PCIS Alliance-Com	Inter-Working certificate		
Date :	02 July 2019	Reference :	rev.2

This document is the result of the certification tests performed by PCIS-Alliance-com between Patton Electronics Co. ("Patton") and Alcatel-Lucent Enterprise ("ALE")'s platforms as per appendix attached.

It certifies proper inter-working between the two platforms.

Alcatel-Lucent Enterprise Communication Platform release	OXO Connect Evolution
Patton Platform release	SmartNode Trinity version 3.15.2
Application Category	SBC Gateway

Test results

☑ Passed ☐ Refused ☐ Postponed

☐ Passed with restrictions

Approvals

Representative	Name	Signature
ALE representative	Laurane Specht	
Patton representative	Brice Imbault	
PCIS Alliance-com representative	Latif Sounfous	
PCIS Alliance-com Director	Thierry Levacher	

Attachment: ALE Application Partner Program - Patton Inter-Working Report

ALE Application Partner Program
Inter-Working Report

Partner: **Patton**

Application type: **eSBC and analogue Gateway**

Application name: **SmartNode Trinity**

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	27th May 2019

ALE representative	Laurane Specht
AAPP member representative	Brice Imbault

Alcatel-Lucent Enterprise Communication Platform	OXO Connect Evolution OmniPCX Office
Alcatel-Lucent Enterprise Communication Platform release	
AAPP member application release	Trinity version 3.15.2
Application Category	SBC Gateway

Author(s): Brice Imbault
Reviewer(s): Laurane Specht

Revision History

Edition 1: 27-May- 2019 – First edition – Initial signed version
Edition 2: 02-July-2019 – Second edition. Added configurations used during the tests in sections 6, 11 and AAPP partner escalation procedure in section 12.

Test results

☑ Passed ☐ Refused ☐ Postponed

☐ Passed with restrictions

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

IWR validity extension

The tests have been done using Patton SmartNode's Trinity software suite 3.15.2 running on all SmartNode product line for eSBC, Gateway and IAD applications. So all SIP to SIP and FXS analogue gateway to SIP tests are valid for the entire product range.
A description of the SmartNode models is available online at: www.patton.com/products/voip-comparison.asp

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

Note 2: *The renewal of the interoperability test (certification) is under the responsibility of the partner except if the certification fee is included in the program fee (e.g. “Application Partner” membership level) in this case ALE will schedule a new certification every two year*

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: Patton Session Border Controller and Analogue gateway

Application version: Smartnode Trinity FW 3.15.X

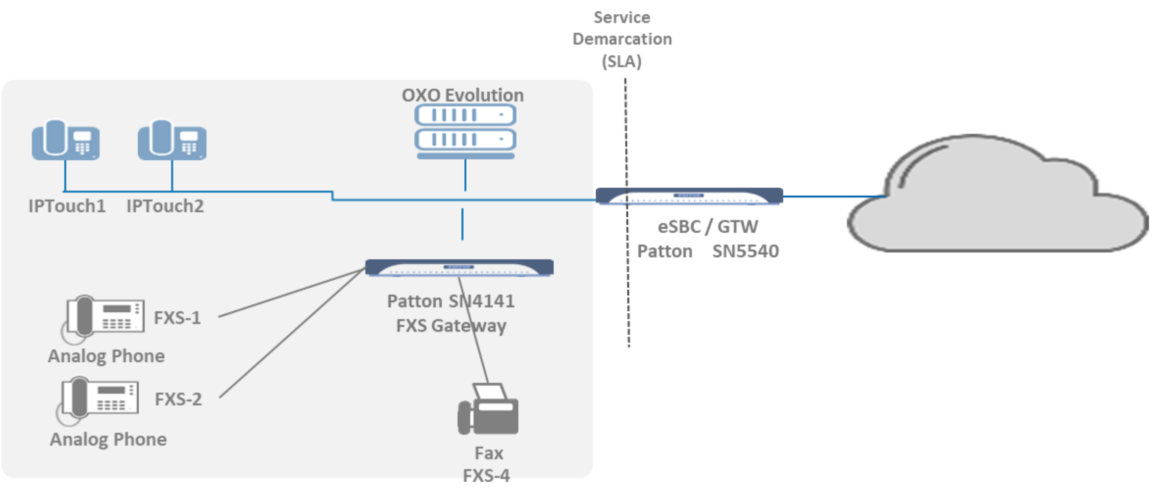
Interface type: SIP and Analogue FXS voice

Brief application description:

The SMARTNODE combines a VOIP Analog adapter and an Enterprise Session Border Controller (eSBC), to control VoIP Media and secures network and access.

5 Test environment

Figure 1 **Test environment**



Note : For the purpose of these tests, the SIP trunk is provided by Synelyans.

5.1 Hardware configuration

List main hardware equipments used for testing

- **OmniPCX Office Evolution**:
  Iptouch: 8068,8058s,8028s
  Fax: MFP sagemcom
- **AHL interface**:
  - TCP/IP

5.2 Software configuration

List main softwares used for testing

- **Alcatel-Lucent Office Evolution: R300_075_001**
- **Partner Application** : SmartNode Trinity version 3.15.x

6 Summary of test results

6.1 Summary of main functions supported

Oxo connect Evolution with SBC

Features	Results	Remarks
Initialisation and network configuration	OK	
Sip registration	OK	
Sip Authentication	OK	
VoIP and RTP support	OK	
Outgoing call	OK	
Incoming call	OK	
Features during conversations	OK	

Oxo connect Evolution with Analogue Gateway

Features	Results	Remarks
Initialisation and network configuration	OK	
Sip registration	OK	
Sip Authentication	OK	
VoIP and RTP support	OK	
Outgoing call	OK	
Incoming call	OK	
Features during conversations	OK	

6.2 Summary of problems

None

6.3 Summary of limitations

None

6.4 Notes, remarks

- Analogues phones are registered in the OXO connect as “Open SIP Phones”
- We only tested with One fax

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 <u>and the expected result</u>
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, <u>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</u>
Comment: to be filled in with any relevant comment. Mandatory in case a test has failed especially the reference number of the issue.

8 SmartNode TEST Results

There are two main test objectives, one for a VoIP Gateway and the other for an eSBC.
The tests are to validate that the SmartNode can handle the configured audio parameters as defined.
Such as codecs, Framing, Voice Activation Detection, …

8.1 eSBC Tests

8.1.1 Test Objectives

The eSBC Configuration:
eSBC is configured to use specific codec
G.722, G.711 A-law, G.711 mu-law, G.729, G.723 in this order
Phone configuration:
Configure IP Touch with codec G.722, G.711 A-law, G.711 mu-law, G.729, G.723 in this order
and to NOT use VAD (unless otherwise stated).
Sip provider:
Configure Sip Provider to use G.722, G.711 A-law, G.711 mu-law, G.729, G.723 in this order

8.1.2 Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
8-1-1	**Codec G711 / G722** Select G711 as first codec on Provider Select G711 as first codec on SBC Select G722 as first codec on IPtouch And G.711 A-law, G.711 mu-law, G.729 as other priority Call from external phone (PSTN) to Ip touch phone Check that call is correctly established *In all Case check audio quality*	☐	☒	☐	
8-1-2	**Codec G729 / G711** Select G729 as first codec on Provider Select G729 as first codec on SBC Select G711 as first codec on IPtouch And G.729 as other priority Call from external phone to Ip touch phone Check that call is correctly established	☐	☒	☐	

	In all Case check audio quality				
8-1-3	**Codec G723 / G711** Select G723 as first codec on Provider Select G723 as first codec on SBC Select G711 as first codec on IPtouch Call from external phone to Ip touch phone Check that call is correctly established *In all Case check audio quality*	☒	☐	☐	
8-1-4	**Codec G723 / G711** Select G723 as first codec on Provider Select G723 as first codec on SBC Select G711 as first codec on IPtouch Call from external phone to Ip touch phone Check that call is correctly established *In all Case check audio quality*	☒	☐	☐	

8.2 Outgoing call

8.2.1 Test Objectives

Generate calls to External PSTN line to check SBC integrity
The outgoing call is generate on an external PSTN phone number

8.2.2 Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
8-2-1	**Outgoing call with DTMF RFC 2833** Call to external attendant using DTMF RFC 2833 Test DTMF return. Call an IVR and navigate to the corresponding menu and verify that DTMF is working Hang-up the call	☐	☒	☐	
8-2-2	**Outgoing call with DTMF Sip Info**	☒	☐	☐	Not support in OXO

	Call to external attendant using DTMF Sip Info Test DTMF return. Call an IVR and navigate to the corresponding menu and verify that DTMF is working Hang-up the call Then Hang-up				
8-2-3	**Outgoing call with DTMF Inband** Call to external attendant using DTMF Inband Test DTMF return. Call an IVR and navigate to the corresponding menu and verify that DTMF is working Hang-up the call Then Hang-up	☐	☐	☒	Inband to Sip info not supported. Inband to RFC 2833 is support with DSP processing
8-2-4	**Call to External number from VPN connected ip touch** Call external number from VPN ip Touch Check audio, then hang-up	☒	☐	☐	
8-2-5	**Outgoing call with DTMF RFC 2833 with VPNIpTouch** Call to external attendant using DTMF RFC 2833 Test DTMF return. Call an IVR and navigate to the corresponding menu and verify that DTMF is working Hang-up the call	☒	☐	☐	
8-2-6	**Outgoing call with DTMF Sip Info with VPNIpTouch** Call to external attendant using DTMF Sip Info Test DTMF return. Call an IVR and navigate to the corresponding menu and verify that DTMF is working Hang-up the call Then Hang-up	☒	☐	☐	
8-2-7	**Outgoing call with DTMF Inband with VPNIpTouch** Call to external attendant using DTMF Inband Test DTMF return. Call an IVR and navigate to the corresponding menu and verify that DTMF is working Hang-up the call Then Hang-up	☒	☐	☐	

8.3 Incoming call

8.3.1 Test Objectives

Generate calls from External PSTN line to check SBC integrity
Called party can be in different states: Free, Busy, Out of services, DND, etc…

8.3.2 Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
8-3-1	**ExtCall to Iptouch Timout** Call from Ext-PSTN to the DID configure on Iptouch1 Answer the call and check audio. Stay online for 5 minutes Then Hang-up	☐	☒	☐	
8-3-2	**ExtCall to Iptouch Display** Call from Ext-PSTN to the DID configure on Iptouch1 Check display Answer the call and check audio. Then Hang-up	☐	☒	☐	
8-3-3	**ExtCall to Iptouch Display multiline** Call from Ext-PSTN to the DID configure on Iptouch1 Check display Answer the call and check audio. Keep the call Call from Ext-PSTN to the DID configure on Iptouch1 Check display Answer the 2d call and check audio. Then Hang-up	☐	☒	☐	
8-3-4	**Call from External number to VPN connected ip touch** Make a call to External number Answer the call on VPN ip Phone Check audio then hang-up	☒	☐	☐	
8-3-5	**Call from VPN Ip touch to Iptouch1** Make a call to internal number allocated to Iptouch1 Answer the call on Iptouch1 Check audio then hang-up	☒	☐	☐	
8-3-6	**Ext call to Unplug VPNPhone** **If Applicable**	☒	☐	☐	

	Unplug VPNIPtouch With Ext PSTN phone call VPNIPtouch Check the ring back then hang-up				
8-3-7	**Ext call to DND VPNPhone** **If Applicable** Enable DND on VPNIPtouch With Ext PSTN phone call VPNIPtouch Check the ring back then hang-up Cancel the DND on VPNIPtouch	☒	☐	☐	

8.4 Features during call

8.4.1 Test Objectives

The objective is to test Features between different users during conversation.
Before test we need to check that dtmf are generated correctly, and multiple sip line is available on devices.

8.4.2 Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
8-4-1	**Hold and resume a current call** From Ext PSTN call IPtouch1 Answer the call and check audio. On IPtouch1press hold. Check tones and display on both parts Resume the call *Keep the call for next test*	☐	☒	☐	
8-4-2	**Switch between calls** With FXS-1 call IPtouch1 With IPtouch1 switch between FXS-1 and Ext PSTN Check tones and display *Keep the calls for next test*	☐	☒	☐	
8-4-3	**Three party conferences initiated from OXO set** With Ext PSTN call IPtouch1 Answer and keep the call	☐	☒	☐	

	With IPtouch1 call IPtouch2 Answer and keep the call With IPtouch1 start a conference Check audio, Display, then hang-up.				

8.5 Call Transfer

8.5.1 Test Objectives

Many sorts of transfer can be requested, the objective is to test several transfer services.

- Unattended transfer
- Semi-attended transfer
- Attended transfer

For each we need to test:

- Audio
- Tone
- Display

Actors:

A- Transferee
B- Transferor
C- Transfer target

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. The transferee call the Transferor
2. The transferor call the transfer target. The transferee is on Hold. The transfer target is ringing.
3. The transferor execute the transfer. The transferor drops the call. The transfer target is already in ringing state, The transfer target answer the call. The Transferee and the Transfer target are now in communication.

Attended Transfer or Transfer on ringing:

1. The transferee call the Transferor
2. The transferor call the transfer target. The transferee is on Hold. Transfer target pick up the call and call is established with the transferor
3. The transferor execute the transfer. The transferor drops the call. The transferee is now on line with the Transfer target.

8.5.2 Test Result

Test Case Id	Sort of transfer	Transferee	Transferor	Transfer Target	N/A	OK	NOK	Comment
8-5-1	Unattended	IPtouch2	IPtouch1	ExtNum	☐	☒	☐	Shown

								number is the IPtouch1
8-5-2	Semi-attended	IPtouch2	IPtouch1	ExtNum	☐	☒	☐	
8-5-3	Attended	IPtouch2	IPtouch1	ExtNum	☐	☒	☐	

9 Analogue VoIP-Gateway TEST Results

9.1 Analogue VoIP Gateway tests

9.1.1 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: ring-back tones, busy tones, voice during the conversation, display (on caller and called party), hang-up phase.

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

9.1.2 Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
9-1-1	**Analogue phone sets – Without authentication** Configure SIP sets MCDU number on the OXO as FXS-1, FXS-2 & FXS-3 to register with the OXO IP address. Authentication is disable for this users Check registration of sets	☐	☒	☐	
9-1-2	**Analogue phone sets – With authentication** Configure SIP sets MCDU number on the OXO as FXS-1, FXS-2 & FXS-3 to register with the OXO IP address. Authentication is **enable** for this users Test with a wrong password and check the phone is rejected	☐	☒	☐	
9-1-3	**Signalling in UDP and TCP** If applicable configure your SIP FXS-2 to use the protocol SIP over UDP and over TCP In the two cases, check the registration and basic calls.	☐	☒	☐	

9.2 Codec negotiation

9.2.1 Test Objectives

These tests check that phones are using the configured audio parameters as defined.
Codecs, Framing, Voice Activation Detection, …
Phone configuration: configure the analog gateway to use
G.722, G.711 A-law, G.711 mu-law, G.729, G.723 in this order
Configure IP Touch with codec G.711 and to NOT use VAD (unless otherwise stated).

9.2.2 Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
9-2-1	**Codec G711** Select G711 as first codec Call from FXS-1 to Ip touch phone Check that call is correctly established in G711 Call from Ip touch to FXS-1 phone Check that call is correctly established in G711 *In all Case check audio quality*	☐	☒	☐	
9-2-2	**Codec G729** Select G729 as first codec Call from FXS-1 to Ip touch phone Check that call is correctly established in G729 Call from Ip touch to FXS-1 phone Check that call is correctly established in G729 *In all Case check audio quality*	☐	☒	☐	
9-2-3	**Codec G723** Select G723 as first codec Call from FXS-1 to Ip touch phone Check that call is correctly established in G723 Call from Ip touch to FXS-1 phone Check that call is correctly established in G723 *In all Case check audio quality*	☒	☐	☐	
9-2-4	**VAD Test** Configure FXS-1 to use VAD Configure IP touch **Not using** VAD	☐	☒	☐	

	Call from Ip touch to FXS-1 in G711 and check audio quality Configure IP touch **using** VAD Call from Ip touch to FXS-1 in G711 and check audio quality				
9-2-5	**Codec Passthrough** In OXO enable codec pass through for SIP Phones. Call from FXS-1 to FXS-2 Check that the call is established using G.722 Check audio quality	☐	☒	☐	

9.3 Outgoing call

9.3.1 Test Objectives

Generate calls from and to several users/devices.
Called party can be in different states: Free, Busy, Out of services, DND, etc…

9.3.2 Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
9-3-1	**Call to User in DND mode** Dial 793 on Ip touch (DND mode) Call from FXS-1 to the Ip touch Check for a ring back tone and display	☐	☒	☐	
9-3-2	**Call to User in CFU mode** Turn CFU in Ip touch phone 1 (791 ip touch phone 2) Call from FXS-1 to the IPtouch1 Check the IPtouch2 is ringing Answer the call and check audio, then hung up. Cancel the CFU mode (790)	☐	☒	☐	
9-3-3	**Call to User in CFNR mode** Turn CFNR in Ip touch phone 1 (CFNR already configure to ip touch phone 2) Call from FXS-1 to the IPtouch1	☒	☐	☐	

	Check the IPtouch1 is ringing after time out check IPtouch2 is ringing Answer the call and check audio, then hung up.				
9-3-4	**Call to User in CFB mode** Setup the CFB on IPtouch1 to IPtouch2 (*62+IPtouch2) Call from FXS-2 to the IPtouch1 and answer to make it buzzy Call from FXS-1 to the IPtouch1 Check IPtouch2 is ringing Answer the call and check audio, then hung up.	☐	☒	☐	
9-3-5	**Call to External number from FXS** Call external number from FXS-1 Check audio, then hang-up	☐	☒	☐	
9-3-6	**Call from External number to FXS** Make a call to External number allocated to FXS-1 Answer the call on FXS-1 Check audio then hang-up	☐	☒	☐	
9-3-7	**Call from VPN Ip touch to FXS-1** Make a call to internal number allocated to FXS-1 Answer the call on FXS-1 Check audio then hang-up	☐	☒	☐	
9-3-8	**Call from FXS-1 to VPN Ip touch** Make a call to internal number allocated to VPN ip touch Answer the call on VPN IP touch Check audio then hang-up	☐	☒	☐	

9.4 Incoming call

9.4.1 Test Objectives

Generate calls from and to several users/devices.
Called party can be in different states: Free, Busy, Out of services, DND, etc…
Network calls are made using SIP private trunk established between two OXO's.

9.4.2 Test Results

Test Case	Test Case	N/A	OK	NOK	Comment

Id					
9-4-1	**Local/Network call to FXS** **Local**: With IPtouch1 call FXS-2 Answer the call and check audio, then hung up. **Network**: With Network IP touch call FXS-2 Answer the call and check audio, then hung up.	☐	☒	☐	
9-4-2	**Local/Network call to Busy FXS** **Local**: With IPtouch2 call FXS-2 Answer the call, don't hang up With IPtouch1 call FXS-2 Check the ring back then hang-up **Network**: With network IPtouch2 call FXS-2 Answer the call, don't hang up With IPtouch1 call FXS-2 Check the ring back then hang-up	☐	☒	☐	
9-4-3	**Local/Network call to Unplug FXS** **Local**: Unplug FXS-1 With IPtouch1 call FXS-1 Check the ring back then hang-up **Network**: Unplug FXS-1 With network IPtouch1 call FXS-1 Check the ring back then hang-up	☐	☒	☐	Back to the VM
9-4-4	**Local/Network call to DND FXS** **Local**: enable DND on FXS-1 With IPtouch1 call FXS-1 Check the ring back then hang-up Cancel the DND on FXS-1 **Network**: enable DND on FXS-1 With Network IPtouch1 call FXS-1 Check the ring back then hang-up Cancel the DND on FXS-1	☐	☒	☐	
9-4-5	**Local/Network call to DND FXS With System function** **Local**: enable DND on FXS-1 (*63) With IPtouch1 call FXS-1 Check the ring back then hang-up Cancel the DND on FXS-1(*63)	☐	☒	☐	

	Network: enable DND on FXS-1(*63) With Network IPtouch1 call FXS-1 Check the ring back then hang-up Cancel the DND on FXS-1(*63)				
9-4-6	**Local/Network call to CFU FXS** **Local**: enable CFU on FXS-1 to IPtouch2 With IPtouch1 call FXS-1 Answer the call on IPtouch2 and check audio, then hung-up. Cancel the CFU on FXS-1 **Network**: enable CFU on FXS-1 to IPtouch2 With Network IPtouch1 call FXS-1 Answer the call on IPtouch2 and check audio, then hung-up. Cancel the CFU on FXS-1	☐	☒	☐	
9-4-7	**Local/Network call to CFU FXS With System Function** **Local**: enable CFU on FXS-1 to IPtouch2 (791+IPtouch2) With IPtouch1 call FXS-1 Answer the call on IPtouch2 and check audio, then hung-up. Cancel the CFU on FXS-1 (*60) **Network**: enable CFU on FXS-1 to IPtouch2 (*61+IPtouch2) With Network IPtouch1 call FXS-1 Answer the call on IPtouch2 and check audio, then hung-up. Cancel the CFU on FXS-1(*60)	☐	☒	☐	
9-4-8	**Local/Network call to CFU FXS to external number** **Local**: enable CFU on FXS-1 to external number (791+ExtNum) With IPtouch1 call FXS-1 Answer the call on external number and check audio, then hung-up. Cancel the CFU on FXS-1(790) **Network**: enable CFU on FXS-1 to external number(791+ExtNum) With Network IPtouch1 call FXS-1 Answer the call on external number and check audio, then hung-up.	☐	☒	☐	

	Cancel the CFU on FXS-1(790)				
9-4-9	**Local/Network call to CFU FXS to external number with system feature** **Local**: enable CFU on FXS-1 to external number With IPtouch1 call FXS-1 Answer the call on external number and check audio, then hung-up. Cancel the CFU on FXS-1 **Network**: enable CFU on FXS-1 to external number With Network IPtouch1 call FXS-1 Answer the call on external number and check audio, then hung-up. Cancel the CFU on FXS-1	☐	☒	☐	
9-4-10	**Local/Network call to CFB FXS state By local feature if applicable (792+iptouch1) and (790)** On FXS-2 enable CFB to IPtouch1 On FXS-2 call the voice mail With FXS-1 call FXS-2 Check that IPtouch1 is ringing	☐	☒	☐	
9-4-11	**Local call to SIP terminal in "forward on no reply" (CFNR) By local feature if applicable** On FXS-3 enable CFNR to IPtouch1 With FXS-2 call FXS-3 Check FXS-3 is ringing and wait timeout After timeout IPtouch1 is ringing Answer the call and check audio, then hung-up.	☒	☐	☐	
9-4-12	**Local/Network call to Busy VPNPhone If Applicable** With IPtouch2 call VPNIPtouch Answer the call, don't hang up With IPtouch1 call VPNIPtouch Check the ring back then hang-up	☒	☐	☐	
9-4-13	**Local/Network call to Unplug VPNPhone If Applicable** Unplug VPNIPtouch With IPtouch1 call VPNIPtouch Check the ring back then hang-up	☒	☐	☐	
9-4-14	**Local/Network call to DND VPNPhone If Applicable**	☒	☐	☐	

	Enable DND on VPNIPtouch With IPtouch1 call VPNIPtouch Check the ring back then hang-up Cancel the DND on VPNIPtouch				

9.5 Features during call

9.5.1 Test Objectives

The objective is to test Features between different users during conversation.
Before test we need to check that DTMF are generated correctly, and multiple sip line is available on devices.

Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
9-5-1	**Hold and resume a current call** With IPtouch1 call FXS-1 Answer the call and check audio. On FXS-1 press hold. Check tones and display on both parts Resume the call *Keep the call for next test*	☐	☒	☐	
9-5-2	**Enquiry call to another local user (if available)** With FXS-1 call IPtouch2 IPtouch1 should be turn on Hold Put IPtouch2 on hold. Check tones and display *Keep the calls for next test*	☐	☒	☐	
9-5-3	**Switch between calls** With FXS-1 switch between IPtouch1 and IPtouch2 Check tones and display *Keep the calls for next test*	☐	☒	☐	
9-5-4	**Release call** Hang-up IPtouch1 Check that FXS-1 and IPtouch2 are still online Check audio, Display, then hang-up.	☐	☒	☐	

9-5-5	**Three party conferences initiated from OXO set** With IPtouch1 call FXS-1 Answer and keep the call With IPtouch1 call IPtouch2 Answer and keep the call With IPtouch1 start a conference Check audio, Display, then hang-up.	☐	☒	☐	
9-5-6	**Meet Me conference** With FXS-3 call Meet me conference Bridge (68) With FXS-2 join the the conference bridge (709 + access code) With IPtouch1 join the the conference bridge (709 + access code) Check that all are in conference Check audio, Display, then hang-up.	☒	☐	☐	

9.6 Call transfert

9.6.1 Test Objectives

Many sorts of transfer can be requested, the objective is to test several transfer services.

- Unattended transfer
- Semi-attended transfer
- Attended transfer

For each we need to test:

- Audio
- Tone
- Display

Actors:

D- Transferee
E- Transferor
F- Transfer target

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:

4. The transferee call the Transferor
5. The transferor call the transfer target. The transferee is on Hold. The transfer target is ringing.

6. The transferor execute the transfer. The transferor drops the call. The transfer target is already in ringing state, The transfer target answer the call. The Transferee and the Transfer target are now in communication.

Attended Transfer or Transfer on ringing:

4. The transferee call the Transferor
5. The transferor call the transfer target. The transferee is on Hold. Transfer target pick up the call and call is established with the transferor
6. The transferor execute the transfer. The transferor drops the call. The transferee is now on line with the Transfer target.

9.6.2 Test Result

Test Case Id	Sort of transfer	Transferee	Transferor	Transfer Target	N/A	OK	NOK	Comment
9-6-1	Unattended	ExtCall	FXS-1	IPtouch1	☒	☐	☐	
9-6-2	Semi-attended	ExtCall	FXS-1	IPtouch1	☐	☒	☐	
9-6-3	Attended	ExtCall	FXS-1	IPtouch1	☐	☒	☐	
9-6-4	Unattended	FXS-1	IPtouch1	ExtCall	☒	☐	☐	
9-6-5	Semi-attended	FXS-1	IPtouch1	ExtCall	☐	☒	☐	
9-6-6	Attended	FXS-1	IPtouch1	ExtCall	☐	☒	☐	
9-6-7	Unattended	IPtouch2	IPtouch1	FXS-1	☒	☐	☐	
9-6-8	Semi-attended	IPtouch2	IPtouch1	FXS-1	☐	☐	☒	ALE fix will be available early Sept 2019
9-6-9	Attended	IPtouch2	IPtouch1	FXS-1	☐	☒	☐	

9.7 Attendant

9.7.1 Test Objectives

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

9.7.2 Test Result

Test Case Id	Test Case	N/A	OK	NOK	Comment
9-7-1	**Call to attendant** From FXS-1 dial 9	☐	☒	☐	

	Check audio and display *Keep the calls for next test*				
9-7-2	**Incoming call During attendant call.** From IPtouch1 call FXS-1 Answer the call Check audio and display	☐	☒	☐	
9-7-3	**SIP set call to attendant, attendant transfers to OXO set** From FXS-1 dial 9 Answer the call Attendant transfer to IPtouch2 Check audio and display	☐	☒	☐	
9-7-4	**SIP set call to attendant, attendant transfers to FXS-1** From IpTouch dial 9 From Attendant transfer "attended" to FXS-1 Check audio and display*	☐	☐	☒	ALE fix will be available early Sept 2019

9.8 Voice Mail

9.8.1 Test Objectives

Voice mail notification, menu consultation, password modification MWI must be checked
VMS must be enable on FXS-1, FXS-2, IPtouch1
DTMF function need to be enable

9.8.2 Test Result

Test Case Id	Test Case	N/A	OK	NOK	Comment
9-8-1	**Password modification** From FXS-1 dial the Voicemail Use the Voice guide to change password Hang-up Recall and test with the old password. Password need to be rejected Hang-up Recall and test with the new password. Password need to be accepted Hang-up	☐	☒	☐	
9-8-2	**MWI test**	☒	☐	☐	

	From IPtouch1 call FXS-1 and leave a message on the Voicemail From FXS-1 call IPtouch1 and leave a message on the Voicemail Check you have MWI on both phones.				
9-8-3	**Message consultation** With FXS-1 all VM and listen previously leaved message 8-8-2 Delete the message With IPtouch1 all VM and listen previously leaved message 8-8-2 Delete the message **Check That MWI is disable on both phones**	☐	☒	☐	
9-8-4	**Sip Call to a Phone forwarded to VoiceMail** Forward IPtouch1 to voicemail (*61+VM number) With FXS-1 call IPtouch1 Call should be forwarded to VM Leave a message On IPtouch1 disable VM forward (*60) Then check that you Have MWI	☐	☒	☐	
9-8-5	**VPNIPtouch Voicemal** Replay Test from 8-8-1 to 8-8-4 with VPNIPtouch	☐	☒	☐	

9.9 Defense

9.9.1 Test Objectives

Test the situation where there is a loss of connection, Ethernet failure, OXO reboot….

9.9.2 Test Result

Test Case Id	Test Case	N/A	OK	NOK	Comment
9-9-1	**Oxo Reboot** From external number call FXS-1 Reboot the OXO After reboot re-establish the same call Check audio	☐	☒	☐	
9-9-2	**Ethernet link failure** From external number call IPtouch1	☐	☒	☐	

	Disconnect ethernet cable of the IPtouch1 Then reconnect After re-establishing do the same call Check audio				
9-9-3	**Power failure on** From external number call IPtouch1 Disconnect ethernet cable of the IPtouch1 From external number call IPtouch1 Check that incoming call go to attendant With IPtouch1 all VM and listen previously leaved message 8-8-2 Delete the message **Check That MWI is disable on both phones**	☐	☒	☐	

9.10 Fax Test

9.10.1 Test Objectives

Fax machine is connected to FXS port of the Smartnode.
FXS-4 port is configuration with g711 only passthrough method.
FXS-3 port is configure the same way.

9.10.2 Test Result

Test Case Id	Test Case	N/A	OK	NOK	Comment
9-10-1	**Fax sending between two fax devices** Send a 4 pages fax from FAXset-1 to FAXset-2	☒	☐	☐	Only 1 internal Fax for test
9-10-2	**External Fax to fax device** Send a 4 pages fax from External fax machine to FAXset-1	☐	☒	☐	
9-10-3	**Fax device to External Fax** Send a 4 pages fax from FAXset-1 to External fax machine	☐	☒	☐	
9-10-4	**Stop sending Fax after the first page** Send a 4 pages fax from External Fax to FAXset-1 Stop the transmission after the 1 page. Check that transmission is correctly stopped	☒	☐	☐	
9-10-5	**Stop receiving Fax after the first page** Send a 4 pages fax from FAXset-1 to External Fax Stop the transmission after the 1 page.	☒	☐	☐	

	Check that transmission is correctly stopped				

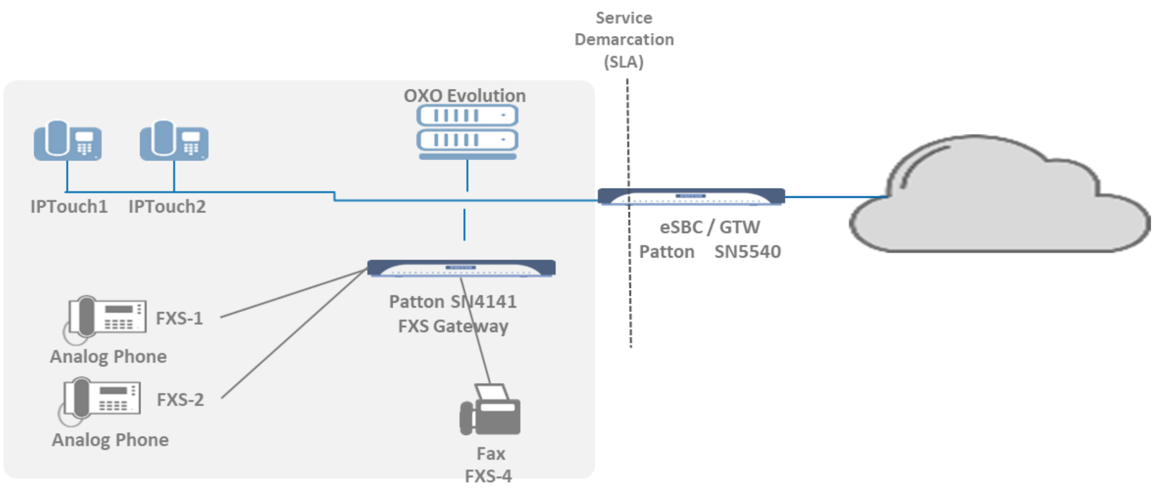
10 Appendix A : AAPP member's Application description

***The SmartNode** is a customer premise Enterprise Session Border Controller, delivering the features for advanced multiservice voice and data network applications. It combines highly flexible SIP routing and manipulation features with powerful quality of service IP routing functions to build professional and reliable VoIP and data networks.*

*SmartNode enables Universal SIP Trunking and provides a single Integrated Access Device with features like IP Routing, VoIP and IP Security and a SIP registrar for survivability. The SN5XXX product series connects to the **OXO system** in the Enterprise's LAN and to an Internet telephony service provider (ITSP), creating a single conduit for multimedia components including voice, video, and data. Whether it is a new installation or an existing deployment, this device will aid in deploying, troubleshooting, logging, and security while increasing the flexibility of the network.*

Applications
SmartNode enables protocol conversion between two VoIP networks to solve interop problems for devices using SIP TCP signaling only. The SmartNode is able to convert SIP TCP or SIP TLS signaling into SIP UDP signaling.

Using the built-in QoS engine, the SmartNode ensures that voice traffic gets top priority resulting in good voice quality across the SIP Trunk over a public network.



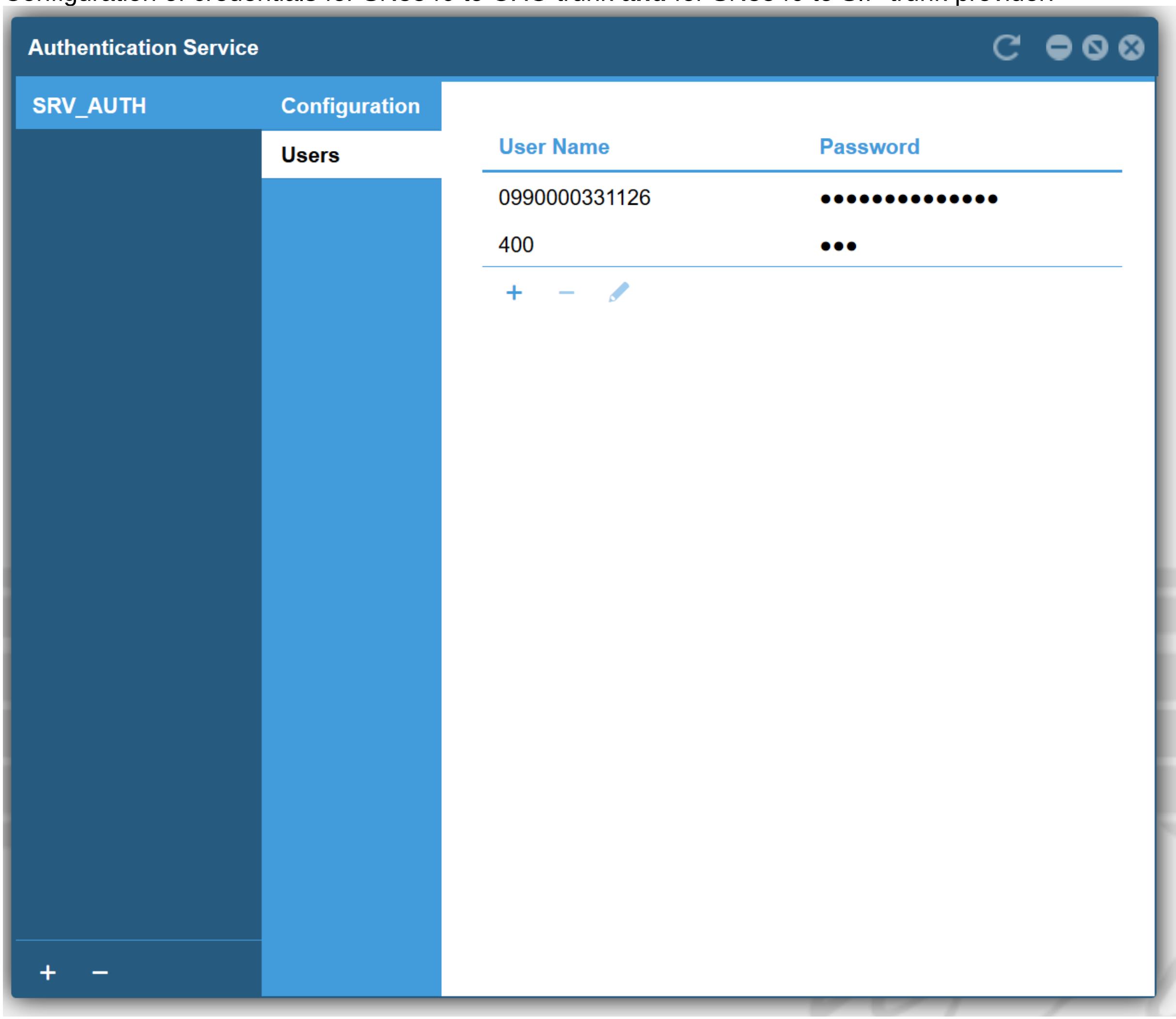
11 Appendix B: Configuration requirements of the AAPP member's application

11.1 PATTON eSBC configuration:

The following configuration has been applied to the SN5540 eSBC.

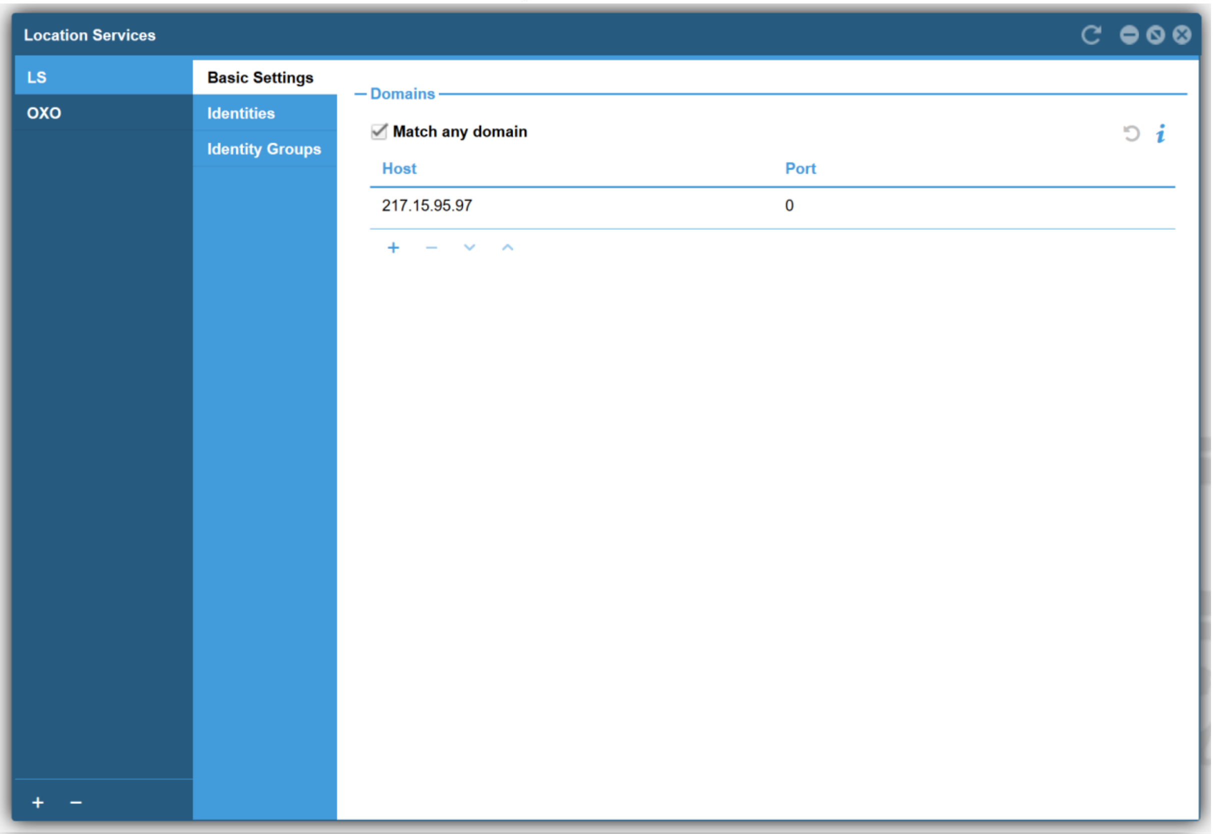
<u>Authentification service:</u>

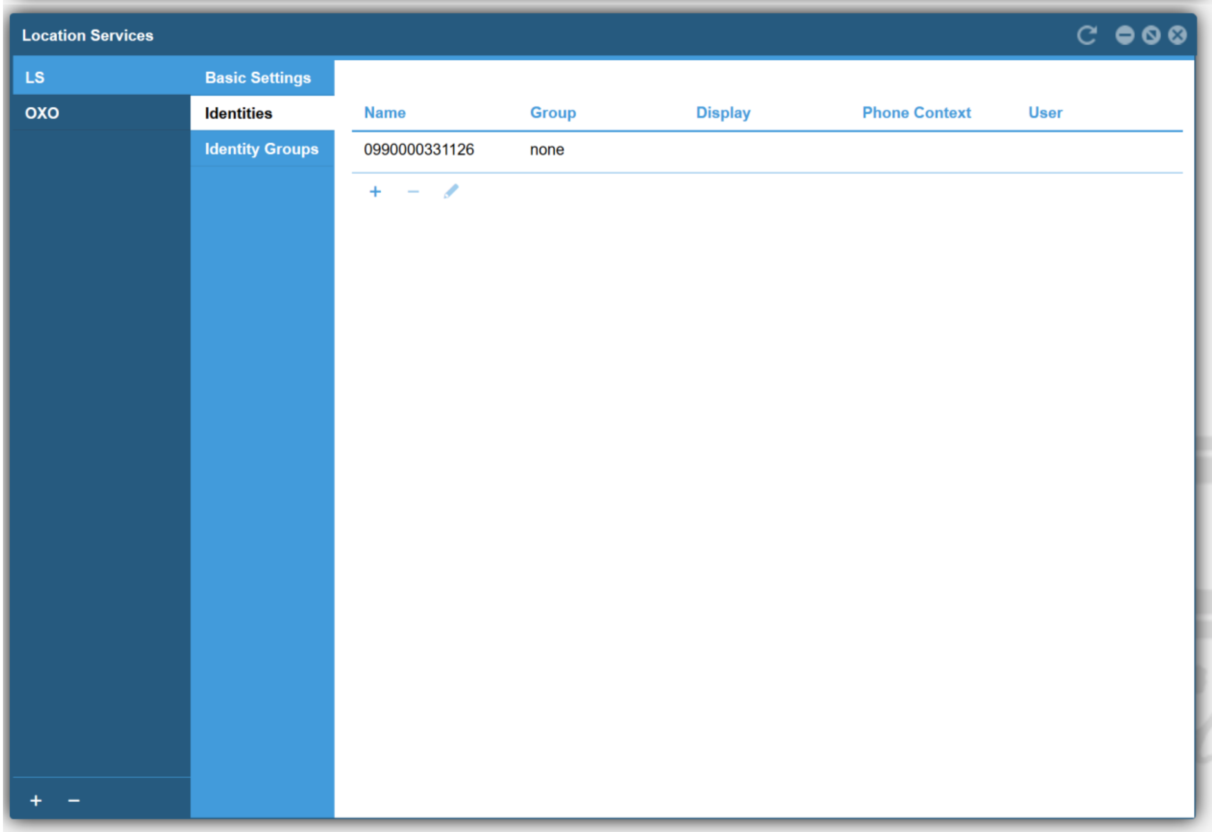
Configuration of credentials for SN5540 to OXO trunk **and** for SN5540 to SIP trunk provider:



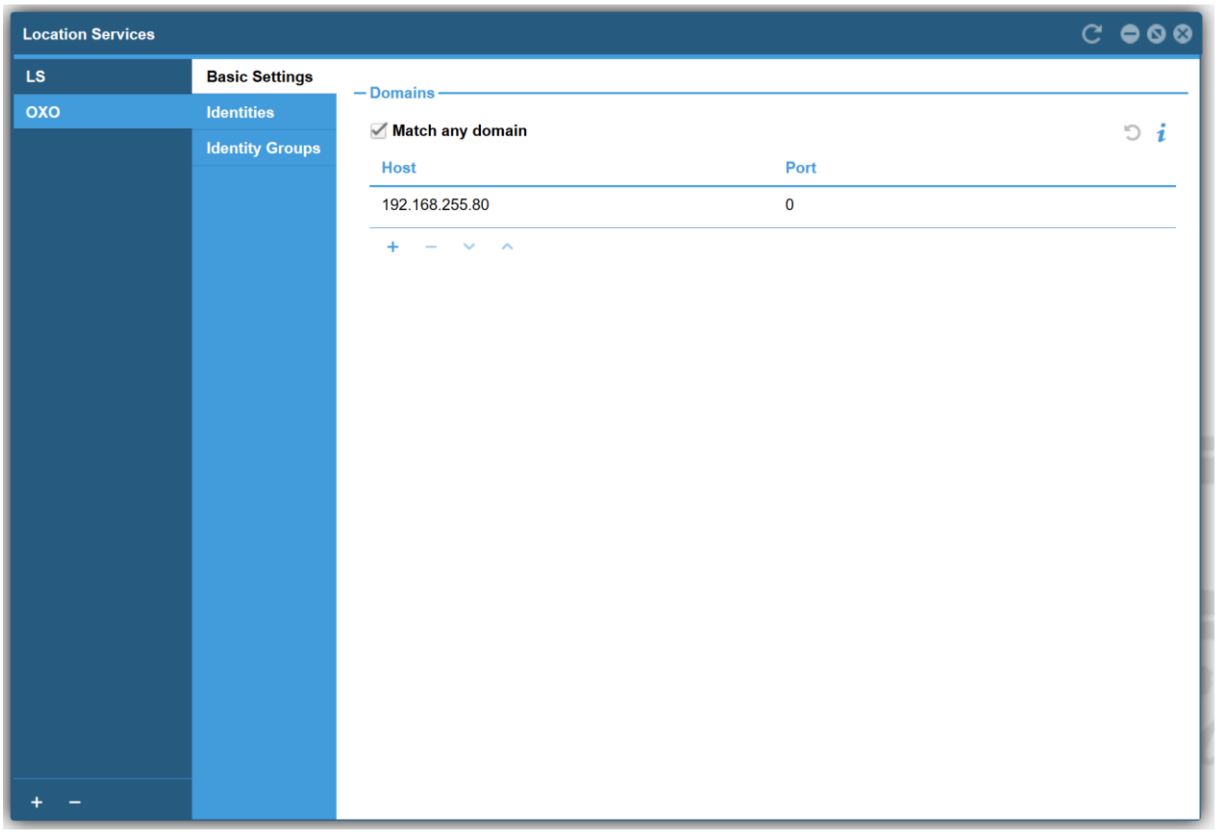
<u>Location service:</u>

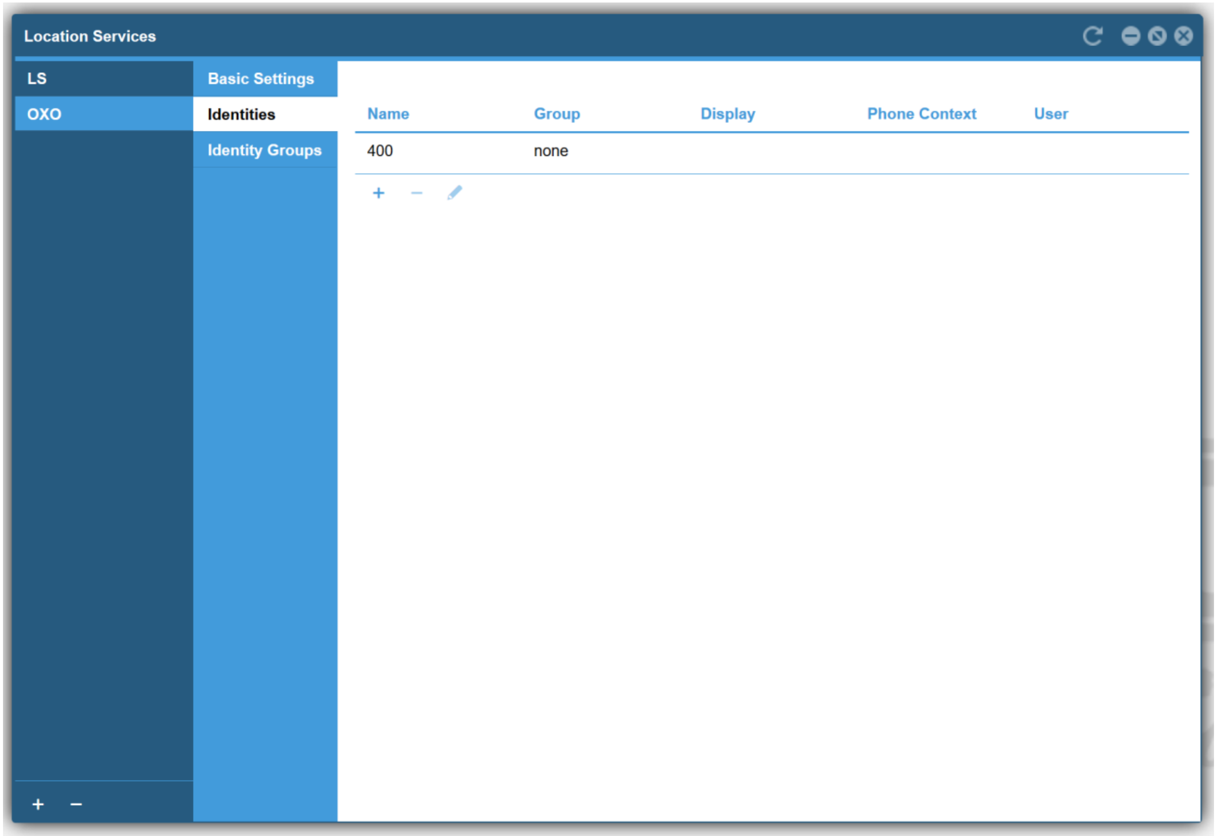
Location Service For the SIP provider, and selection of the identity:


Location Services
LS
OXO
Basic Settings
Identities
Identity Groups
Domains
Match any domain
Host
Port
217.15.95.97
0


Location Services
LS
OXO
Basic Settings
Identities
Identity Groups
Name
Group
Display
Phone Context
User
0990000331126
none

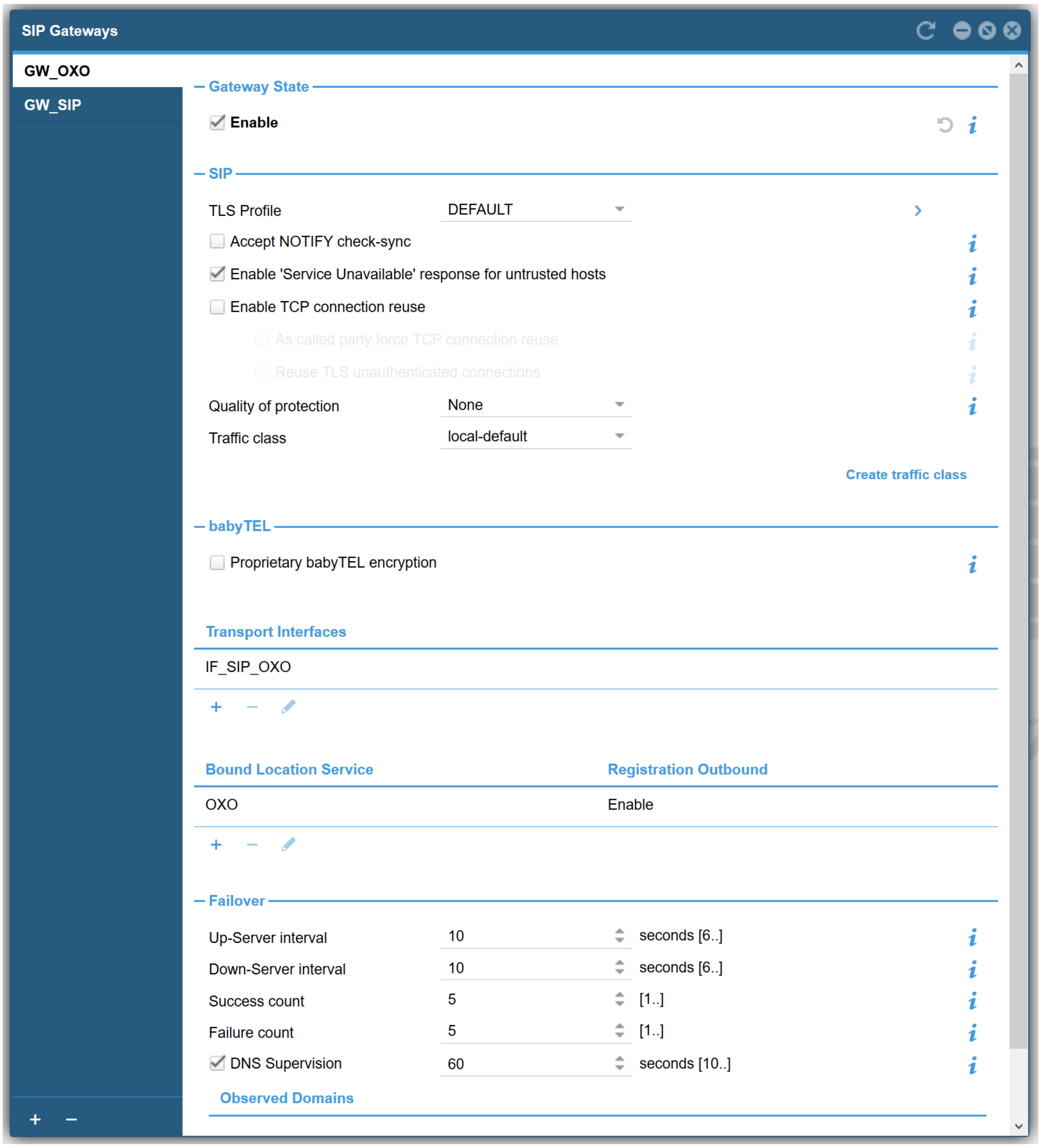
Location Service For the OXO, and selection of the identity:

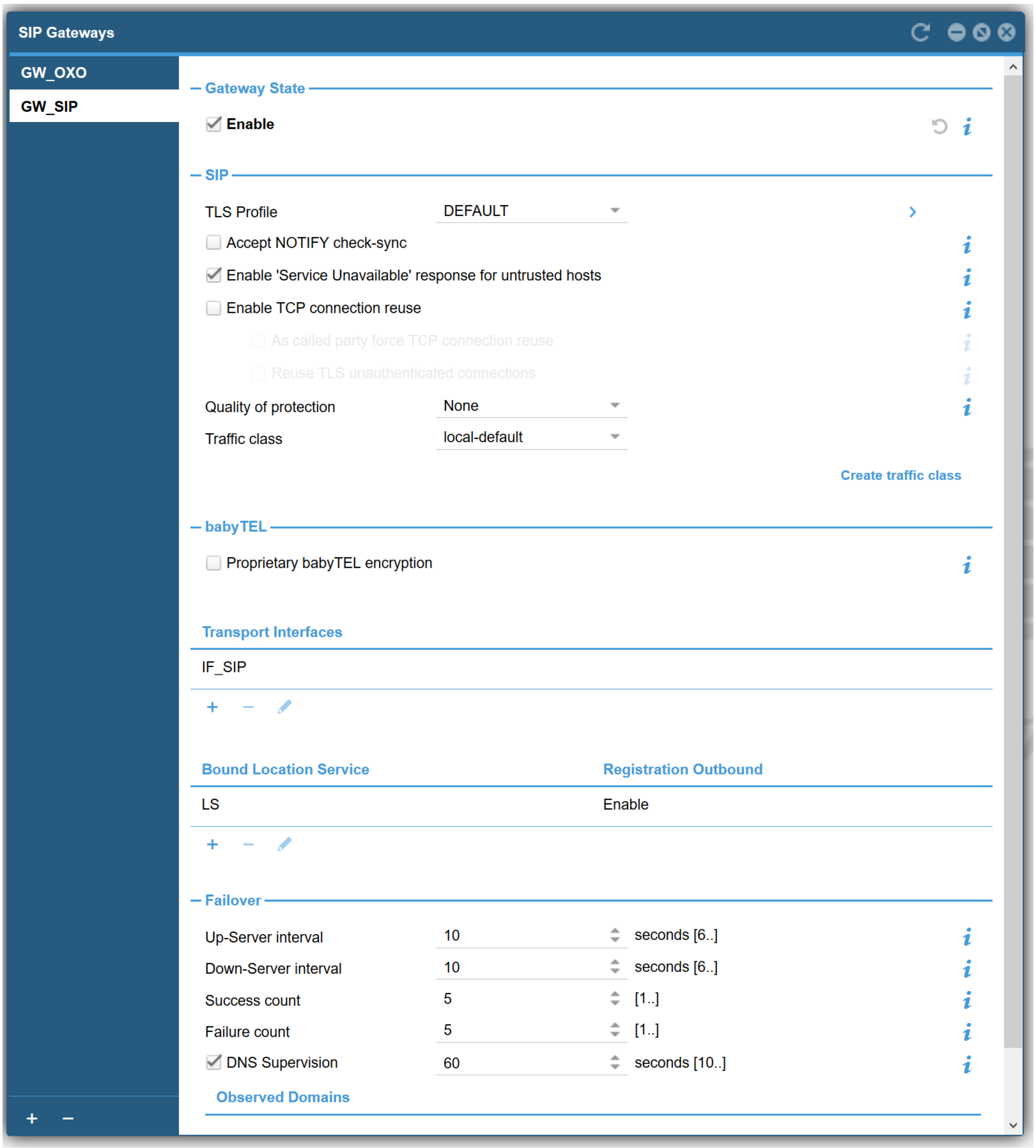




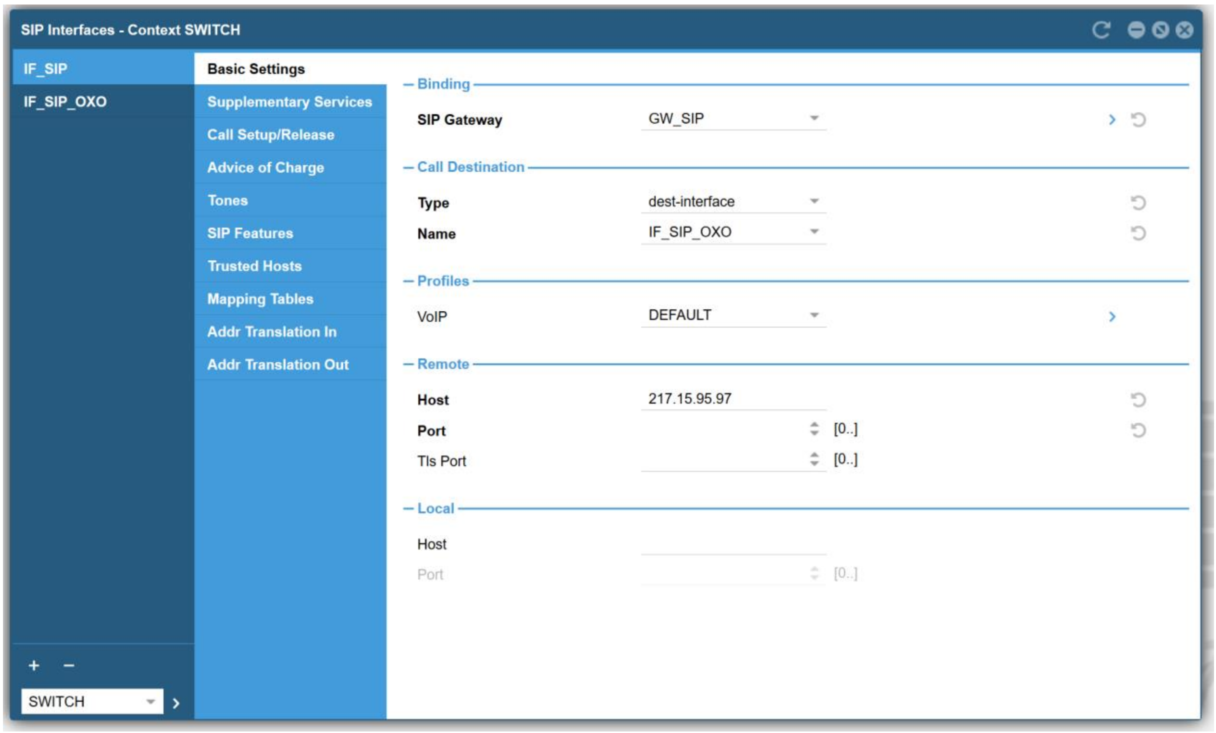
SIP Gateways:

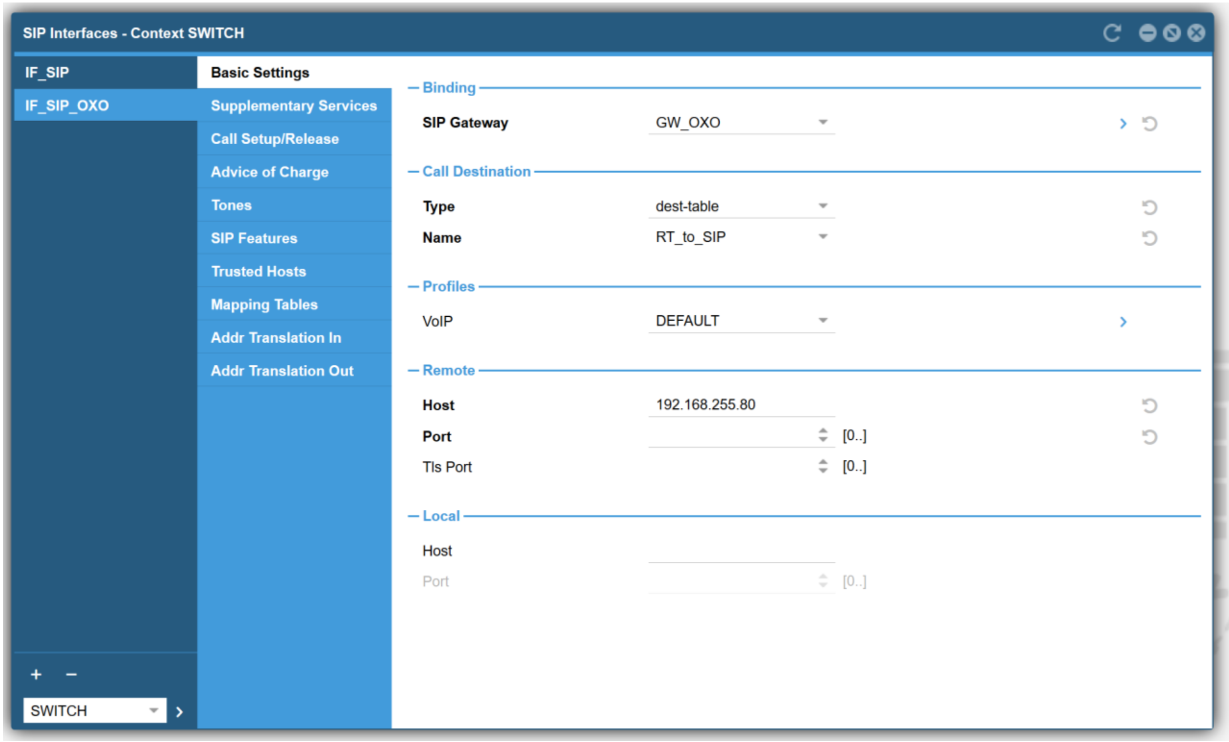
Configuration of Gateways and link to “the transport interfaces”



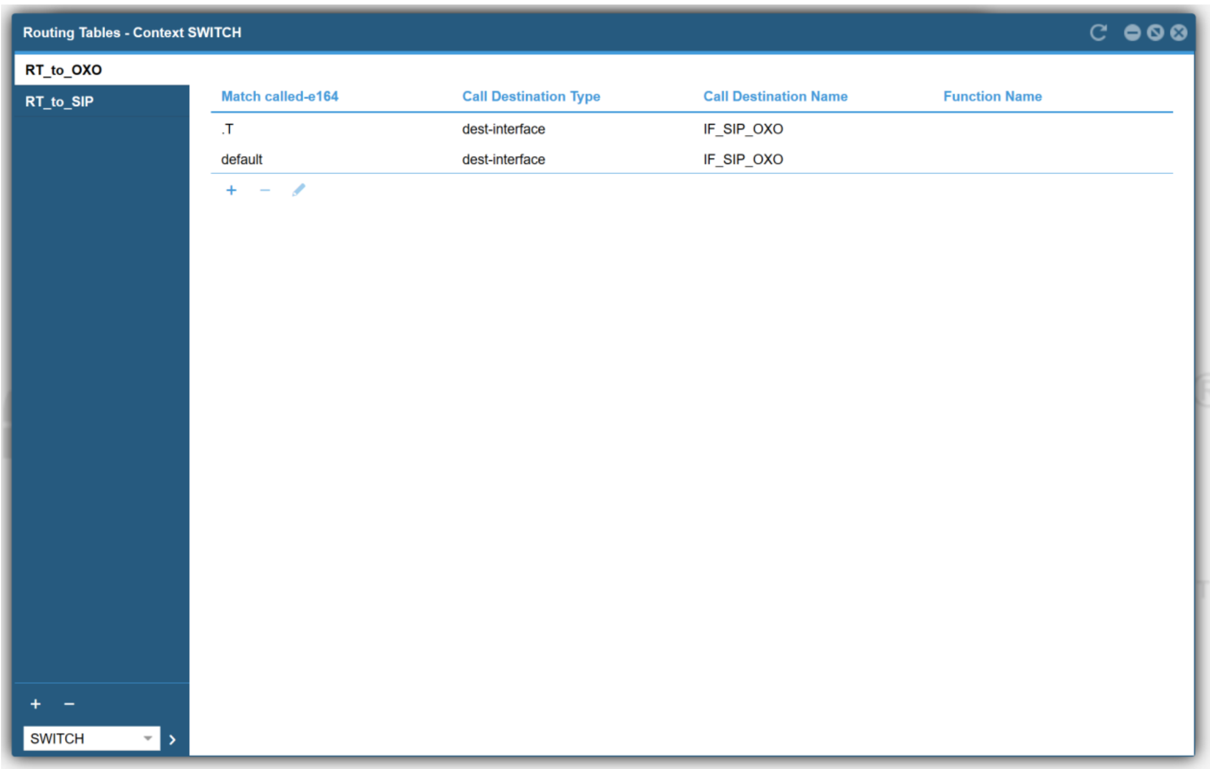

SIP Gateways
GW_OXO
GW_SIP
Gateway State
Enable
SIP
TLS Profile
DEFAULT
Accept NOTIFY check-sync
Enable 'Service Unavailable' response for untrusted hosts
Enable TCP connection reuse
As called party force TCP connection reuse
Reuse TLS unauthenticated connections
Quality of protection
None
Traffic class
local-default
Create traffic class
babyTEL
Proprietary babyTEL encryption
Transport Interfaces
IF_SIP
Bound Location Service
Registration Outbound
LS
Enable
Failover
Up-Server interval
10
seconds [6..]
Down-Server interval
10
seconds [6..]
Success count
5
[1..]
Failure count
5
[1..]
DNS Supervision
60
seconds [10..]
Observed Domains

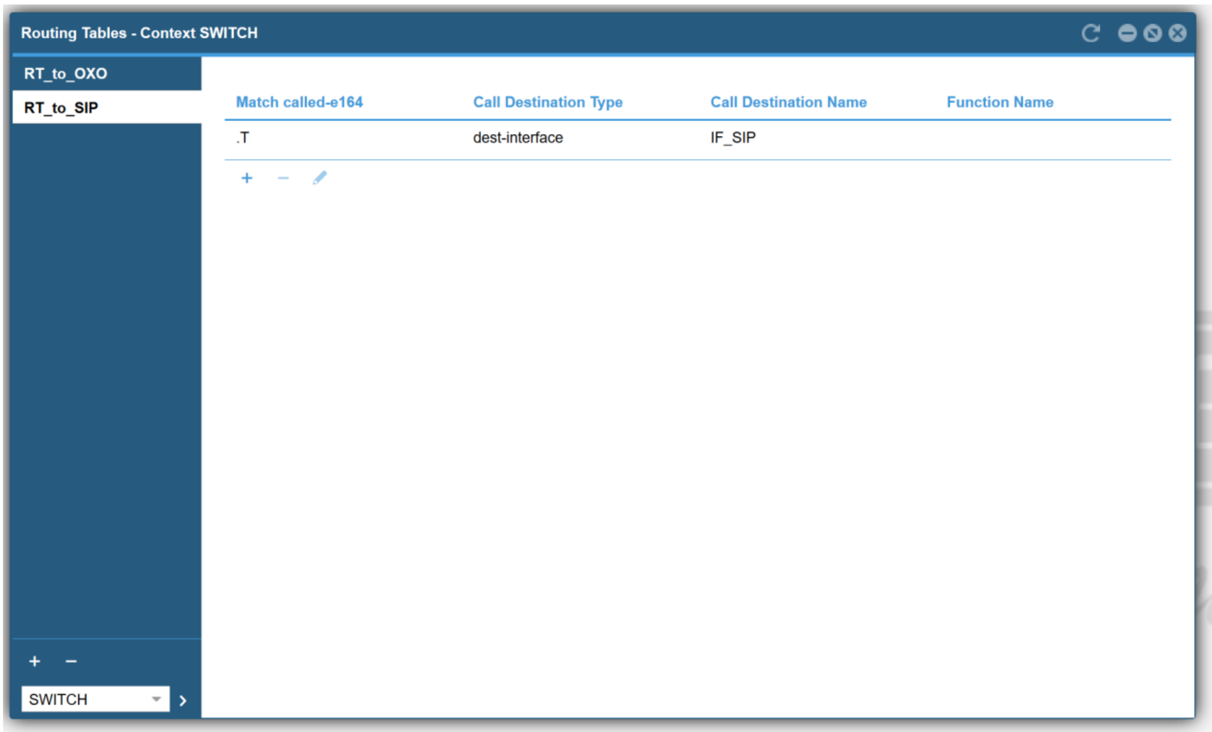
SIP Interfaces:





Routing tables :

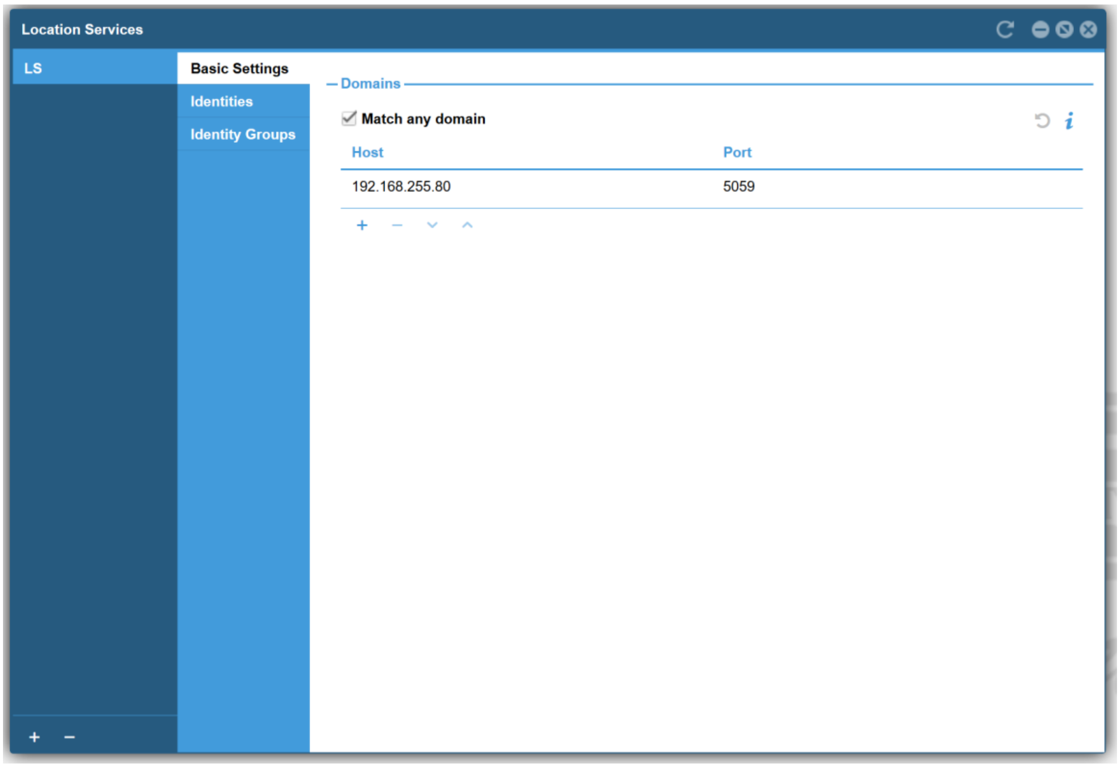


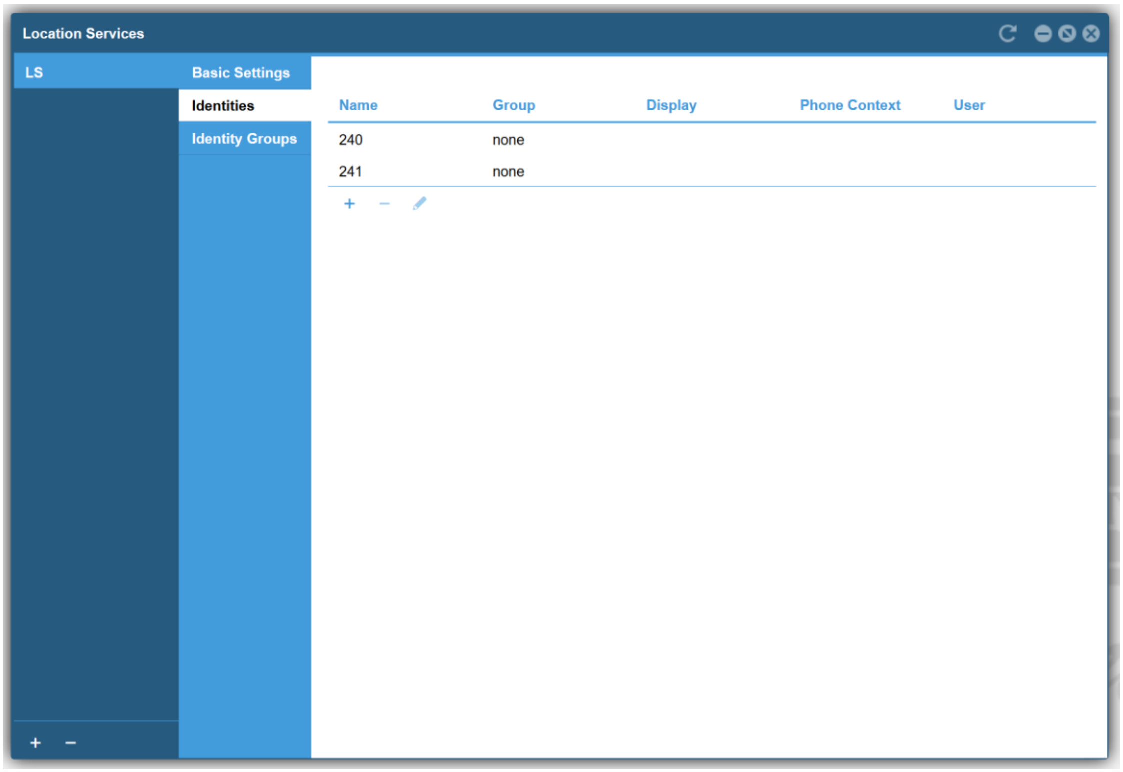


11.2 PATTON FXS analog gateway configuration:

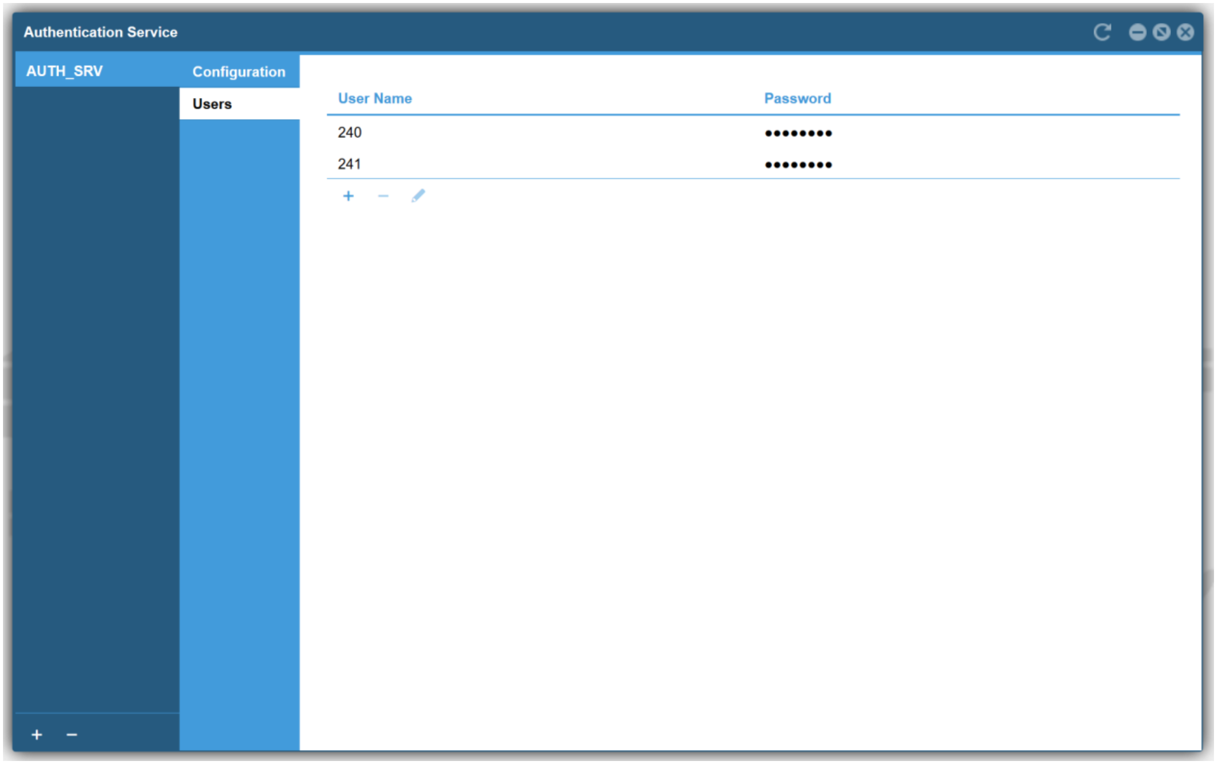
The following configuration has been applied to the SN4141 gateway.

Location Services:

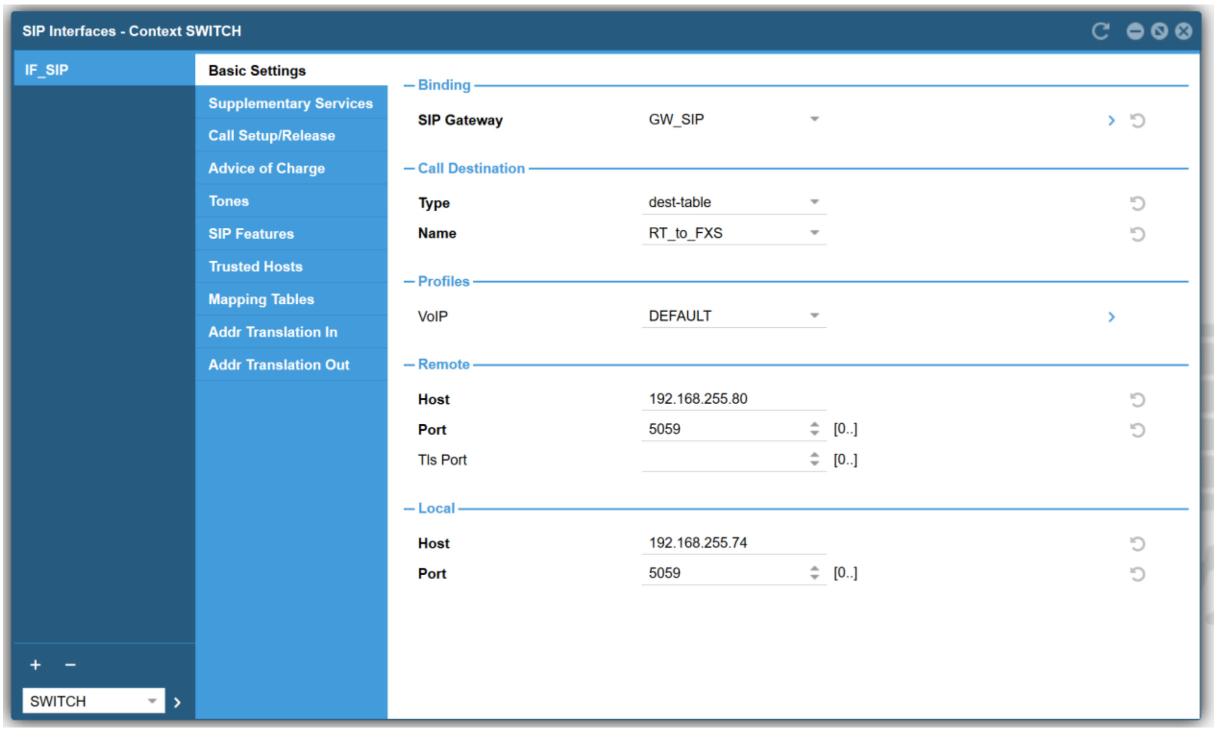




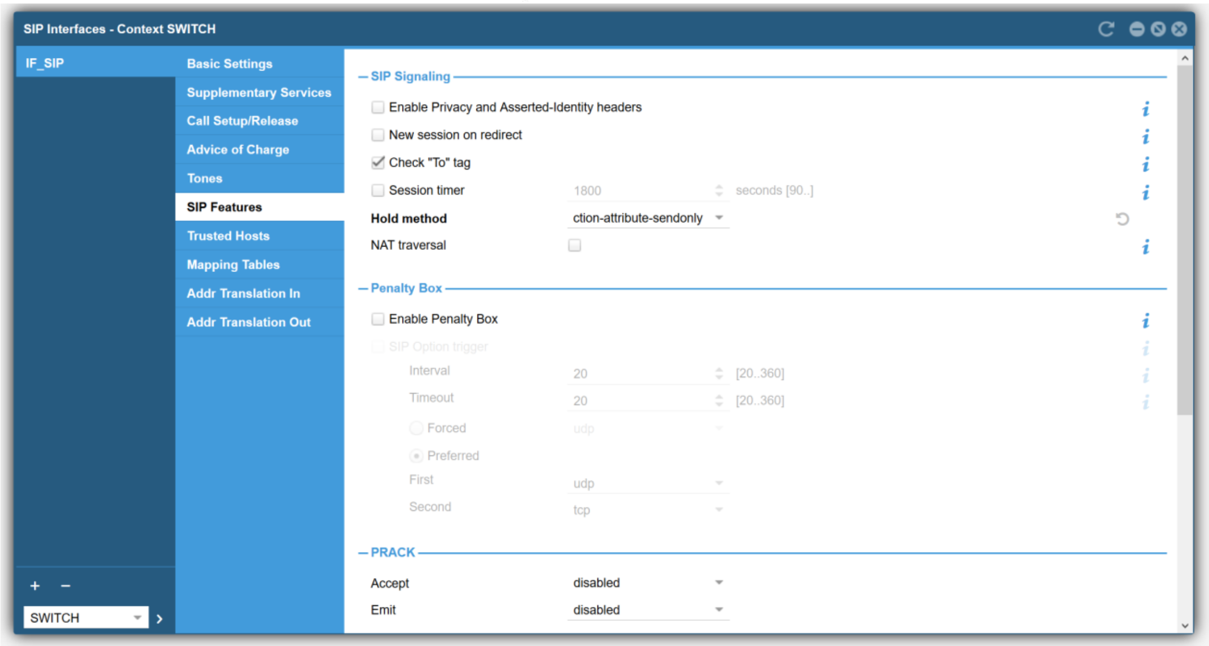
<u>Authentication service:</u>

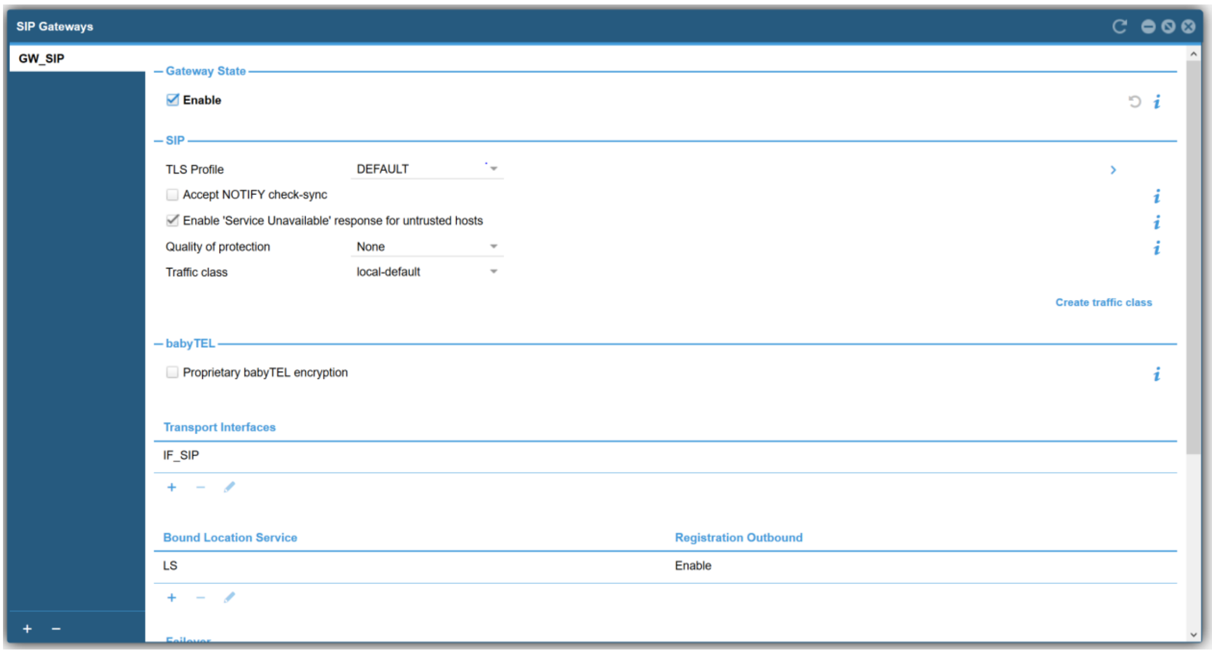


<u>SIP Interfaces:</u>

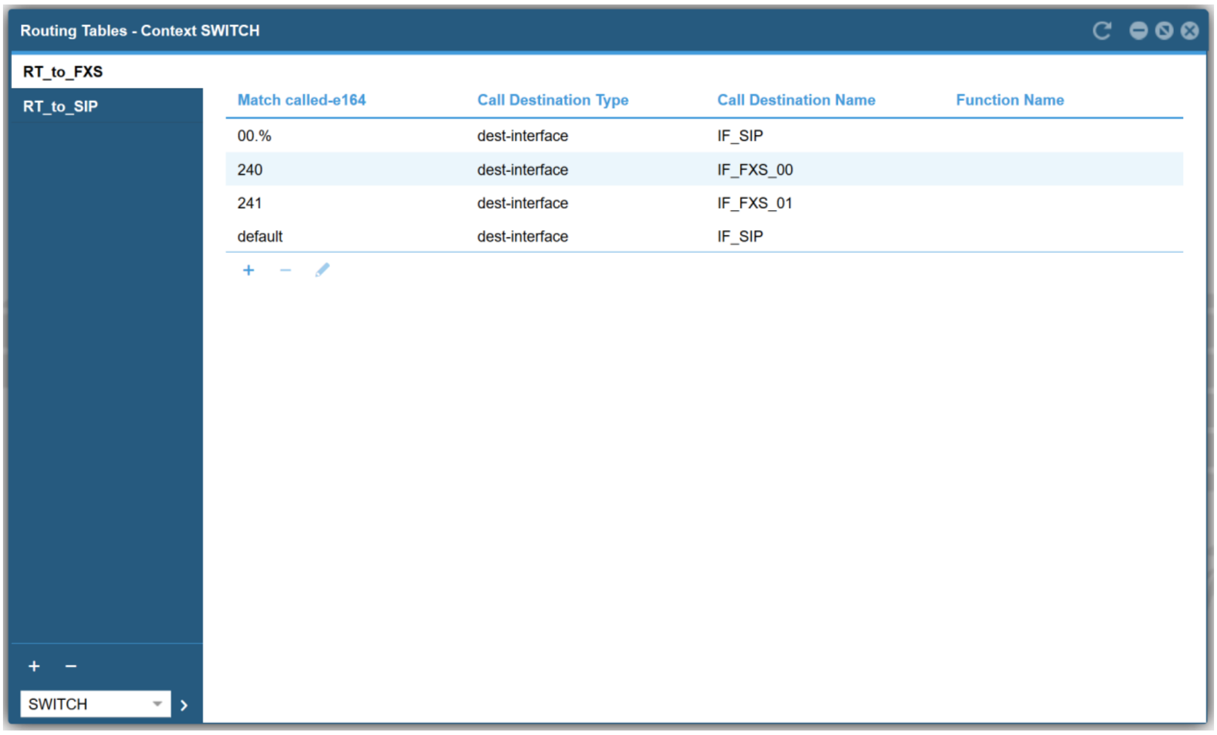


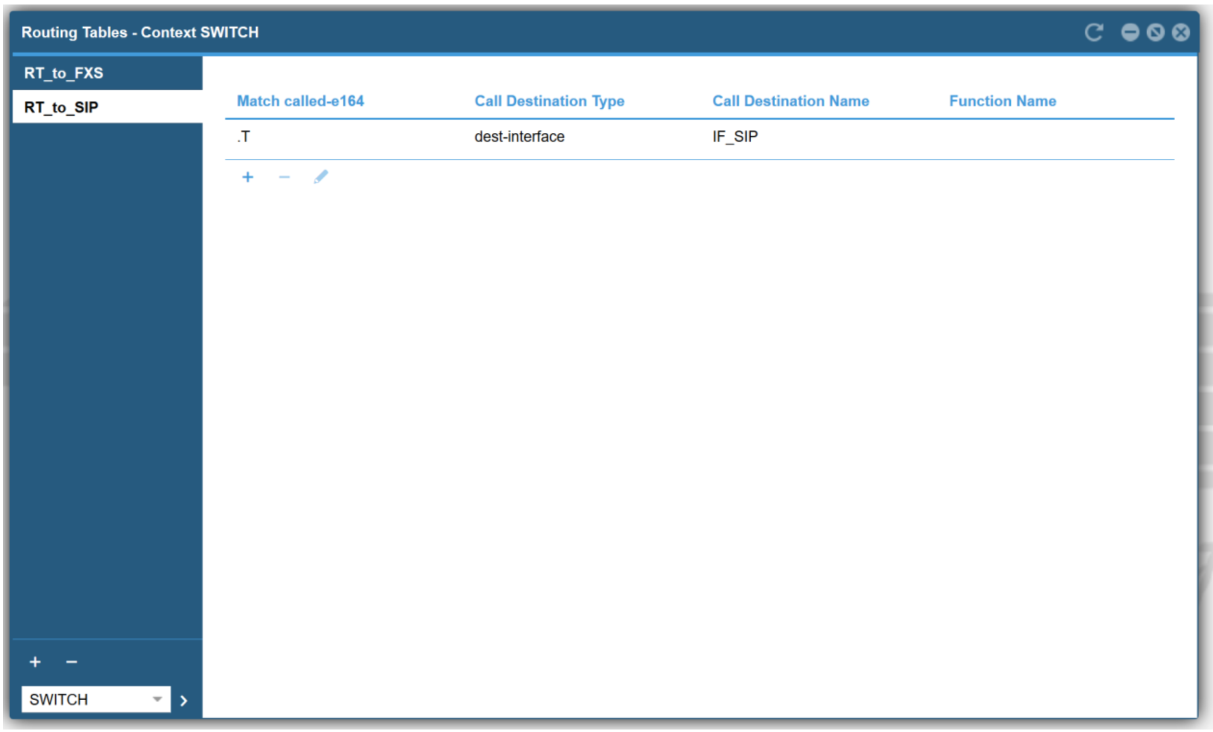
In **Sip Features**, **Hold method** select “direction-attribute-sendonly”


SIP Interfaces - Context SWITCH
IF_SIP
Basic Settings
Supplementary Services
Call Setup/Release
Advice of Charge
Tones
SIP Features
Trusted Hosts
Mapping Tables
Addr Translation In
Addr Translation Out
SWITCH
SIP Signaling
Enable Privacy and Asserted-Identity headers
New session on redirect
Check "To" tag
Session timer
1800
seconds [90..]
Hold method
ction-attribute-sendonly
NAT traversal
Penalty Box
Enable Penalty Box
SIP Option trigger
Interval
20
[20..360]
Timeout
20
[20..360]
Forced
udp
Preferred
First
udp
Second
tcp
PRACK
Accept
disabled
Emit
disabled

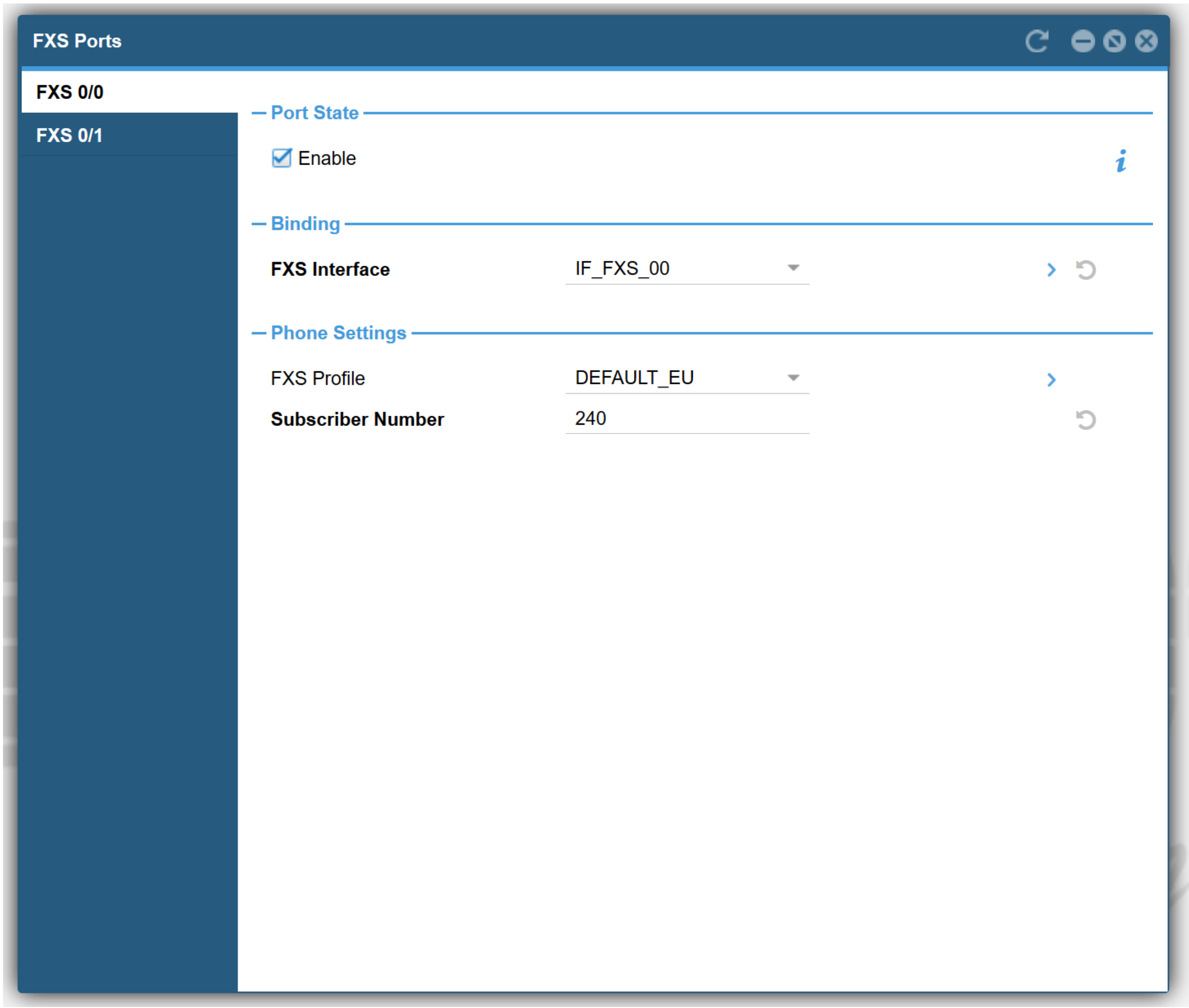

SIP Gateways
GW_SIP
Gateway State
Enable
SIP
TLS Profile
DEFAULT
Accept NOTIFY check-sync
Enable 'Service Unavailable' response for untrusted hosts
Quality of protection
None
Traffic class
local-default
Create traffic class
babyTEL
Proprietary babyTEL encryption
Transport Interfaces
IF_SIP
Bound Location Service
Registration Outbound
LS
Enable

Routing tables :

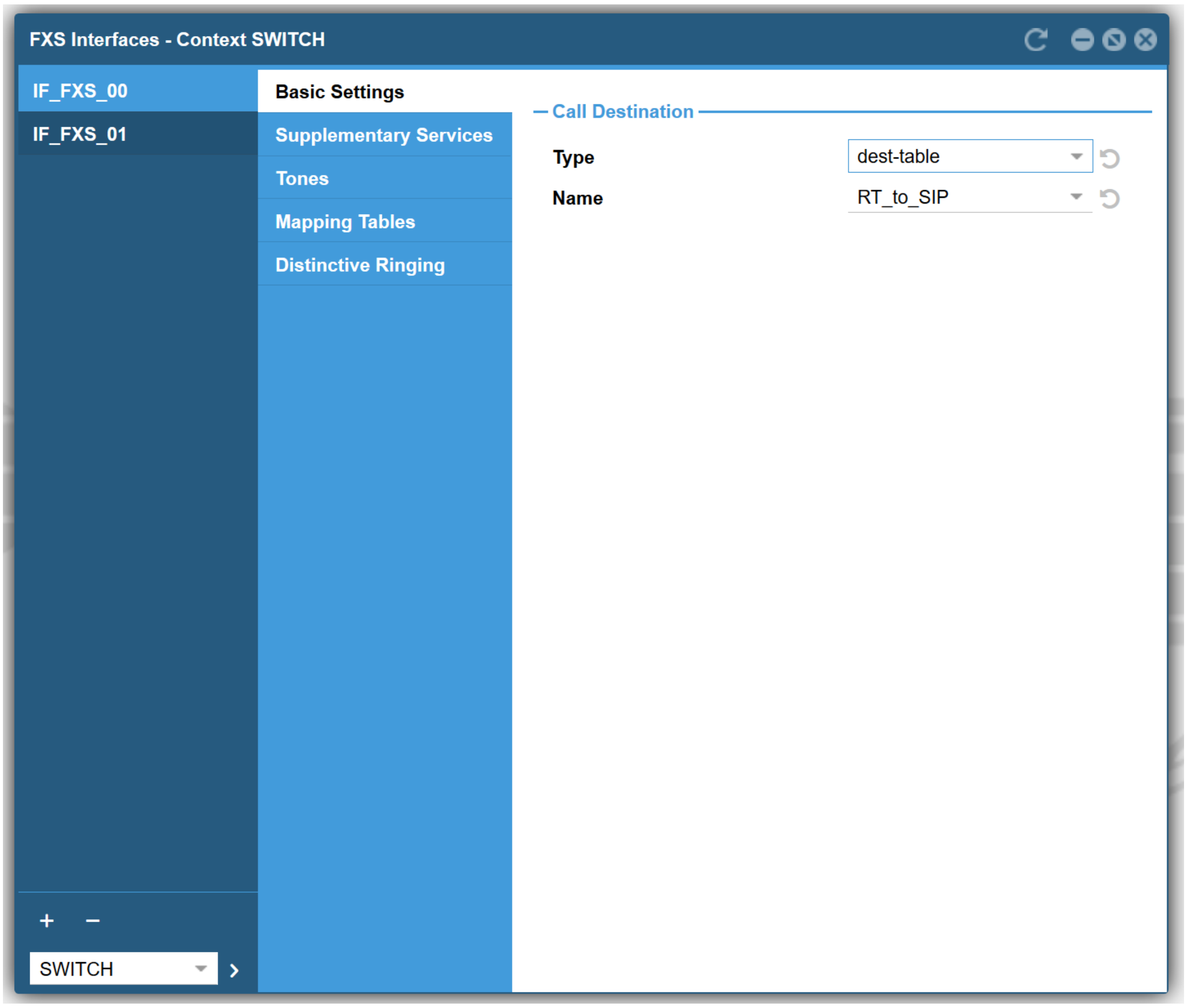




FXS Ports:


FXS Ports
FXS 0/0
FXS 0/1
Port State
Enable
Binding
FXS Interface
IF_FXS_00
Phone Settings
FXS Profile
DEFAULT_EU
Subscriber Number
240

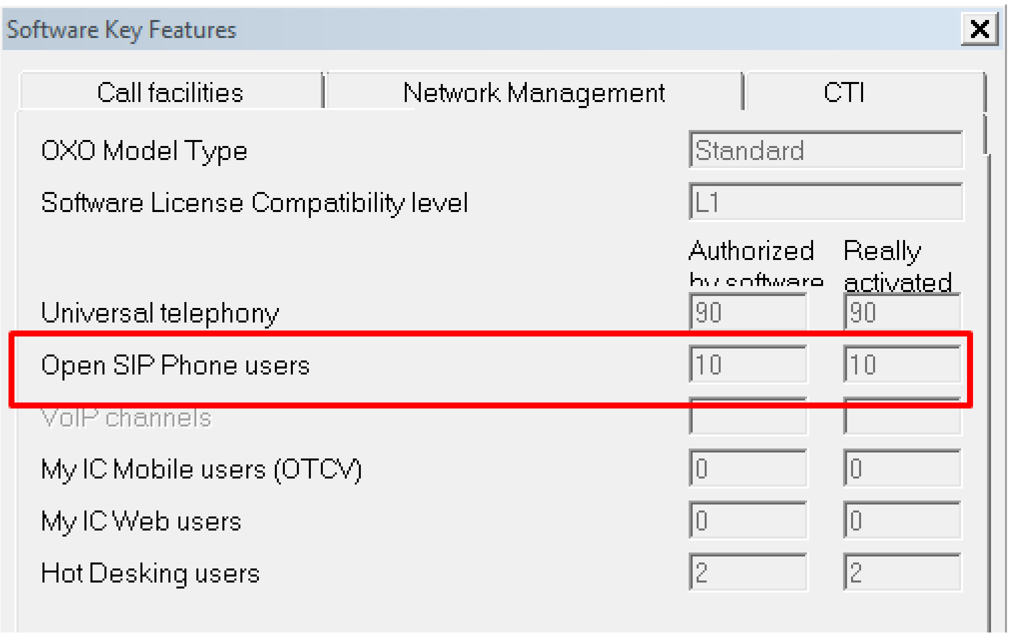
FXS Interfaces:



12 Appendix C: Alcatel-Lucent Enterprise Communication Platform: OXO Connect Evolution configuration requirements

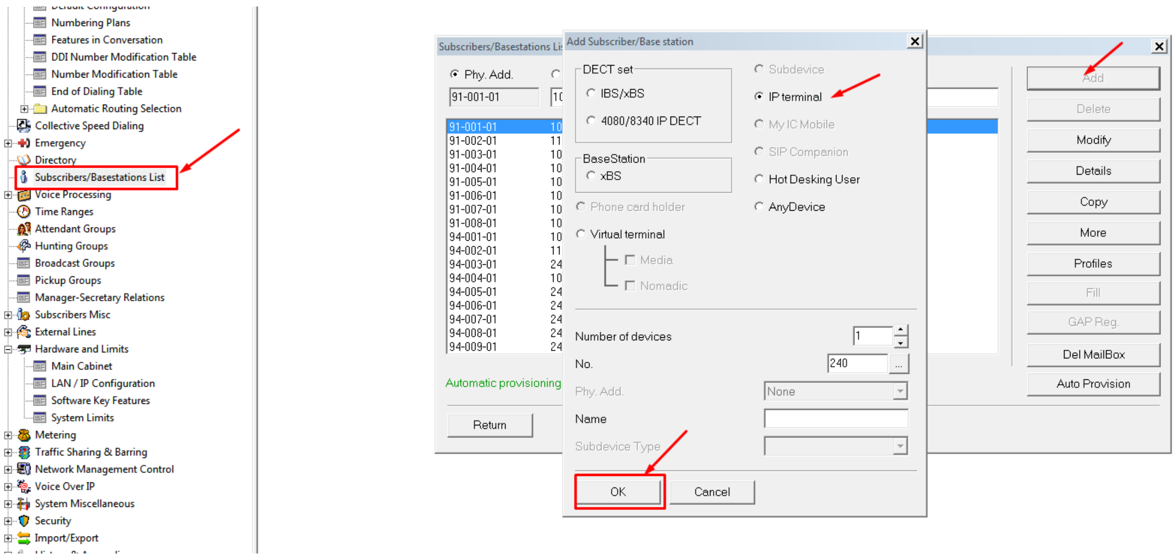
Licences:

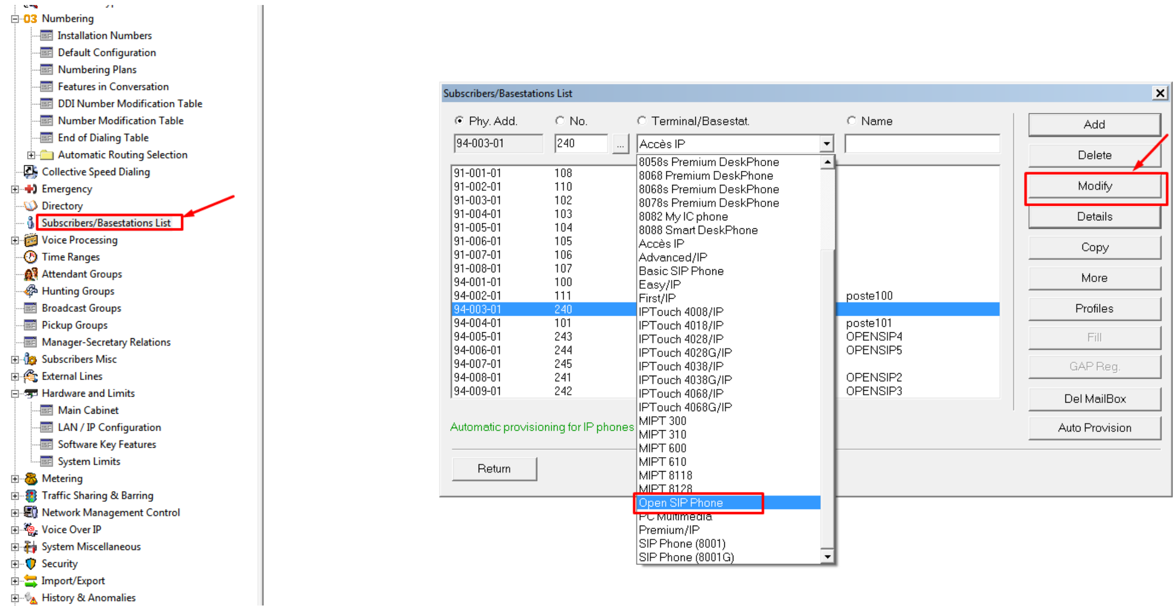
- Open SIP licence is required for Pattaon smartnode analogue interface integration

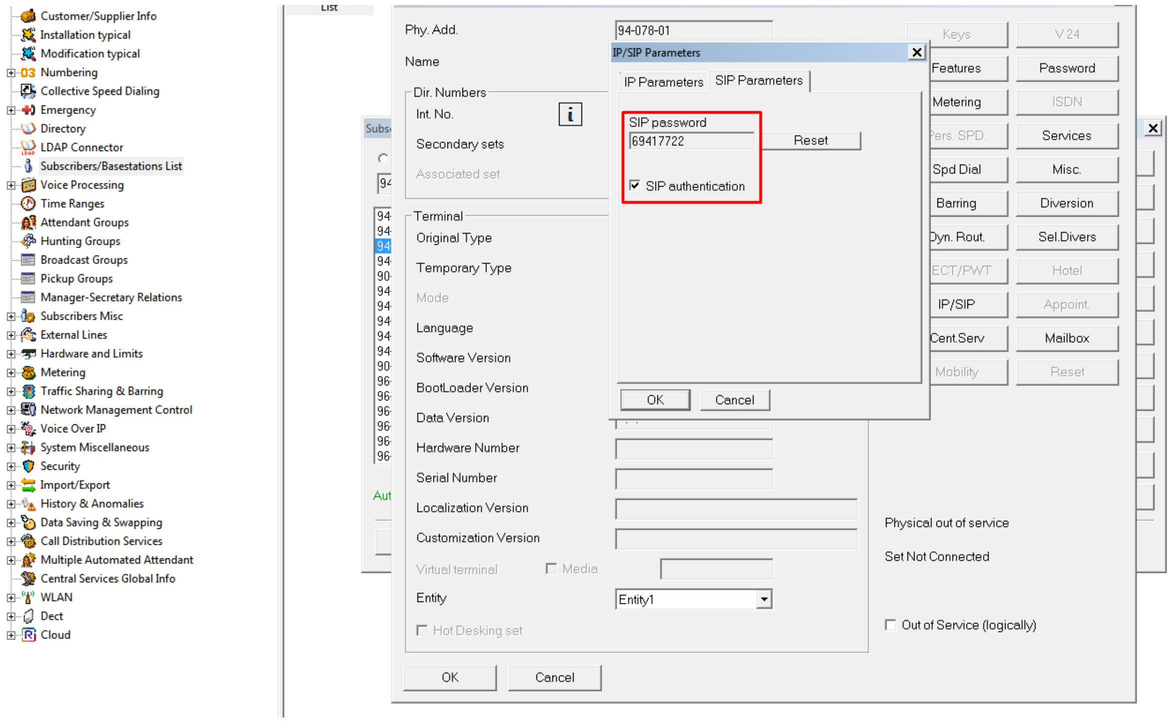


Sip set configuration

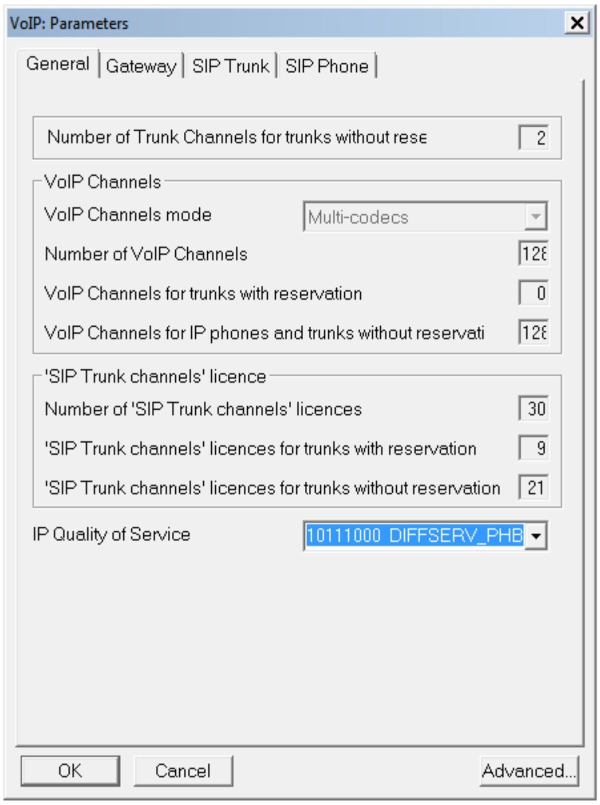
1- Configure Sip user as Open SIP phone

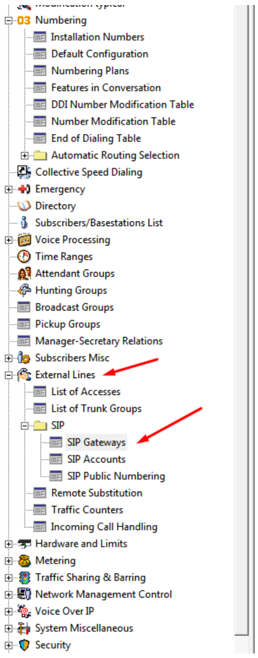


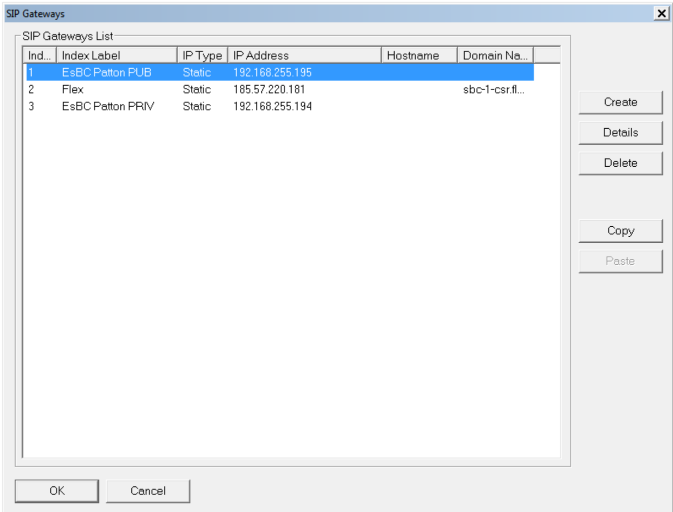

03 Numbering
Installation Numbers
Default Configuration
Numbering Plans
Features in Conversation
DDI Number Modification Table
Number Modification Table
End of Dialing Table
Automatic Routing Selection
Collective Speed Dialing
Emergency
Directory
Subscribers/Basestations List
Voice Processing
Time Ranges
Attendant Groups
Hunting Groups
Broadcast Groups
Pickup Groups
Manager-Secretary Relations
Subscribers Misc
External Lines
Hardware and Limits
Main Cabinet
LAN / IP Configuration
Software Key Features
System Limits
Metering
Traffic Sharing & Barring
Network Management Control
Voice Over IP
System Miscellaneous
Security
Import/Export
History & Anomalies
Subscribers/Basestations List
Phy. Add.
No.
Terminal/Basestat.
Name
94-003-01
240
Accès IP
8058s Premium DeskPhone
8068 Premium DeskPhone
8068s Premium DeskPhone
8078s Premium DeskPhone
8082 My IC phone
8088 Smart DeskPhone
Accès IP
Advanced/IP
Basic SIP Phone
Easy/IP
First/IP
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)
poste100
poste101
OPENSIP4
OPENSIP5
OPENSIP2
OPENSIP3
Automatic provisioning for IP phones
Return
Add
Delete
Modify
Details
Copy
More
Profiles
Fill
GAP Reg.
Del MailBox
Auto Provision


Customer/Supplier Info
Installation typical
Modification typical
Numbering
Collective Speed Dialing
Emergency
Directory
LDAP Connector
Subscribers/Basestations List
Voice Processing
Time Ranges
Attendant Groups
Hunting Groups
Broadcast Groups
Pickup Groups
Manager-Secretary Relations
Subscribers Misc
External Lines
Hardware and Limits
Metering
Traffic Sharing & Barring
Network Management Control
Voice Over IP
System Miscellaneous
Security
Import/Export
History & Anomalies
Data Saving & Swapping
Call Distribution Services
Multiple Automated Attendant
Central Services Global Info
WLAN
Dect
Cloud
Phy. Add.
94-078-01
Name
Dir. Numbers
Int. No.
Secondary sets
Associated set
Terminal
Original Type
Temporary Type
Mode
Language
Software Version
BootLoader Version
Data Version
Hardware Number
Serial Number
Localization Version
Customization Version
Virtual terminal
Media
Entity
Entity1
Hot Desking set
OK
Cancel
IP/SIP Parameters
IP Parameters
SIP Parameters
SIP password
69417722
Reset
SIP authentication
OK
Cancel
Keys
V 24
Features
Password
Metering
ISDN
Pers. SPD.
Services
Spd Dial
Misc.
Barring
Diversion
Dyn. Rout.
Sel.Divers
ECT/PWT
Hotel
IP/SIP
Appoint.
Cent.Serv
Mailbox
Mobility
Reset
Physical out of service
Set Not Connected
Out of Service (logically)

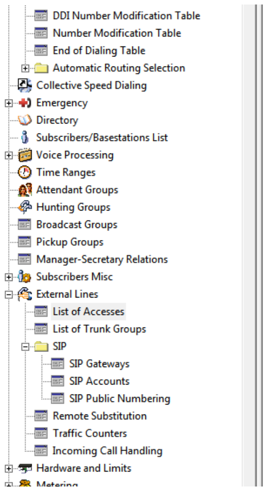
2- Check the VoIP channel availability

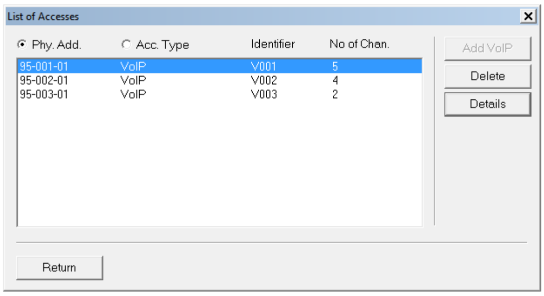




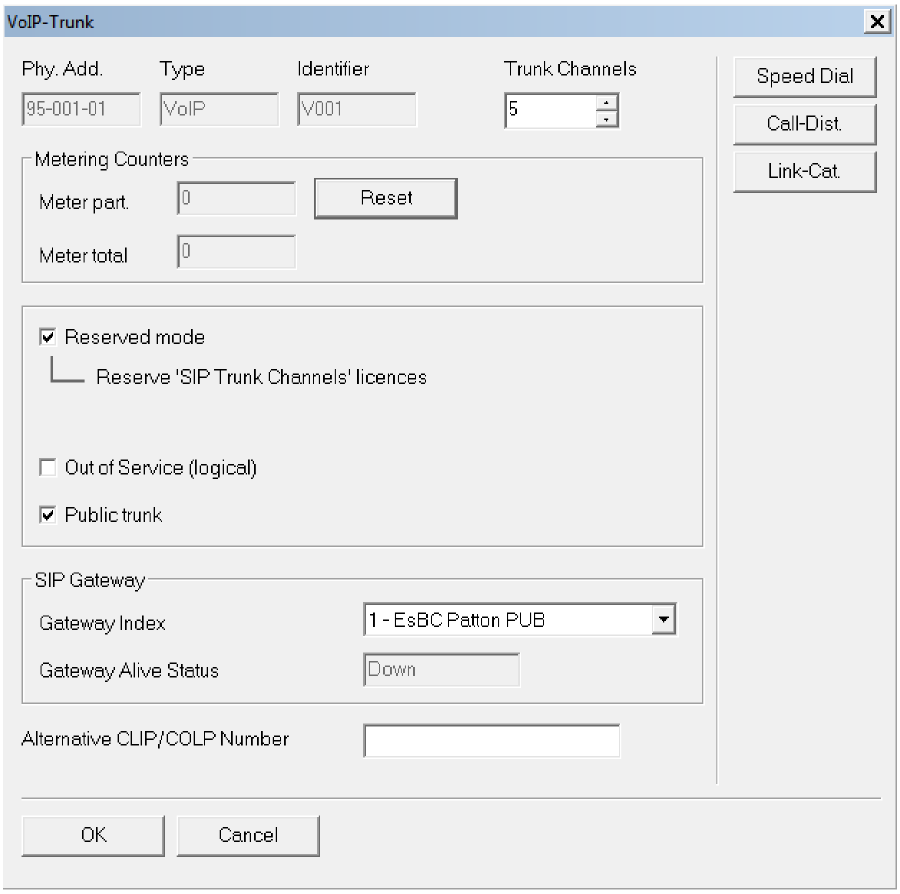


3- Add the VOIP channel to the list of external lines





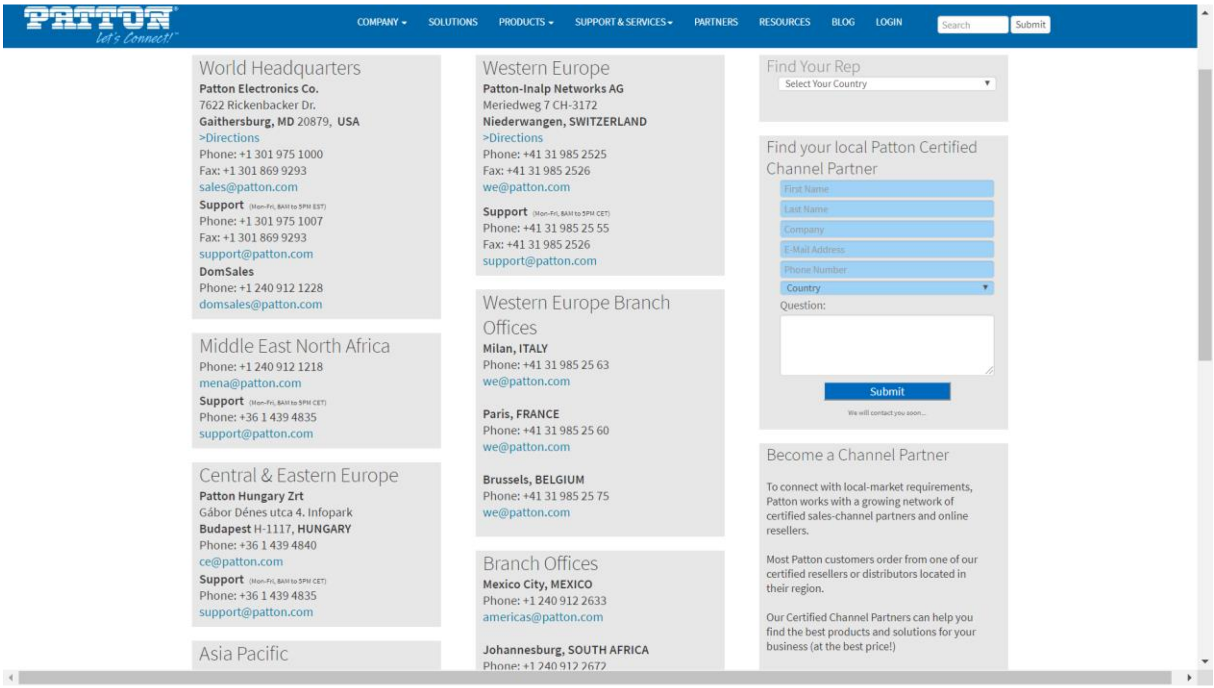
4- Select the gateway in the list of access



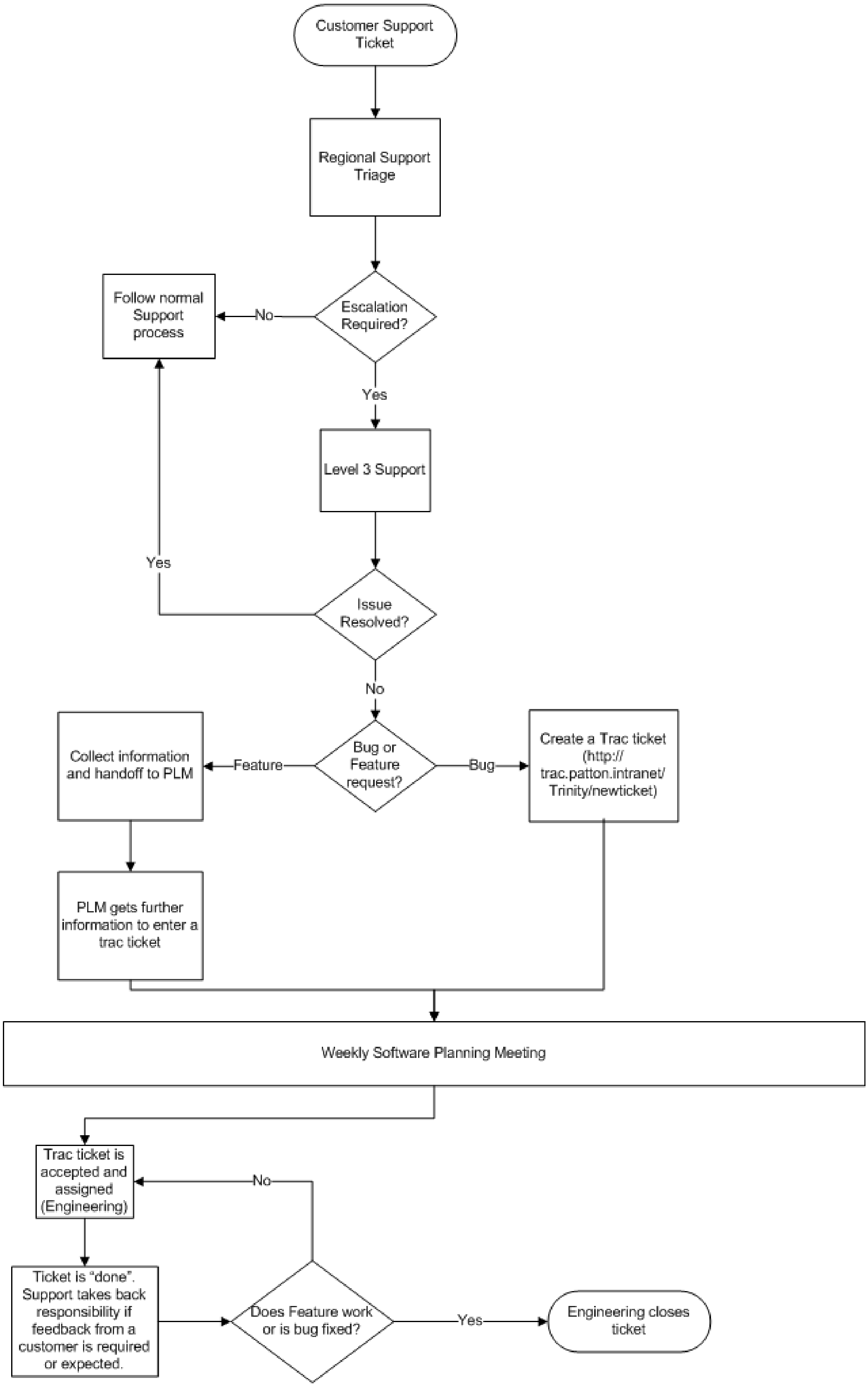
13 Appendix D: AAPP member's escalation process

PATTON customers open tickets through Patton's web portal at: www.patton.com/support/#portal.

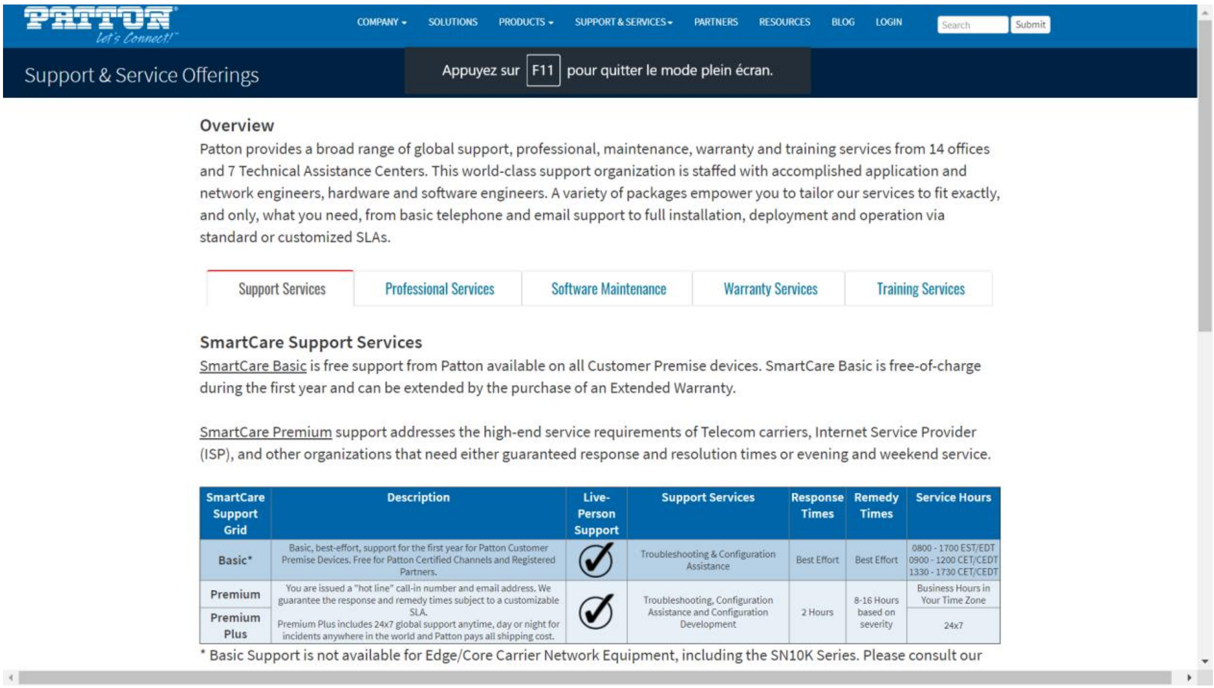
Or by sending an email to support@patton.com or by contacting their assigned support centre available at www.patton.com/company/contact-us.asp.



Escalation process:



The following escalation levels are applied to customer calls in line with Patton published support level agreements available at www.patton.com/support/support.asp.



Overview

Patton provides a broad range of global support, professional, maintenance, warranty and training services from 14 offices and 7 Technical Assistance Centers. This world-class support organization is staffed with accomplished application and network engineers, hardware and software engineers. A variety of packages empower you to tailor our services to fit exactly, and only, what you need, from basic telephone and email support to full installation, deployment and operation via standard or customized SLAs.

Support Services | Professional Services | Software Maintenance | Warranty Services | Training Services

SmartCare Support Services

SmartCare Basic is free support from Patton available on all Customer Premise devices. SmartCare Basic is free-of-charge during the first year and can be extended by the purchase of an Extended Warranty.

SmartCare Premium support addresses the high-end service requirements of Telecom carriers, Internet Service Provider (ISP), and other organizations that need either guaranteed response and resolution times or evening and weekend service.

SmartCare Support Grid	Description	Live-Person Support	Support Services	Response Times	Remedy Times	Service Hours
Basic*	Basic, best-effort, support for the first year for Patton Customer Premise Devices. Free for Patton Certified Channels and Registered Partners.	✓	Troubleshooting & Configuration Assistance	Best Effort	Best Effort	0800 - 1700 EST/EDT 0900 - 1200 CET/CEDT 1330 - 1730 CET/CEDT
Premium	You are issued a "hot line" call-in number and email address. We guarantee the response and remedy times subject to a customizable SLA.	✓	Troubleshooting, Configuration Assistance and Configuration Development	2 Hours	8-16 Hours based on severity	Business Hours in Your Time Zone
Premium Plus	Premium Plus includes 24x7 global support anytime, day or night for incidents anywhere in the world and Patton pays all shipping cost.					24x7

* Basic Support is not available for Edge/Core Carrier Network Equipment, including the SN10K Series. Please consult our

Patton Contacts

1	TAC Email: support@patton.com Europe TAC Phone: +41 31 985 25 55
2	Technical Support for Alcatel-Lucent Enterprise account: Brice Imbault Email: Brice.imbault@patton-inalp.com Phone: +41 31 985 25 24
3	Customer Support Manager Europe: Miklos Szabo Email: mszabo@patton.com Phone: +36 1 439 4831
4	VP EMEA: Marjan Torkar Email: Marjan.Torkar@patton-inalp.com Phone: +41 31 985 2525

14 Appendix E: AAPP program

14.1 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:

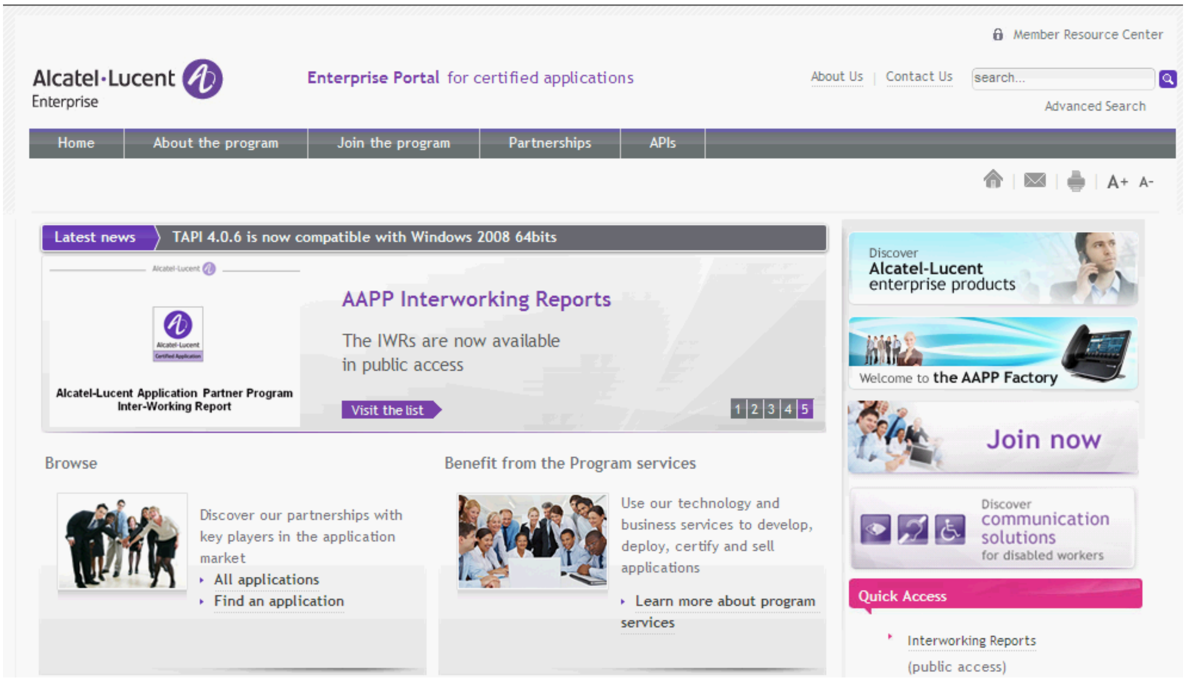
- **Provide easy interfacing for Alcatel-Lucent Enterprise communication products**: Alcatel-Lucent Enterprise's communication products for the enterprise market include infrastructure elements, platforms and software suites. To ensure easy integration, the AAPP provides a full array of standards-based 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



14.2 Enterprise.Alcatel-Lucent.com

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

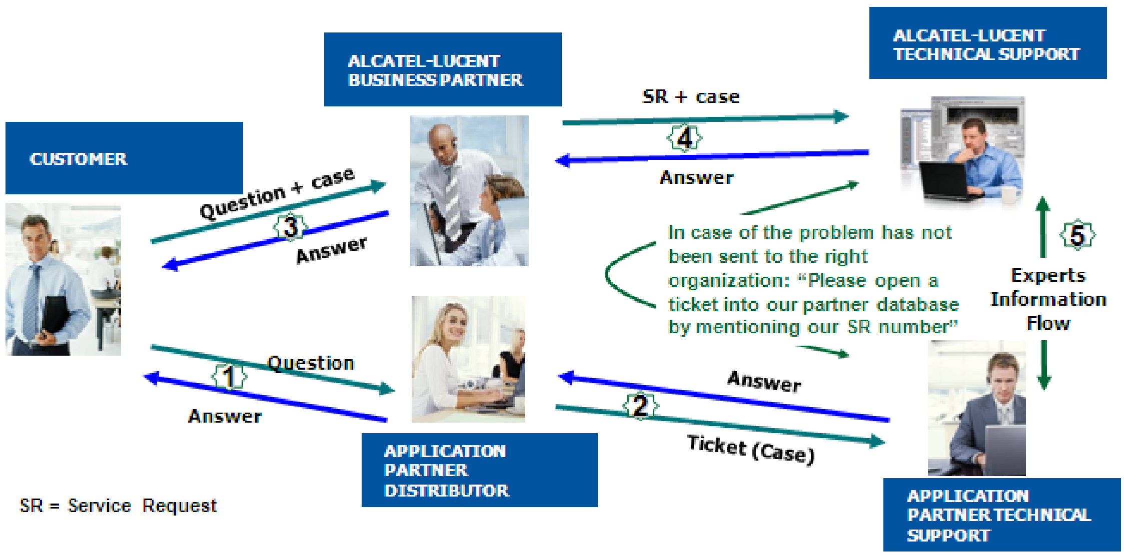
15 Appendix F: AAPP Escalation process

15.1 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

15.2 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.

15.3 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.

15.4 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		
Luxembourg		
Germany	German	
Austria		
Switzerland		
United Kingdom	English	
Italy		
Australia		
Denmark		
Ireland		
Netherlands		
South Africa		
Norway		
Poland		
Sweden		
Czech Republic		
Estonia		
Finland		
Greece		
Slovakia		
Portugal		
Spain	Spanish	

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

END OF DOCUMENT