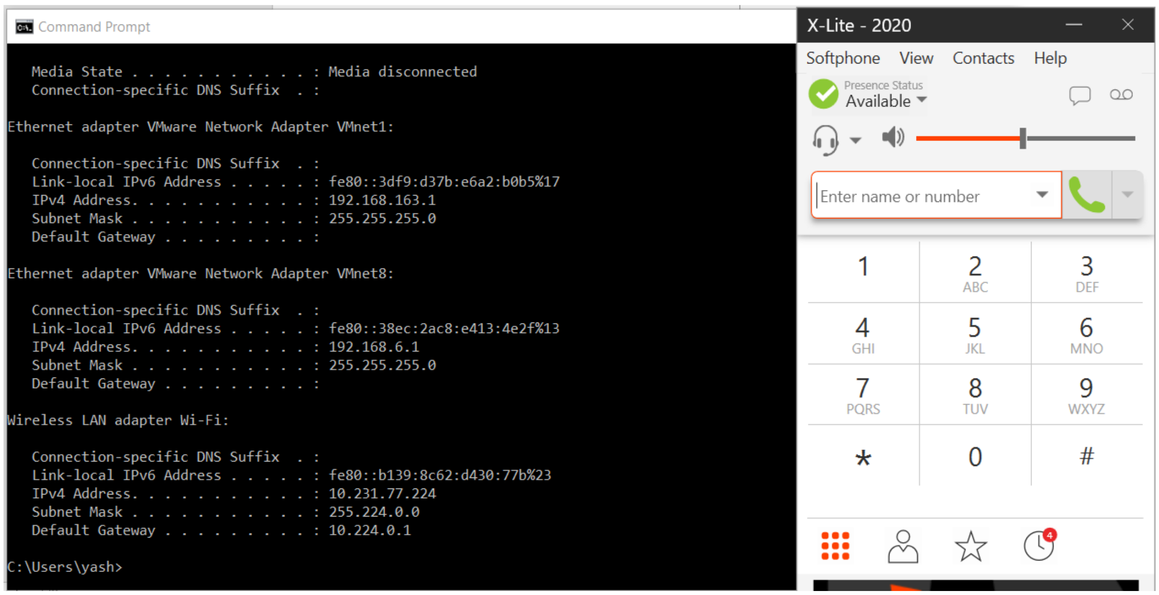
STEP2: Configure extensions.conf file as below

```
extensions.conf
/etc/asterisk

exten => _X.,n,SayUnixTime(${FUTURETIME},${timezone},HNS)
exten => _X.,n,Playback(spy-local)
exten => _X.,n,WaitUntil(${FUTURETIME})
exten => _X.,n,Playback(beep)
exten => _X.,n,Return()

;
; ANI context: use in the same way as "time" above
;

[ani]
exten => _X.,40000(ani),NoOp(ANI: ${EXTEN})
exten => _X.,n,Wait(0.25)
exten => _X.,n,Answer()
exten => _X.,n,Playback(vm-from)
exten => _X.,n,SayDigits(${CALLERID(ani)})
exten => _X.,n,Wait(1.25)
exten => _X.,n,SayDigits(${CALLERID(ani)})      ; playback again in case of missed digit
exten => _X.,n,Return()

; For more information on applications, just type "core show applications" at your
; friendly Asterisk CLI prompt.
;
; "core show application <command>" will show details of how you
; use that particular application in this file, the dial plan.
; "core show functions" will list all dialplan functions
; "core show function <COMMAND>" will show you more information about
; one function. Remember that function names are UPPER CASE.
[from-sip]
exten => 2000,1,Dial(SIP/2000,20)
exten => 2000,2,Hangup
exten => 2010,1, Wait(0.05)
exten => 2010, 2, Dial(SIP/2020,20)
exten => 2010, 2, Dial(SIP/2020,20)
exten => 2010,3,Hangup
exten => 2020,1,Dial(SIP/2020,20)
exten => 2020,2,Hangup
```

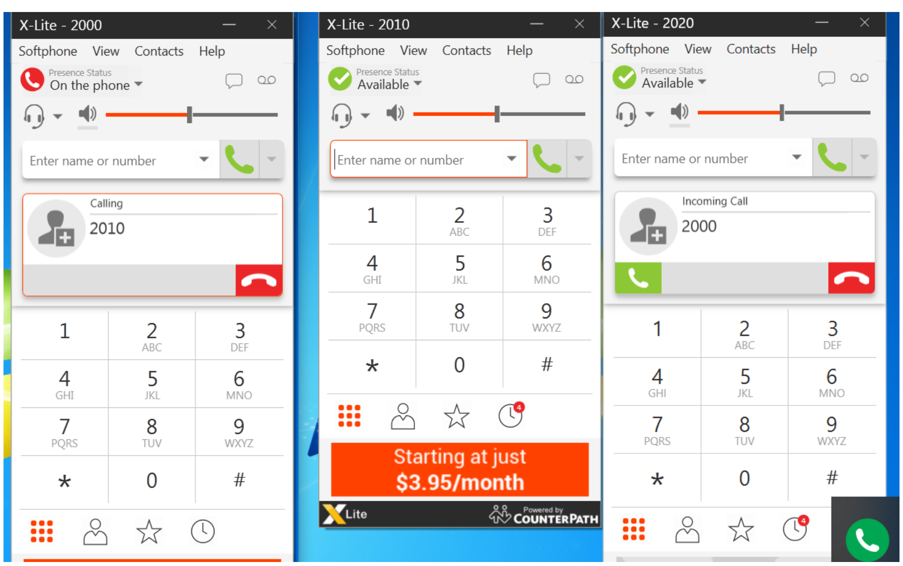
STEP3: Configure X-LITE Soft phone with the parameters similar to Part1

STEP4 : Reload to update the changes

In this part of the experiment we will use third user[2020] ,having IP addresss192.168.163.1 as shown below



The user 2000 tries to call user 2010 but the call is forwarded to user 2020 as shown below.



User 2000 sends invite to Proxy server using SIP message: INVITE sip: 2010|192.168.6.133

sip

No.	Time	Source	Destination	Protocol	Length	Info
14	20.316501865	192.168.6.135	192.168.6.133	SIP/SDP	952	Request: INVITE sip:2010@192.168.6.133 |
15	20.316881995	192.168.6.133	192.168.6.135	SIP	618	Status: 401 Unauthorized |
16	20.319101745	192.168.6.135	192.168.6.133	SIP	374	Request: ACK sip:2010@192.168.6.133 |
17	20.326876176	192.168.6.135	192.168.6.133	SIP/SDP	1112	Request: INVITE sip:2010@192.168.6.133 |
18	20.329438766	192.168.6.133	192.168.6.135	SIP	561	Status: 100 Trying |
19	20.385444640	192.168.6.133	192.168.6.1	SIP/SDP	987	Request: INVITE sip:2020@192.168.6.1:52163;rinstance=95bd101e247107ed |
20	20.604692866	192.168.6.1	192.168.6.133	SIP	340	Status: 100 Trying |
21	21.106311724	192.168.6.1	192.168.6.133	SIP	492	Status: 180 Ringing |
22	21.106587545	192.168.6.133	192.168.6.135	SIP	577	Status: 180 Ringing |
29	29.982765865	192.168.6.1	192.168.6.133	SIP/SDP	799	Status: 200 OK |
30	29.982963073	192.168.6.133	192.168.6.1	SIP	503	Request: ACK sip:2020@192.168.6.1:52163;rinstance=95bd101e247107ed |
31	29.983252805	192.168.6.133	192.168.6.135	SIP/SDP	885	Status: 200 OK |
32	29.984092285	192.168.6.133	192.168.6.1	SIP/SDP	963	Request: INVITE sip:2020@192.168.6.1:52163;rinstance=95bd101e247107ed, in-dialog |
45	30.031526577	192.168.6.135	192.168.6.133	SIP	493	Request: ACK sip:2010@192.168.6.133:5060 |
46	30.031732950	192.168.6.133	192.168.6.135	SIP/SDP	889	Request: INVITE sip:2000@192.168.6.135:55505;rinstance=3cd78b673e9dd0d7, in-dialog |
50	30.069086422	192.168.6.1	192.168.6.133	SIP	353	Status: 100 Trying |
82	30.276302339	192.168.6.1	192.168.6.133	SIP/SDP	799	Status: 200 OK |

Proxy server sends invite to user 2020 since the call is forwarded from 2010 to 2020 using SIP message: INVITE sip: 2020@192.168.6.1

	Time	Source	Destination	Protocol	Length	Info
14	20.316501865	192.168.6.135	192.168.6.133	SIP/SDP	952	Request: INVITE sip:2010@192.168.6.133 |
15	20.316881995	192.168.6.133	192.168.6.135	SIP	618	Status: 401 Unauthorized |
16	20.319101745	192.168.6.135	192.168.6.133	SIP	374	Request: ACK sip:2010@192.168.6.133 |
17	20.326876176	192.168.6.135	192.168.6.133	SIP/SDP	1112	Request: INVITE sip:2010@192.168.6.133 |
18	20.329438766	192.168.6.133	192.168.6.135	SIP	561	Status: 100 Trying |
19	20.385444640	192.168.6.133	192.168.6.1	SIP/SDP	987	Request: INVITE sip:2020@192.168.6.1:52163;rinstance=95bd101e247107ed |
20	20.604692866	192.168.6.1	192.168.6.133	SIP	340	Status: 100 Trying |
21	21.106311724	192.168.6.1	192.168.6.133	SIP	492	Status: 180 Ringing |
22	21.106587545	192.168.6.133	192.168.6.135	SIP	577	Status: 180 Ringing |
29	29.982765865	192.168.6.1	192.168.6.133	SIP/SDP	799	Status: 200 OK |
30	29.982963073	192.168.6.133	192.168.6.1	SIP	503	Request: ACK sip:2020@192.168.6.1:52163;rinstance=95bd101e247107ed |
31	29.983252805	192.168.6.133	192.168.6.135	SIP/SDP	885	Status: 200 OK |
32	29.984092285	192.168.6.133	192.168.6.1	SIP/SDP	963	Request: INVITE sip:2020@192.168.6.1:52163;rinstance=95bd101e247107ed, in-dialog |
45	30.031526577	192.168.6.135	192.168.6.133	SIP	493	Request: ACK sip:2010@192.168.6.133:5060 |
46	30.031732950	192.168.6.133	192.168.6.135	SIP/SDP	889	Request: INVITE sip:2000@192.168.6.135:55505;rinstance=3cd78b673e9dd0d7, in-dialog |
50	30.069086422	192.168.6.1	192.168.6.133	SIP	353	Status: 100 Trying |
82	30.276302339	192.168.6.1	192.168.6.133	SIP/SDP	799	Status: 200 OK |
83	30.276527743	192.168.6.133	192.168.6.1	SIP	503	Request: ACK sip:2020@192.168.6.1:52163;rinstance=95bd101e247107ed |

User 2020 sends OK msg to SIP Proxy server and the call is established

sip

	Time	Source	Destination	Protocol	Length	Info
14	20.316501865	192.168.6.135	192.168.6.133	SIP/SDP	952	Request: INVITE sip:2010@192.168.6.133 |
15	20.316881995	192.168.6.133	192.168.6.135	SIP	618	Status: 401 Unauthorized |
16	20.319101745	192.168.6.135	192.168.6.133	SIP	374	Request: ACK sip:2010@192.168.6.133 |
17	20.326876176	192.168.6.135	192.168.6.133	SIP/SDP	1112	Request: INVITE sip:2010@192.168.6.133 |
18	20.329438766	192.168.6.133	192.168.6.135	SIP	561	Status: 100 Trying |
19	20.385444640	192.168.6.133	192.168.6.1	SIP/SDP	987	Request: INVITE sip:2020@192.168.6.1:52163;rinstance=95bd101e247107ed |
20	20.604692866	192.168.6.1	192.168.6.133	SIP	340	Status: 100 Trying |
21	21.106311724	192.168.6.1	192.168.6.133	SIP	492	Status: 180 Ringing |
22	21.106587545	192.168.6.133	192.168.6.135	SIP	577	Status: 180 Ringing |
29	29.982765865	192.168.6.1	192.168.6.133	SIP/SDP	799	Status: 200 OK |
30	29.982963073	192.168.6.133	192.168.6.1	SIP	503	Request: ACK sip:2020@192.168.6.1:52163;rinstance=95bd101e247107ed |
31	29.983252805	192.168.6.133	192.168.6.135	SIP/SDP	885	Status: 200 OK |
32	29.984092285	192.168.6.133	192.168.6.1	SIP/SDP	963	Request: INVITE sip:2020@192.168.6.1:52163;rinstance=95bd101e247107ed, in-dialog |
45	30.031526577	192.168.6.135	192.168.6.133	SIP	493	Request: ACK sip:2010@192.168.6.133:5060 |
46	30.031732950	192.168.6.133	192.168.6.135	SIP/SDP	889	Request: INVITE sip:2000@192.168.6.135:55505;rinstance=3cd78b673e9dd0d7, in-dialog |
50	30.069086422	192.168.6.1	192.168.6.133	SIP	353	Status: 100 Trying |
82	30.276302339	192.168.6.1	192.168.6.133	SIP/SDP	799	Status: 200 OK |
83	30.276527743	192.168.6.133	192.168.6.1	SIP	503	Request: ACK sip:2020@192.168.6.1:52163;rinstance=95bd101e247107ed |

When the call has to be disconnected the user 2020 sends and the call is terminated as shown below

	Time	Source	Destination	Protocol	Length	Info
32	29.984092285	192.168.6.133	192.168.6.1	SIP/SDP	963	Request: INVITE sip:2020@192.168.6.1:52163;rinstance=95bd101e247107ed, in-dialog |
45	30.031526577	192.168.6.135	192.168.6.133	SIP	493	Request: ACK sip:2010@192.168.6.133:5060 |
46	30.031732950	192.168.6.133	192.168.6.135	SIP/SDP	889	Request: INVITE sip:2000@192.168.6.135:55505;rinstance=3cd78b673e9dd0d7, in-dialog |
50	30.069086422	192.168.6.1	192.168.6.133	SIP	353	Status: 100 Trying |
82	30.276302339	192.168.6.1	192.168.6.133	SIP/SDP	799	Status: 200 OK |
83	30.276527743	192.168.6.133	192.168.6.1	SIP	503	Request: ACK sip:2020@192.168.6.1:52163;rinstance=95bd101e247107ed |
98	30.330453400	192.168.6.135	192.168.6.133	SIP	329	Status: 100 Trying |
99	30.330989165	192.168.6.135	192.168.6.133	SIP/SDP	787	Status: 200 OK |
100	30.331237022	192.168.6.133	192.168.6.135	SIP	476	Request: ACK sip:2000@192.168.6.135:55505;rinstance=3cd78b673e9dd0d7 |
114	37.576319718	192.168.6.1	192.168.6.133	SIP	524	Request: BYE sip:2000@192.168.6.133:5060 |
115	37.576582574	192.168.6.133	192.168.6.1	SIP	560	Status: 200 OK |
116	37.576852397	192.168.6.133	192.168.6.135	SIP/SDP	891	Request: INVITE sip:2000@192.168.6.135:55505;rinstance=3cd78b673e9dd0d7, in-dialog |
117	37.659091979	192.168.6.135	192.168.6.133	SIP	329	Status: 100 Trying |
118	37.676964958	192.168.6.135	192.168.6.133	SIP/SDP	787	Status: 200 OK |
119	37.677178618	192.168.6.133	192.168.6.135	SIP	476	Request: ACK sip:2000@192.168.6.135:55505;rinstance=3cd78b673e9dd0d7 |
120	37.677251069	192.168.6.133	192.168.6.135	SIP	675	Request: BYE sip:2000@192.168.6.135:55505;rinstance=3cd78b673e9dd0d7 |
121	37.795618150	192.168.6.135	192.168.6.133	SIP	436	Status: 200 OK |

The figure shows the Wireshark capture for Call Forwarding