ALE Application Partner Program
Inter-Working Report

Partner: **Audiocodes**
Application type: **Analog Media Gateway**
Application name: **MediaPack MP118**
Alcatel-Lucent Enterprise Platform:
OXO Connect™
OXO Connect Evolution™

The product and release listed have been tested with the Alcatel-Lucent Enterprise Communication Platform and the release specified hereinafter. The tests concern only the inter-working between the AAPP member's product and the Alcatel-Lucent Enterprise Communication Platform. The inter-working report is valid until the AAPP member's product issues a new major release of such product (incorporating new features or functionality), or until ALE issues a new major release of such Alcatel-Lucent Enterprise product (incorporating new features or functionalities), whichever first occurs.

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Certification overview

Date of the certification	August 2018

ALE representative	Rachid Himmi
AAPP member representative	Eran Battat

Alcatel-Lucent Enterprise Communication Platform	OXO Connect
Alcatel-Lucent Enterprise Communication Platform release	OXO Connect (PowerCPU EE) R3.0/045.001 OXO Connect Evolution R3.0/045.001
AAPP member application release	6.60A.347.002
Application Category	Gateway

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Reviewer(s): Rachid Himmi, Thierry Chevert

Revision History

Edition 1: creation of the document – August 2018 (OXO Connect -Power CPU)
Edition 2: Additional tests with OXO connect evolution (IP BOX) – August 2018 (OXO Connect Evolution)

Test results

☑ Passed ☐ Refused ☐ Postponed

☐ Passed with restrictions

Refer to the section 6 for a summary of the test results.

IWR validity extension

Only the MP118 hardware has been tested. Behavior should be the same with all the MP11x family

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1 Introduction

This document is the result of the certification tests performed between the AAPP member's application and Alcatel-Lucent Enterprise's platform.

It certifies proper inter-working with the AAPP member's application.

Information contained in this document is believed to be accurate and reliable at the time of printing. However, due to ongoing product improvements and revisions, ALE cannot guarantee accuracy of printed material after the date of certification nor can it accept responsibility for errors or omissions. Updates to this document can be viewed on:

- the Technical Support page of the Enterprise Business Portal (https://businessportal.alcatel-lucent.com) in the Application Partner Interworking Reports corner (restricted to Business Partners)

- the Application Partner portal (https://www.al-enterprise.com/en/partners/aapp) with free access.

2 Validity of the InterWorking Report

This InterWorking report specifies the products and releases which have been certified.

This inter-working report is valid unless specified until the AAPP member issues a new major release of such product (incorporating new features or functionalities), or until ALE issues a new major release of such Alcatel-Lucent Enterprise product (incorporating new features or functionalities), whichever first occurs.

A new release is identified as following:

- a “Major Release” is any x. enumerated release. Example Product 1.0 is a major product release.
- a “Minor Release” is any x.y enumerated release. Example Product 1.1 is a minor product release

The validity of the InterWorking report can be extended to upper major releases, if for example the interface didn’t evolve, or to other products of the same family range. Please refer to the “IWR validity extension” chapter at the beginning of the report.

Note: *The InterWorking report becomes automatically obsolete when the mentioned product releases are end of life.*

3 Limits of the Technical support

For certified AAPP applications, Technical support will be provided within the scope of the features which have been certified in the InterWorking report. The scope is defined by the InterWorking report via the tests cases which have been performed, the conditions and the perimeter of the testing and identified limitations. All those details are documented in the IWR. The Business Partner must verify an InterWorking Report (see above "Validity of the InterWorking Report) is valid and that the deployment follows all recommendations and prerequisites described in the InterWorking Report.

The certification does not verify the functional achievement of the AAPP member's application as well as it does not cover load capacity checks, race conditions and generally speaking any real customer's site conditions.

Any possible issue will require first to be addressed and analyzed by the AAPP member before being escalated to ALE. Access to technical support by the Business Partner requires a valid ALE maintenance contract

For details on all cases (3rd party application certified or not, request outside the scope of this IWR, etc.), please refer to Appendix F "AAPP Escalation Process".

3.1 Case of additional Third party applications

In case at a customer site an additional third party application NOT provided by ALE is included in the solution between the certified Alcatel-Lucent Enterprise and AAPP member products such as a Session Border Controller or a firewall for example, ALE will consider that situation as to that where no IWR exists. ALE will handle this situation accordingly (for more details, please refer to Appendix F "AAPP Escalation Process").

4 Application information

Application commercial name: Telephone Adapter / VoIP Gateway for Analog equipment

Application version: 6.60A.347.002

Interface type: SIP/Ethernet

Brief application description:

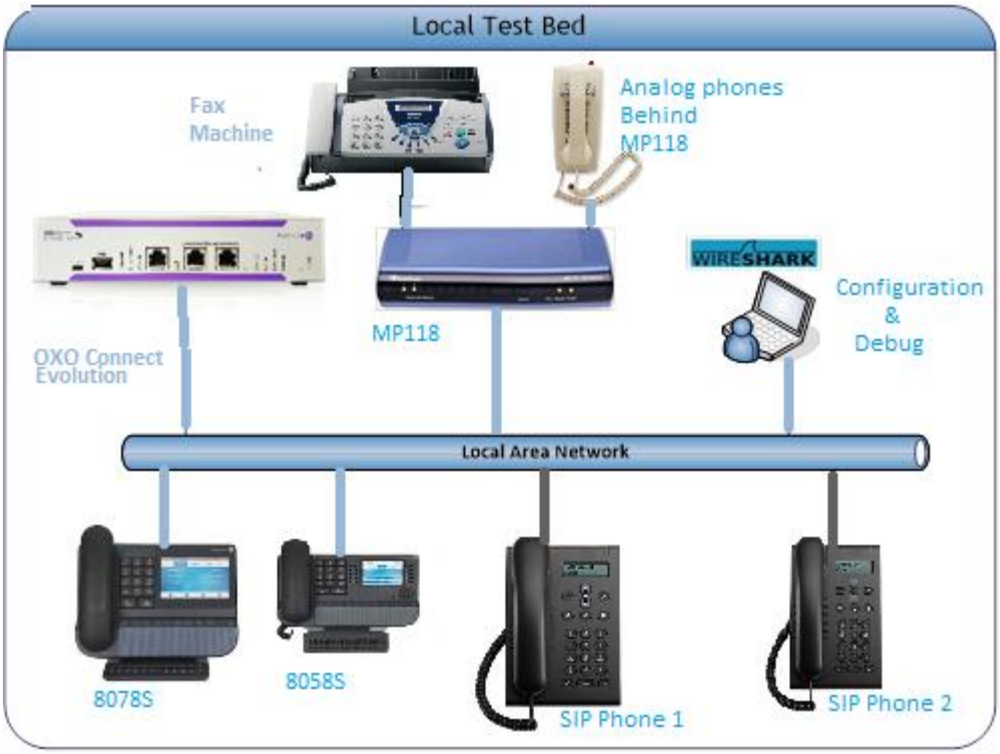
AudioCodes' Media Pack 1xx series of analog VoIP gateways offer service providers and enterprises superior voice technology for connecting legacy telephones, fax machines and PBX systems with IP telephony networks and IP-PBX systems.

The Media Pack 1xx gateways are fully interoperable with leading soft switches and SIP servers.

- Leverage investment in legacy analog telephone, modem, and fax systems – easing VoIP migration
- Secured zero-touch provisioning, useful for large-scale deployments
- Standalone Survivability (SAS) keeps your business running in the event of a network failure

5 Test environment

Figure 1 **Test environment**



5.1 Hardware configuration

List main hardware equipments used for testing

- **OXO Connect**:
  - ➢ OXO Connect Medium / PowerCPU EE (for OXO Connect)

  - ➢ **OXO IP box for OXO connect Evolution software.**

  - ➢ Release: R3.0/045.001
  - ➢ OMC : 30.0_17.1a

Setup Details:

OXO Connect Power CPU EE

Setup Information OXO 1	
OXO 1 IP address	10.9.224.220
Voicemail No	114 -121
Attendant No	9
OXO Extension Details used for test	
IP Touch extension numbers	IPset-1 : 130 IPset-2 : 129 IPset-3 : 131
SIP extension numbers These are analog extension numbers connected behind MP118 Fax extension number connected behind MP118	SIP set-1 :120 SIP Set-2:121 GWFAXset-1 :122

OXO Connect Evolution

Setup Information OXO 2	
OXO 1 IP address	10.9.223.140
Voicemail No	114 -121
Attendant No	9
OXO Extension Details used for test	
IP Touch extension numbers	IPset-1 : 118 IPset-2 : 119
SIP extension numbers These are analog extension numbers connected behind MP118 Fax extension number connected behind MP118	SIP set-1 :120 SIP Set-2:121 GWFAXset-1 :122

5.2 Software configuration

List main softwares used for testing

- **Alcatel-Lucent Communication Platform**:
  OXO Connect Power CPU EE softwareR3.0/045.001
  OXO Connect Evolution softwareR3.0/045.001
- **Partner Application**: Audio Codes Media pack 112 6.60A.347.002

Note: Analog phones are registered in the OXO Connect as "Open SIP phone".
Fax phones are registered in the OXO Connect as "Basic SIP phone".

6 Summary of test results

6.1 Summary of main functions supported

6.1.1 OXO Connect R3.0 (power CPU EE)

Feature	Results	Remarks
Initialization including network configuration	OK	
SIP registration	OK	
SIP authentication	OK	
Voice over IP and RTP codec support	OK	
Outgoing Call	OK	
Incoming Call	OK	
Features During Conversation	OK	

6.1.2 OXO Connect Evolution R3.0

Feature	Results	Remarks
Initialization including network configuration	OK	
SIP registration	OK	
SIP authentication	OK	
Voice over IP and RTP codec support	OK	
Outgoing Call	OK	We used SIP trunk calls for external calling related scenarios
Incoming Call	OK	
Features During Conversation	OK	

6.2 Summary of problems

- None

6.3 Summary of limitations

- None

6.4 Notes, remarks

- Analog phones are registered in the OXO as "Open SIP phone".
- Fax extensions are suggested to be configured as “Basic SIP phone”
- We have tested fax only between two gateway devices that is behind the Audio Codes. We did not test scenarios involving external fax to gateway fax device. We need to confirm the validity of the tests involving external fax
- Only the MP118 hardware has been tested. Behavior should be the same with all the MP11x family
- When using OXO prefix to set a forward, Do not disturb, wake-up, there is no indication on the remote office analog sets. Only tones are heard when programming There is also no indication when an OXO forward is set,
- Message waiting indication has to be enabled separately in the configuration.
- Network calls are tested between OXO Connect power CPU EE and OXO Connect R3.0 Evolution.
- Extensions connected behind the AudioCodes are referred as SIP phone Z-set
- OXO connect evolution supports only 80X8 series phones. We tested with 8028 and 8068 phones during our testing.

When mentioning OXO in this document means system used for the tests (OXO Connect and/or OXO Connect Evolution)

7 Test Result Template

The results are presented as indicated in the example below:

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Test case 1** • Action • Expected result	☐	☒	☐	
2	**Test case 2** • Action • Expected result	☐	☒	☐	The application waits for PBX timer or phone set hangs up
3	**Test case 3** • Action • Expected result	☒	☐	☐	Relevant only if the CTI interface is a direct CSTA link
4	**Test case 4** • Action • Expected result	☐	☐	☒	No indication, no error message
…	…	☐	☐	☐	

Test Case Id: a feature testing may comprise multiple steps depending on its complexity. Each step has to be completed successfully in order to conform to the test.
Test Case: describes the test case with the detail of the main steps to be executed the and the expected result
N/A: when checked, means the test case is not applicable in the scope of the application
OK: when checked, means the test case performs as expected
NOK: when checked, means the test case has failed. In that case, describe in the field "Comment" the reason for the failure and the reference number of the issue either on ALE side or on AAPP member side
Comment: to be filled in with any relevant comment. Mandatory in case a test has failed especially the reference number of the issue.

8 Test Results

8.1 Analog phones tests

8.1.1 Test Objectives

In this section analog phones are connected as Open SIP device on OXO though the analog gateway. These phones acts as OXO sets, so system features are available (prefix, suffix for example)

These tests shall verify that the different components are properly connected and can communicate together (the external application and the Alcatel-Lucent Communication Platform is connected and the interface link is operational).

8.1.2 Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
AN1	**SIP sets** Configure your SIP sets MCDU number on the OXO as Zset-1, Zset-2 & Zset-3 to register with the OXO IP address Check the registration on your sets and the display Note that authentication is disabled for these users, the password doesn't matter.	☐	☒	☐	
AN2	**SIP set registration to OXO in static IP addressing** For this test we will try to register the SIP phone with authentication enabled. SIP phones Zset-1, Zset-2 & Zset-3 are configured with a static IP address of OXO. Check the phone registration and display. Redo the same test on one IP phone with a wrong password and check that the phone is rejected.	☐	☒	☐	
AN3	**Support of "423 Interval Too Brief" (1)** The SIP phone Zset-2 is configured with a value lower than 120 seconds. Check the phone registration and display	☐	☒	☐	
AN4	**Signaling TCP-UDP** If applicable configure your SIP set Zset-2 to use the protocol SIP over UDP and other TCP In the two cases, check the registration and basic calls.	☐	☒	☐	

8.2 Audio codec negotiations/ VAD / Framing

8.2.1 Test Objectives

These tests check that the phones are using the configured audio parameters (codec, VAD, framing).

Phone configuration: configure the analog gateway to use G.711 A-law, G.711 mu-law, G.729, G.723 in this order (unless otherwise stated).
Configure IP Touch with codec G.711 and to NOT use VAD (unless otherwise stated).

8.2.2 Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
AU1	Select G711 A-law as 1st codec in the analog gateway Call from SIP Zset-2 to IP Touch IPset-2 Check that the call is established in G711 A-law. Check audio quality Call from IP Touch IPset-2 to SIP Zset-2 Check that the call is established in G711 A-law. Check audio quality	☐	☒	☐	
AU2	Select G729 as 1st codec in the analog gateway Call from SIP Zset-2 to IP Touch IPset-2 Check that the call is established in G729 Check audio quality Call from IP Touch IPset-2 to SIP Zset-2 Check that the call is established in G729 Check audio quality	☐	☒	☐	
AU3	Select G723 as 1st codec in the analog gateway Call from SIP Zset-2 to IP Touch IPset-2 Check that the call is established in G723 Check audio quality Call from IP Touch IPset-2 to SIP Zset-2 Check that the call is established in G723 Check audio quality	☒	☐	☐	G723 is not supported in OXO Connect evolution 3.0

AU4	**Configure** Zset-2 **to use VAD** **Configure IP Touch** IPset-2 **NOT to use VAD** Call from SIP Zset-2 to IP Touch IPset-2 Check that the call is established in G711 A-law. Check audio quality **Configure SIP** Zset-2 **to use VAD** **Configure IP Touch** IPset-2 **to use VAD** Redo the same tests **Configure SIP** Zset-2 **NOT to use VAD** **Configure IP Touch** IPset-2 **to use VAD** Redo the same tests	☐	☒	☐	
AU5	**In OXO enable codec pass through for SIP phones** **Call from SIP** Zset-1 **to SIP** Zset-2 Check that the call is established using G.722 Check audio quality	☐	☒	☐	
AU6	**In OXO 1 and OXO 2 enable codec pass through for SIP phone: direct RTP and codec pass through for SIP trunk. G723 is preferred codec in the analog gateway** **Call from SIP** Zset-1 **to Network SIP** NwkZset-1 Check that the call is established using direct RTP in G723. Check audio quality	☐	☒	☐	
AU7	**In OXO 1 and OXO 2 disable direct RTP and codec pass through for SIP trunk ARS table is configured with "default" codec. G723 is preferred codec in the analog gateway** **Call from SIP** Zset-1 **to Network SIP** NwkZset-1 Check that the call is established in G711. Check audio quality	☐	☒	☐	
AU8	**In OXO 1 and OXO 2 disable direct RTP and codec pass through for SIP trunk** **ARS table is configured with codec G729_30** **Call from SIP** Zset-1 **to Network SIP** NwkZset-1 Check that the call is established in G729. Check audio quality	☐	☒	☐	

8.3 Outgoing calls

8.3.1 Test Objectives

The calls are generated to several users belonging to the same network.
Called party can be in different states: free, busy, out of service, do not disturb, etc.
Calls to data devices are refused.
Points to be checked: tones, voice during the conversation, display (on caller and called party), hang-up phase.
OXO prefixes are mandatory for several tests of this section. For more information refer to the appendix C.

Note: dialing will be based on direct dialing number but also using programming numbers on the SIP phone (if available).

8.3.2 Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
OU1	**Call to a local user** With SIP Phone Zset-2 call the IP Touch IPset-1. Check that IPset-1 is ringing. Take the call and check ring back tone audio and display.	☐	☒	☐	
OU2	**Call to local user with no answer** With SIP Phone Zset-3 call the IP Touch IPset-1. And never take the call. Check time out (if any) and display. Note that IPset-1 don't have a Voice Mail	☐	☒	☐	
OU3	**Call to another SIP set** With the SIP phone Zset-2 call the other SIP Phone Zset-3 Check the display and audio during all steps (dialing, ring back tone, conversation, and release).	☐	☒	☐	
OU4	**Call to wrong number** (SIP: "404 Not Found") With the SIP phone Zset-2 call a wrong number Check the ring back tone and display	☐	☒	☐	
OU5	**Call to busy user** **(SIP: "182 Queued")** With the SIP phone Zset-2 call IP Touch IPset-1, take the call and don't hang up. With other SIP phone Zset-3 call IPset-1 which is busy Check the ring back tone and display	☐	☒	☐	
OU6	**Call to user in "Out of Service" state** **(SIP: "480 Temporarily Unavailable")** With the SIP phone Zset-3 call the IP Touch IPset-1 which is in "Out of Service State"	☐	☒	☐	Instead of out of service message, the call is automatically forwarded with 181 forwarded from OXO

Test Case Id	Test Case	N/A	OK	NOK	Comment
	Check the display and ring back tone				
OU7	**Call to user in "Do not Disturb" (DND) state (SIP: "480 Temporarily not available")** Dial "*63" on the IP Touch IPset-1 in order to enable the DND. Wait for acknowledgement ring back tone from OXO. With the SIP phone Zset-2 call IPset-1. Check ring back tone and display. Redial *60 on IPset-1 to cancel the DND	☐	☒	☐	
OU8	**Call to local user, immediate forward (CFU). (SIP: "181 Forwarded")(1)** On IP Touch IPset-1 dial the *61IPset-2 to activate the CFU. Wait for acknowledgement ring back tone from OXO. With the SIP phone Zset-2 call the IPset-1. Check that IPset-2 is ringing and the display. Take the call check audio and hung up. Dial *60 on IPset-1 for forward cancellation.	☐	☒	☐	
OU9	**Call to local user, forward on no reply (CFNR). (1)** On IP Touch IPset-1 configure with OMC the CFNR using dynamic routing to IPset-2. With Zset-2 call the IPset-1. Check that IPset-1 is ringing but don't take the call and wait the time out (about 30 sec). Time out is defined in IPset-1 dynamic routing of Timer 1. After time out check that IPset-2 is ringing and take the call. Check the audio and display.	☐	☒	☐	
OU10	**Call to local user, forward on busy (CFB). (1)** On IP Touch IPset-1 dial the *62IPset-2 (*62+<target MCDU number>) to activate the CFB. Wait for acknowledgement ring back tone from OXO. With SIP phone Zset-2 call IPset-1 and take the call to make it busy. With other SIP phone Zset-3 call IPset-1. Check that IPset-2 is ringing and take the call. Check the audio and display. Dial *60 on IPset-1 for forward cancellation.	☐	☒	☐	
OU11	**Call to external number (Check ring back tone, called party display)** **With SIP set Zset-2 dial 9 (9 prefix +external number)** **Take the call and check audio, display and call release.**	☐	☒	☐	**OXO Connect** We used ISDN T0 line **OXO Connect evolution** We used SIP trunk for testing this scenario
OU12	**SIP session timer expiration: Check if call is maintained or released after the session timer has expired** With SIP set Zset-2 call IP Touch IPset-1. Take the call on IPset-1 and never hang up, wait for time out expiration. Check that call is maintained or release.	☐	☒	☐	

Notes:

(1) For test cases with call to forwarded user: User is forwarded to another local user. Special case of forward to Voice Mail is tested in another section.

8.4 Incoming calls

8.4.1 Test Objectives

Calls will be generated using the numbers or the name of the SIP user.
SIP terminal will be called in different states: free, busy, out of service, forward.
The states are to be set by the appropriate system prefixes unless otherwise noted.
Points to be checked: tones, voice during the conversation, display (on caller and called party), hang-up phase.

Network calls are made using SIP private trunk established between two OXO's.
OXO prefixes are mandatory for several tests of this section. For more information refer to the appendix C.

8.4.2 Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
IN1	**Local /network call to free SIP terminal** Local: with IP Touch IPset-1 call SIP set Zset-2. Check that Zset-2 is ringing and take the call Check ring back tone and called party display. Network: with IP Touch IPset-1 call SIP set NwkZset-2 on another Node. Check that NwkZset-2 is ringing and take the call. Check ring back tone and called party display.	☐	☒	☐	
IN2	**Local/network call to busy SIP terminal** Local: With SIP set Zset-3 call other SIP set Zset-2 and take the call to make it busy, don't hang up. With IP Touch IPset-2 call Zset-2 which is busy Check the ring back tone and display. Network: With SIP set Zset-2 call SIP set NwkZset-2 and take the call to make it busy, don't hang up. With IPset-1 call NwkZset-2 which is busy Check ring back tone and called party display.	☐	☒	☐	
IN3	**Local/network call to unplugged SIP terminal** Local: Unplug the Zset-2 SIP set and call it with IP Touch IPset-1. Check the ring back tone and display Network: Unplug the SIP set NwkZset-2 and call it with IPset-1 Check the ring back tone and display	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
IN4A	**Local/network call to SIP terminal in Do Not Disturb (DND) mode** **By local feature if applicable:** Local: Enable DND on SIP set Zset-2 and call it with IP Touch IPset-1 Check the ring back tone and display Cancel the DND on Zset-2. Network: Enable DND on SIP set NwkZset-2 and call it with IP Touch IPset-1 Check the ring back tone and display Cancel the DND on Zset-2.	☐	☒	☐	603 Declined message is sent
IN4B	**By system feature** Local: Enable DND on SIP set Zset-2 using the *63 prefix.. Wait for acknowledgement ring back tone from OXO. With IP Touch IPset-1 call Zset-2 Check the ring back tone and display Cancel the DND on Zset-2 using *63 prefix. Network: Enable DND on SIP set NwkZset-2 using the *63 prefix. Wait for acknowledgement ring back tone from OXO. With IP Touch IPset-1 call NwkZset-2 Check the ring back tone and display Cancel the DND on NwkZset-2 using * 60 prefix.	☐	☒	☐	Call goes to voicemail
IN5A	**Local/network/SIP call to SIP terminal in immediate forward (CFU) to local user:** **By local feature if applicable:** Local: On SIP set Zset-2 enable CFU to IP Touch IPset-1 With SIP set Zset-3 call Zset-2. Check that IPset-1 is ringing. Take the call and check audio and display. Disable CFU on Zset-2. Network: On SIP set NwkZset-2 enable CFU to IP Touch NwkIPset-1. With SIP set Zset-2 call NwkZset-2. Check that NwkIPset-1 is ringing. Take the call and check audio and display. Disable CFU on NwkZset-2.	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
IN5B	**By system feature:** <u>Local</u>: On SIP set Zset-2 enable CFU to IP Touch IPset-1 using *61IPset-1 prefix (*61 + <target MCDU number>). Wait for acknowledgement ring back tone from OXO. With SIP set Zset-3 call Zset-2. Check that IPset-1 is ringing. Take the call and check audio and display. Disable CFU on Zset-2 using *60 prefix. <u>Network</u>: On SIP Set NwkZset-2 enable CFU to IP Touch NwkIPset-1 using *61 + <target MCDU number>. Wait for acknowledgement ring back tone from OXO. With SIP Set Zset-3 call NwkZset-2. Check that NwkIPset-1 is ringing. Take the call and check audio and display. Disable CFU on NwkZset-2 using *60 prefix.	☐	☒	☐	
IN6A	**Local/network/SIP call to SIP terminal in immediate forward (CFU) to network number:** By local feature if applicable: Local: On SIP Set Zset-3 enable CFU to SIP Set NwkZset-1.With SIP set Zset-2 call Zset-3. Check that NwkZset-1 is ringing. Take the call and check audio and display. Disable CFU on Zset-3. Network: On SIP Set Zset-2 enable CFU to IP Touch NwkIPset-1. With SIP Set NwkZset-2 call Zset-2. Check that NwkIPset-1 is ringing. Take the call and check audio and display. Disable CFU on Zset-2.	☐	☒	☐	
IN6B	By system feature: Local: On SIP Set Zset-2 enable CFU to SIP Set NwkZset-1 using *61NwkZset-1 prefix (*61 + <target MCDU number>). Wait for acknowledgement ring back tone from OXO. With SIP set Zset-3 call Zset-2. Check that NwkZset-1 is ringing. Take the call and check audio and display. Disable CFU on Zset-2 using *60 prefix. Network: On SIP Set Zset-2 enable CFU to IP Touch NwkIPset-1 using *61 + <target MCDU number>. Wait for acknowledgement ring back tone from OXO. With SIP Set NwkZset-2 call Zset-2. Check that NwkIPset-1 is ringing. Take the call and check audio and display. Disable CFU on Zset-2 using *60 prefix.	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
IN7A	**Local/network/SIP call to SIP terminal in immediate forward (CFU) to another SIP user** **By local feature if applicable:** Local: On SIP set Zset-2 enable CFU to SIP set NwkZset-1. With Zset-3 call Zset-2. Check that NwkZset-1 is ringing. Take the call and check audio and display. Disable CFU on Zset-2. Network: On SIP set Zset-2 enable CFU to IP Touch NwkIPset-3. With SIP Set NwkZset-1 call Zset-2. Check that NwkIPset-3 is ringing. Take the call and check audio and display. Disable CFU on Zset-2.	☐	☒	☐	
IN7B	**By system feature:** Local: On SIP Set Zset-3 enable CFU to SIP Set NwkZset-1 using *61 + <target MCDU number>. Wait for acknowledgement ring back tone from OXO. With SIP Set Zset-2 call Zset-3. Check that NwkZset-1 is ringing. Take the call and check audio and display. Disable CFU on Zset-3 using *60 prefix. Network: On SIP Set Zset-3 enable CFU to IP Touch NwkIPset-3 using *61 + <target MCDU number>. Wait for acknowledgement ring back tone from OXO. With SIP Set NwkZset-1 call Zset-3. Check that NwkIPset-3 is ringing. Take the call and check audio and display. Disable CFU on Zset-3 using *60 prefix	☐	☒	☐	
IN8A	**Local call to SIP terminal in “forward on busy” (CFB) state:** **By local feature if applicable** On SIP Set Zset-2 enable CFB to IP Touch IPset-1 With Zset-2 call the voice mail to make it busy. With SIP Set Zset-3 call Zset-2 which is busy. Check that IPset-1 is ringing Take the call and check audio and display. Disable CFU on Zset-2.	☐	☒	☐	
IN8B	By system feature: On SIP Set Zset-2 enable CFB to IP Touch IPset-1 using *62 + <target MCDU number>. Wait for acknowledgement ring back tone from OXO. With Zset-2 call the voice mail to make it busy. With SIP Set Zset-3 call Zset-2 which is busy. Check that IPset-1 is ringing Take the call and check audio and display. Disable CFB on Zset-2 using *60 prefix.	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
IN9A	**Local call to SIP terminal in "forward on no reply" (CFNR)** **By local feature if applicable** On SIP Set Zset-3 enable CFNR to IP Touch IPset-1 With SIP Set Zset-2 call Zset-3. Check that Zset-3 is ringing and don't take the call, wait for time out (about 30 seconds). After time out expiration the IPset-1 is ringing, take the call and check audio and display.	☐	☒	☐	
IN9B	**By system feature** CNFR via prefix not available on OXO (dynamic routing has to be used)	☐	☒	☐	
IN10	**Call to busy user, Call waiting.** (Camp-on), local feature if applicable: With SIP Set Zset-2 call other SIP Set Zset-3 (multiline set) to make it busy, take the call and don't hang up. With IP Touch IPset-2 call Zset-3 (on Zset-3 camp-on feature is enabled). Check the Call waiting or ring back tones and display	☐	☒	☐	Busy tone is heard
IN11	**External call to SIP terminal.** Check that external call back number is shown correctly: With SIP Set Zset-3 dial 9 + target MCDU number. Check that external is ringing and the external call number is shown correctly Take the call and check audio, display and call release.	☐	☒	☐	
IN12	**Calling Line Identity Restriction (CLIR): Local call to SIP terminal.** On IP Touch IPset-2 enable mask Identity and call SIP Set Zset-3 in order to hide IPset-2 identity. Check that Zset-3 is ringing, take the call and check that IPset-2 identity is hidden.	☐	☒	☐	
IN13	Display: Call to free SIP terminal from IP Touch user with a name containing non-ASCII characters (eg éëêèè). Check caller display. Check that SIP set is ringing and check on its display that the characters are correctly printed.	☐	☒	☐	
IN14	Display: Call from IP Touch to SIP which has the name containing non-ASCII characters, eg &@(#?+)=. Check caller display. Check that SIP set is ringing and check that the characters are correctly printed.	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
IN15	SIP set is part of a sequential hunt group **(1)**. Call to hunt group. Check call/release. With IP Touch IPset-1 call the sequential hunt group MCDU number 328 Check that Zset-2 is ringing Take the call and don't hang up. And with IP Touch IPset-2 call the sequential hunt group MCDU number 328 Check that IPset-2 is ringing Take the call and don't hang up. And with SIP Set Zset-1 call the sequential hunt group MCDU number 328 Check that Zset-3 is ringing Take the call and don't hang up.	☐	☒	☐	
IN16	SIP set is part of a cyclic hunt group **(2)**. Call to hunt group. Check call/release. With IP Touch IPset-1 call the cyclic hunt group MCDU number IPset-2 Check that 301 is ringing Take the call and hang up. And with IPset-1 call the cyclic hunt group MCDU number IPset-2 Check that Zset-3 is ringing Take the call and hang up. And with SIP Set Zset-1 call the cyclic hunt group MCDU number IPset-2 Check that Zset-2 is ringing Take the call and don't hang up.	☐	☒	☐	
IN17	SIP set is declared as a MultiSet. Call to main set and see if twin set rings. Take call with twin set. With IP Touch IPset-2 call IP Touch IPset-1 which is in MultiSet with SIP Set Zset-3. Check that Zset-3 and IPset-1 both ringing. Take the call from Zset-3 and check that IPset-1 stop ringing. Check audio and display.	☐	☒	☐	

8.5 Features during Conversation

8.5.1 Test Objectives

Features during conversation between local user and SIP user must be checked.
Check that right tones are generated on the SIP phone. A multiline SIP set is mandatory for tests 2, 3, 4 and 8.
OXO prefixes are mandatory for several tests of this section. For more information refer to the appendix C.

8.5.2 Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
FE1A	**Hold and resume with local feature** (if applicable) With Zset-3 call IPset-1 take the call, check audio and display. With Zset-3 put IPset-1 on hold check tones and display on both and resume the call. With IPset-1 put Zset-3 on hold check tones and display on both and resume the call. Keep this call for the next test.	☐	☒	☐	CPT file was generated using the Audiocodes CPTwizard and has to be loaded to the gateway to get the tones.
FE1B	**Enquiry call to another local user** (if applicable) Distant user is put on hold with local feature With Zset-3 (multi-lines) call IPset-2 and take the call. IPset-1 will be put on hold when making second call to IPset-2 Put IPset-2 on hold and check tones and display on both. Keep these two calls for the next test.	☐	☒	☐	
FE1C	**Broker request, toggle back and forth between both lines with local feature** (if applicable) With Zset-3 switch between IPset-1 and IPset-2 lines. Check the tones and display on sets on hold state. Keep these two calls for the next test.	☐	☒	☐	
FE1D	Release first call. Keep second call. Hang up IPset-1 and only Zset-3 and IPset-2 are in call Check that Zset-3 & IPset-2 are still in a call, check display.	☐	☒	☐	
FE2	Repeat the test 1C to 1D but using the call server feature	☐	☒	☐	
FE3	**Three party conferences initiated from OXO set** With IPset-1 call Zset-2, take the call and don't release it. With IPset-1 call IPset-2, take the call and don't release it too.	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
	With IPset-1 start a conference. Check that IPset-1, IPset-2 and Zset-2 are in conference. Check audio and display.				
FE4A	**Three party conferences initiated from SIP set with local feature** (if applicable) With Zset-2 call IPset-1 take the call and don't release it. With Zset-2 call IPset-2, take the call and don't release it too. With Zset-2 start a conference by the local feature Check that IPset-1, IPset-2 and Zset-2 are in conference. Check audio and display.	☐	☒	☐	
FE4B	Three party conferences initiated from SIP set with OXO feature	☐	☒	☐	
FE5	**Meet Me conference** With Zset-3 call the Meet me Conference bridge dialing prefix 68 and follow instruction to open the bride. With Zset-2 join the conference bridge by dialing prefix 69 and enter access code. With IPset-1 join the conference bridge by dialing prefix 69 and enter access code. Check that IPset-1, Zset-2 and Zset-3 are in conference.	☐	☒	☐	

8.6 Call Transfer

8.6.1 Test Objectives

During the consultation call step, the transfer service can be requested and must be tested.
Several transfer services exist: Unattended Transfer, Semi-Attended Transfer and Attended Transfer.
Audio, tones and display must be checked.

We use the following scenario, terminology and notation:

There are three actors in a given transfer event:

- A – *Transferee*: the party being transferred to the Transfer Target.
- B – *Transferor*: the party doing the transfer.
- C – *Transfer Target*: the new party being introduced into a call with the Transferee.

There are three sorts of transfers in the SIP world:

- ***Unattended Transfer*** or *Blind transfer* : The Transferor provides the Transfer Target's contact to the Transferee. The Transferee attempts to establish a session using that contact and reports the results of that attempt to the Transferor.

- ***Semi-Attended Transfer*** or *Transfer on ringing*:

1. A (Transferee) calls B (Transferor).
2. B (Transferor) calls C (Transfer Target). A is on hold during this phase. C is in ringing state (does not pick up the call).
3. B executes the transfer. B drops out of the communication. A is now in contact with C, in ringing state. When C picks up the call it is in conversation with A.

- ***Attended Transfer*** or *Consultative Transfer* or *Transfer in conversation*:
  1. A (Transferee) calls B (Transferor).
  2. B (Transferor) calls C (Transfer Target). A is on hold during this phase. C picks up the call and goes in conversation with B.
  3. B executes the transfer. B drops out of the communication. A is now in conversation with C.

Note: Unattended and Semi Attended transfer are not supported for SIP phones on OXO Connect.

In the below table, SIP means a partner SIP set, OXO means a proprietary OXO (Z/UA/IP) set, Ext. Call means an External Call, ISDN for example.

8.6.2 Test Results

Test case ID	Test case			N/A	OK	NOK	Comment
	A Transferee	B Transferor	C Transfer Target				
CT1	OXO	SIP	OXO	☐	☒	☐	

CT2	Ext Call	SIP	OXO	☐	☒	☐	**OXO Connect** We used ISDN T0 line **OXO Connect evolution** We used SIP trunk for testing this scenario
CT3	Ext Call	SIP	Ext Call	☐	☒	☐	**OXO Connect** We used ISDN T0 line **OXO Connect evolution** We used SIP trunk for testing this scenario
CT4	SIP	SIP	SIP	☐	☒	☐	
CT5	SIP	OXO	OXO	☐	☒	☐	
CT6	Ext Call	OXO	SIP	☐	☒	☐	**OXO Connect** We used ISDN T0 line **OXO Connect evolution** We used SIP trunk for testing this scenario
CT7	SIP	OXO	SIP	☐	☒	☐	

8.7 Attendant

8.7.1 Test Objectives

An attendant console is defined on the system. Call going to and coming from the attendant console are tested.

8.7.2 Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
AT1	**SIP set call to attendant** From SIP set Zset-2 dial “9” (attendant call prefix) Check audio and display	☐	☒	☐	
AT2	**2nd incoming call while in conversation with attendant** While SIP set Zset-2 is in conversation with the attendant, from IP Touch IPset-2 call Zset-2 Answer the call and check audio and display	☐	☒	☐	Beep sound is heard when there is second incoming call.
AT3	**SIP set call to attendant, attendant transfers to OXO set, semi-attended** From SIP set Zset-2 dial “9” (attendant call prefix) and answer. Attendant transfer semi-attended to IP Touch IPset-2 Answer the call and check audio and display	☐	☒	☐	
AT4	**SIP set call to attendant, attendant transfers to OXO set, attended** From SIP set Zset-2 dial “9” (attendant call prefix) and answer Attendant transfer attended to IP Touch IPset-2 Check audio and display	☐	☒	☐	
AT5	**OXO set calls to attendant, attendant transfers to SIP set, attended** From IP Touch IPset-2 dial “9” (attendant call prefix) and answer Attendant transfer attended to SIP set Zset-2 Check audio and display	☐	☒	☐	
AT6	**External ISDN Call to attendant, attendant transfers to SIP set, attended** ISDN incoming call to the attendant. From the attendant call SIP set Zset-2 and transfer attended Check audio and display	☐	☒	☐	
AT7	**SIP set call to attendant, attendant transfers to External** From SIP set Zset-2, dial “9” (attendant call prefix) and answer	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
	From the attendant, call an external ISDN destination and transfer semi-attended Answer and check audio and display.				

8.8 Voice Mail

8.8.1 Test Objectives

Voice Mail notification, consultation and password modification must be checked.
MWI (Message Waiting Indication) has to be checked.

Voice mail service is enabled on SIP sets Zset-2, Zset-3 and OXO IPset-1.

For these tests, DTMF sending (RFC 2833) has to be validated in order to use Voice Mail menu.

8.8.2 Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
VO1	Password modification With SIP set Zset-3 call the Voice Mail and follow the Voice guide in order to modify the default password. When modification is accepted hang-up. Recall the voice mail and try to log with a wrong password. Check the rejection. Recall the voice mail and try to log with the right password. Check the service access.	☐	☒	☐	Password modification is successful
VO2	Message display activation, MWI (1): With SIP set Zset-2 call the Voice Mail. Follow the instructions in order to send a voice message in SIP set Zset-3 boxes. Check that the MWI on Zset-3 is activated.	☐	☒	☐	
VO3	Message consultation With SIP set Zset-3 call the Voice Mail. Follow the instructions in order to listen your voice message leaved during the previous test. Check that your can listen it and delete. Check that MWI display is disabled on Zset-3 after message cancellation.	☐	☒	☐	
VO4	SIP call to a OXO user forwarded to Voice Mail Forward the IP Touch IPset-1 to Voice Mail by dialing *61 prefix + <Voice Mail number>. With SIP set Zset-3 call IPset-1 and check that you are immediately forwarded to Voice Mail. Check that you can leave a message On IPset-1 disable Voice Mail forwarding with *60 prefix.	☐	☒	☐	Call forward can be enabled from the codes configured in the MP11x device
VO5	OXO set call to a SIP user forwarded to Voice Mail Forward the SIP set Zset-3 to Voice Mail by dialing *61 prefix + <Voice Mail number>. With IP Touch IPset-1 call Zset-3 and check that you are immediately forwarded to Voice Mail. Check that you can leave a message On Zset-3 disable Voice Mail forwarding with *60 prefix.	☐	☒	☐	Call forward can be enabled from the codes configured in the MP11x device

8.9 Defence

8.9.1 Test Objectives

Checks how the SIP set will react in case of an OXO reboot, Ethernet link failure.

8.9.2 Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
DE1	**OXO Reboot** Establish an incoming ISDN call with SIP set-1. Reboot the OXO. When the OXO is up again, re-establish an incoming ISDN call with SIPset-2 and check the audio.	☐	☒	☐	**OXO Connect** We tested this scenario with OXO connect R3.0 Medium cabinet and mix board. **OXO Connect evolution** For OXO connect evolution we used SIP trunk for this scenario.
DE2	**Ethernet link failure** Establish an incoming ISDN call with SIP set-1. Disconnect the Ethernet link of SIP set-1. Check that the incoming call is presented to the attendant. Reconnect the Ethernet link of SIP set-1. Re-establish an incoming ISDN call with SIP set-1 and check the audio.	☐	☒	☐	**OXO Connect** This scenario is applicable only for OXO connect (Power CPU EE).
DE3	**Power failure** In OXO connect Evolution the Establish an incoming ISDN call with SIP set-1. Disconnect the Ethernet link of SIP set-1. Check that the incoming call is presented to the attendant. Reconnect the Ethernet link of SIP set-1. Re-establish an incoming ISDN call with SIP set-1 and check the audio.				**OXO Connect evolution** We used SIP trunk instead of ISDN for OXO connect evolution. OXO connect evolution is powered by PoE. So once the cable is disconnected, the evolution box is switched off. Once it is connected it boots up and calls are successful.

8.10 Fax tests

8.10.1 Test Objectives

In this section fax modules are connected as Basic SIP phone on OXO though the analog gateway. These fax modules are limited to G711 pass-through sending method.

These tests shall verify that the basic communication between FAX can be made on different conditions.

8.10.2 Test Results

Basic Fax Tests

Test Case Id	Test Case	N/A	OK	NOK	Comment
FA1	**Register with no authentication** On OXO and on the analog gateway, configure GWFAXset-1 and GWFAXset-2 registration with no authentication. Check registration.	☐	☒	☐	Registration fax sets behind MP118
FA2	**Register with authentication** On OXO and on the analog gateway, configure GWFAXset-1 and GWFAXset-2 registration with digest authentication mode. Check registration.	☐	☒	☐	Registration fax sets behind MP118
FA3	**Register time out** On the analog gateway, configure 120 seconds as registration period. Wait for a registration timeout and check that the gateway registers again	☐	☒	☐	Registration fax sets behind MP118

Loop-back communication from GATEWAY Fax to GATEWAY Fax through OXO.

Test Case Id	Test Case	N/A	OK	NOK	Comment
FA4	**Fax sending to an between two gateway fax devices** Send a fax from GWFAXset-1 to GWFAXset-2	☐	☒	☐	We sent single fax between two MP118 phones.
FA5	**Fax sending with no authentication** Send a fax from an FAXset-1 to GWFAXset-1	☒	☐	☐	The scenario is not valid for OXO connect Evolution
FA6	**Fax receiving with no authentication** Send a fax from GWFAXset-1 to FAXset-1	☒	☐	☐	The scenario is not valid for OXO connect Evolution
FA7	**Register with authentication** On OXO and on the analog gateway, configure GWFAXset-1 and GWFAXset-2 registration with digest authentication mode. Check registration.	☒	☐	☐	The scenario is not valid for OXO connect Evolution
FA8	**Fax receiving with authentication** Send a fax from an FAXset-1 to GWFAXset-1	☒	☐	☐	The scenario is not valid for OXO connect Evolution

FA9	**Receiving fax with authentication** Send a fax from GWFAXset-1 to FAXset-1	☒	☐	☐	The scenario is not valid for OXO connect Evolution

Basic communication between Gateway fax and External fax

Test Case Id	Test Case	N/A	OK	NOK	Comment
FA10	**Fax sending to an external fax** Send a fax from an External fax to GWFAXset-1	☒	☐	☐	The scenario is not valid for OXO connect Evolution
FA11	**Fax receiving to an external fax** -0 Send a fax from GWFAXset-1 to an External fax	☒	☐	☐	

Multiple pages exchanged between GATEWAY Fax and OXO.

Test Case Id	Test Case	N/A	OK	NOK	Comment
FA12	**Fax sending with 5 pages to an between two gateway fax devices** The test is to confirm proper working while sending multi page fax between two Send a fax (5 pages) from GWFAXset-1 to GWFAXset-2	☐	☒	☐	
FA13	**Fax receiving with 5 pages** Send a fax (5pages) from FAXset-1 to GWFAXset-1	☒	☐	☐	The scenario is not valid for OXO connect Evolution
FA14	**Fax sending with 5 pages** Send a fax (5 pages) from GWFAXset-1 to FAXset-1	☒	☐	☐	The scenario is not valid for OXO connect Evolution
FA15	**Fax receiving with 5 pages from an external fax** Send a fax (5pages) from an external fax device to GWFAXset-1	☒	☐	☐	The scenario is not valid for OXO connect Evolution
FA16	**Fax sending with 5 pages to an external fax** Send a fax (5pages) from GWFAXset-1 to an external fax device	☒	☐	☐	The scenario is not valid for OXO connect Evolution

8.11 Surveillance/Recovery

8.11.1 Test objectives

These tests shall verify that the basic communication between faxes can be made when the network or equipment are stressed.

8.11.2 Perturbations

Description: Check the solution behaviors when network is perturb

Test Case Id	Test Case	N/A	OK	NOK	Comment
SU1	**Fax receiving stop after the first page** Send a fax from GWFAXset-1 to GWFAXset-2. Stop the transmission after sending the first page. Check the fax receiving is correctly stopped.	☐	☒	☐	
SU2	**Fax sending stop after the first page** Send a fax from GWFAXset-1 to FAXset-1. Stop the transmission after sending the first page. Check the fax sending is correctly stopped.	☒	☐	☐	The scenario is not valid for OXO connect Evolution
SU3	**Fax receiving when busy** Send one fax from GWFAXset-1 to GWFAXset-2 Send one fax from GWFAXset-3 to GWFAXset-2 Check the GWFAXset-3 receives a busy tone.	☐	☒	☐	
SU4	**Fax sending when no answer** Send one fax from GWFAXset-1 to GWFAXset-2. Verify that the behavior is correct when there is no answer	☐	☒	☐	

OXO system phones call GATEWAY Fax

Test Case Id	Test Case	N/A	OK	NOK	Comment
SU5	**Fax receiving stop after the first page** Make a call from the Ipset-1 to the GWFAXset-1, verify that the call is released after a time out Verify that no issues are generated	☐	☒	☐	

9 Appendix A: AAPP member's Application description

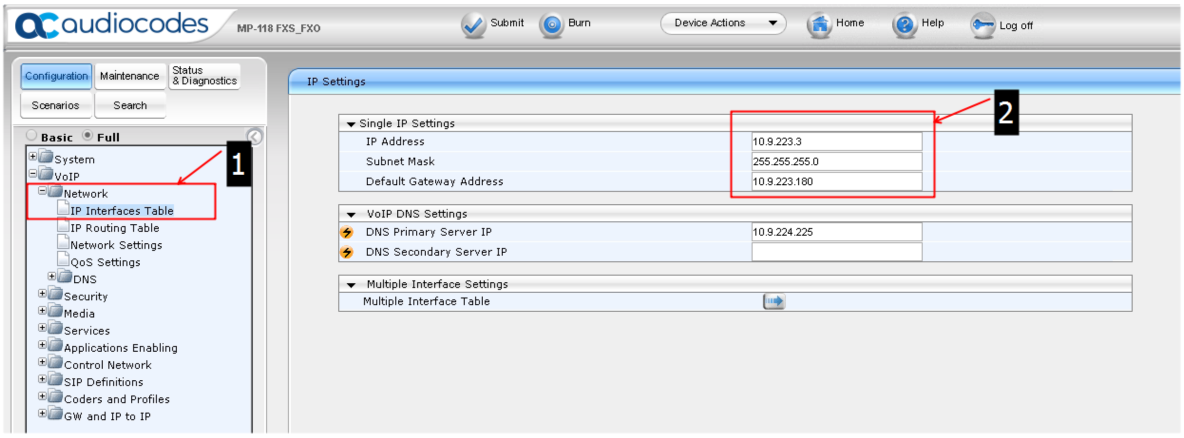
MediaPack 1xx

The Media Pack Series Analog VoIP Gateways are cost-effective, best-of-breed technology products. These stand-alone analog VoIP Gateways provide superior voice technology for connecting legacy telephones, fax machines and PBX systems with IP-based telephony networks, as well as for integration with new IP-based PBX systems. These products are designed and tested to be fully interoperable with leading Soft switches, SIP servers and H.323

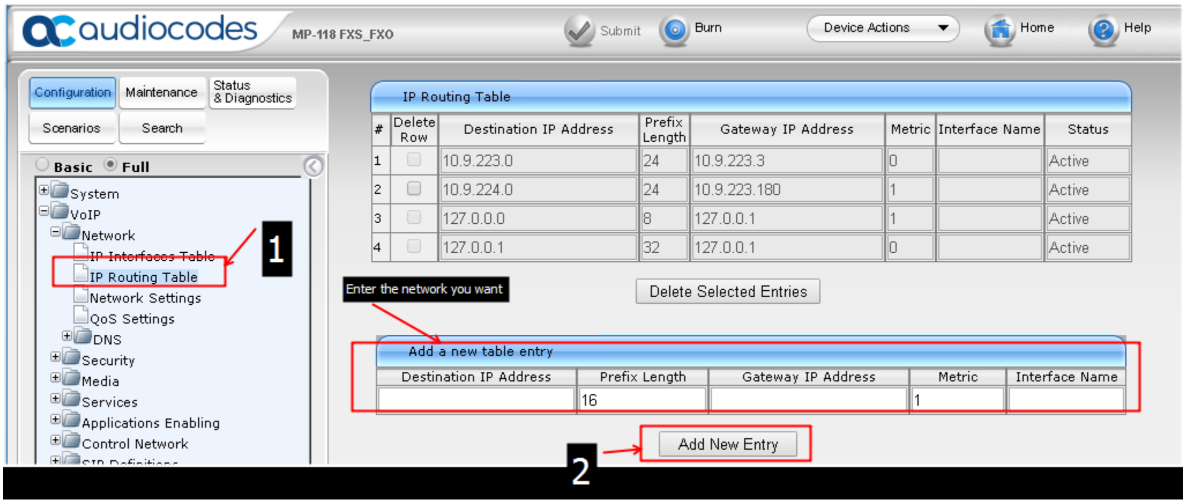
10 Appendix B: Configuration requirements of the AAPP member's application

To start with the configuration, we intially started with factory reset of the device.

1) Configure the IP interface table of the audiocodes.

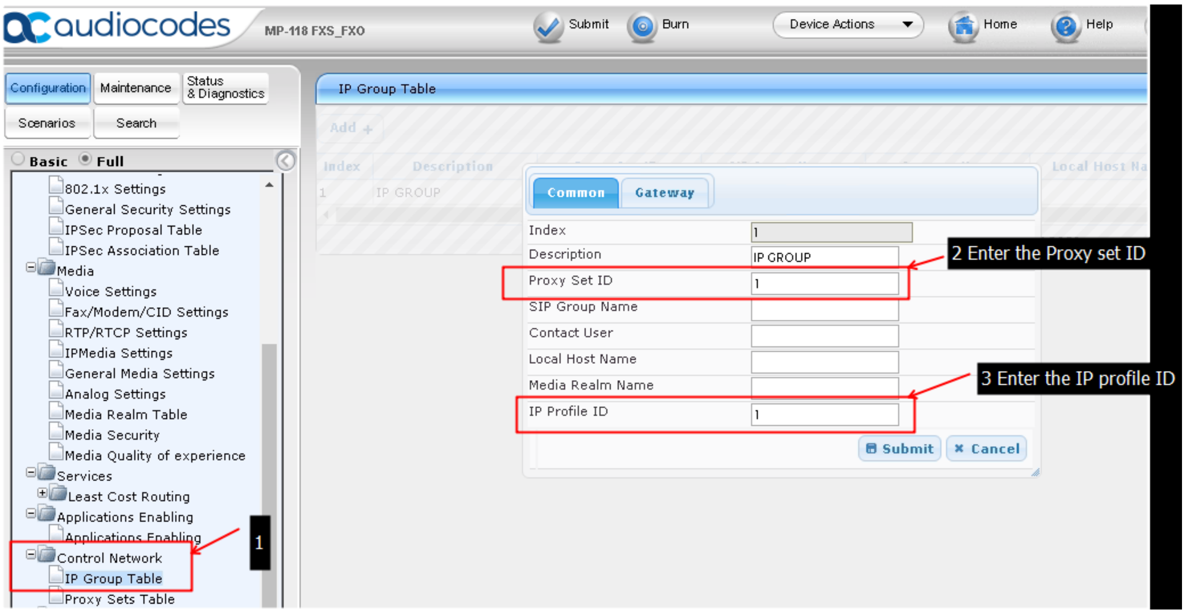


2) Check the IP routing table and add appropriate routes for your testing network



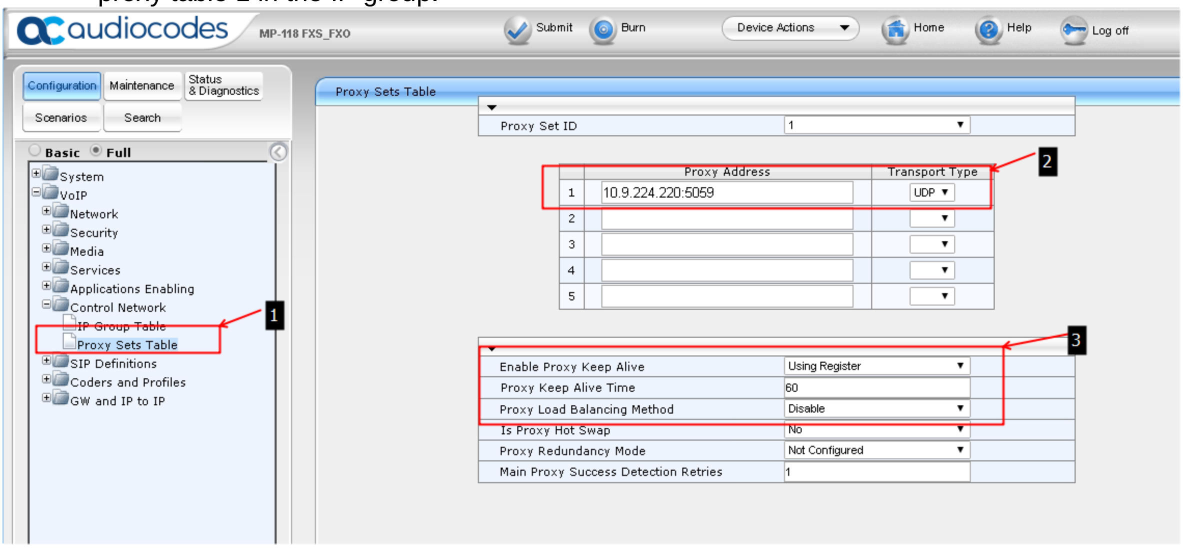
3) The registration can work even with default proxy settings.
But for ease of understading We used IP group and a separate Proxy Set.
We have given both configuration with default proxy and with separate IP group , proxy set.

Configuration with IP group and separate proxy ID

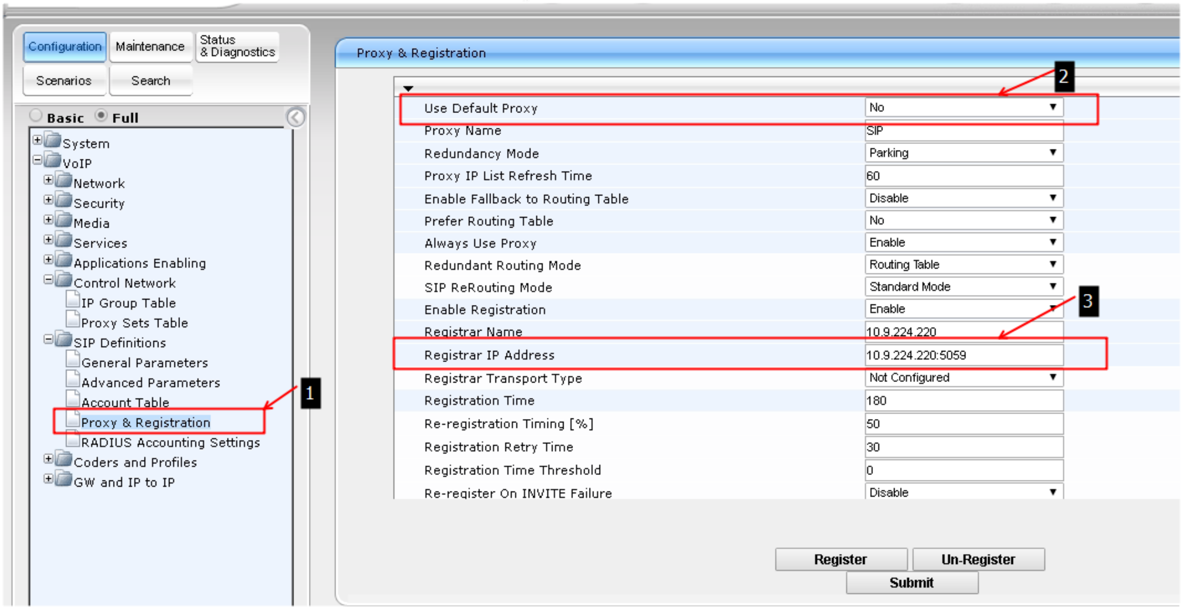


Click the gateway tab and select the SIP re-routing method. Select yes in the “always Use Route table”

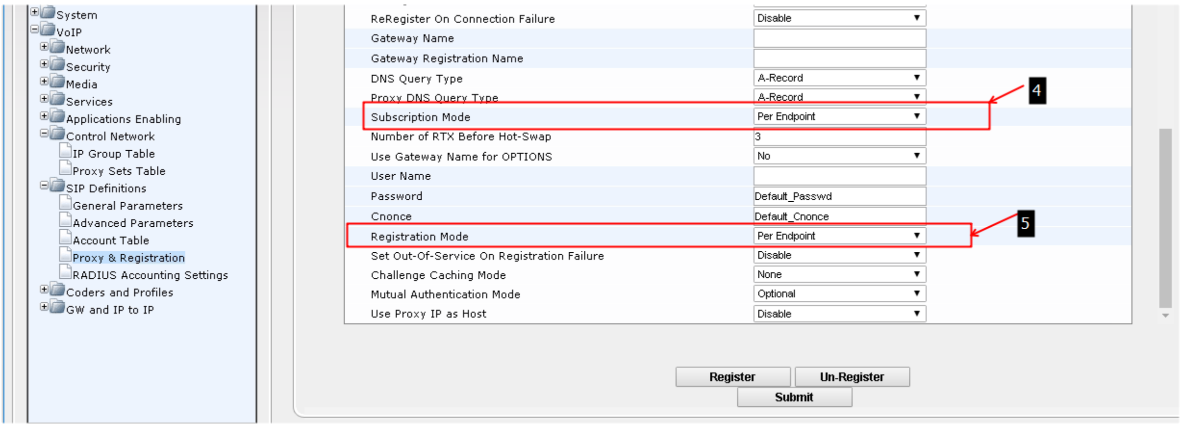
Add your OXO IP address and port number in the proxy table 1. Select the same options as shown in the picture below. We are not using default proxy table 0 here as we specified proxy table 1 in the IP group.



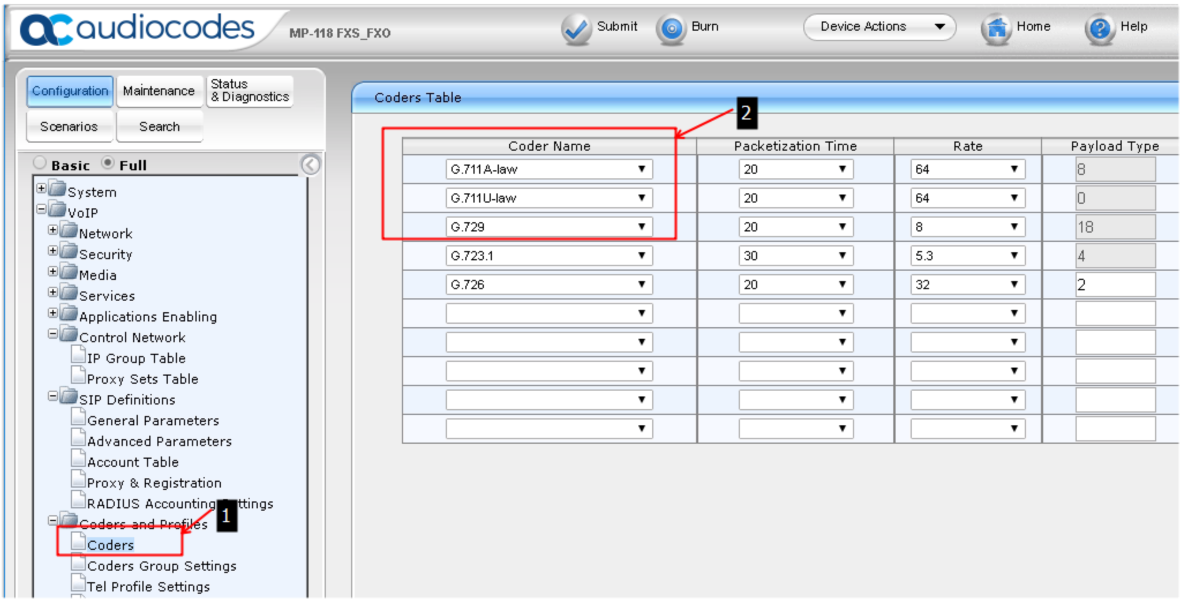
Goto SIP definitions and select the Proxy & registration menu. Enter the Registrar IP address. We have added to screenshots of this menu as it is many options.



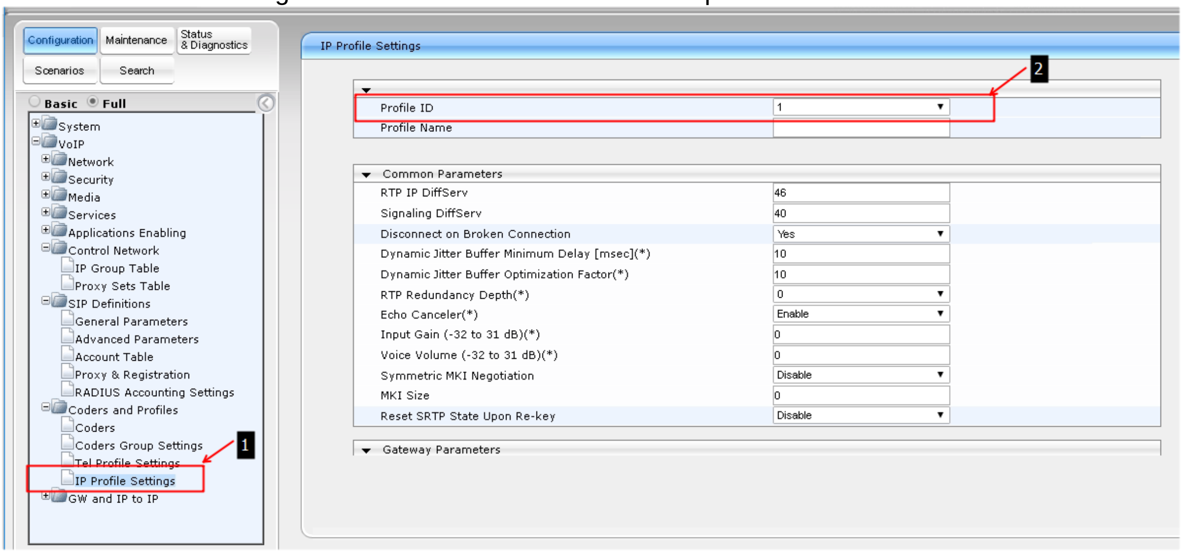
4) Remaining SIP parameters are

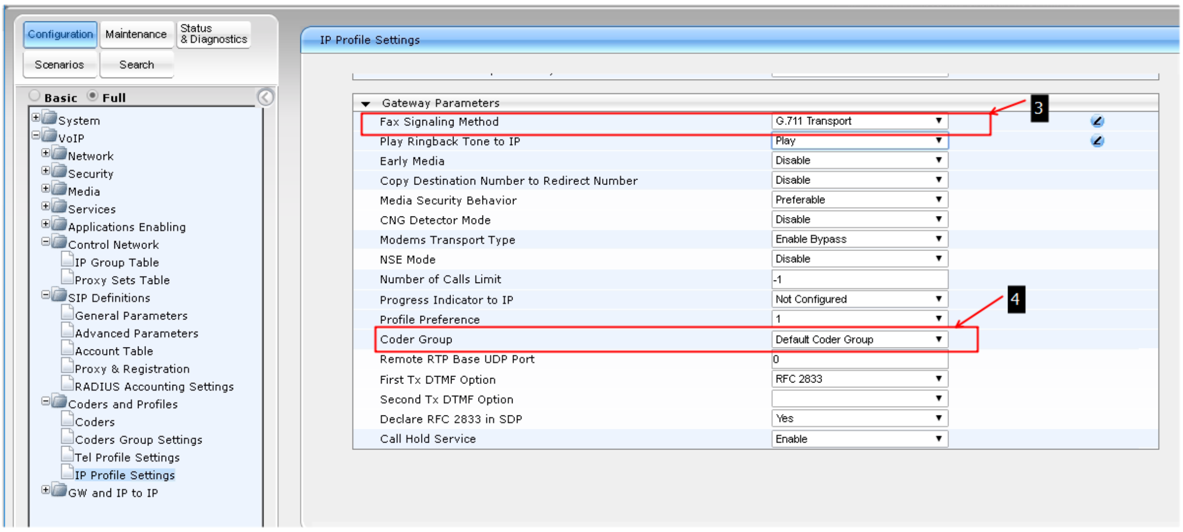


5) Select the supported codecs in OXO.

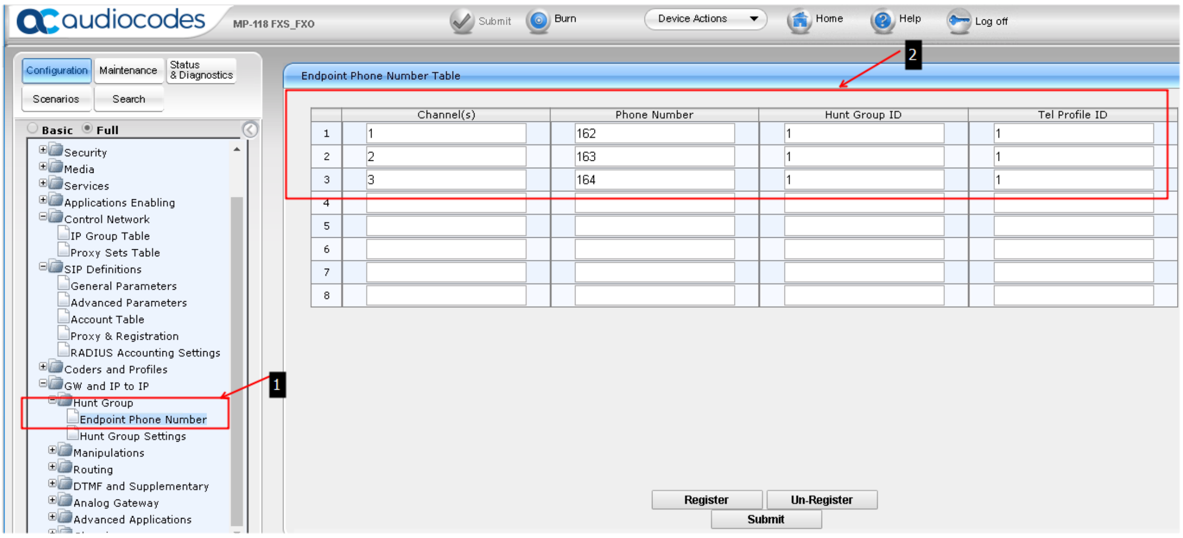


6) Select the IP profile declared in the IP group and check the approriate values are selected. For basic calling functionalities we use the default options.



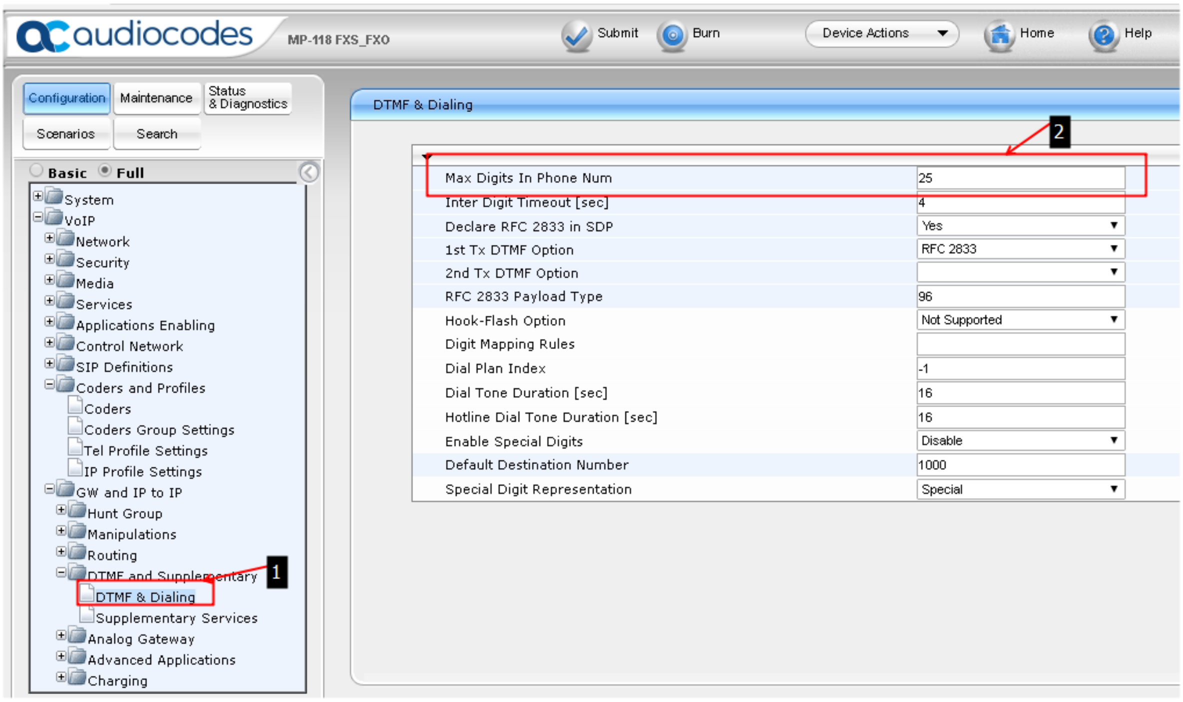


7) Under hunt group settings endter the end point numbers. The end point numbers are declared in OXO as Open SIP numbers. Declare the channels and enter the telephone profile IP and hunt group ID.

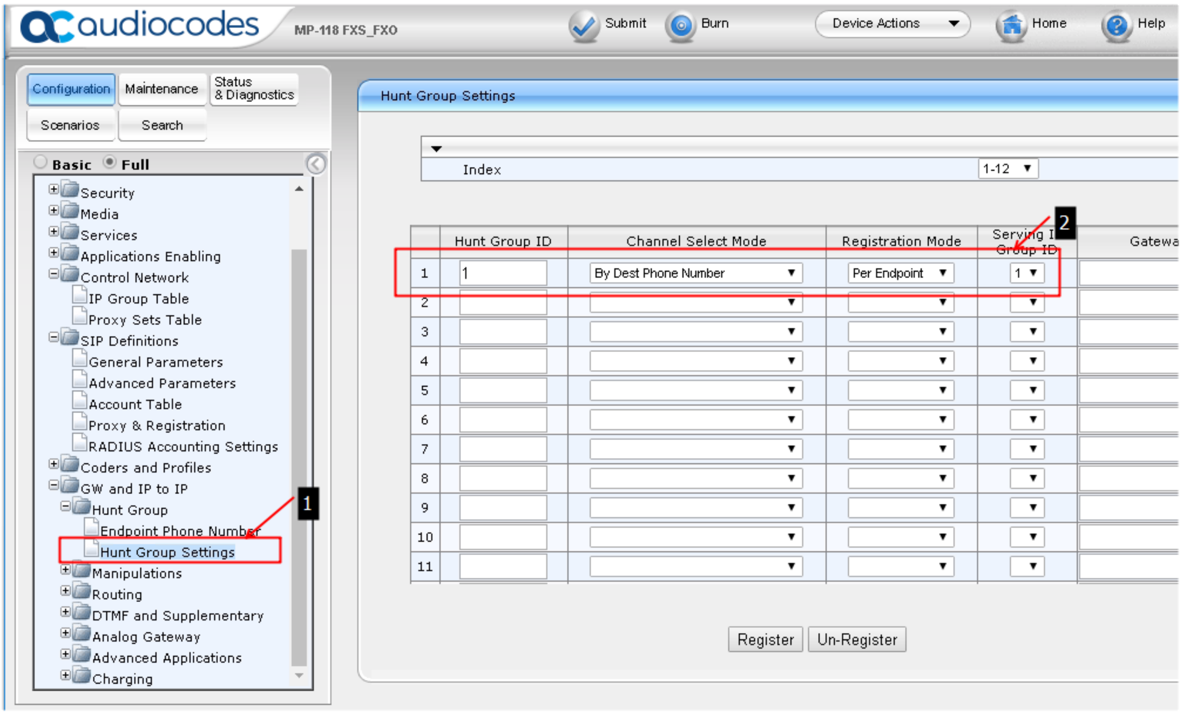


DTMF and Dialling

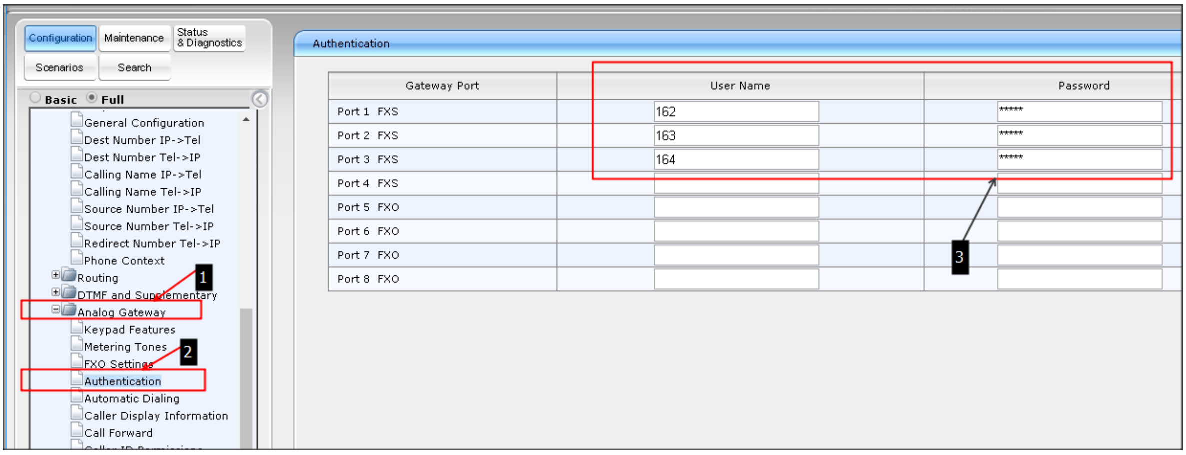
Please enter maximum digits to be sent via the gateway. The default value is very less and make sure to change this maximum number of digits to a value that matches the Prefix + length of number to be dialled.



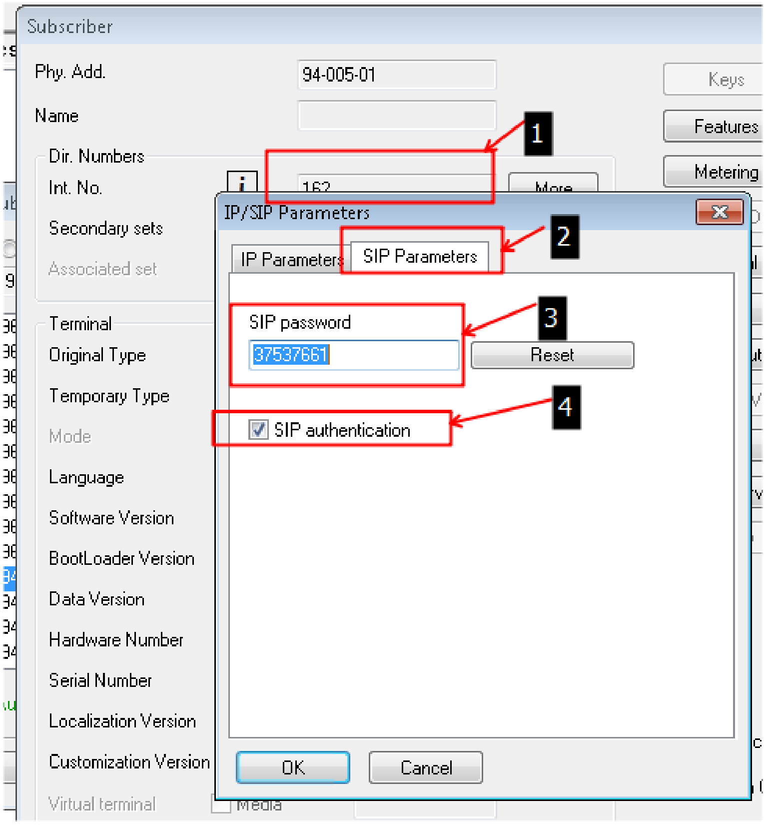
8) Hunt group settings select the registration mode.



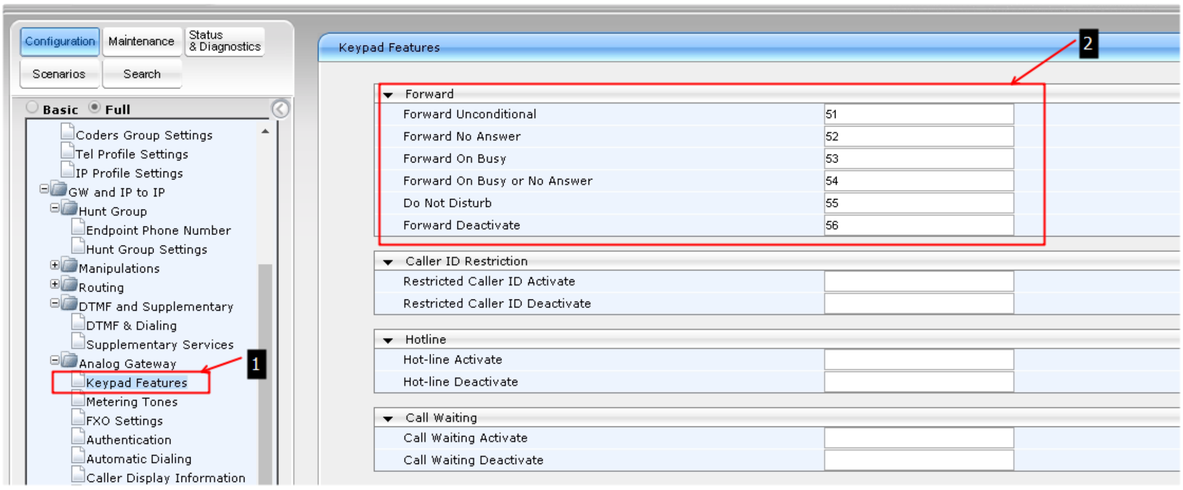
9) Under the analog gateway please enter password for OXO declared open SIP extensions.



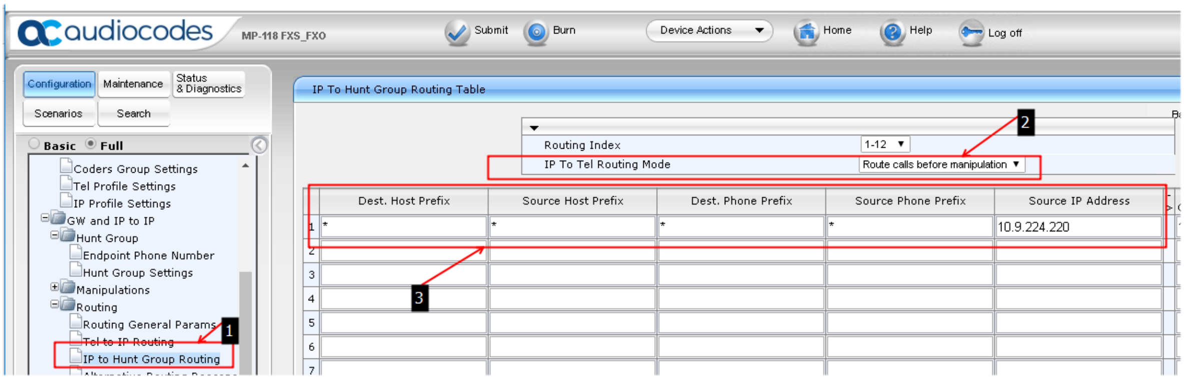
Password can be found in the IP/SIP options authentication shows below.

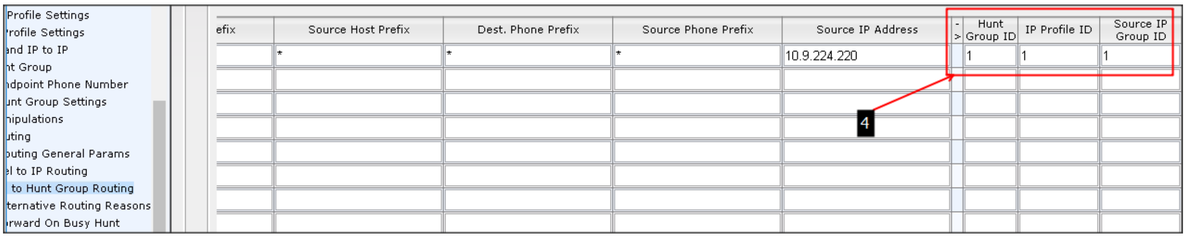


10) Keypad features need to be configured to enable forward for the analog phones behind MP11x

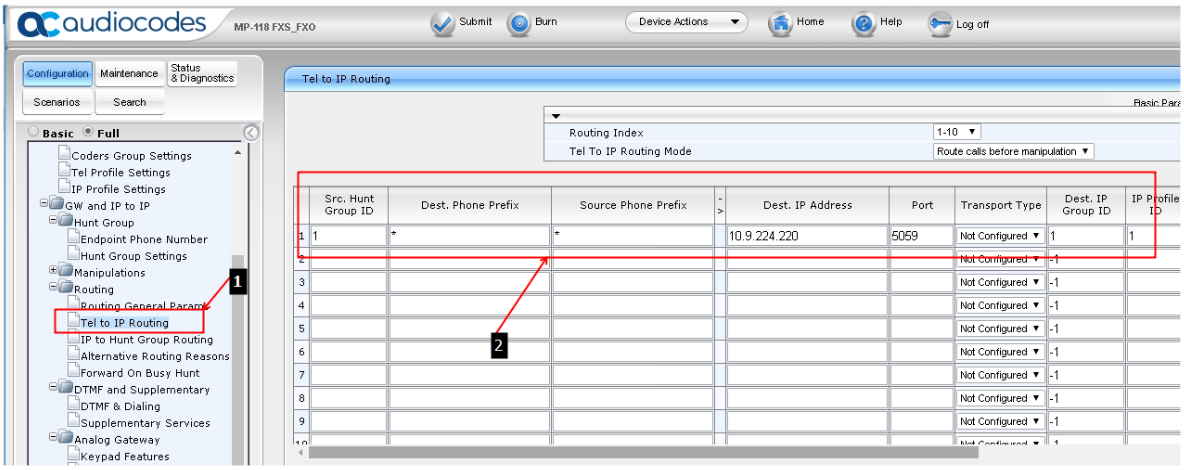


11) For calling to work we need to manage the IP to tel routing in routing menu. Give the source IP and huntgoup ID proeperly.

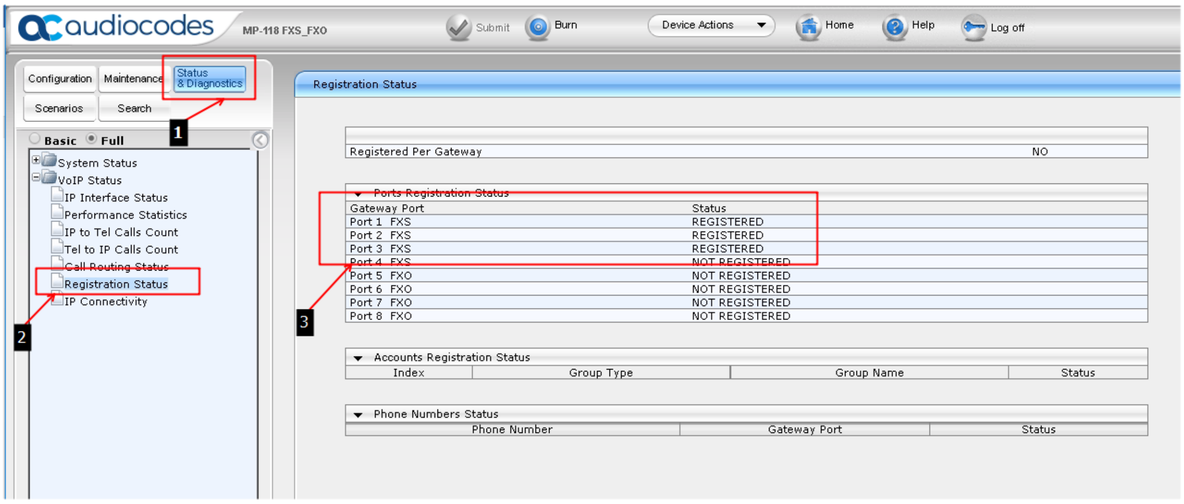




12) Same way we need to manage routing from the analog extensions behind MP11x to OXO .

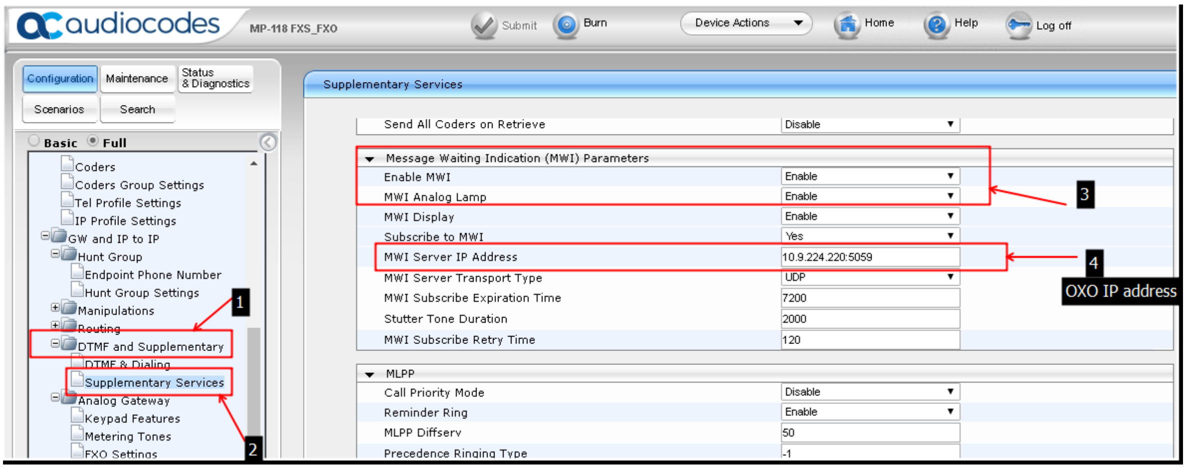


Once the configuration is done the enpoint numbers should be registered properly.

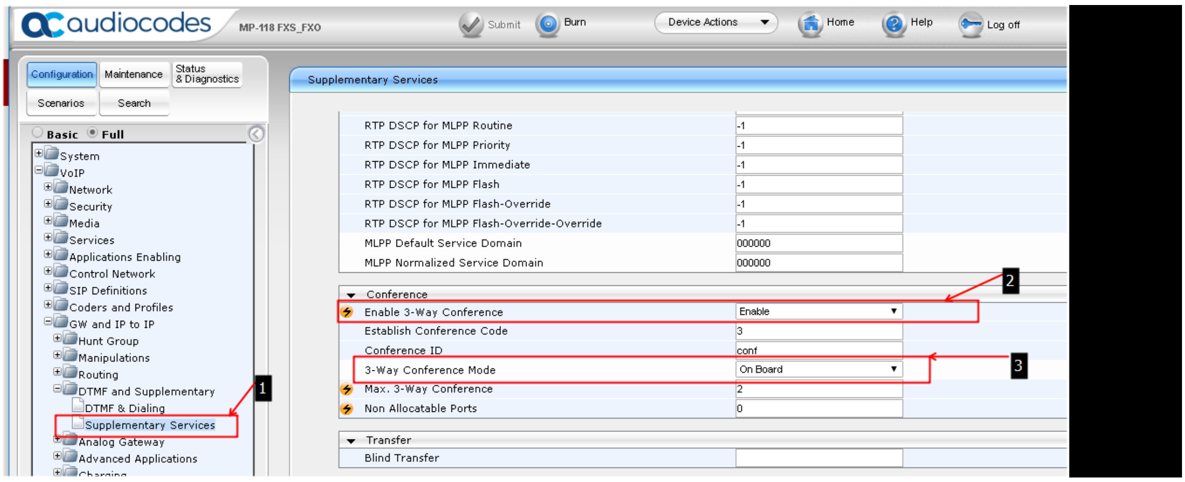


13) Message waiting Indication coniguration.

To enable MWI support (Message Waiting Indicator) go to the below mentioned link.
Configuration->VoIP->GW and IP to IP->DTMF and Supplementary ->Supplementary Services



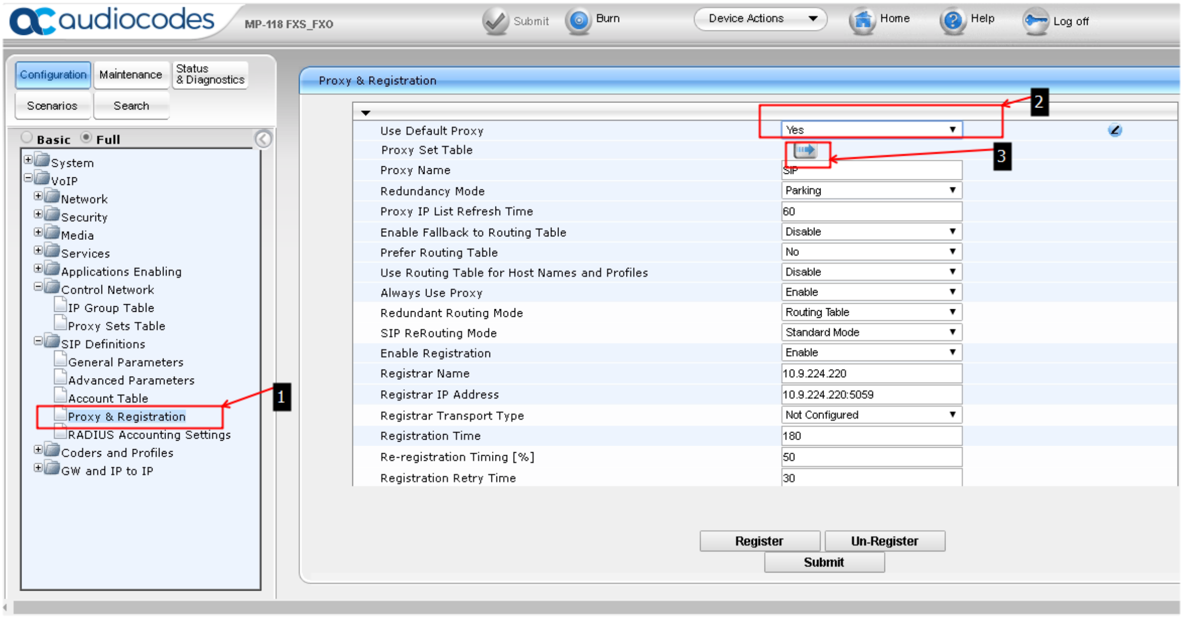
14) Conference Feature.



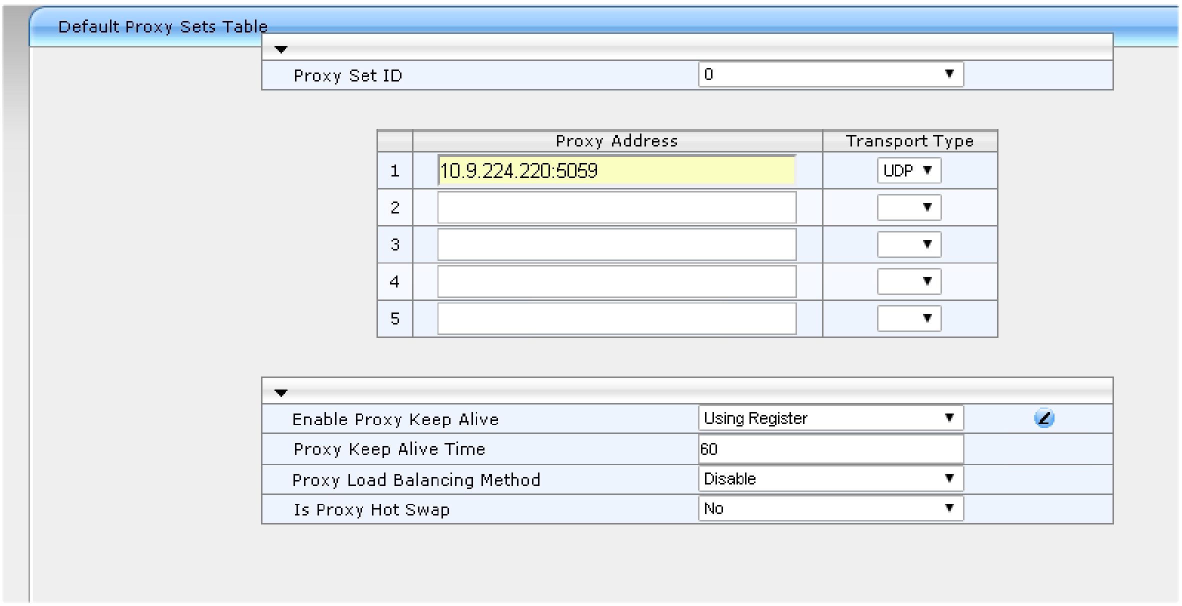
Configuration with default proxy

If you are using the above configuration then please ignore this section. The endpoint management and extension number declaration is same desribed above.

The other general features can be declared



Default Proxy settings

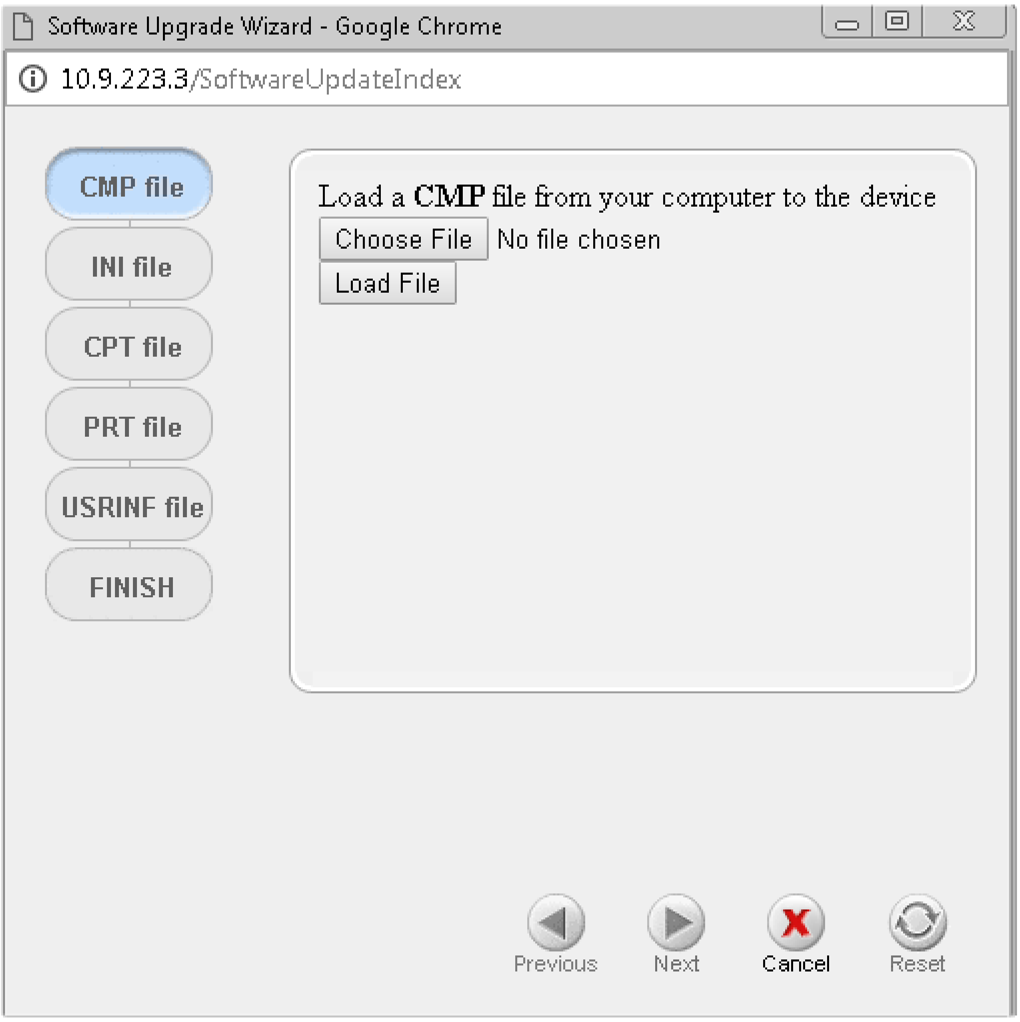


Procedure to load the tones.

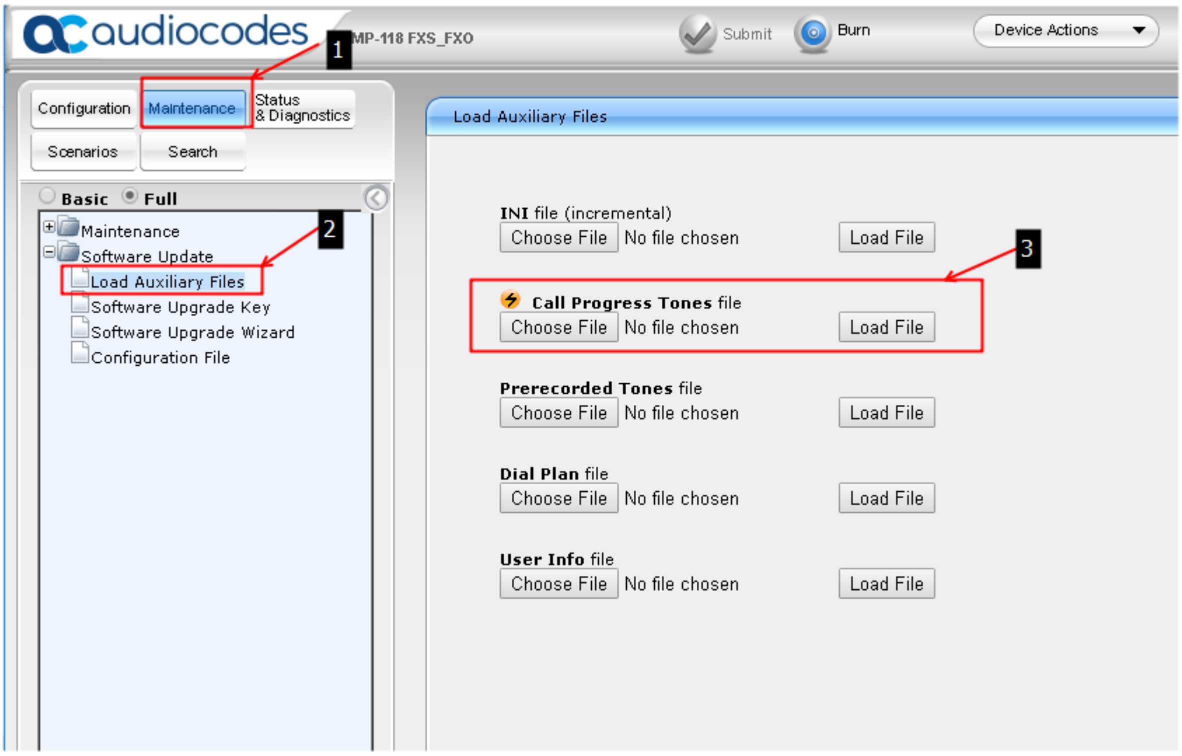
The tones can be loaded in two ways.

1) With firmware and other tones using the Software configuration wizard.

Launch the software upgrade the wizard.



2) Or using the load auxillary files option in the maintainance tab as show below.



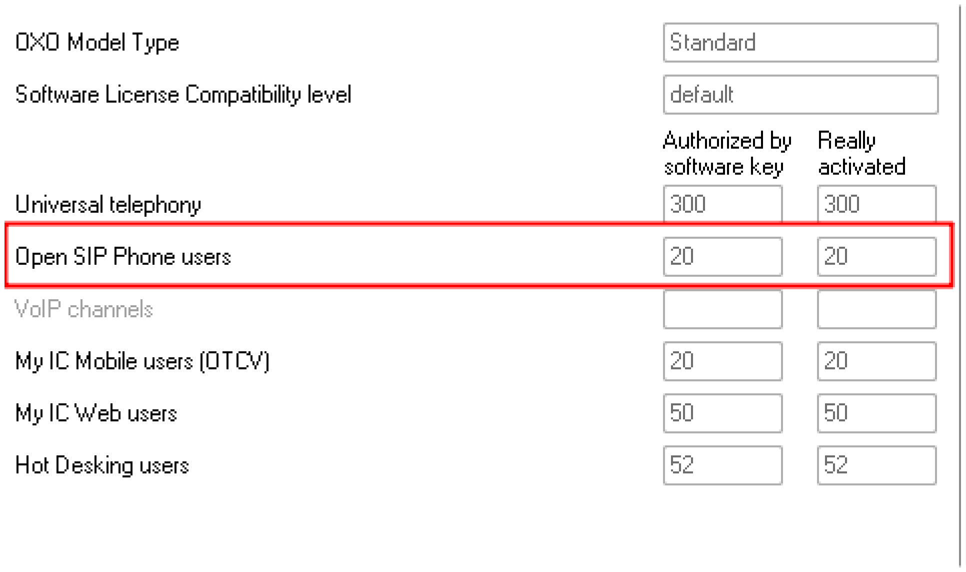
11 Appendix C: Alcatel-Lucent Enterprise Communication Platform: configuration requirements

The tests were performed with both OXO connect (Power CPU EE) Software and OXO connect Evolution software.

Both softwares were of same version R3.0/045.001

License requirment.

OPEN SIP licenses are needed only if all call features are needed.



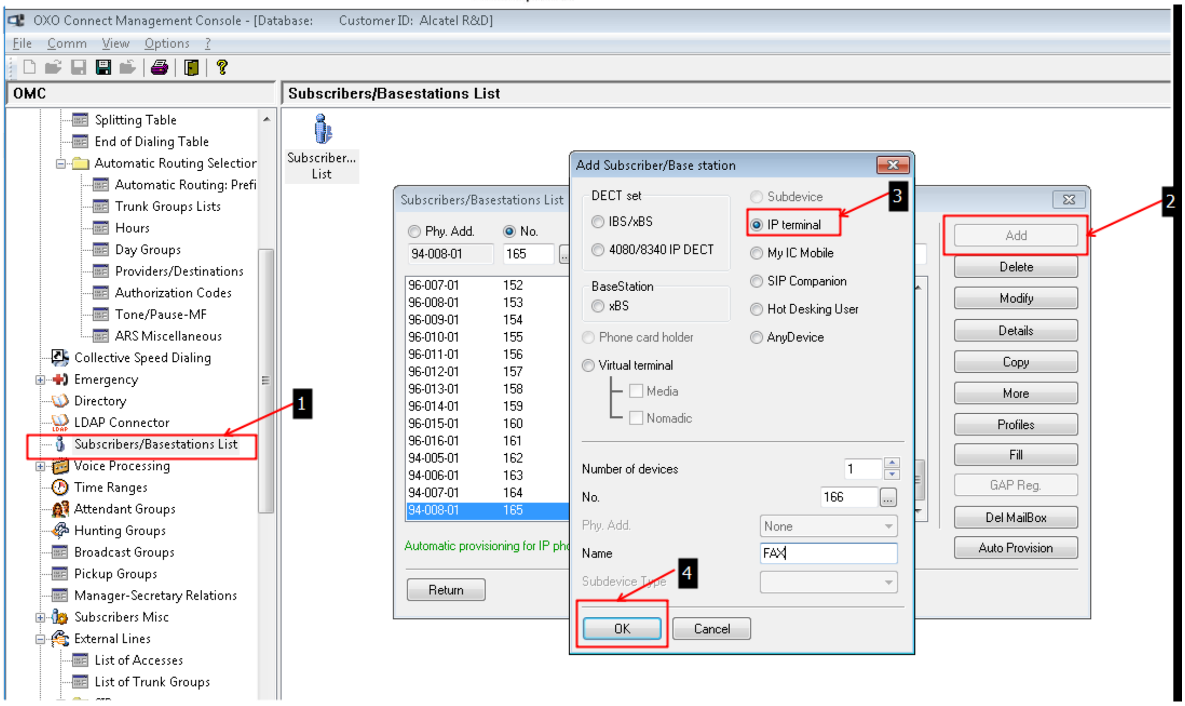
The configuration is same for both the versions of OXO software (OXO connect and OXO connection evolution)

SIP Set Configuration

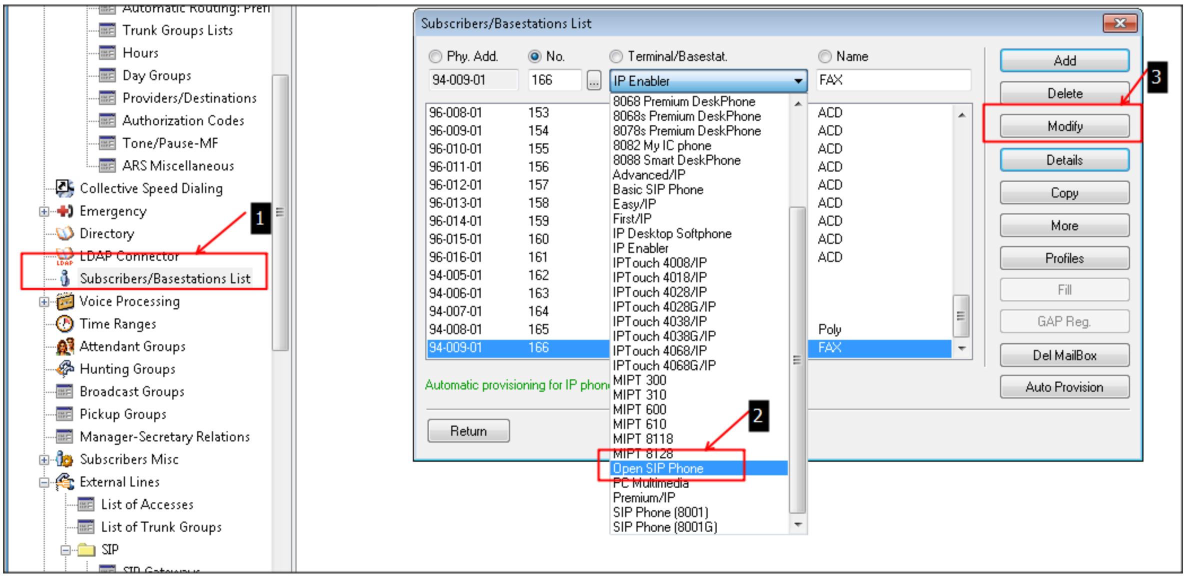
<u>Normal calls & FAX calls</u>

For normal calls we need to configure the sip user as Open SIP phone

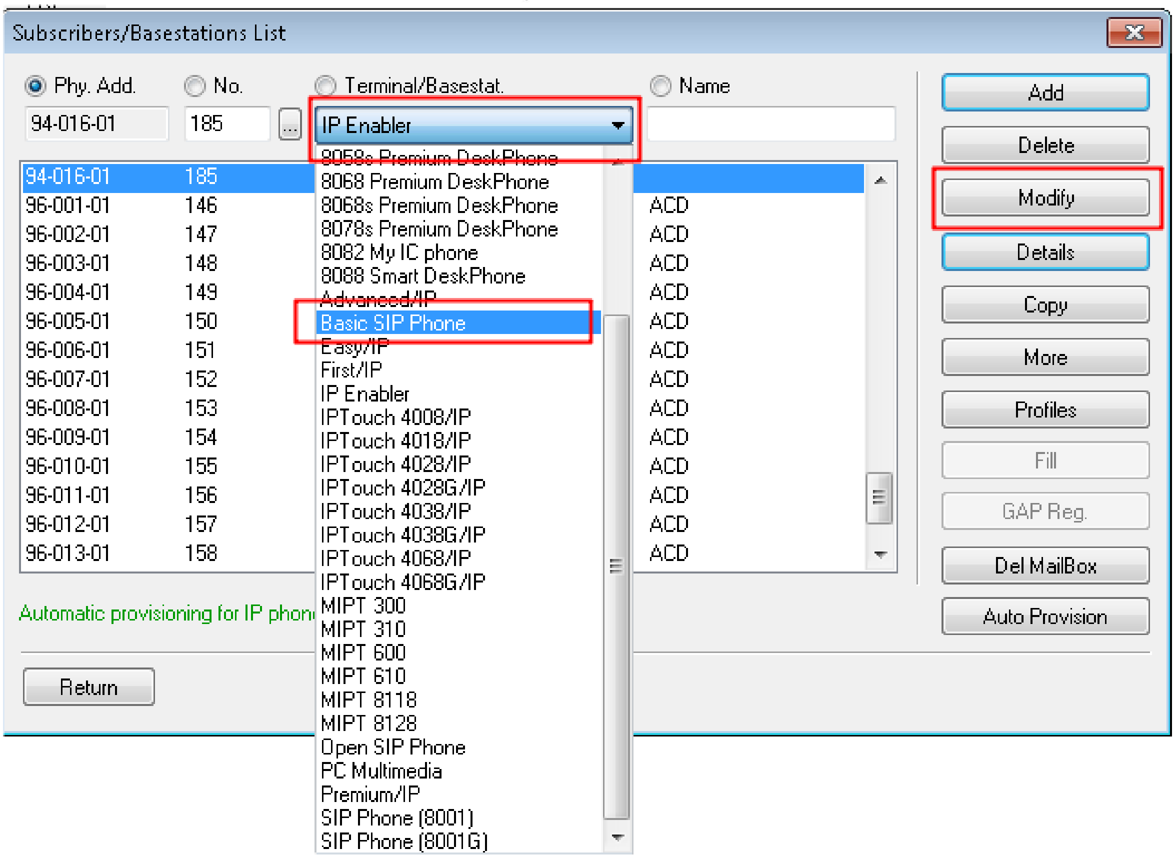
1)Open the User/Base stations List in the OMC. And click on Add

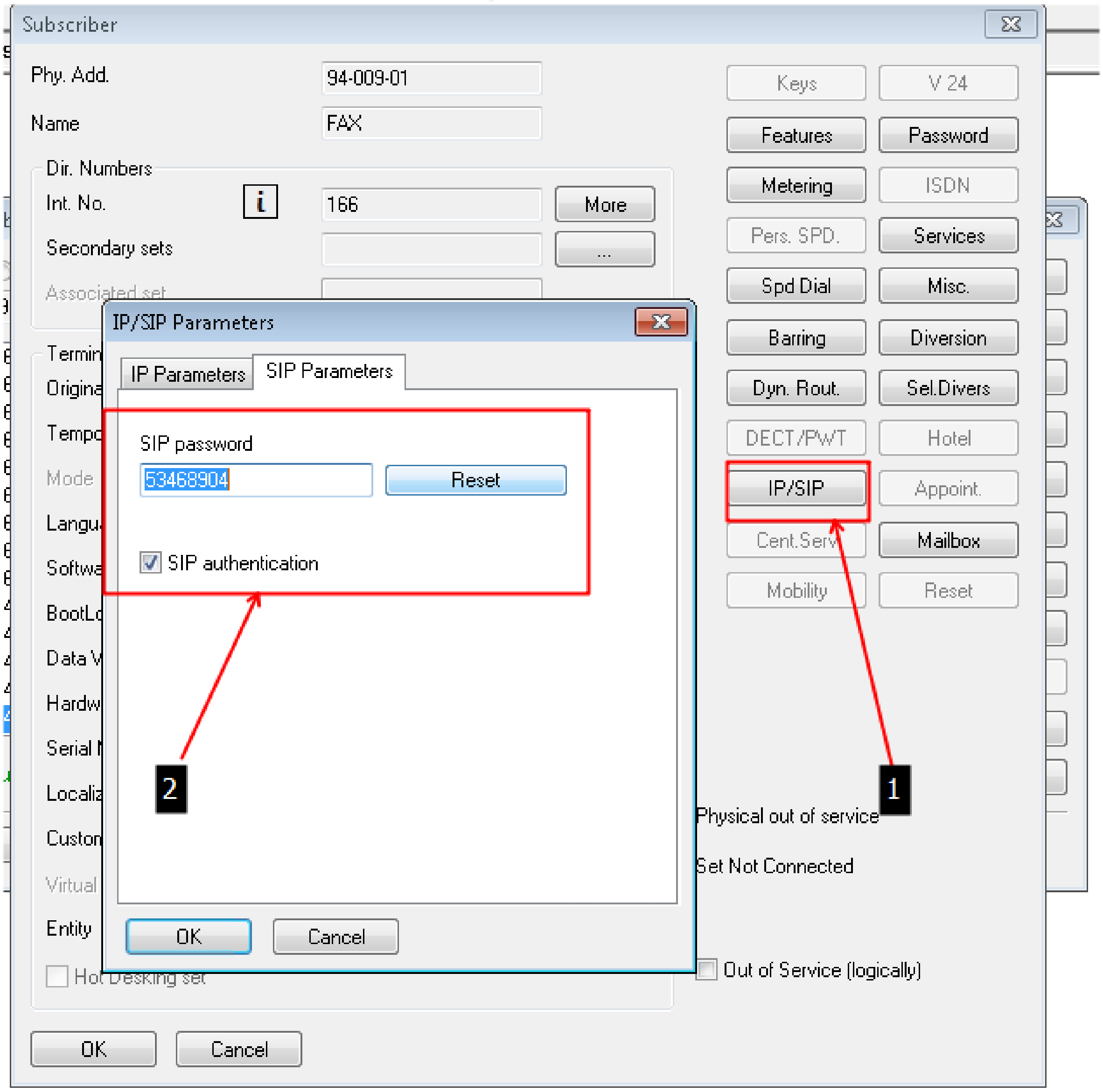


For Normal calling functionalities we used Open SIP extension type



For fax purpose we need to use BASIC SIP type.


Subscribers/Basestations List
Phy. Add.
No.
Terminal/Basestat.
Name
94-016-01
185
IP Enabler
Add
Delete
Modify
Details
Copy
More
Profiles
Fill
GAP Reg.
Del MailBox
Auto Provision
94-016-01 185
96-001-01 146 ACD
96-002-01 147 ACD
96-003-01 148 ACD
96-004-01 149 ACD
96-005-01 150 ACD
96-006-01 151 ACD
96-007-01 152 ACD
96-008-01 153 ACD
96-009-01 154 ACD
96-010-01 155 ACD
96-011-01 156 ACD
96-012-01 157 ACD
96-013-01 158 ACD
8058s Premium DeskPhone
8068 Premium DeskPhone
8068s Premium DeskPhone
8078s Premium DeskPhone
8082 My IC phone
8088 Smart DeskPhone
Advanced/IP
Basic SIP Phone
Easy/IP
First/IP
IP Enabler
IPTouch 4008/IP
IPTouch 4018/IP
IPTouch 4028/IP
IPTouch 4028G/IP
IPTouch 4038/IP
IPTouch 4038G/IP
IPTouch 4068/IP
IPTouch 4068G/IP
MIPT 300
MIPT 310
MIPT 600
MIPT 610
MIPT 8118
MIPT 8128
Open SIP Phone
PC Multimedia
Premium/IP
SIP Phone (8001)
SIP Phone (8001G)
Automatic provisioning for IP phon
Return



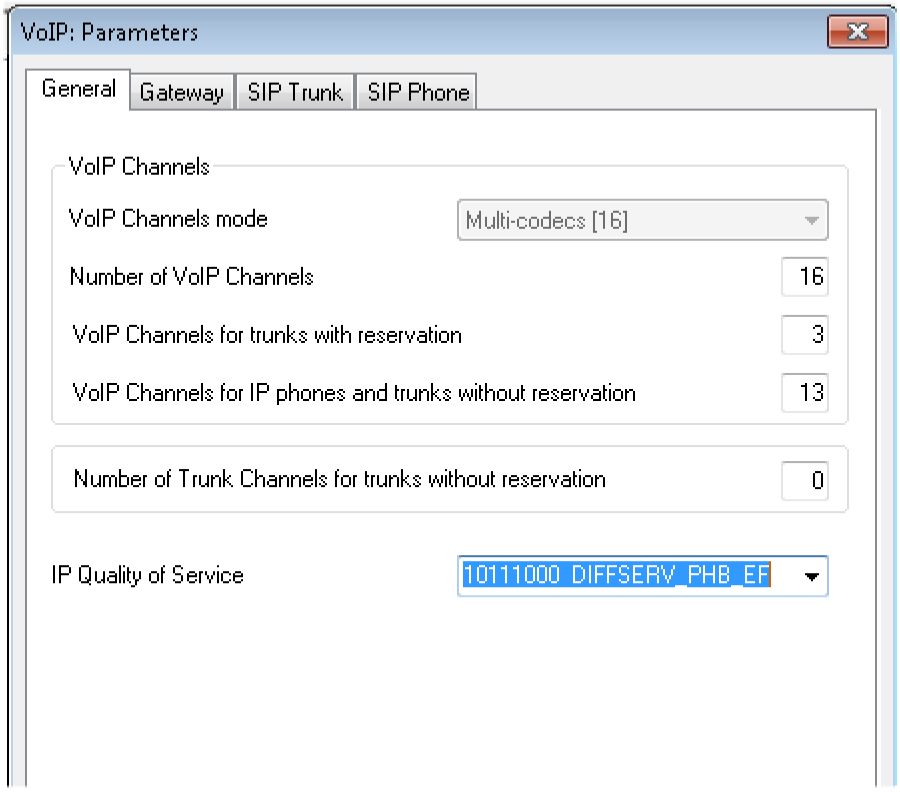
Network OXO Configuration.

This configuration was needed for the setup that we performed above.

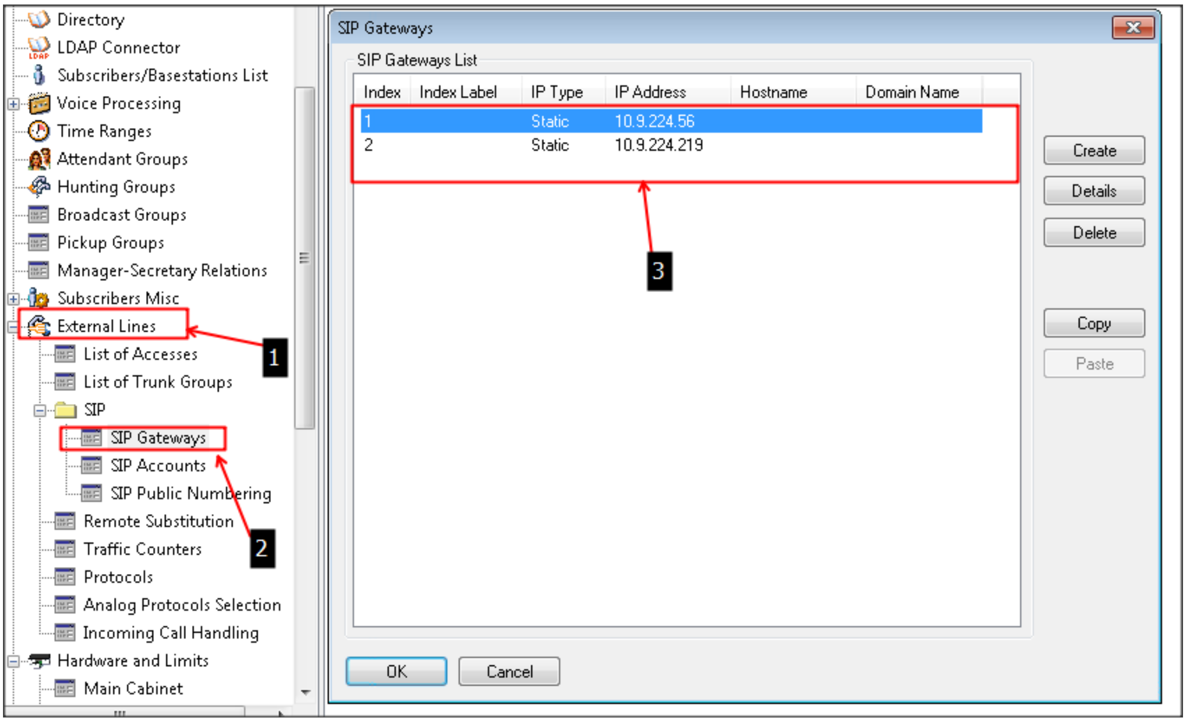
The network OXO setup was done for executing certain specific test cases and can be used for checking external calling facility as OXO connect evolution does not have ISDN option.

Basically network OXO can be considered for external calling scenarios.

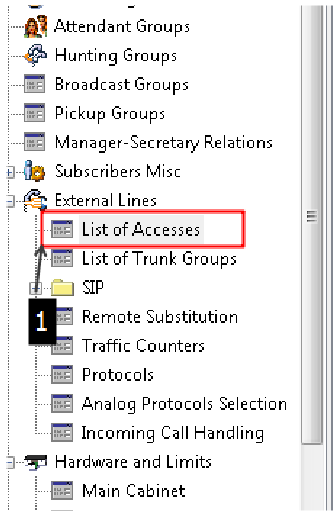
1) Check the VOIP channel availability.

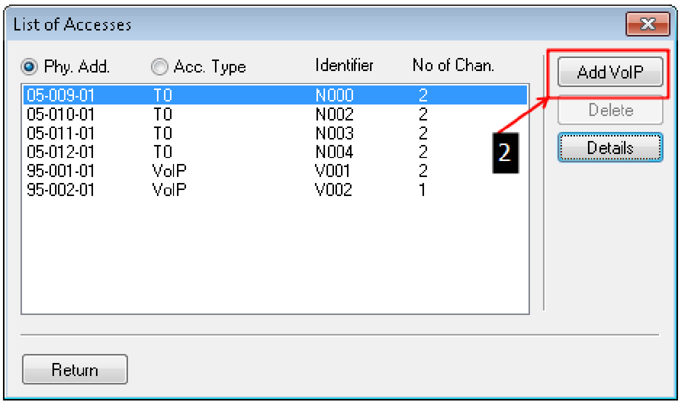


2) Add the gateway under the SIP setting in external lines.

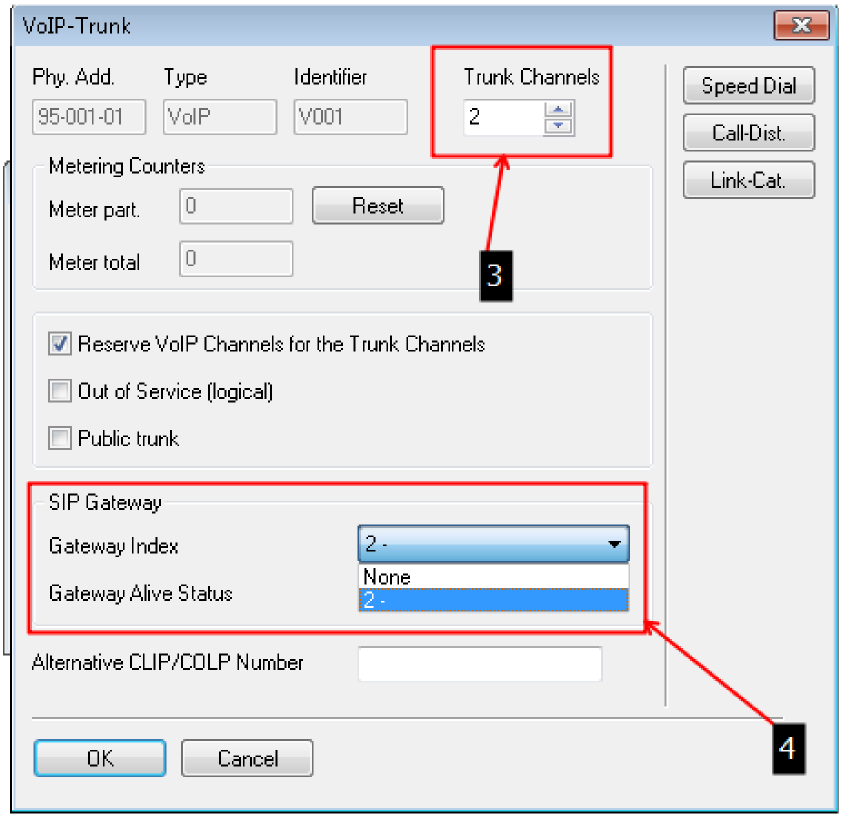


3) Add the VOIP channel to the list of access in external lines.

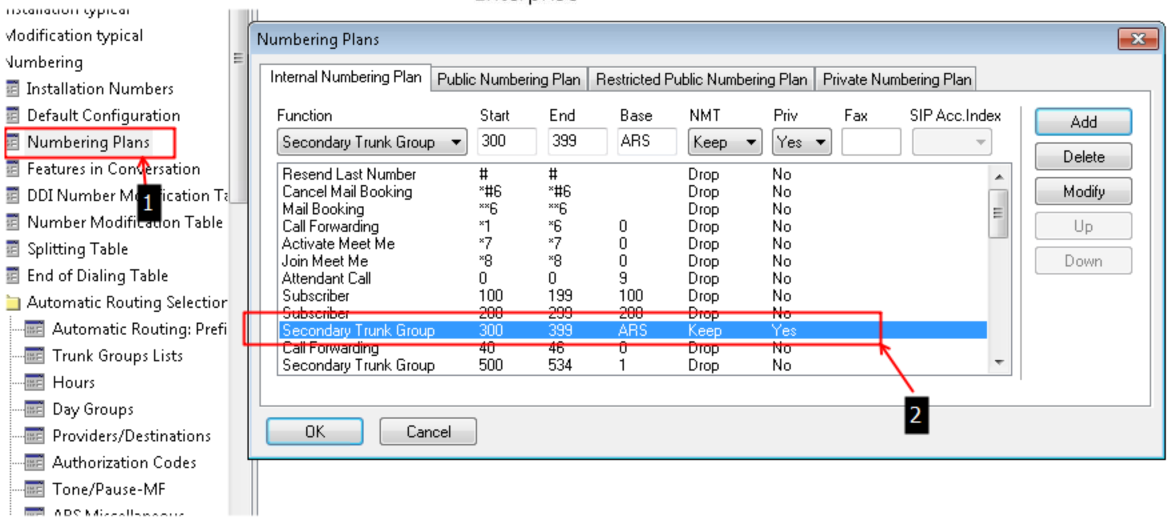




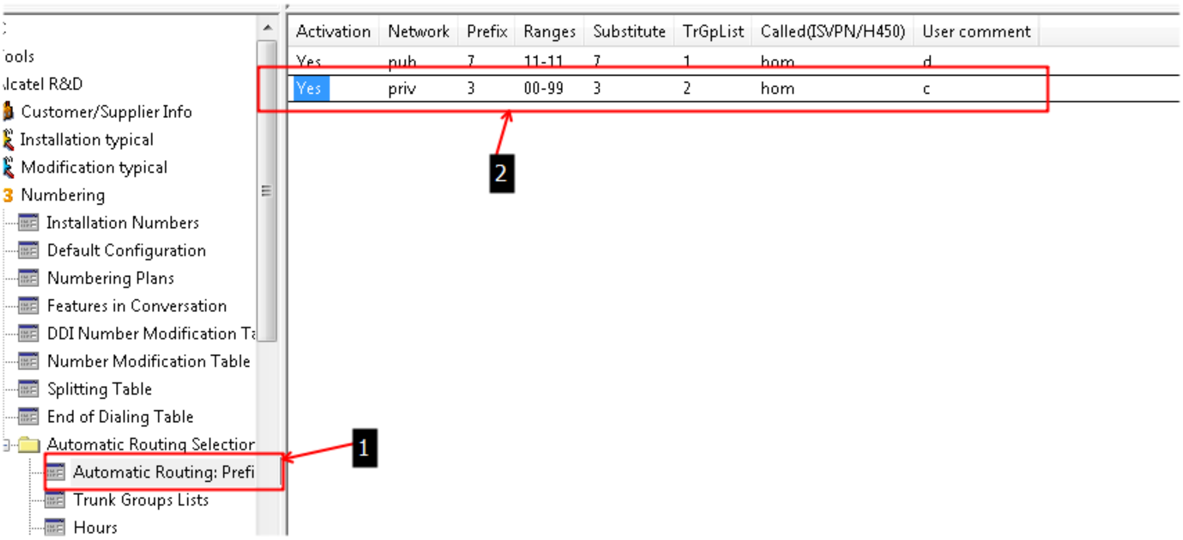
4) Select the gateway in the list of access as shown below.



5) In the ARS manage the numbering range of the Other OXO so that it can be reached.



Manage the Automatic routing prefixes



12 Appendix D: AAPP member’s escalation process

In case you would need technical assistance, please contact the reseller/distributor where you purchased your AudioCodes products. They have been trained on the products to give you 1st and 2nd levels of support. They are in plus in direct relation with 3rd level AudioCodes support in case an escalation would be needed.

13 Appendix E: AAPP program

13.1 Alcatel-Lucent Application Partner Program (AAPP)

The Application Partner Program is designed to support companies that develop communication applications for the enterprise market, based on Alcatel-Lucent Enterprise's product family. The program provides tools and support for developing, verifying and promoting compliant third-party applications that complement Alcatel-Lucent Enterprise's product family. ALE facilitates market access for compliant applications.

The Alcatel-Lucent Application Partner Program (AAPP) has two main objectives:

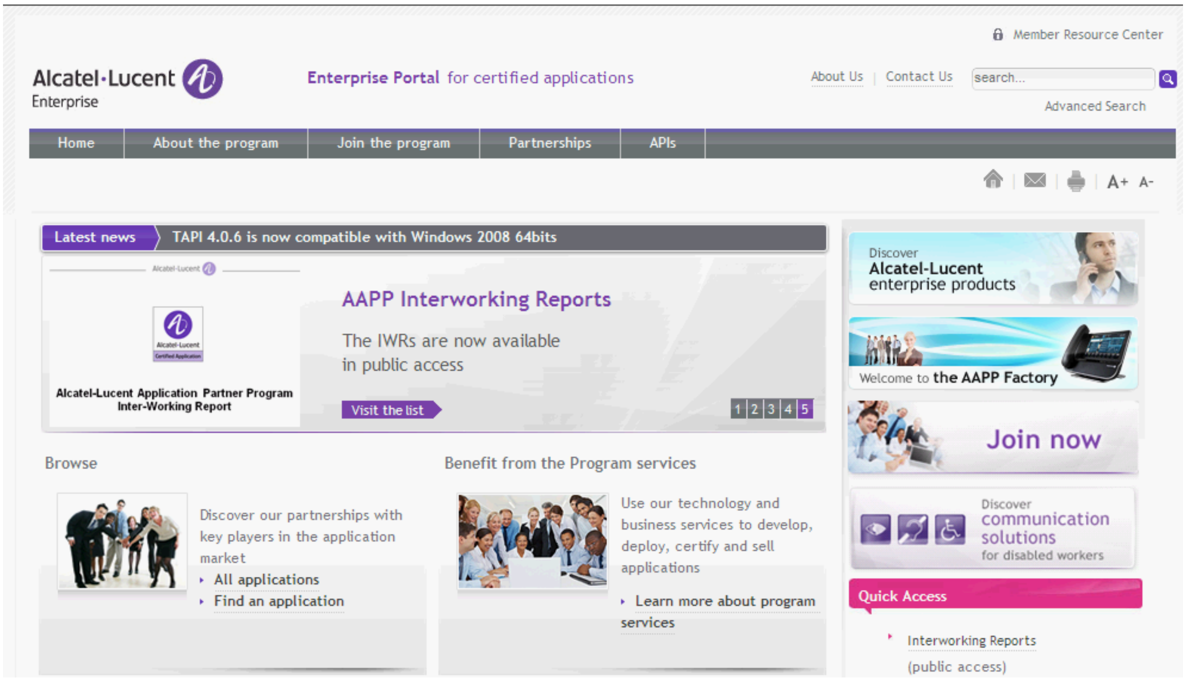
- **Provide easy interfacing for Alcatel-Lucent Enterprise communication products**: Alcatel-Lucent Enterprise's communication products for the enterprise market include infrastructure elements, platforms and software suites. To ensure easy integration, the AAPP provides a full array of standards-based application programming interfaces and fully-documented proprietary interfaces. Together, these enable third-party applications to benefit fully from the potential of Alcatel-Lucent Enterprise products.

- **Test and verify a comprehensive range of third-party applications**: to ensure proper inter-working, ALE tests and verifies selected third-party applications that complement its portfolio. Successful candidates, which are labelled Alcatel-Lucent Enterprise Compliant Application, come from every area of voice and data communications.

The Alcatel-Lucent Application Partner Program covers a wide array of third-party applications/products designed for voice-centric and data-centric networks in the enterprise market, including terminals, communication applications, mobility, management, security, etc.

Web site

The Application Partner Portal is a website dedicated to the AAPP program and where the InterWorking Reports can be consulted. Its access is free at https://www.al-enterprise.com/en/partners/aapp



13.2 Enterprise.Alcatel-Lucent.com

You can access the Alcatel-Lucent Enterprise website at this URL: https://www.al-enterprise.com

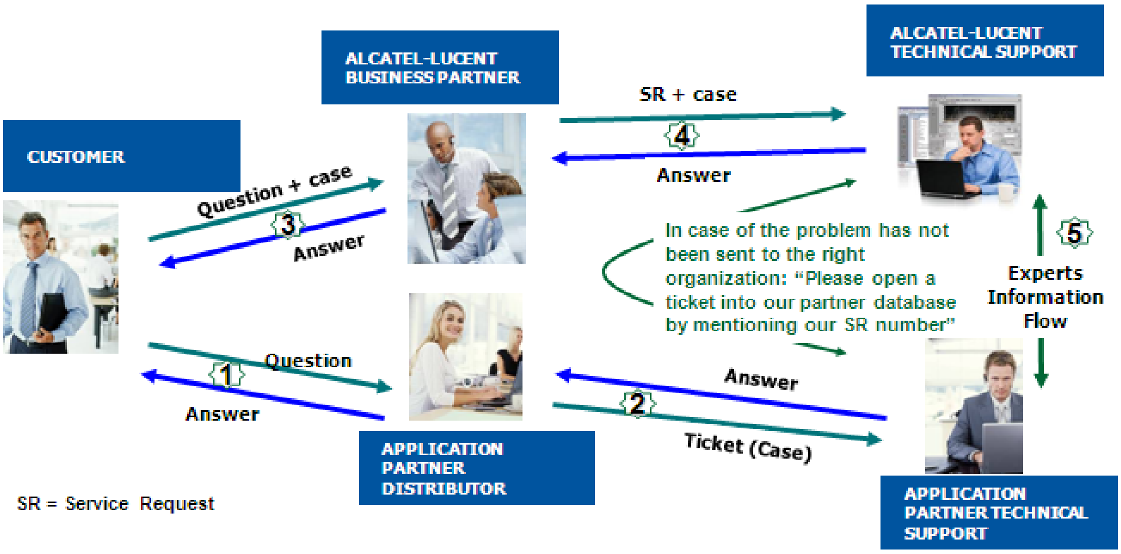
14 Appendix F: AAPP Escalation process

14.1 Introduction

The purpose of this appendix is to define the escalation process to be applied by the ALE Business Partners when facing a problem with the solution certified in this document.

The principle is that ALE Technical Support will be subject to the existence of a valid InterWorking Report within the limits defined in the chapter “Limits of the Technical support”.

In case technical support is granted, ALE and the Application Partner, are engaged as following:



(*) The Application Partner Business Partner can be a Third-Party company or the ALE Business Partner itself

14.2 Escalation in case of a valid Inter-Working Report

The InterWorking Report describes the test cases which have been performed, the conditions of the testing and the observed limitations.

This defines the scope of what has been certified.

If the issue is in the scope of the IWR, both parties, ALE and the Application Partner, are engaged:

Case 1: the responsibility can be established 100% on ALE side.
In that case, the problem must be escalated by the ALE Business Partner to the ALE Support Center using the standard process: open a ticket (eService Request –eSR)

Case 2: the responsibility can be established 100% on Application Partner side.
In that case, the problem must be escalated directly to the Application Partner by opening a ticket through the Partner Hotline. In general, the process to be applied for the Application Partner is described in the IWR.

Case 3: the responsibility can not be established.
In that case the following process applies:

- The Application Partner shall be contacted first by the Business Partner (responsible for the application, see figure in previous page) for an analysis of the problem.

- The ALE Business Partner will escalate the problem to the ALE Support Center only if the Application Partner has demonstrated with traces a problem on the ALE side or if the Application Partner (not the Business Partner) needs the involvement of ALE

In that case**,** the ALE Business Partner must provide the reference of the Case Number on the Application Partner side. The Application Partner must provide to ALE the results of its investigations, traces, etc, related to this Case Number.

ALE reserves the right to close the case opened on his side if the investigations made on the Application Partner side are insufficient or do not exist.

Note: Known problems or remarks mentioned in the IWR will not be taken into account.

For any issue reported by a Business Partner outside the scope of the IWR, ALE offers the "On Demand Diagnostic" service where ALE will provide 8 hours assistance against payment .

IMPORTANT NOTE 1: The possibility to configure the Alcatel-Lucent Enterprise PBX with ACTIS quotation tool in order to interwork with an external application is not
the guarantee of the availability and the support of the solution. The reference remains the existence of a valid InterWorking Report.

Please check the availability of the Inter-Working Report on the AAPP (URL: https://www.al-enterprise.com/en/partners/aapp) or Enterprise Business Portal (Url: Enterprise Business Portal) web sites.

IMPORTANT NOTE 2: Involvement of the ALE Business Partner is mandatory, the access to the Alcatel-Lucent Enterprise platform (remote access, login/password) being the Business Partner responsibility.

14.3 Escalation in all other cases

For non-certified AAPP applications, no valid InterWorking Report is available and the integrator is expected to troubleshoot the issue. If the ALE Business Partner finds out the reported issue is maybe due to one of the Alcatel-Lucent Enterprise solutions, the ALE Business Partner opens a ticket with ALE Support and shares all trouble shooting information and conclusions that shows a need for ALE to analyze.

Access to technical support requires a valid ALE maintenance contract and the most recent maintenance software revision deployed on site. The resolution of those non-AAPP solutions cases is based on best effort and there is no commitment to fix or enhance the licensed Alcatel-Lucent Enterprise software.

For information, for non-certified AAPP applications and if the ALE Business Partner is not able to find out the issues, ALE offers an “On Demand Diagnostic” service where assistance will be provided for a fee.

14.4 Technical support access

The ALE **Support Center** is open 24 hours a day; 7 days a week:

- e-Support from the Application Partner Web site (if registered Alcatel-Lucent Application Partner): https://www.al-enterprise.com/en/partners/aapp
- e-Support from the ALE Business Partners Web site (if registered Alcatel-Lucent Enterprise Business Partners): https://businessportal2.alcatel-lucent.com click under "Contact us" the *eService Request* link
- e-mail: Ebg_Global_Supportcenter@al-enterprise.com
- Fax number: +33(0)3 69 20 85 85
- Telephone numbers:

ALE Business Partners Support Center for countries:

Country	**Supported language**	**Toll free number**
France	French	+800-00200100
Belgium	French	+800-00200100
Luxembourg	French	+800-00200100
Germany	German	+800-00200100
Austria	German	+800-00200100
Switzerland	German	+800-00200100
United Kingdom	English	+800-00200100
Italy	English	+800-00200100
Australia	English	+800-00200100
Denmark	English	+800-00200100
Ireland	English	+800-00200100
Netherlands	English	+800-00200100
South Africa	English	+800-00200100
Norway	English	+800-00200100
Poland	English	+800-00200100
Sweden	English	+800-00200100
Czech Republic	English	+800-00200100
Estonia	English	+800-00200100
Finland	English	+800-00200100
Greece	English	+800-00200100
Slovakia	English	+800-00200100
Portugal	English	+800-00200100
Spain	Spanish	+800-00200100

For other countries:

English answer:	+ 1 650 385 2193
French answer:	+ 1 650 385 2196
German answer:	+ 1 650 385 2197
Spanish answer:	+ 1 650 385 2198

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