

TELEPHONY OVER IP SOLUTION

Travaux pratiques – 1

SIP REGISTER

Classe : E4CCSN



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1) Installation et configuration :

1.1 First step is to download the software from this link:

<https://www.microsip.org/downloads>

The screenshot shows the MicroSIP website's download page. The main heading is "MicroSIP Downloads - Installer and Portable version". Under "Project Information", it lists the code license as GNU GPL v2 and provides labels for SIP, PJSIP, Windows, Softphone, STUN, and ICE. It also shows compatibility logos for Windows 8, Windows 10, and Windows XP. The "File" section lists two download options: "MicroSIP-3.21.3.exe" (8 MB, portable.zip) and "MicroSIP-Lite-3.21.3.exe" (5 MB, portable.zip). A callout box with a checkmark points to the first file, stating: "We select this version as the professor mentioned the 'MicroSIP-3.21.3.exe'".

File	Size	Format	Download Count	Total
MicroSIP-3.21.3.exe	8 MB	portable.zip	497164	4,305,365
MicroSIP-Lite-3.21.3.exe	5 MB	portable.zip	71267	753,740

1.2 After we download the "MicroSIP" we install it on our machine than we open it and we go for the configuration (account configuration, server and credential ...):

The screenshot shows the MicroSIP application interface. The main window has a menu bar with "Phone", "Logs", and "Contacts". Below the menu bar is a numeric keypad with letters assigned to numbers (1-9, *, 0, #). At the bottom, there are status indicators for "Online" and "1008". A context menu is open over the application, listing options such as "Make Active", "Edit Account", "Settings", "Shortcuts", "Always on Top", "View Log File", "Visit Website", "Help", and "Exit".

The screenshot shows the "Account" configuration dialog box in MicroSIP. It contains the following fields and values:

- Account Name: Mohammed BOUCLAGHEM
- SIP Server: 194.5.159.151
- SIP Proxy: 194.5.159.151
- Username*: 1008
- Domain*: agram01
- Login: 1008
- Password: *****
- Display Name: 1008
- Voicemail Number: (empty)
- Dialing Prefix: (empty)
- Dial Plan: (empty)
- Hide Caller ID:
- Media Encryption: Disabled
- Transport: UDP
- Public Address: Auto
- Register Refresh: 300
- Keep-Alive: 15
- Publish Presence:
- Allow IP Rewrite:
- ICE:
- Disable Session Timers:

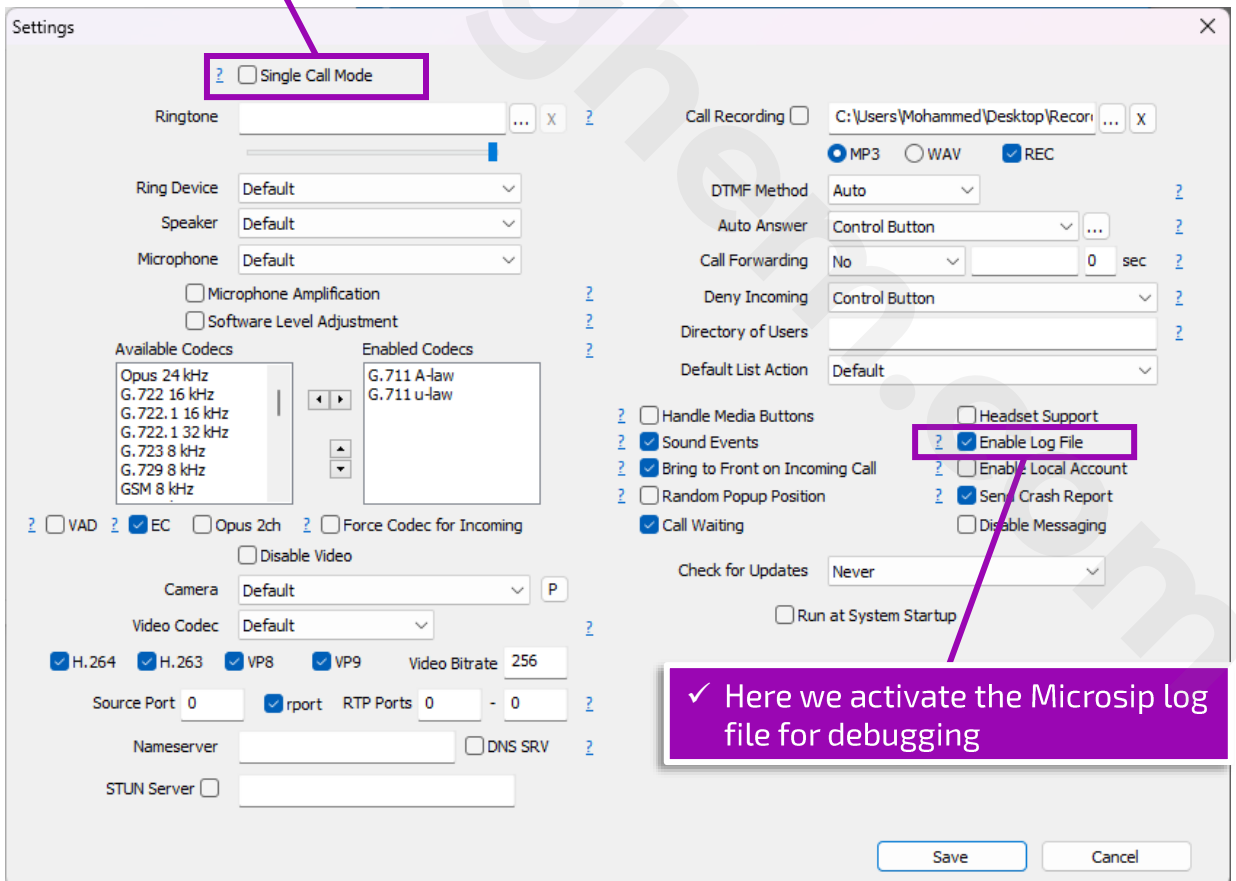
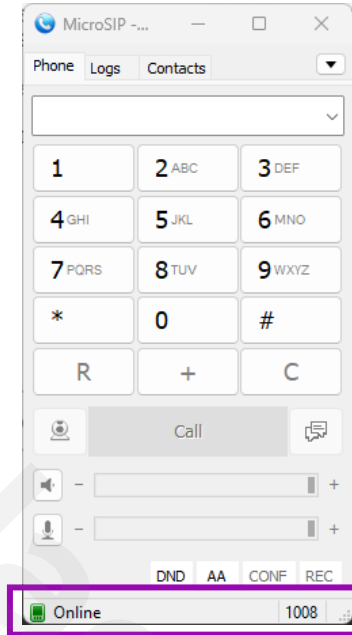
Buttons for "Save" and "Cancel" are visible at the bottom.

1) Installation et configuration :

1.3 Now the “MicroSIP” is *online* and ready to start calls and receive is but we still need to configure some parameters for the logs to check them later:

✓ We must disable this it will allow us to manage multiple calls, and conference calls

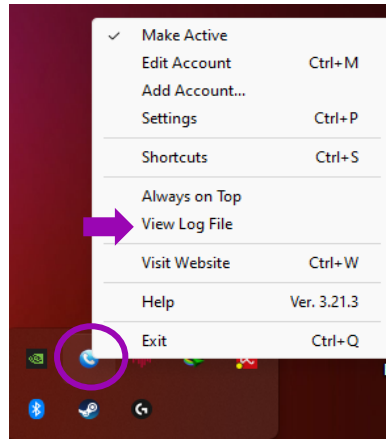
✓ Our softphone is connected to the server and ready to receive and make calls



✓ Here we activate the Microsip log file for debugging

2) Phase REGISTER :

2.1 To open log file right click on tray icon.



2.2 Than we look for the REGISTER PHASE as the screenshot shows:

```
19:20:24.155 sip_endpoint.c .Module "mod-pjsua-10
63 REGISTER sip:194.5.159.151 SIP/2.0
64 Via: SIP/2.0/UDP 192.168.1.51:63181;rport;branch=z9hG4bKPj692766057af1446ca686162f21db4700
65 Route: <sip:194.5.159.151;lr>
66 Max-Forwards: 70
67 From: "1008" <sip:1008@agraham01>;tag=2af10a3751d34eca99f551ed8978b1f8
68 To: "1008" <sip:1008@agraham01>
69 Call-ID: fb4b323151eb4021a86b6a4494a1ad9e
70 CSeq: 36210 REGISTER
71 User-Agent: MicroSIP/3.21.3
72 Contact: "1008" <sip:1008@192.168.1.51:63181;ob>
73 Expires: 300
74 Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, INFO, SUBSCRIBE, NOTIFY, REFER, MESSAGE, OPTIONS
75 Content-Length: 0
76
```

- 1 My address IP (Private @IP) : 192.168.1.51
- 2 My public address IP because I am outside : 91.174.23*.*** (got it from a web site : <https://whatismyipaddress.com/>).

I couldn't find it on my own log I think is my router configuration or something like that hide it the only addresses I found are the private of my computer 192.168.1.19 and the server address 194.5.159.151.

2) Phase REGISTER :

2 But if we look into the professor log file we will find the public address IP of ESTIAM : **178.16.174.1**

```
874 REGISTER sip:194.5.159.151 SIP/2.0
875 Via: SIP/2.0/UDP 10.13.14.29:52685;rport;branch=z9hG4bKPj4640fecfd49b4171a2d22587dadbd938
876 Route: <sip:194.5.159.151;lr>
877 Max-Forwards: 70
878 From: "1000" <sip:1000@agraham01>;tag=5f10cc6564a648c8a0229cc74efb94ae
879 To: "1000" <sip:1000@agraham01>
880 Call-ID: 1c51473e258243789564951603486e9f
881 CSeq: 33740 REGISTER
882 User-Agent: MicroSIP/3.21.3
883 Contact: "1000" <sip:1000@10.13.14.29:52685;ob>
884 Expires: 300
885 Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, INFO, SUBSCRIBE, NOTIFY, REFER, MESSAGE, OPTIONS
886 Authorization: Digest username="1000", realm="agraham01", nonce="6244dd7b-1a0a-47ca-a174-58aab2e4f595", uri="sip:194.
response="5760fceca7773cbf1ab8ef821ed4af6a", algorithm=MD5, cnonce="6ee316cd776b4e0ab0b101f10f935deb", qop=auth, nc=0
887 Content-Length: 0
888
889
890 --end msg--
891 16:11:29.189 tsx03C1BF14 ...State changed from Null to Calling, event=TX_MSG
892 16:11:29.212 sip_endpoint.c Processing incoming message: Response msg 200/REGISTER/cseq=33740 (rdata03BFD084)
893 16:11:29.213 pjsua_core.c .RX 636 bytes Response msg 200/REGISTER/cseq=33740 (rdata03BFD084) from UDP 194.5.159.15
894 SIP/2.0 200 OK
895 Via: SIP/2.0/UDP 10.13.14.29:52685;rport=9105;branch=z9hG4bKPj4640fecfd49b4171a2d22587dadbd938;received=178.16.174.1
896 From: "1000" <sip:1000@agraham01>;tag=5f10cc6564a648c8a0229cc74efb94ae
```

The register phase allows us to follow and track the routing of the call.

We can see that the call leaves first of all from the Sip server with the domain

agraham01 which has the address **194.5.159.151** then goes through the internet network and finally enters the **ESTIAM network** and retrieves the IP address of the network which is **178.16.174.1**, and finally routes the traffic to the corresponding private address .

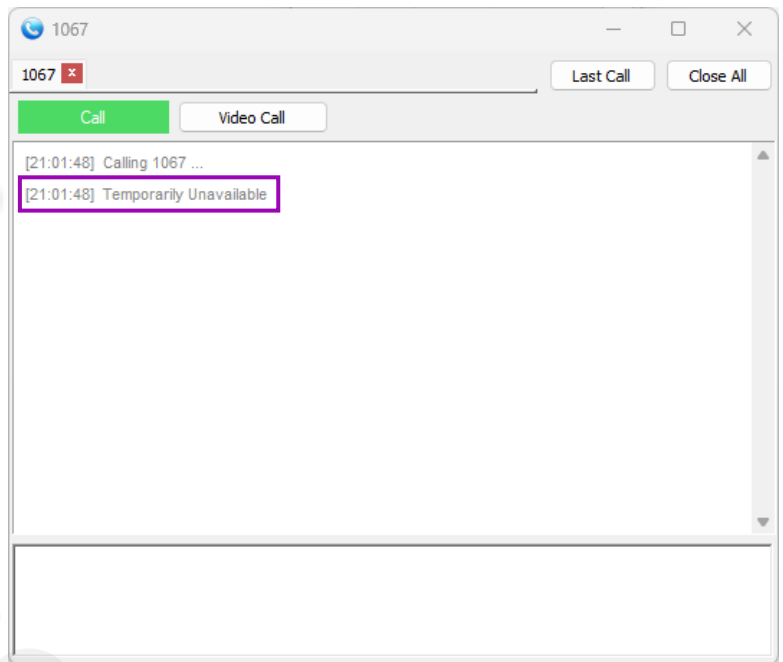
3) Phase INVITE :

3.1 Now we just try to call an absent colleague (Calling 1067) :

1

As we can see here the 1067 is temporarily unavailable

1



```
7014 INVITE sip:1067@agraham01 SIP/2.0
7015 Via: SIP/2.0/UDP 192.168.1.51:63181;rport;branch=z9hG4bKJcc093f71eb3a46909f480f25f23d4560
7016 Max-Forwards: 70
7017 From: "1008" <sip:1008@agraham01>;tag=b1c7803650e04e309caad17284112555
7018 To: <sip:1067@agraham01>
7019 Contact: "1008" <sip:1008@192.168.1.51:63181;ob>
7020 Call-ID: 68b8913583c948428c4d442c3c144033
7021 CSeq: 22049 INVITE
7022 Route: <sip:194.5.159.151;lr>
7023 Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, INFO, SUBSCRIBE, NOTIFY, REFER, MESSAGE, OPTIONS
7024 Supported: replaces, 100rel, timer, norefersub
7025 Session-Expires: 1800
7026 Min-SE: 90
7027 User-Agent: MicroSIP/3.21.3
7028 Proxy-Authorization: Digest username="1008", realm="agraham01", nonce="a4e9a8ba-6078-4d64-81c6-ddd002d69f2e", uri="sip:1067@agraham01",
response="a3ef0a6b515e8d8a59b24371426d654b", algorithm=MD5, cnonce="9688705ec17f4164905b2d04ca9ed1e7", qop=auth, nc=00000001
7029 Content-Type: application/sdp
7030 Content-Length: 339
```

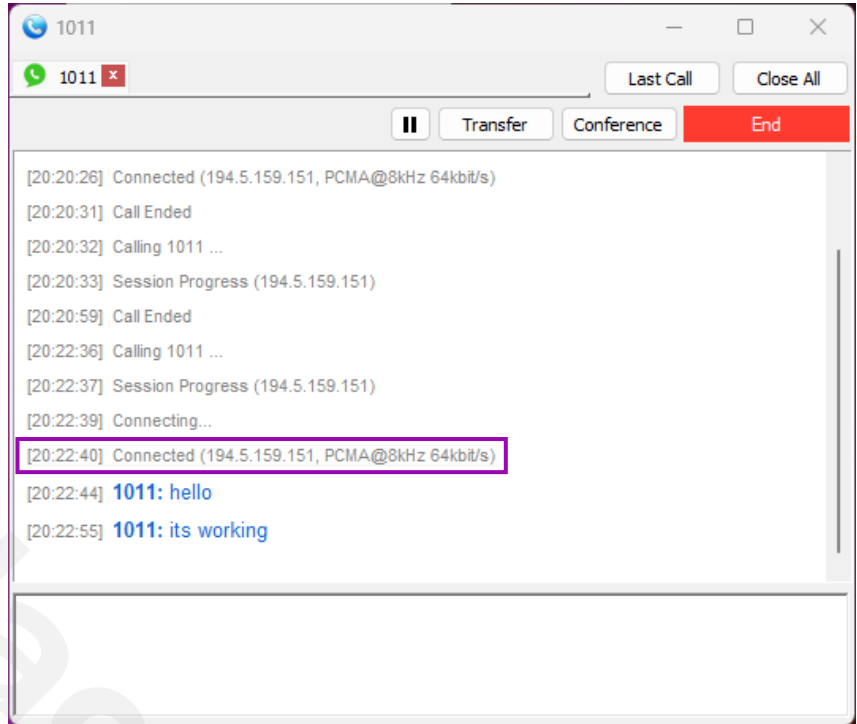
...

```
7070 21:01:48.878 _pisa core.c .RX 880 bytes Response msg 480/INVITE/cseq=22049 (rdata04865EDC) from UDP 194.5.159.151:5060:
7071 SIP/2.0 480 Temporarily Unavailable
7072 Via: SIP/2.0/UDP 192.168.1.51:63181;rport=63181;branch=z9hG4bKJcc093f71eb3a46909f480f25f23d4560
7073 Max-Forwards: 69
7074 From: "1008" <sip:1008@agraham01>;tag=b1c7803650e04e309caad17284112555
7075 To: <sip:1067@agraham01>;tag=4NaK0NmV7yBQF
7076 Call-ID: 68b8913583c948428c4d442c3c144033
7077 CSeq: 22049 INVITE
7078 User-Agent: FreeSWITCH-mod_sofia/1.10.8-release.14~64bit
7079 Accept: application/sdp
7080 Allow: INVITE, ACK, BYE, CANCEL, OPTIONS, MESSAGE, INFO, UPDATE, REGISTER, REFER, NOTIFY, PUBLISH, SUBSCRIBE
7081 Supported: timer, path, replaces
7082 Allow-Events: talk, hold, conference, presence, as-feature-event, dialog, line-seize, call-info, sla, include-session-description, presence.winfo,
message-summary, refer
7083 Reason: Q.850;cause=16;text="NORMAL_CLEARING"
7084 Content-Length: 0
7085 Remote-Party-ID: "1067" <sip:1067@agraham01>;party=calling;privacy=off;screen=no
```

So he we tried to call a non-registered number the 1067 and the log file shows us that it try to call but got no reply so it mentioned it as unavailable for the moment

3) Phase INVITE :

3.2 Now we just try to call a registered colleague (Calling 1011) :



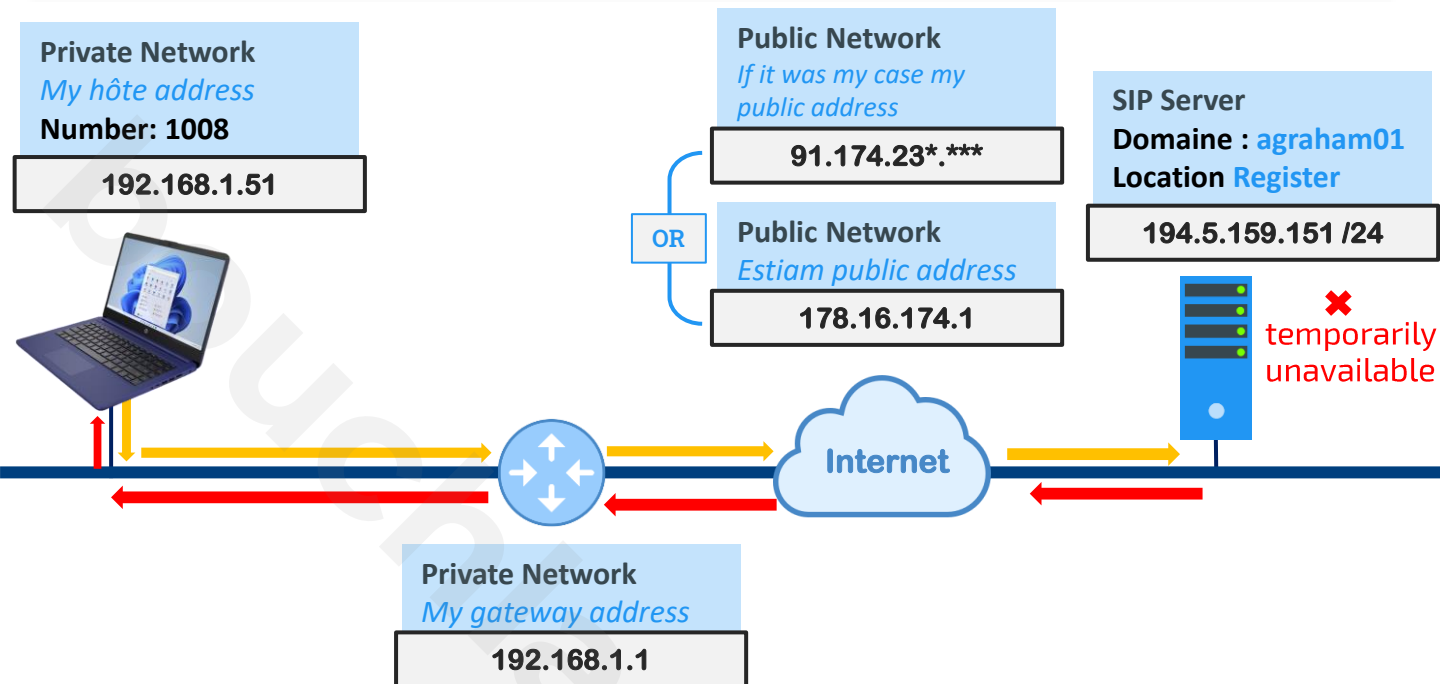
1 As we can see here the 1011 is available and the call started, and we get a SIP/2.0 487 Request Terminated,

```
3400 SIP/2.0 200 OK
3401 Via: SIP/2.0/UDP 192.168.1.51:63181;rport=63181;branch=z9hG4bKpj17bad698ff354c3fad151f31eaa2de42
3402 From: "1008" <sip:1008@agraham01>;tag=dc351032561241f2a7d087543117ef7d
3403 To: <sip:1011@agraham01>;tag=NKUm8FX0K7c5r
3404 Call-ID: 2533e0d85c7c41bcac476cef7598fb21
3405 CSeq: 9726 ACK
3406 Content-Length: 0
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3423 20:20:59.819 pjsua_core.c .RX 702 bytes Response msg 487/INVITE/cseq=9726 (rdata04865EDC) from UDP 194.5.159.151:5060:
3424 SIP/2.0 487 Request Terminated
3425 Via: SIP/2.0/UDP 192.168.1.51:63181;rport=63181;branch=z9hG4bKpj17bad698ff354c3fad151f31eaa2de42
3426 From: "1008" <sip:1008@agraham01>;tag=dc351032561241f2a7d087543117ef7d
3427 To: <sip:1011@agraham01>;tag=NKUm8FX0K7c5r
3428 Call-ID: 2533e0d85c7c41bcac476cef7598fb21
3429 CSeq: 9726 INVITE
3430 User-Agent: FreeSWITCH-mod_sofia/1.10.8-release.14~64bit
3431 Allow: INVITE, ACK, BYE, CANCEL, OPTIONS, MESSAGE, INFO, UPDATE, REGISTER, REFER, NOTIFY, PUBLISH, SUBSCRIBE
3432 Supported: timer, path, replaces
3433 Allow-Events: talk, hold, conference, presence, as-feature-event, dialog, line-seize, call-info, sla, include-session-descri
message-summary, refer
3434 Content-Length: 0
3435
3436
3437 --end msg--
3438 20:20:59.819 tsx0486F4EC .Incoming Response msg 487/INVITE/cseq=9726 (rdata04865EDC) in state Proceeding
3439 20:20:59.819 endpoint ..Request msg ACK/cseq=9726 (tdta0333512C) created.
3440 20:20:59.819 pjsua_core.c ..TX 370 bytes Request msg ACK/cseq=9726 (tdta0333512C) to UDP 194.5.159.151:5060:
3441 ACK sip:1011@agraham01 SIP/2.0
3442 Via: SIP/2.0/UDP 192.168.1.51:63181;rport;branch=z9hG4bKpj17bad698ff354c3fad151f31eaa2de42
3443 Max-Forwards: 70
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```

So he we called a registered number the 1011 and the log file shows us that it try to call and it succeed when we get the Request Terminated in the SIP level.

4) Media Stream Path :

Calling a non-registered number the (1067)



```

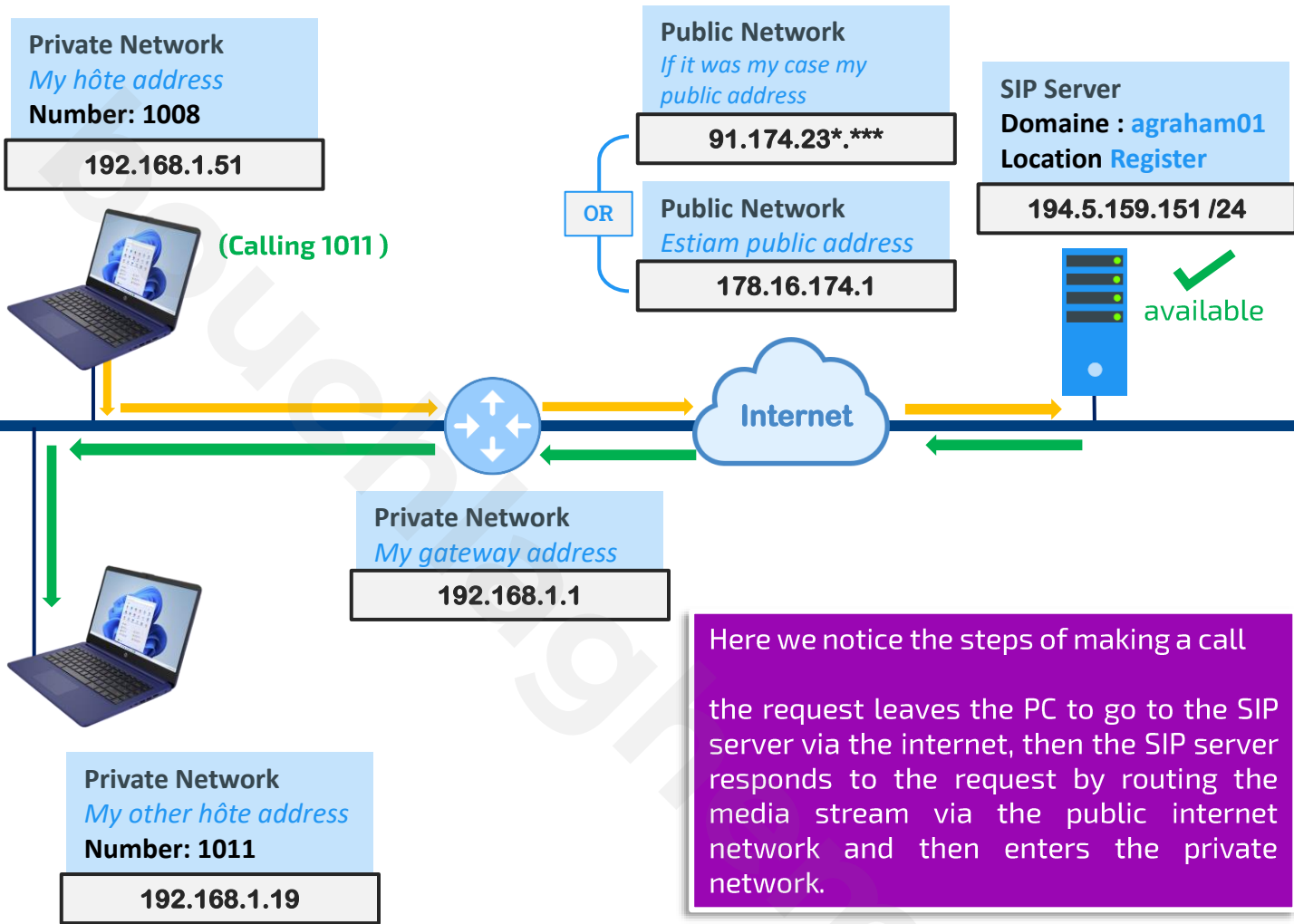
7049 --end msg--
7050 21:01:48.803 tsx0487352C .....State changed from Null to Calling, event=TX_MSG
7051 21:01:48.803 dlG0486F4EC .....Transaction tsx0487352C state changed to Calling
7052 21:01:48.846 sip_endpoint.c Processing incoming message: Response msg 100/INVITE/cseq=22049 (rdata04865EDC)
7053 21:01:48.846 pjsua_core.c .RX 358 bytes Response msg 100/INVITE/cseq=22049 (rdata04865EDC) from UDP 194.5.159.151:5060:
7054 SIP/2.0 100 Trying
7055 Via: SIP/2.0/UDP 192.168.1.51:63181;rport=63181;branch=z9hG4bKPjcc093f71eb3a46909f480f25f23d4560
7056 From: "1008" <sip:1008@agraham01>;tag=b1c7803650e04e309caad17284112555
7057 To: <sip:1067@agraham01>
7058 Call-ID: 68b8913583c948428c4d442c3c144033
7059 CSeq: 22049 INVITE
7060 User-Agent: FreeSWITCH-mod_sofia/1.10.8-release.14~64bit
7061 Content-Length: 0
7062
7063
7064 --end msg--
7065 21:01:48.846 tsx0487352C .Incoming Response msg 100/INVITE/cseq=22049 (rdata04865EDC) in state Calling
7066 21:01:48.846 tsx0487352C ..State changed from Calling to Proceeding, event=RX_MSG
7067 21:01:48.846 dlG0486F4EC ...Received Response msg 100/INVITE/cseq=22049 (rdata04865EDC)
7068 21:01:48.846 dlG0486F4EC ...Transaction tsx0487352C state changed to Proceeding
7069 21:01:48.878 sip_endpoint.c Processing incoming message: Response msg 480/INVITE/cseq=22049 (rdata04865EDC)
7070 21:01:48.878 pjsua_core.c .RX 880 bytes Response msg 480/INVITE/cseq=22049 (rdata04865EDC) from UDP 194.5.159.151:5060:
7071 SIP/2.0 480 Temporarily Unavailable
    
```

We notice the change in state from Null to Calling, then it goes to Proceeding then our softphone process the incoming message, then we get the SIP/2.0 480 Temporarily Unavailable.

So Firstly the request leaves the PC to go to the SIP server via the internet, then the SIP server responds to the request by routing the media stream via the public internet network and then enters the private network.

4) Media Stream Path :

Calling a registered colleague (Calling 1011)



Here we notice the steps of making a call
the request leaves the PC to go to the SIP server via the internet, then the SIP server responds to the request by routing the media stream via the public internet network and then enters the private network.

```

3452 --end msg--
3453 20:20:59.819 tsx0486F4EC ..State changed from Proceeding to Completed, event=RX_MSG
3454 20:20:59.819 dlg048704FC ...Received Response msg 487/INVITE/cseq=9726 (rdata04865EDC)
3455 20:20:59.819 dlg048704FC ...Transaction tsx0486F4EC state changed to Completed
3456 20:20:59.819 dlg048704FC ...Session count dec to 1 by mod-invite
3457 20:20:59.819 tdtta0484CE84 ..Destroying txdata Request msg INVITE/cseq=9726 (tdta0484CE84)
3458 20:20:59.919 wav_player.c !File port C:\Users\Mohammed\AppData\Local\MicroSIP\hangup.wav EOF
3459 20:20:59.919 pjsua_aud.c !Conf disconnect: 1 -x- 0
3460 20:20:59.919 conference.c .Port 1 (C:\Users\Mohammed\AppData\Local\MicroSIP\hangup.wav) stop transmitting to port 0 (Wave mapper)
3461 20:20:59.919 pjsua_aud.c Destroying player 0..
3462 20:21:00.920 pjsua_aud.c !Closing sound device after idle for 1 second(s)
3463 20:21:00.920 pjsua_aud.c .Closing Wave mapper sound playback device and Wave mapper sound capture device
3464 20:21:00.921 wmmme_dev.c .Stopped WMME playback stream
3465 20:21:00.922 wmmme_dev.c .Stopped WMME capture stream
3466 20:21:00.922 wmmme_dev.c .Stopped WMME playback stream
3467 20:21:00.922 wmmme_dev.c .Stopped WMME capture stream
3468 20:21:00.922 wmmme_dev.c !WMME: thread stopping..
3469 20:21:00.922 wmmme Thread stack max usage=9473 by ..\src\pj\string.c:483
3470 20:21:04.426 tsx0486ECE4 !Timeout timer event
3471 20:21:04.426 tsx0486ECE4 .State changed from Completed to Terminated, event=TIMER
3472 20:21:04.426 dlg048704FC ..Transaction tsx0486ECE4 state changed to Terminated
3473 20:21:04.426 tsx0486ECE4 !Timeout timer event
3474 20:21:04.426 tsx0486ECE4 .State changed from Terminated to Destroyed, event=TIMER
3475 20:21:04.426 tdtta0330B5EC ..Destroying txdata Request msg ACK/cseq=9725 (tdta0330B5EC)
3476 20:21:04.426 tsx0486ECE4 Transaction destroyed!
3477 20:21:04.818 tsx0487453C !Timeout timer event
3478 20:21:04.818 tsx0487453C .State changed from Completed to Terminated, event=TIMER
3479 20:21:04.818 dlg048704FC ..Transaction tsx0487453C state changed to Terminated
3480 20:21:04.818 tsx0487453C !Timeout timer event
3481 20:21:04.818 tsx0487453C .State changed from Terminated to Destroyed, event=TIMER
3482 20:21:04.818 tdtta0332A184 ..Destroying txdata Request msg CANCEL/cseq=9726 (tdta0332A184)
3483 20:21:04.818 tsx0487453C Transaction destroyed!
3484 20:21:07.263 pjsua_acc.c Sending 2 bytes keep-alive packet for acc 0 to 194.5.159.151:5060
3485 20:21:07.263 tsx0333014E ..Transaction tsx0333014E state changed to Completed
  
```

5) Video CALL :

Here is the log file or the lines from starting the video call and the state changes that happened during the initiation of the video call

```
12545 BYE sip:1011@194.5.159.151:5060;transport=udp SIP/2.0
12546 Via: SIP/2.0/UDP 192.168.1.51:63181;rport;branch=z9hG4bKPje2433915c061464f866a432e8f50ab78
12547 Max-Forwards: 70
12548 From: "1008" <sip:1008@agraham01>;tag=ba33477d41964d458fbb4be0de6959bc
12549 To: <sip:1011@agraham01>;tag=Uj77XcUymDtrS
12550 Call-ID: 15a4bed0c06a43a599e9ea58f0271e2e
12551 CSeq: 10545 BYE
12552 User-Agent: MicroSIP/3.21.3
12553 Content-Length: 0
12554
12555 --end msg--
12556 23:03:52.594 tsx0487352C ...State changed from Null to Calling, event=TX_MSG
12558 23:03:52.594 dlg0487251C ....Transaction tsx0487352C state changed to Calling
12559 23:03:52.607 pjsua_aud.c Creating file player: C:\Users\Mohammed\AppData\Local\MicroSIP\hangup.wav..
12560 23:03:52.608 wav_player.c .File player 'C:\Users\Mohammed\AppData\Local\MicroSIP\hangup.wav' created: samp.rate=8000, ch=1, bufsize=3KB, filesize=3KB
12561 23:03:52.608 pjsua_aud.c .Player created, id=0, slot=1
12562 23:03:52.608 wav_player.c pjmedia_wav_player_set_eof_cb() is deprecated. Use pjmedia_wav_player_set_eof_cb2() instead.
12563 23:03:52.608 pjsua_aud.c Set sound device: capture=-1, playback=-2, mode=0
12564 23:03:52.608 pjsua_aud.c .No changes in capture and playback devices
12565 23:03:52.608 pjsua_aud.c Conf connect: 1 --> 0
12566 23:03:52.608 conference.c .Port 1 (C:\Users\Mohammed\AppData\Local\MicroSIP\hangup.wav) transmitting to port 0 (Wave mapper)
12567 23:03:52.623 pjsua_vid.c Stopping preview for cap_dev=1
12568 23:03:52.623 vid_conf.c .Port 0 (OBS Virtual Camera) stop transmitting to port 1 (SDL renderer)
12569 23:03:52.623 sdl_dev.c .Stopping sdl video stream
12570 23:03:52.623 pjsua_vid.c .Window 0: destroying..
12571 23:03:52.623 vid_conf.c ..Removed port 0 (OBS Virtual Camera)
12572 23:03:52.624 dshow_dev.c ..Stopping dshow video stream
12573 23:03:52.624 vid_port.c ..Closing OBS Virtual Camera..
12574 23:03:52.624 dshow_dev.c ..Stopping dshow video stream
12575 23:03:52.630 sip_endpoint.c !Processing incoming message: Response msg 200/BYE/cseq=10545 (rdata04865EDC)
12576 23:03:52.630 pjsua_core.c .RX 513 bytes Response msg 200/BYE/cseq=10545 (rdata04865EDC) from UDP 194.5.159.151:5060:
12577 SIP/2.0 200 OK
12578 Via: SIP/2.0/UDP 192.168.1.51:63181;rport=63181;branch=z9hG4bKPje2433915c061464f866a432e8f50ab78
12579 From: "1008" <sip:1008@agraham01>;tag=ba33477d41964d458fbb4be0de6959bc
12580 To: <sip:1011@agraham01>;tag=Uj77XcUymDtrS
12581 Call-ID: 15a4bed0c06a43a599e9ea58f0271e2e
12582 CSeq: 10545 BYE
12583 User-Agent: FreeSWITCH-mod sofia/1.10.8-release.14-64bit
12584 Allow: INVITE, ACK, BYE, CANCEL, OPTIONS, MESSAGE, INFO, UPDATE, REGISTER, REFER, NOTIFY, PUBLISH, SUBSCRIBE
12585 Supported: timer, path, replaces
12586 Content-Length: 0
12587
```

1 My virtual camera

```
12473 [CONFIRMED] To: <sip:1011@agraham01>;tag=Uj77XcUymDtrS
12474 Call time: 00h:00m:12s, 1st res in 179 ms, conn in 6473ms
12475 #0 audio PCMA @8kHz, sendrecv, peer=194.5.159.151:40289
12476 EC stat: WebRTC delay metric: median=0, std=40, frac of poor delay=0,49
12477 SRTP status: Not active Crypto-suite:
12478 RX pt=8, last update:00h:00m:03.667s ago
12479 total 898pkt 143.6KB (179.6KB +IP hdr) @avg=62.5Kbps/78.2Kbps
12480 pkt loss=0 (0,0%), discrd=0 (0,0%), dup=0 (0,0%), reord=0 (0,0%)
12481 (msec) min avg max last dev
12482 loss period: 0,000 0,000 0,000 0,000 0,000
12483 jitter : 0,000 8,921 396,750 0,125 6,155
12484 TX pt=8, ptime=20, last update:00h:00m:00.255s ago
12485 total 919pkt 147.0KB (183.8KB +IP hdr) @avg=64.0Kbps/80.0Kbps
12486 pkt loss=3 (0,3%), dup=0 (0,0%), reorder=0 (0,0%)
12487 (msec) min avg max last dev
12488 loss period: 20,000 30,000 40,000 40,000 10,000
12489 jitter : 0,000 0,021 0,125 0,000 0,046
12490 RTT msec : 52,856 53,019 53,298 53,298 0,198
12491 #1 video VP8, sendrecv, peer=194.5.159.151:41848
12492 SRTP status: Not active Crypto-suite:
12493 RX pt=100, size=720x480, fps=30,00, last update:00h:00m:04.353s ago
12494 total 279pkt 68.1KB (79.2KB +IP hdr) @avg=29.6Kbps/34.5Kbps
12495 pkt loss=0 (0,0%), discrd=0 (0,0%), dup=0 (0,0%), reord=0 (0,0%)
12496 (msec) min avg max last dev
12497 loss period: 0,000 0,000 0,000 0,000 0,000
12498 jitter : 0,322 5,730 22,244 3,122 4,359
12499 TX pt=100, size=720x480, fps=15,00, last update:00h:00m:00.007s ago
12500 total 275pkt 13.7KB (24.7KB +IP hdr) @avg=6.0Kbps/10.7Kbps
12501 pkt loss=1 (0,4%), dup=0 (0,0%), reorder=0 (0,0%)
12502 (msec) min avg max last dev
12503 loss period: 0,000 0,000 0,000 0,000 0,000
12504 jitter : 4,700 54,117 66,000 66,000 22,399
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4) Video CALL :

Here is the log file or the lines from starting the video call and the state changes that happened during the initiation of the video call

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12416 --end msg--
12417 23:03:40.508 tsx0487352C Timeout timer event
12418 23:03:40.508 tsx0487352C .State changed from Terminated to Destroyed, event=TIMER
12419 23:03:40.508 tsx0487352C Transaction destroyed!
12420 23:03:40.537 strm03224A74 !Jitter buffer empty (prefetch=0), plc invoked
12421 23:03:40.597 strm03224A74 Jitter buffer starts returning normal frames (after 3 empty/lost)
12422 23:03:40.617 strm03224A74 Jitter buffer empty (prefetch=0), plc invoked
12423 23:03:40.657 strm03224A74 Jitter buffer starts returning normal frames (after 2 empty/lost)
12424 23:03:40.828 vpx.c !Frame size changed: 352x288 --> 720x480
12425 23:03:40.828 vstdec0331B614 codec decode() error: Codec frame is too short (PJMEDIA_CODEC_EFRMTOOSHORT)
12426 23:03:40.845 vstdec0331B614 Decoding format changed: 720x480 I420<- 30/1(~30)fps
12427 23:03:40.845 pjsua_media.c !Call 3: Media 1: Received media event, type=FMCH, src=03222564, epub=0331BF9C
12428 23:03:40.845 vid_conf.c Port 3 (vstdec0331B614): updated frame rate 50 -> 30
12429 23:03:40.845 vid_conf.c Updating render state for port id 2 (1 sources)..
12430 23:03:40.845 vid_conf.c Created render state for connection 3->2
12431 23:03:40.845 vid_conf.c src#0=I420/720x480->586x480@67,0 dst=352x288@0,0
12432 23:03:40.846 vid_conf.c Port 3 (vstdec0331B614): updated frame size 352x288 -> 720x480
12433 23:03:40.846 sdl_dev.c Stopping sdl video stream
12434 23:03:40.852 mainDlg.cpp !Event FMCH
12435 23:03:40.896 sdl_dev.c !Starting sdl video stream
12436 23:03:40.896 pjsua_media.c Call 3: Media 1: Received media event, type=FMCH, src=03222564, epub=047F7F34
12437 23:03:40.897 vid_conf.c !Port 2 (SDL renderer): updated frame rate 75 -> 45
12438 23:03:40.898 vid_conf.c Updating render state for port id 2 (1 sources)..
12439 23:03:40.898 vid_conf.c Cleaned up render state for connection 3->2
12440 23:03:40.898 vid_conf.c This port only has single source with matched format & size, no conversion needed
12441 23:03:40.898 vid_conf.c Port 2 (SDL renderer): updated frame size 352x288 -> 720x480
12442 23:03:40.915 mainDlg.cpp !Event FMCH
12443 23:03:41.677 strm03224A74 !Frame lost, recovered!
12444 23:03:41.677 strm03224A74 Jitter buffer starts returning normal frames (after 1 empty/lost)
12445 23:03:41.877 strm03224A74 Jitter buffer empty (prefetch=0), plc invoked
12446 23:03:42.138 strm03224A74 Jitter buffer starts returning normal frames (after 13 empty/lost)
12447 23:03:42.312 pjsua_media.c !Call 3: Media 1: Received media event, type=RTFB, src=0331BB64, epub=0331B614
12448 23:03:42.313 pjsua_media.c !Call 3: Media 1: Received media event, type=RTFB, src=0331BB64, epub=0331B614
12449 23:03:42.322 mainDlg.cpp !Event RTFB
12450 23:03:42.322 mainDlg.cpp Event RTFB
12451 23:03:42.361 vstenc0331B614 !Forcing encoder to generate keyframe
12452 23:03:42.364 vstenc0331B614 Keyframe generated
12453 23:03:42.667 pjsua_acc.c !Sending 2 bytes keep-alive packet for acc 0 to 194.5.159.151:5060
12454 23:03:42.667 tdt12742ECC Destroying txdata raw
12455 23:03:43.837 strm03224A74 !Jitter buffer empty (prefetch=0), plc invoked
12456 23:03:43.877 strm03224A74 Jitter buffer starts returning normal frames (after 2 empty/lost)
12457 23:03:46.271 udp032E85F8 !Remote RTP address switched to 194.5.159.151:23501
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