TELEPHONY OVER IP SOLUTION

Travaux pratiques – 1

SIP REGISTER

Classe : E4CCSN

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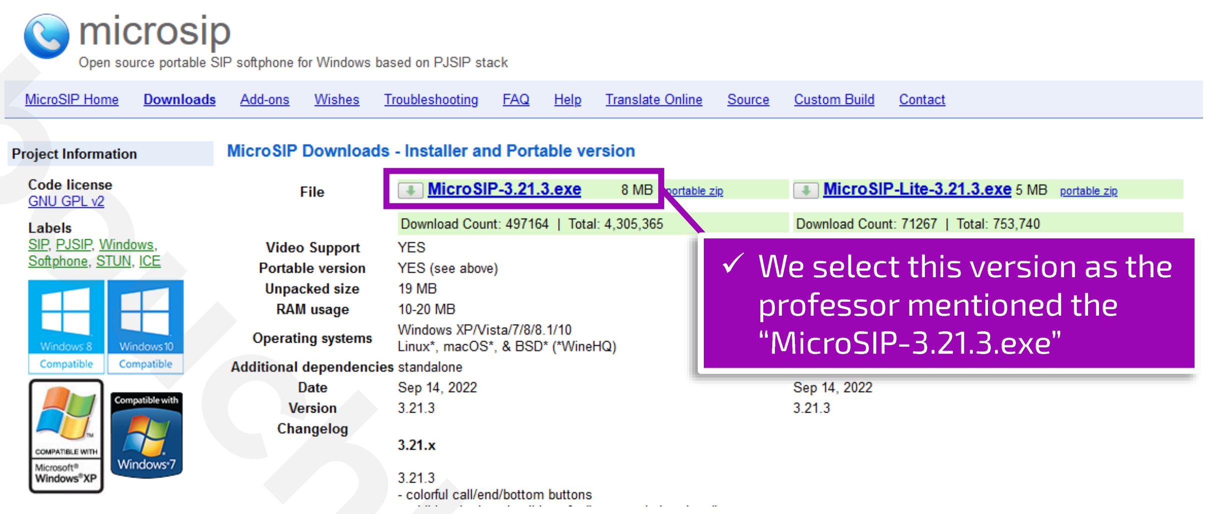
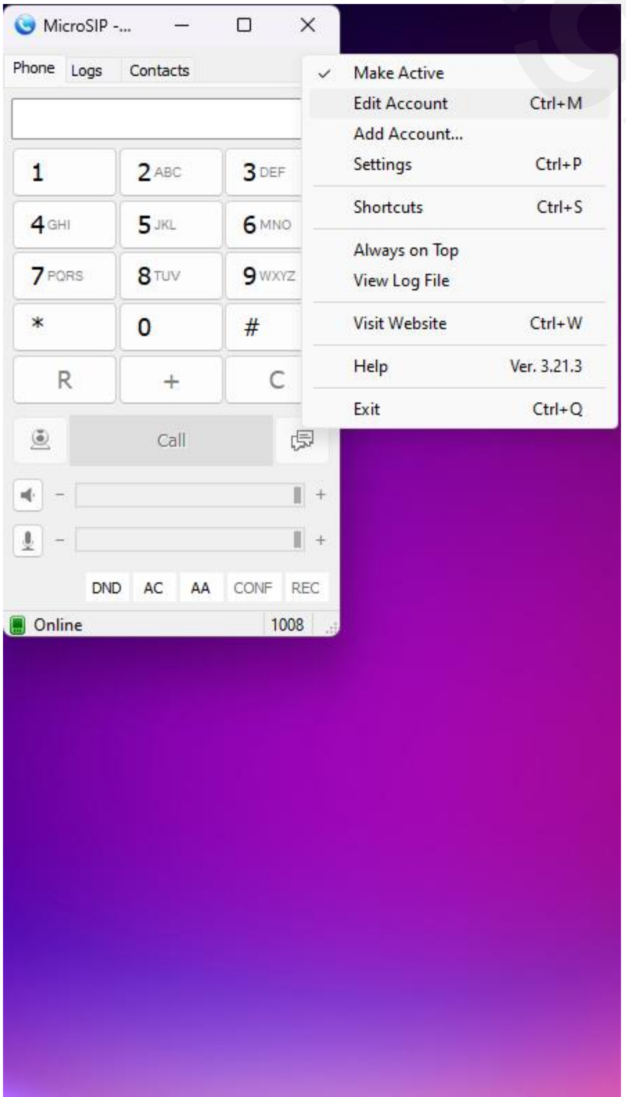
1) Installation et configuration :

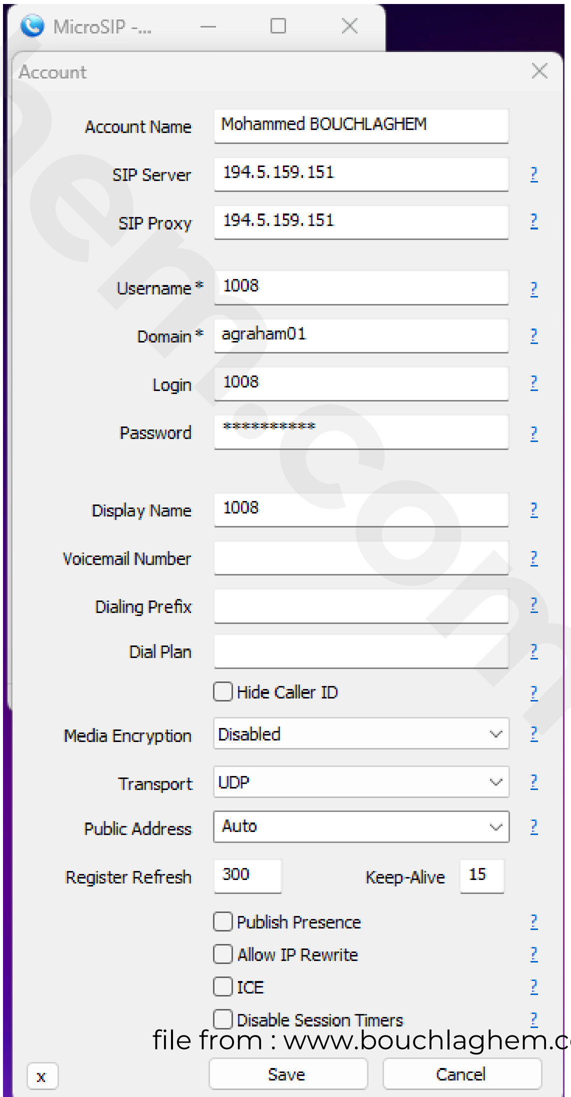
1.1 First step is to download the software from this link:

https://www.microsip.org/downloads



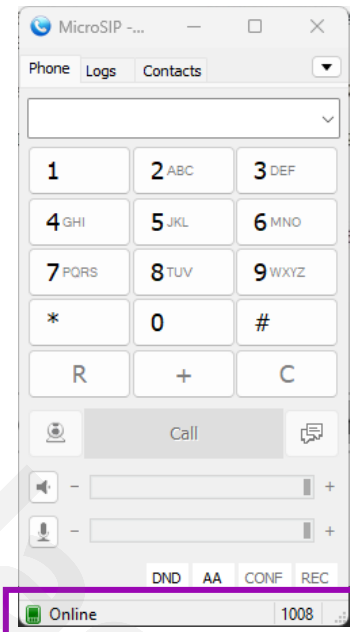
1.2 After we download the "MicroSIP" we install it on our machine than we open it and we go for the configuration (account configuration, server and credential …):





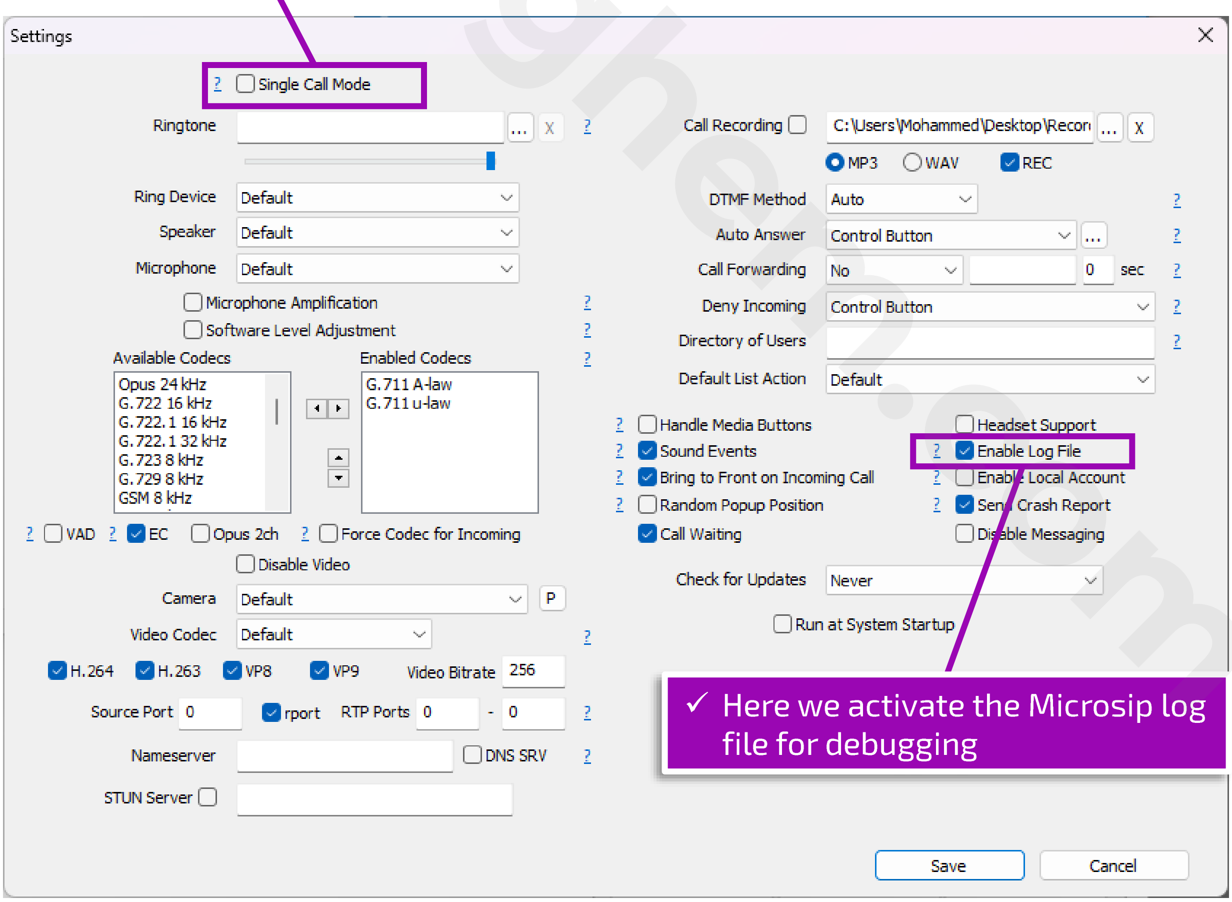
1) Installation et configuration :

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:



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

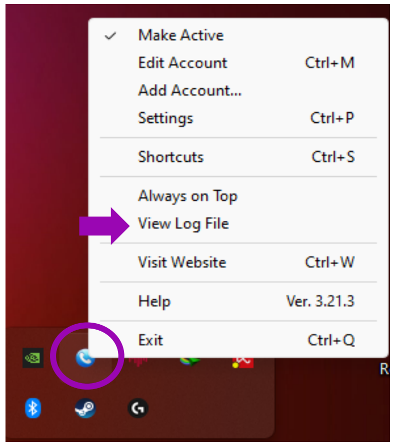
✓ We must disable this it will allow us to manage multiple calls, and conference calls



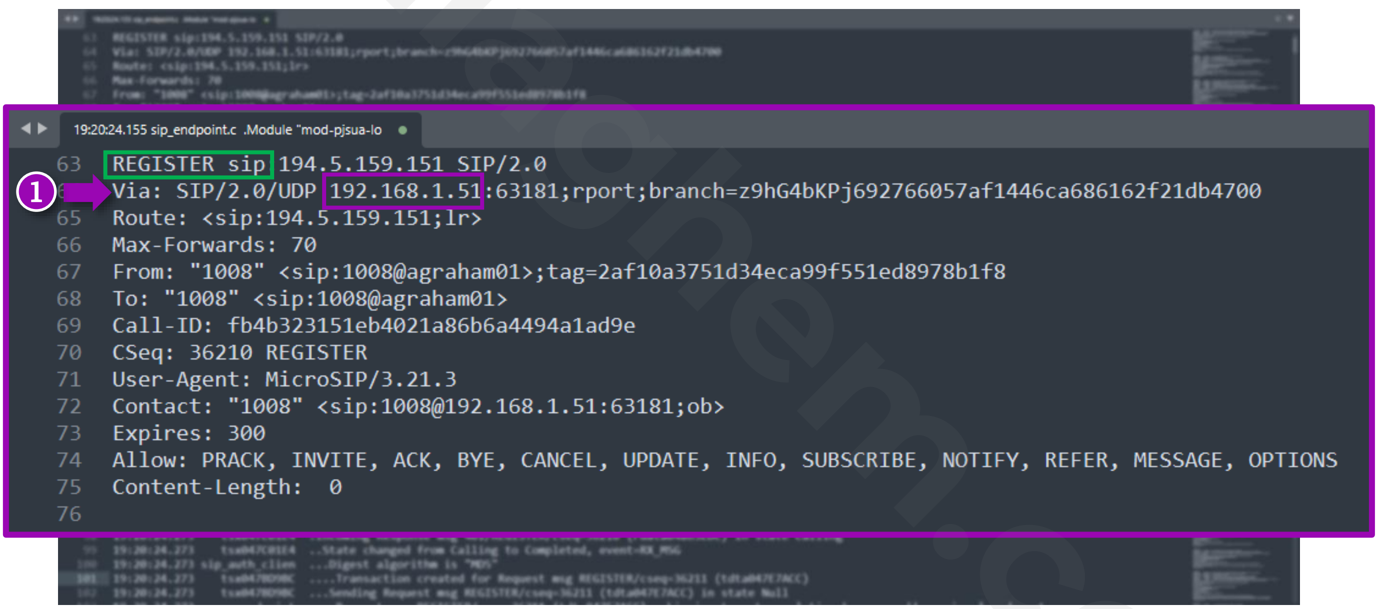
✓ Here we activate the Microsip log file for debugging

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:



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

```
REGISTER sip:194.5.159.151 SIP/2.0
Via: SIP/2.0/UDP 10.13.14.29:52685;rport;branch=z9hG4bKPj4640fecfd49b4171a2d22587dadbd938
Route: <sip:194.5.159.151;lr>
Max-Forwards: 70
From: "1000" <sip:1000@agraham01>;tag=5f10cc6564a648c8a0229cc74efb94ae
To: "1000" <sip:1000@agraham01>
Call-ID: 1c51473e258243789564951603486e9f
CSeq: 33740 REGISTER
User-Agent: MicroSIP/3.21.3
Contact: "1000" <sip:1000@10.13.14.29:52685;ob>
Expires: 300
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, INFO, SUBSCRIBE, NOTIFY, REFER, MESSAGE, OPTIONS
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
Content-Length:  0


--end msg--
16:11:29.189    tsx03C1BF14  ....State changed from Null to Calling, event=TX_MSG
16:11:29.212 sip_endpoint.c  Processing incoming message: Response msg 200/REGISTER/cseq=33740 (rdata03BFD084)
16:11:29.213   pjsua_core.c  .RX 636 bytes Response msg 200/REGISTER/cseq=33740 (rdata03BFD084) from UDP 194.5.159.15
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.13.14.29:52685;rport=9105;branch=z9hG4bKPj4640fecfd49b4171a2d22587dadbd938;received=178.16.174.1
From: "1000" <sip:1000@agraham01>;tag=5f10cc6564a648c8a0229cc74efb94ae
```

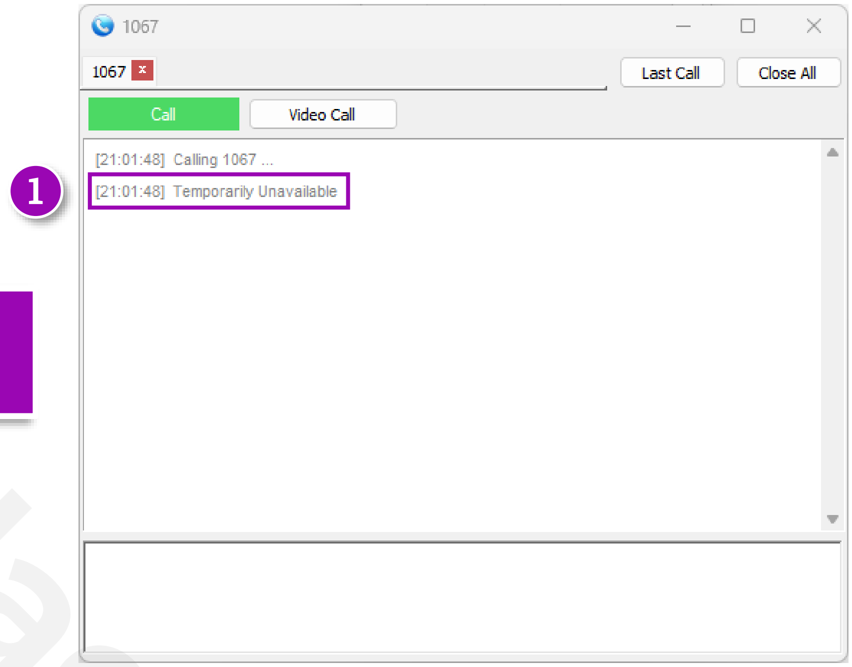
2

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

```
INVITE sip:1067@agraham01 SIP/2.0
Via: SIP/2.0/UDP 192.168.1.51:63181;rport;branch=z9hG4bKPjcc093f71eb3a46909f480f25f23d4560
Max-Forwards: 70
From: "1008" <sip:1008@agraham01>;tag=b1c7803650e04e309caad17284112555
To: <sip:1067@agraham01>
Contact: "1008" <sip:1008@192.168.1.51:63181;ob>
Call-ID: 68b8913583c948428c4d442c3c144033
CSeq: 22049 INVITE
Route: <sip:194.5.159.151;lr>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, INFO, SUBSCRIBE, NOTIFY, REFER, MESSAGE, OPTIONS
Supported: replaces, 100rel, timer, norefersub
Session-Expires: 1800
Min-SE: 90
User-Agent: MicroSIP/3.21.3
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
Content-Type: application/sdp
Content-Length:   339
```

...

1

```
21:01:48.878   pjsua_core.c  .RX 880 bytes Response msg 480/INVITE/cseq=22049 (rdata04865EDC) from UDP 194.5.159.151:5060:
SIP/2.0 480 Temporarily Unavailable
Via: SIP/2.0/UDP 192.168.1.51:63181;rport=63181;branch=z9hG4bKPjcc093f71eb3a46909f480f25f23d4560
Max-Forwards: 69
From: "1008" <sip:1008@agraham01>;tag=b1c7803650e04e309caad17284112555
To: <sip:1067@agraham01>;tag=4NaK0Nmv7yBQF
Call-ID: 68b8913583c948428c4d442c3c144033
CSeq: 22049 INVITE
User-Agent: FreeSWITCH-mod_sofia/1.10.8-release.14~64bit
Accept: application/sdp
Allow: INVITE, ACK, BYE, CANCEL, OPTIONS, MESSAGE, INFO, UPDATE, REGISTER, REFER, NOTIFY, PUBLISH, SUBSCRIBE
Supported: timer, path, replaces
Allow-Events: talk, hold, conference, presence, as-feature-event, dialog, line-seize, call-info, sla, include-session-description, presence.winfo,
message-summary, refer
Reason: Q.850;cause=16;text="NORMAL_CLEARING"
Content-Length: 0
Remote-Party-ID: "1067" <sip:1067@agraham01>;party=calling;privacy=off;screen=no
```

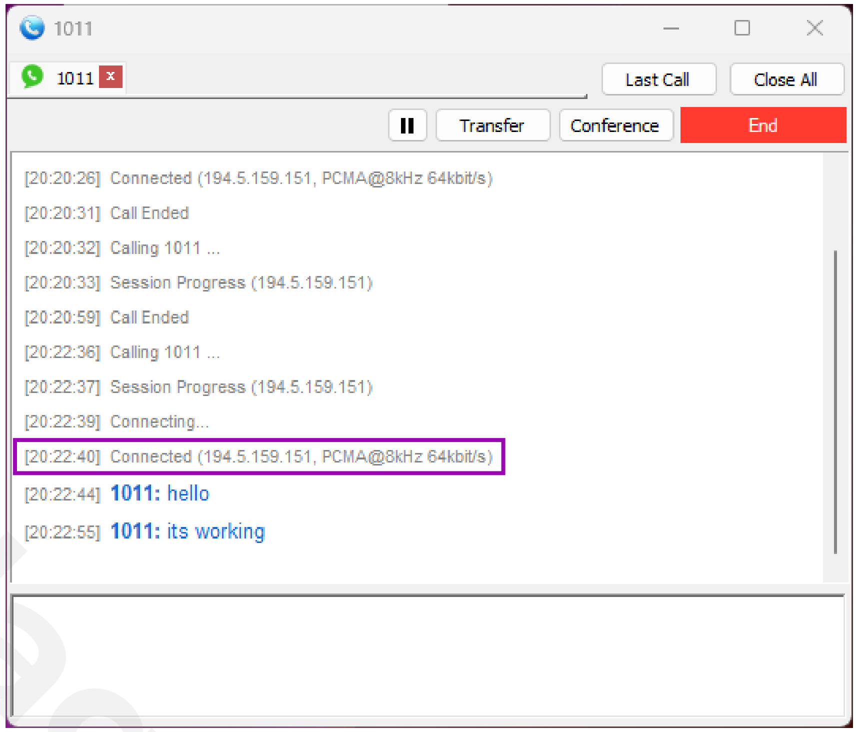
So he we tried to call a non-registered number the 1067 and the log file shows us that it try to call but got no reply so it mentioned it as unavailable for the moment

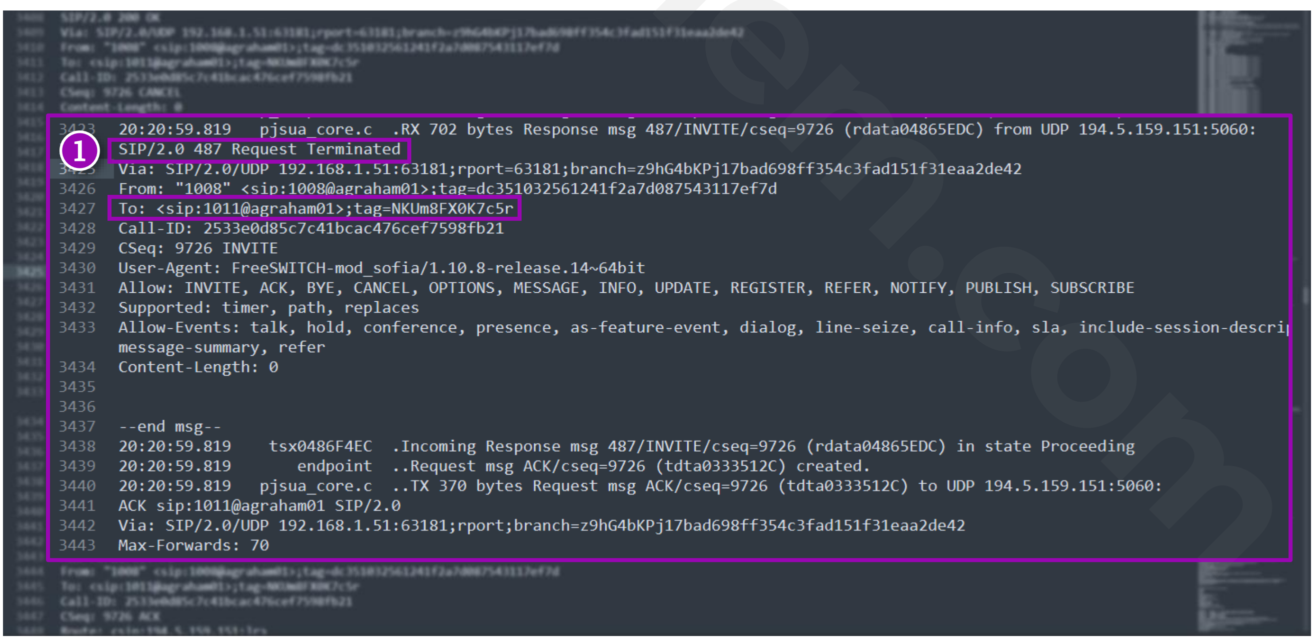
3) Phase INVITE :

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,

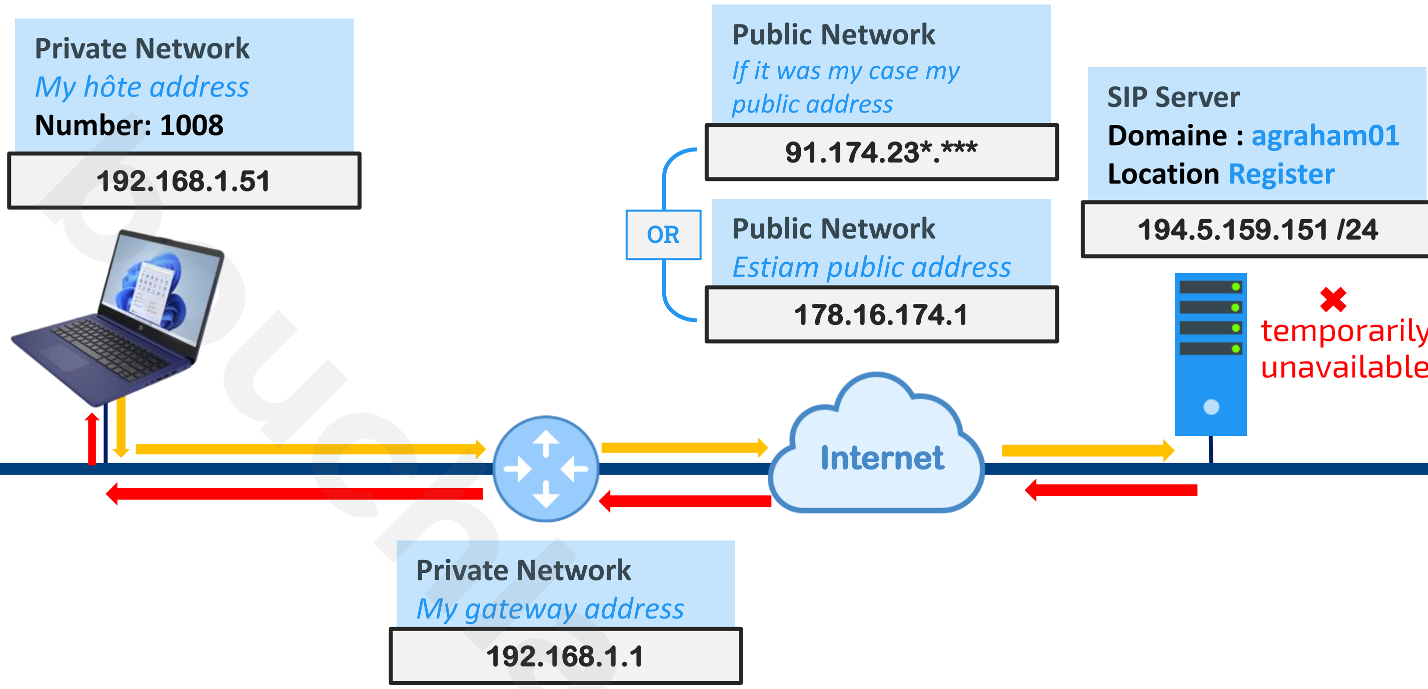




So he we called a registered number the 1011 and the log file shows us that it try to call and it succeed when we get the Request Terminated in the SIP level.

4) Media Stream Path :

Calling a non-registered number the (1067)



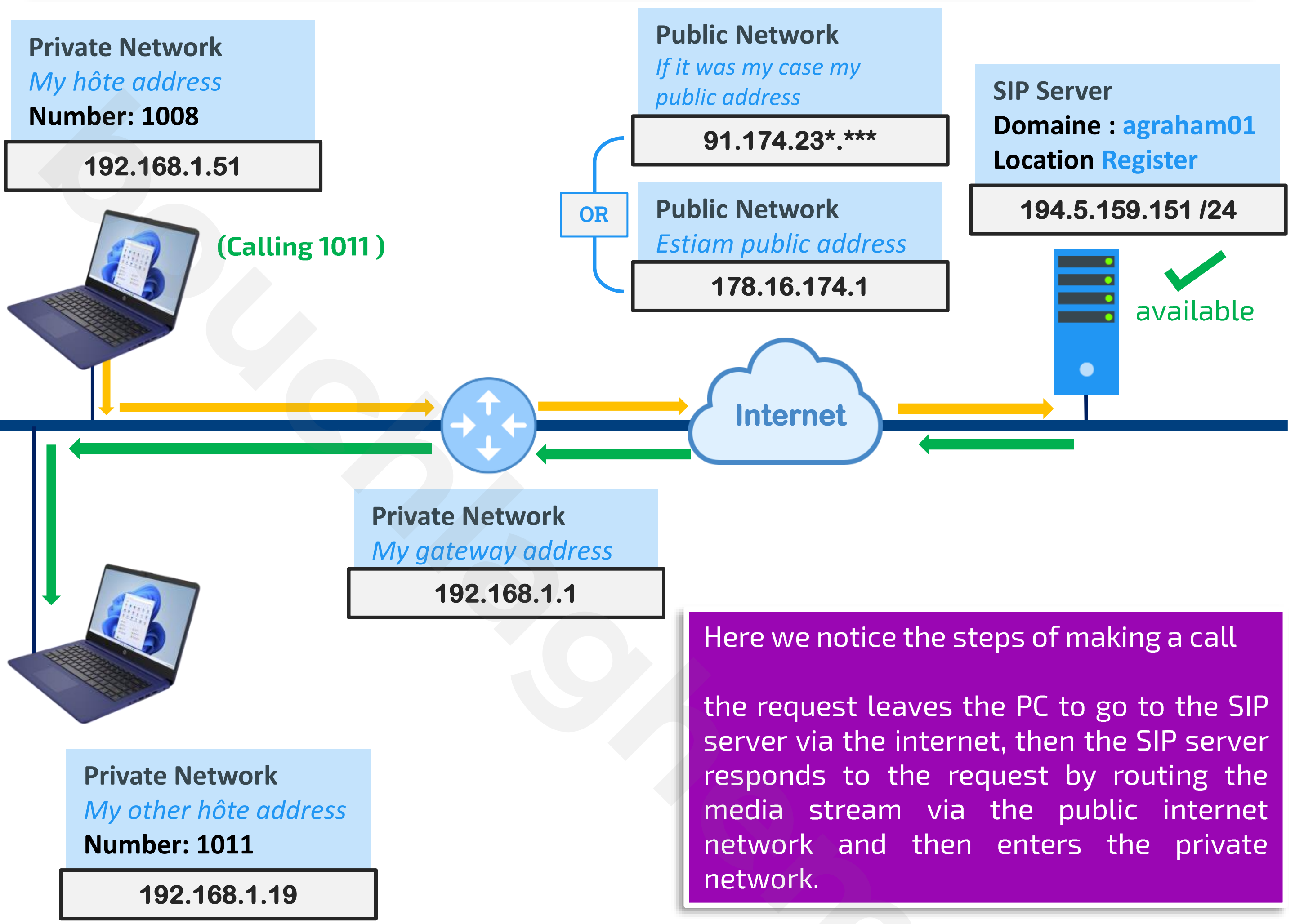

```
--end msg--
21:01:48.803    tsx0487352C  .......State changed from Null to Calling, event=TX_MSG
21:01:48.803    dlg0486F4EC  ........Transaction tsx0487352C state changed to Calling
21:01:48.846 sip_endpoint.c  Processing incoming message: Response msg 100/INVITE/cseq=22049 (rdata04865EDC)
21:01:48.846   pjsua_core.c  .RX 358 bytes Response msg 100/INVITE/cseq=22049 (rdata04865EDC) from UDP 194.5.159.151:5060:
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 192.168.1.51:63181;rport=63181;branch=z9hG4bKPjcc093f71eb3a46909f480f25f23d4560
From: "1008" <sip:1008@agraham01>;tag=b1c7803650e04e309caad17284112555
To: <sip:1067@agraham01>
Call-ID: 68b8913583c948428c4d442c3c144033
CSeq: 22049 INVITE
User-Agent: FreeSWITCH-mod_sofia/1.10.8-release.14~64bit
Content-Length: 0


--end msg--
21:01:48.846    tsx0487352C  .Incoming Response msg 100/INVITE/cseq=22049 (rdata04865EDC) in state Calling
21:01:48.846    tsx0487352C  ..State changed from Calling to Proceeding, event=RX_MSG
21:01:48.846    dlg0486F4EC  ...Received Response msg 100/INVITE/cseq=22049 (rdata04865EDC)
21:01:48.846    dlg0486F4EC  ...Transaction tsx0487352C state changed to Proceeding
21:01:48.878 sip_endpoint.c  Processing incoming message: Response msg 480/INVITE/cseq=22049 (rdata04865EDC)
21:01:48.878   pjsua_core.c  .RX 880 bytes Response msg 480/INVITE/cseq=22049 (rdata04865EDC) from UDP 194.5.159.151:5060:
SIP/2.0 480 Temporarily Unavailable
```

We notice the change in state from Null to Calling, than it goes to Proceeding than our softphone process the incoming message, than 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 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.

```
--end msg--
20:20:59.819    tsx0486F4EC  ..State changed from Proceeding to Completed, event=RX_MSG
20:20:59.819    dlg048704FC  ...Received Response msg 487/INVITE/cseq=9726 (rdata04865EDC)
20:20:59.819    dlg048704FC  ...Transaction tsx0486F4EC state changed to Completed
20:20:59.819    dlg048704FC  .....Session count dec to 1 by mod-invite
20:20:59.819   tdta0484CE84  ..Destroying txdata Request msg INVITE/cseq=9726 (tdta0484CE84)
20:20:59.919   wav_player.c !File port C:\Users\Mohammed\AppData\Local\MicroSIP\hangup.wav EOF
20:20:59.919    pjsua_aud.c !Conf disconnect: 1 -x- 0
20:20:59.919   conference.c  .Port 1 (C:\Users\Mohammed\AppData\Local\MicroSIP\hangup.wav) stop transmitting to port 0 (Wave mapper)
20:20:59.919    pjsua_aud.c  Destroying player 0..
20:21:00.920    pjsua_aud.c !Closing sound device after idle for 1 second(s)
20:21:00.920    pjsua_aud.c  .Closing Wave mapper sound playback device and Wave mapper sound capture device
20:21:00.921     wmme_dev.c  .Stopped WMME playback stream
20:21:00.922     wmme_dev.c  .Stopped WMME capture stream
20:21:00.922     wmme_dev.c  .Stopped WMME playback stream
20:21:00.922     wmme_dev.c  .Stopped WMME capture stream
20:21:00.922     wmme_dev.c !WMME: thread stopping..
20:21:00.922           wmme  Thread stack max usage=9473 by ..\src\pj\string.c:483
20:21:04.426    tsx0486ECE4 !Timeout timer event
20:21:04.426    tsx0486ECE4  .State changed from Completed to Terminated, event=TIMER
20:21:04.426    dlg048704FC  ..Transaction tsx0486ECE4 state changed to Terminated
20:21:04.426    tsx0486ECE4  Timeout timer event
20:21:04.426    tsx0486ECE4  .State changed from Terminated to Destroyed, event=TIMER
20:21:04.426   tdta0330B5EC  ..Destroying txdata Request msg ACK/cseq=9725 (tdta0330B5EC)
20:21:04.426    tsx0486ECE4  Transaction destroyed!
20:21:04.818    tsx0487453C  Timeout timer event
20:21:04.818    tsx0487453C  .State changed from Completed to Terminated, event=TIMER
20:21:04.818    dlg048704FC  ..Transaction tsx0487453C state changed to Terminated
20:21:04.818    tsx0487453C  Timeout timer event
20:21:04.818    tsx0487453C  .State changed from Terminated to Destroyed, event=TIMER
20:21:04.818   tdta0332A184  ..Destroying txdata Request msg CANCEL/cseq=9726 (tdta0332A184)
20:21:04.818    tsx0487453C  Transaction destroyed!
20:21:07.263    pjsua_acc.c  Sending 2 bytes keep-alive packet for acc 0 to 194.5.159.151:5060
```

5) Video CALL :

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

```
19:20:24.155 sip_endpoint.c .Module "mod-pjsua-lo    19:20:24.155 sip_endpoint.c .Module "mod-pjsua-lo
BYE sip:1011@194.5.159.151:5060;transport=udp SIP/2.0
Via: SIP/2.0/UDP 192.168.1.51:63181;rport;branch=z9hG4bKPje2433915c061464f866a432e8f50ab78
Max-Forwards: 70
From: "1008" <sip:1008@agraham01>;tag=ba33477d41964d458fbb4be0de6959bc
To: <sip:1011@agraham01>;tag=Uj77XcUymDtrS
Call-ID: 15a4bed0c06a43a599e9ea58f0271e2e
CSeq: 10545 BYE
User-Agent: MicroSIP/3.21.3
Content-Length:  0


--end msg--
23:03:52.594    tsx0487352C  ....State changed from Null to Calling, event=TX_MSG
23:03:52.594    dlg0487251C  .....Transaction tsx0487352C state changed to Calling
23:03:52.607    pjsua_aud.c  Creating file player: C:\Users\Mohammed\AppData\Local\MicroSIP\hangup.wav..
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
23:03:52.608    pjsua_aud.c  .Player created, id=0, slot=1
23:03:52.608   wav_player.c  pjmedia_wav_player_set_eof_cb() is deprecated. Use pjmedia_wav_player_set_eof_cb2() instead.
23:03:52.608    pjsua_aud.c  Set sound device: capture=-1, playback=-2, mode=0
23:03:52.608    pjsua_aud.c  .No changes in capture and playback devices
23:03:52.608    pjsua_aud.c  Conf connect: 1 --> 0
23:03:52.608   conference.c  .Port 1 (C:\Users\Mohammed\AppData\Local\MicroSIP\hangup.wav) transmitting to port 0 (Wave mapper)
23:03:52.623    pjsua_vid.c  Stopping preview for cap_dev=-1
23:03:52.623     vid_conf.c  .Port 0 (OBS Virtual Camera) stop transmitting to port 1 (SDL renderer)
23:03:52.623      sdl_dev.c  .Stopping sdl video stream
23:03:52.623    pjsua_vid.c  .Window 0: destroying..
23:03:52.623     vid_conf.c  ..Removed port 0 (OBS Virtual Camera)
23:03:52.624     dshow_dev.c ..Stopping dshow video stream
23:03:52.624     vid_port.c  ..Closing OBS Virtual Camera..
23:03:52.624     dshow_dev.c ..Stopping dshow video stream
23:03:52.630 sip_endpoint.c !Processing incoming message: Response msg 200/BYE/cseq=10545 (rdata04865EDC)
23:03:52.630   pjsua_core.c  .RX 513 bytes Response msg 200/BYE/cseq=10545 (rdata04865EDC) from UDP 194.5.159.151:5060:
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.1.51:63181;rport=63181;branch=z9hG4bKPje2433915c061464f866a432e8f50ab78
From: "1008" <sip:1008@agraham01>;tag=ba33477d41964d458fbb4be0de6959bc
To: <sip:1011@agraham01>;tag=Uj77XcUymDtrS
Call-ID: 15a4bed0c06a43a599e9ea58f0271e2e
CSeq: 10545 BYE
User-Agent: FreeSWITCH-mod_sofia/1.10.8-release.14~64bit
Allow: INVITE, ACK, BYE, CANCEL, OPTIONS, MESSAGE, INFO, UPDATE, REGISTER, REFER, NOTIFY, PUBLISH, SUBSCRIBE
Supported: timer, path, replaces
Content-Length: 0
```

1 My virtual camera

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

4) Video CALL :

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

```
--end msg--
23:03:40.508    tsx0487352C  Timeout timer event
23:03:40.508    tsx0487352C  .State changed from Terminated to Destroyed, event=TIMER
23:03:40.508    tsx0487352C  Transaction destroyed!
23:03:40.537   strm03224A74 !Jitter buffer empty (prefetch=0), plc invoked
23:03:40.597   strm03224A74  Jitter buffer starts returning normal frames (after 3 empty/lost)
23:03:40.617   strm03224A74  Jitter buffer empty (prefetch=0), plc invoked
23:03:40.657   strm03224A74  Jitter buffer starts returning normal frames (after 2 empty/lost)
23:03:40.828          vpx.c !Frame size changed: 352x288 --> 720x480
23:03:40.828 vstdec0331B614  codec decode() error: Codec frame is too short (PJMEDIA_CODEC_EFRMTOOSHORT)
23:03:40.845 vstdec0331B614   Decoding format changed: 720x480 I420<- 30/1(~30)fps
23:03:40.845  pjsua_media.c !Call 3: Media 1: Received media event, type=FMCH, src=03222564, epub=0331BF9C
23:03:40.845     vid_conf.c  Port 3 (vstdec0331B614): updated frame rate 50 -> 30
23:03:40.845     vid_conf.c  Updating render state for port id 2 (1 sources)..
23:03:40.845     vid_conf.c  Created render state for connection 3->2
23:03:40.845     vid_conf.c  src#0=I420/720x480->586x480@67,0 dst=352x288@0,0
23:03:40.846     vid_conf.c  Port 3 (vstdec0331B614): updated frame size 352x288 -> 720x480
23:03:40.846      sdl_dev.c  Stopping sdl video stream
23:03:40.852    mainDlg.cpp !Event FMCH
23:03:40.896      sdl_dev.c !Starting sdl video stream
23:03:40.896  pjsua_media.c  Call 3: Media 1: Received media event, type=FMCH, src=03222564, epub=047F7F34
23:03:40.897     vid_conf.c !Port 2 (SDL renderer): updated frame rate 75 -> 45
23:03:40.898     vid_conf.c  Updating render state for port id 2 (1 sources)..
23:03:40.898     vid_conf.c  Cleaned up render state for connection 3->2
23:03:40.898     vid_conf.c  This port only has single source with matched format & size, no conversion needed
23:03:40.898     vid_conf.c  Port 2 (SDL renderer): updated frame size 352x288 -> 720x480
23:03:40.915    mainDlg.cpp !Event FMCH
23:03:41.677   strm03224A74 !Frame lost, recovered!
23:03:41.677   strm03224A74  Jitter buffer starts returning normal frames (after 1 empty/lost)
23:03:41.877   strm03224A74  Jitter buffer empty (prefetch=0), plc invoked
23:03:42.138   strm03224A74  Jitter buffer starts returning normal frames (after 13 empty/lost)
23:03:42.312  pjsua_media.c !Call 3: Media 1: Received media event, type=RTFB, src=0331BB64, epub=0331B614
23:03:42.313  pjsua_media.c !Call 3: Media 1: Received media event, type=RTFB, src=0331BB64, epub=0331B614
23:03:42.322    mainDlg.cpp !Event RTFB
23:03:42.322    mainDlg.cpp  Event RTFB
23:03:42.361 vstenc0331B614 !Forcing encoder to generate keyframe
23:03:42.364 vstenc0331B614  Keyframe generated
23:03:42.667    pjsua_acc.c !Sending 2 bytes keep-alive packet for acc 0 to 194.5.159.151:5060
23:03:42.667   tdta12742ECC  Destroying txdata raw
23:03:43.837   strm03224A74 !Jitter buffer empty (prefetch=0), plc invoked
23:03:43.877   strm03224A74  Jitter buffer starts returning normal frames (after 2 empty/lost)
23:03:46.271    udp032E85F8 !Remote RTCP address switched to 194.5.159.151:23501
```