Alcatel Lucent Application Partner Program
Inter-Working Report

Partner: Ascom
Application type: VoWiFi handset
Application name: Ascom i62
Alcatel-Lucent Platform: OmniPCX Office™

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

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Certification overview

Date of the certification	May 2014

Alcatel-Lucent's representative	Alain Botti
AAPP member representative	Peter Åstrand

Alcatel-Lucent Communication Platform	OmniPCX Office
Alcatel-Lucent Communication Platform Release	R920/041.001
AAPP member application version	i62 V5.1.22 OAW: 5.0.4.2 with AP126/12x
Application Category	Terminals Mobility DECT / Wi-Fi

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Reviewer(s): Denis Lienhart, Rachid Himmi, Florian Residori

Revision History

Edition 1: creation of the document –March *2014*

Test results

☐ Passed ☐ Refused ☐ Postponed

☑ Passed with restrictions

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

IWR validity extension

None

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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'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, Alcatel-Lucent 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://applicationpartner.alcatel-lucent.com) with free access.

2 Validity of the Inter-Working Report

This inter-working 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 Alcatel-Lucent issues a new major release of such Alcatel-Lucent 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 inter-working report becomes automatically obsolete when the mentioned product releases are end of life.*

3 Limits of the Technical support

Technical support will be provided only in case of a valid Interworking Report (see chapter 2 "Validity of the Interworking Report) and in the scope of the features which have been certified. That scope is defined by the Interworking report via the tests cases which have been performed, the conditions and the perimeter of the testing as well as the observed limitations. All this being documented in the IWR. 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 Alcatel-Lucent.

For any request outside the scope of this IWR, Alcatel-Lucent offers the "On Demand Diagnostic" service where assistance will be provided against payment.

For more details, 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 Alcatel-Lucent is included in the solution between the certified Alcatel-Lucent and AAPP member products such as a Session Border Controller or a firewall for example, Alcatel-Lucent will consider that situation as to that where no IWR exists. Alcatel-Lucent will handle this situation accordingly (for more details, please refer to Appendix F "AAPP Escalation Process").

4 Application information

Application type: VoWiFi handset

Application commercial name: Ascom i62

Application version: V 5.1.22

Interface type : SIP/Ethernet

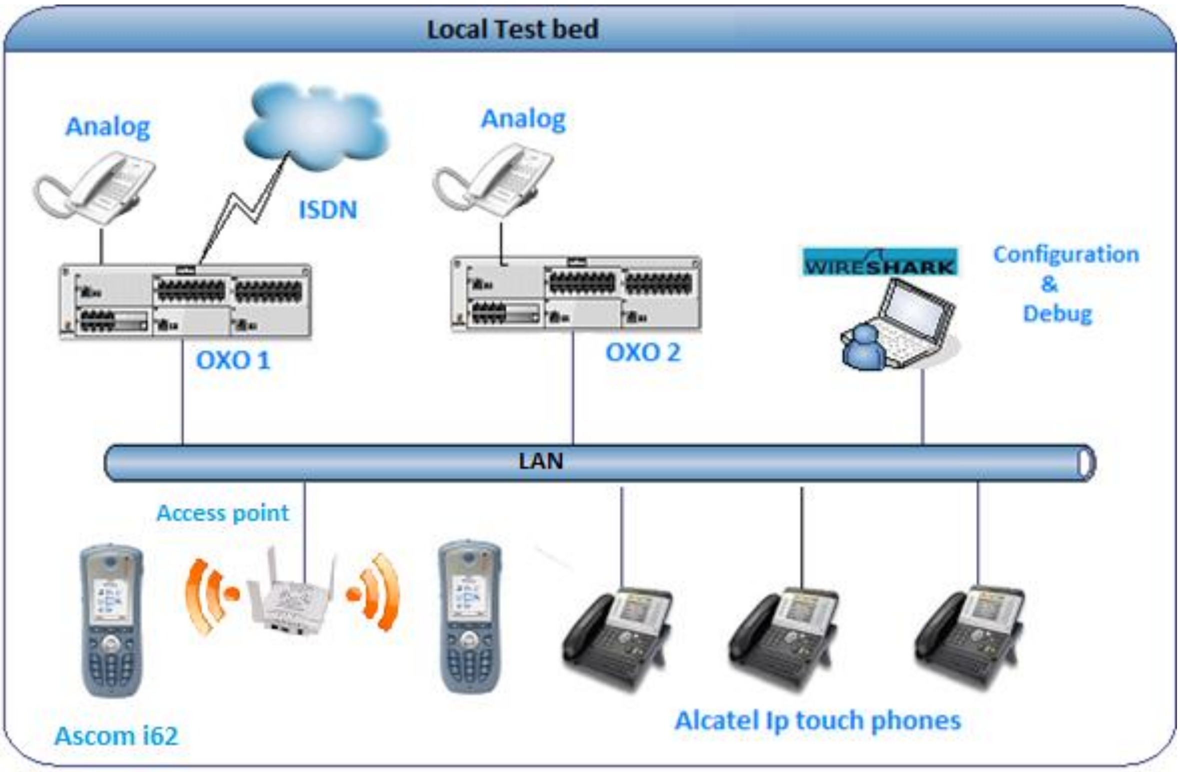
Brief application description:

The Ascom i62 VoWiFi handset delivers wireless telephony service and integrates smoothly in the Alcatel-Lucent OXO/OAW environment through the standardized SIP and WiFi endpoint interfaces. All telephony features provided by Alcatel-Lucent OXO are available to the Ascom i62 handsets, including Broker call, Transfer, Conference, Waiting, Forwards. The solution also provides attendant calls and access to the OXO Voice mail boxes and DTMF dialing for navigation.

The Alcatel-Lucent OAW infrastructure and Ascom i62 forms a high-performance mobility solution with excellent speech quality and campus-wide handover while maintaining high network security through the use of WPA2 and 802.1X port-based authentication.

Type of application/product:

5 Test Environment



5.1 Hardware configuration

<u>Alcatel-Lucent Communication Platform:</u>

- OmniPCX Office Rack
- PowerCPU
- Release: R920/041.001
- OMC: R921/25.1a

<u>Setup Details</u>

Setup Information OXO 1	
OXO 1 IP address	10.9.223.49
Domain name	Oxoone.testandvalidate.com
Voicemail No	500
Attendant No	100
OXO Extension Details used for test	
IP Touch numbers	124,123,122
Ascom Dir numbers	126,127 ,134
UA Set No	101

Setup Information OXO 2	
Network OXO address	10.9.223.251
Network OXO Domain name	Oxotwo.testandvalidate.com
Network OXO Extension Details used for test	
Ascom Dir number	333
IP Touch numbers	324 , 325 , 326
UA Set No	301

Note:

1) The Two OXO systems are connected via private SIP Trunk.
2) For some tests we will change the set type from IP Touch to UA set.

5.2 Software configuration

- **Alcatel-Lucent Communication Platform**: OmniPCX Office R920/041.001

- **Partner Application**: Ascom V5.1.22

Note: Ascom is registered in the OmniPCX Office as "Open SIP phone".

6 Summary of test results

6.1 Summary of main functions supported

Features	Status	Comments
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	
Call Transfer	OK	
Attendant	OK	
Voice mail interaction and indication	OK	

6.2 Summary of problems

- Hold tone is not heard in Ascom when initiated by the other extension in the call.
- The hold reinvite sent by the i62 set is not triggered by OXO due to an OXO nonce caching issue.

6.3 Summary of limitations

- In conference we are unable to see the other user information other than the one who initiated the conference.
- Call feature activation in the call server (CFU/CFB) for SIP extension is not displayed on the Ascom
- Call back from sip phones is not supported by OXO.
- MWI is only available from OXO version 920.041.001

6.4 Notes, remarks

- Ascom is registered in the OmniPCX Office as "Open SIP phone".

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 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 Alcatel-Lucent side or on Application Partner 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 Connectivity and Setup

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).

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**SIP sets** Configure your SIP sets MCDU number on the OXO as 126, 127 & 134 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.	☐	☒	☐	
2	**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 126, 127 & 134 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.	☐	☒	☐	
3	**DHCP registration (with OXO internal DHCP server)**	☒	☐	☐	
4	**NTP registration** The SIP phone 134 is configured to retrieve the date and time from the OXO IP address. Check the phone retrieves the right date and time information and displays it.	☒	☐	☐	
5	**Support of "423 Interval Too Brief" (1)** The SIP phone 127 is configured with a value lower than 120 seconds. Check the phone registration and display	☐	☒	☐	
6	**Signaling TCP-UDP** If applicable configure your SIP set 127 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

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

Phone configuration: configure Ascom to use G.722, 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).

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	Select G711 A-law as 1st codec in Ascom Call from SIP 127 to IP Touch 123 Check that the call is established in G711 A-law. Check audio quality Call from IP Touch 123 to SIP 127 Check that the call is established in G711 A-law. Check audio quality	☐	☒	☐	
2	Select G729 as 1st codec in Ascom Call from SIP 127 to IP Touch 123 Check that the call is established in G729 Check audio quality Call from IP Touch 123 to SIP 127 Check that the call is established in G729 Check audio quality	☐	☒	☐	
3	**Select G723 as 1st codec in Ascom** **Check that the call is established in G723** **Check audio quality** **Call from IP Touch 123 to SIP 127** **Check that the call is established in G723** **Check audio quality**	☒	☐	☐	G723 is not available in Ascom
4	**Configure** 127 **to use VAD** **Configure IP Touch** 123 **NOT to use VAD** Call from SIP 127 to IP Touch 123 Check that the call is established in G711 A-law. Check audio quality **Configure SIP** 127 **to use VAD** **Configure IP Touch** 123 **to use VAD** Redo the same tests **Configure SIP** 127 **NOT to use VAD** **Configure IP Touch** 123 **to use VAD** Redo the same tests	☐	☒	☐	
5	In OXO enable codec pass through for SIP phones Call from SIP 126 to SIP 127 Check that the call is established using G.722 Check audio quality	☐	☒	☐	

6	**In OXO 1 and OXO 2 enable codec pass through for SIP phone; direct RTP and codec pass through for SIP trunk. G722 is preferred codec in Ascom** **Call from SIP** 126 **to Network SIP** 324 Check that the call is established using direct RTP in G722. Check audio quality	☐	☒	☐	
7	**In OXO 1 and OXO 2 disable direct RTP and codec pass through for SIP trunk ARS table is configured with “default” codec. G722 is preferred codec in Ascom** **Call from SIP** 126 **to Network SIP** 324 Check that the call is established in G711. Check audio quality	☐	☒	☐	
8	**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** 126 **to Network SIP** 324 Check that the call is established in G729. Check audio quality	☐	☒	☐	

8.3 Outgoing Calls

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).

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Call to a local user** With SIP Phone 127 call the IP Touch 124. Check that 124 is ringing. Take the call and check ring back tone audio and display.	☐	☒	☐	
2	**Call to local user with no answer** With SIP Phone 134 call the IP Touch 124. And never take the call. Check time out (if any) and display. Note that 124 don't have a Voice Mail	☐	☒	☐	124 is ringing until 134 releases the call
3	**Call to another SIP set** With the SIP phone 127 call the other SIP Phone 134 Check the display and audio during all steps (dialing, ring back tone, conversation, and release).	☐	☒	☐	
4	**Call to wrong number** (SIP: "404 Not Found") With the SIP phone 127 call a wrong number Check the ring back tone and display	☐	☒	☐	"Invalid number" is displayed on the phone.
5	**Call to busy user** **(SIP: "486 Busy Here")** With the SIP phone 127 call IP Touch 124, take the call and don't hang up. With other SIP phone 134 call 124 which is busy Check the ring back tone and display	☐	☒	☐	
6	**Call to user in "Out of Service" state** **(SIP: "480 Temporarily Unavailable")** With the SIP phone 134 call the IP Touch 124 which is in "Out of Service State" Check the display and ring back tone	☐	☒	☐	No response message is displayed

Test Case Id	Test Case	N/A	OK	NOK	Comment
7	**Call to user in "Do not Disturb" (DND) state** **(SIP: "480 Temporarily not available")** Dial "*63" on the IP Touch 124 in order to enable the DND. Wait for acknowledgement ring back tone from OXO. With the SIP phone 127 call 124. Check ring back tone and display. Redial *60 on 124 to cancel the DND	☐	☒	☐	User busy status is displayed if voice mail is disbled.
8	**Call to local user, immediate forward (CFU).** **(SIP: "181 Forwarded")(1)** On IP Touch 124 dial the *61123 (*61 + 123) to activate the CFU. Wait for acknowledgement ring back tone from OXO. With the SIP phone 127 call the 124. Check that 123 is ringing and the display. Take the call check audio and hung up. Dial *60 on 124 for forward cancellation.	☐	☒	☐	The user information is shown properly
9	**Call to local user, forward on no reply (CFNR). (1)** On IP Touch 124 configure with OMC the CFNR using dynamic routing to 123. With 127 call the 124. Check that 124 is ringing but don't take the call and wait the time out (about 30 sec). Time out is defined in 124 dynamic routing of Timer 1. After time out check that 123 is ringing and take the call. Check the audio and display.	☐	☒	☐	The user information is shown properly
10	**Call to local user, forward on busy (CFB). (1)** On IP Touch 124 dial the *62123 (*62+<target MCDU number>) to activate the CFB. Wait for acknowledgement ring back tone from OXO. With SIP phone 127 call 124 and take the call to make it busy. With other SIP phone 134 call 124. Check that 123 is ringing and take the call. Check the audio and display. Dial *60 on 124 for forward cancellation.	☐	☒	☐	
11	**Call to external number** **(Check ring back tone, called party display)** With SIP set 127 dial 9 (9 prefix +external number) Take the call and check audio, display and call release.	☐	☒	☐	
12	**SIP session timer expiration: Check if call is maintained or released after the session timer has expired** With SIP set 127 call IP Touch 124. Take the call on 124 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

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.

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Local /network call to free SIP terminal** Local: with IP Touch 124 call SIP set 127. Check that 127 is ringing and take the call Check ring back tone and called party display. Network: with IP Touch 124 call SIP set 323 on another Node. Check that 323 is ringing and take the call. Check ring back tone and called party display.	☐	☒	☐	
2	**Local/network call to busy SIP terminal** Local: With SIP set 134 call other SIP set 127 and take the call to make it busy, don't hang up. With IP Touch 123 call 127 which is busy Check the ring back tone and display. Network: With SIP set 127 call SIP set 123 and take the call to make it busy, don't hang up. With 324 call 127 which is busy Check ring back tone and called party display.	☐	☒	☐	
3	**Local/network call to unplugged SIP terminal** Local: Unplug the 127 SIP set and call it with IP Touch 124. Check the ring back tone and display Network: Unplug the SIP set 123 and call it with 124 Check the ring back tone and display	☐	☒	☐	Call to SIP phone is not routed to its VMU until OXO detects SIP phone is unregistered (after register time out)
4A	**Local/network call to SIP terminal in Do Not Disturb (DND) mode** **By local feature if applicable:** Local: Enable DND on SIP set 127 and call it with IP Touch 124 Check the ring back tone and display Cancel the DND on 127. Network: Enable DND on SIP set 123 and call it with IP Touch 124 Check the ring back tone and display Cancel the DND on 127.	☐	☒	☐	.

Test Case Id	Test Case	N/A	OK	NOK	Comment
4B	**By system feature** <u>Local</u>: Enable DND on SIP set 127 using the *63 prefix. Wait for acknowledgement ring back tone from OXO. With IP Touch 124 call 127 Check the ring back tone and display Cancel the DND on 127 using *63 prefix. <u>Network</u>: Enable DND on SIP set 123 using the *63 prefix. Wait for acknowledgement ring back tone from OXO. With IP Touch 124 call 123 Check the ring back tone and display Cancel the DND on 123 using * 60 prefix.	☐	☒	☐	
5A	**Local/network/SIP call to SIP terminal in immediate forward (CFU) to local user:** **By local feature if applicable:** <u>Local</u>: On SIP set 127 enable CFU to IP Touch 124 With SIP set 134 call 127. Check that 124 is ringing. Take the call and check audio and display. Disable CFU on 127. <u>Network</u>: On SIP set 123 enable CFU to IP Touch 102. With SIP set 127 call 123. Check that 102 is ringing. Take the call and check audio and display. Disable CFU on 123.	☐	☒	☐	
5B	**By system feature:** <u>Local</u>: On SIP set 127 enable CFU to IP Touch 124 using *61124 prefix (*61 + <target MCDU number>). Wait for acknowledgement ring back tone from OXO. With SIP set 134 call 127. Check that 124 is ringing. Take the call and check audio and display. Disable CFU on 127 using *60 prefix. <u>Network</u>: On SIP Set 123 enable CFU to IP Touch 102 using *61122 prefix (*61 + <target MCDU number>). Wait for acknowledgement ring back tone from OXO. With SIP Set 134 call 123. Check that 102 is ringing. Take the call and check audio and display. Disable CFU on 123 using *60 prefix.	☐	☒	☐	
B	**Local/network/SIP call to SIP terminal in immediate forward (CFU) to network number:** **By local feature if applicable:** <u>Local</u>: On SIP Set 134 enable CFU to SIP Set122.With SIP set 127 call 134. Check that 122 is ringing. Take the call and check audio and display. Disable CFU on 134. <u>Network:</u> On SIP Set 127 enable CFU to IP Touch 102. With SIP Set 123 call 127. Check that 102 is ringing. Take the call and check audio and display. Disable CFU on 127.	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
6B	**By system feature:** <u>Local</u>: On SIP Set 127 enable CFU to SIP Set 122 using *61122 prefix (*61 + <target MCDU number>). Wait for acknowledgement ring back tone from OXO. With SIP set 134 call 127. Check that 122 is ringing. Take the call and check audio and display. Disable CFU on 127 using *60 prefix. <u>Network</u>: On SIP Set 127 enable CFU to IP Touch 102 using *61102 prefix (*61 + <target MCDU number>). Wait for acknowledgement ring back tone from OXO. With SIP Set 123 call 127. Check that 102 is ringing. Take the call and check audio and display. Disable CFU on 127 using *60 prefix.	☐	☒	☐	
7A	**Local/network/SIP call to SIP terminal in immediate forward (CFU) to another SIP user** **By local feature if applicable:** <u>Local</u>: On SIP set 127 enable CFU to SIP set 122. With 134 call 127. Check that 122 is ringing. Take the call and check audio and display. Disable CFU on 127. <u>Network</u>: On SIP set 127 enable CFU to IP Touch 103. With SIP Set 122 call 127. Check that 103 is ringing. Take the call and check audio and display. Disable CFU on 127.	☐	☒	☐	
7B	**By system feature:** <u>Local</u>: On SIP Set 134 enable CFU to SIP Set 122 using *61122 prefix (*61 + <target MCDU number>). Wait for acknowledgement ring back tone from OXO. With SIP Set 127 call 134. Check that 122 is ringing. Take the call and check audio and display. Disable CFU on 134 using *60 prefix. <u>Network</u>: On SIP Set 134 enable CFU to IP Touch 103 using *61123 prefix (*61 + <target MCDU number>). Wait for acknowledgement ring back tone from OXO. With SIP Set 122 call 134. Check that 103 is ringing. Take the call and check audio and display. Disable CFU on 134 using *60 prefix	☐	☒	☐	CFU is activated and working in the call server but OXO returns a SIP msg “487 Request Terminated”.
8A	**Local call to SIP terminal in “forward on busy” (CFB) state:** **By local feature if applicable** On SIP Set 127 enable CFB to IP Touch 124 With 127 call the voice mail at 500 to make it busy. With SIP Set 134 call 127 which is busy. Check that 124 is ringing Take the call and check audio and display. Disable CFU on 127.	☐	☒	☐	Call waiting has to be disabled in Ascom device

Test Case Id	Test Case	N/A	OK	NOK	Comment
8B	**By system feature:** On SIP Set 127 enable CFB to IP Touch 124 using *62124 prefix (*62 + <target MCDU number>). Wait for acknowledgement ring back tone from OXO. With 127 call the voice mail at 500 to make it busy. With SIP Set 134 call 127 which is busy. Check that 124 is ringing Check the Call waiting or ring back tones and display Disable CFB on 127 using *60 prefix.	☐	☒	☐	
9A	**Local call to SIP terminal in "forward on no reply" (CFNR)** **By local feature if applicable** On SIP Set 134 enable CFNR to IP Touch 124 With SIP Set 127 call 134. Check that 134 is ringing and don't take the call, wait for time out (about 30 seconds). After time out expiration the 124 is ringing, take the call and check audio and display.	☐	☒	☐	
9B	**By system feature** CNFR via prefix not available on OXO (dynamic routing has to be used)	☐	☒	☐	
10	**Call to busy user, Call waiting.** (Camp-on), local feature if applicable: With SIP Set 127 call other SIP Set 134 (multiline set) to make it busy, take the call and don't hang up. With IP Touch 123 call 134 (on 134 camp-on feature is enabled).	☐	☒	☐	
11	**External call to SIP terminal.** Check that external call back number is shown correctly: With SIP Set 134 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.	☐	☒	☐	
12	**Calling Line Identity Restriction (CLIR): Local call to SIP terminal.** On IP Touch 123 enable mask Identity and call SIP Set 134 in order to hide 123 identity. Check that 134 is ringing, take the call and check that 123 identity is hidden.	☐	☒	☐	Identity secrecy to be enabled in Features
13	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.	☐	☒	☐	
14	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
15	SIP set is part of a sequential hunt group **(1)**. Call to hunt group. Check call/release. With IP Touch 124 call the sequential hunt group MCDU number 328 Check that 127 is ringing Take the call and don't hang up. And with IP Touch 123 call the sequential hunt group MCDU number 328 Check that 123 is ringing Take the call and don't hang up. And with SIP Set 126 call the sequential hunt group MCDU number 328 Check that 134 is ringing Take the call and don't hang up.	☐	☒	☐	
16	SIP set is part of a cyclic hunt group **(2)**. Call to hunt group. Check call/release. With IP Touch 124 call the cyclic hunt group MCDU number 123 Check that 126 is ringing Take the call and hang up. And with 124 call the cyclic hunt group MCDU number 123 Check that 134 is ringing Take the call and hang up. And with SIP Set 126 call the cyclic hunt group MCDU number 123 Check that 127 is ringing Take the call and don't hang up.	☐	☒	☐	
17	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 123 call IP Touch 124 which is in MultiSet with SIP Set 134. Check that 134 and 124 both ringing. Take the call from 134 and check that 124 stop ringing. Check audio and display.	☐	☒	☐	

Notes:
(1) Sequential Hunt Group behavior: the endpoint n+1 is ringing **only** if the endpoint n is now in call (busy).

(2) Cyclic Hunt Group behavior: the endpoint n+1 is ringing if previously the endpoint n has been reached (ringing only or in call). The actual state of the n endpoint doesn't matter.

8.5 Features during Conversation

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.

Test Case Id	Test Case	N/A	OK	NOK	Comment
1A	**Hold and resume with local feature** (if applicable) With 134 call 124 take the call, check audio and display. With 134 put 124 on hold check tones and display on both and resume the call. With 124 put 134 on hold check tones and display on both and resume the call. Keep this call for the next test.	☐	☐	☒	When we initiate hold from other user to Ascom phone, there is no hold tone heard on Ascom. The hold reinvite sent by the i62 is not triggered by OXO due to an OXO nonce caching issue.
1B	**Enquiry call to another local user** (if applicable) Distant user is put on hold with local feature With 134 (multi-lines) call 123 and take the call. 124 will be put on hold when making second call to 123 Put 123 on hold and check tones and display on both. Keep these two calls for the next test.	☐	☒	☐	
1C	**Broker request, toggle back and forth between both lines with local feature** (if applicable) With 134 switch between 124 and 123 lines. Check the tones and display on sets on hold state. Keep these two calls for the next test.	☐	☒	☐	
1D	Release first call. Keep second call. Hang up 124 and only 134 and 123 are in call Check that 134 & 123 are still in a call, check display.	☐	☒	☐	
2	Repeat the test 1C to 1D but using the call server feature	☒	☐	☐	Hold, enquiry, broker call functionality are not supported within call server for SIP extension
3	**Three party conferences initiated from OXO set** With 124 call 127, take the call and don't release it. With 124 call 122, take the call and don't release it too. With 124 start a conference. Check that 124, 123 and 127 are in conference. Check audio and display.	☐	☒	☐	SIP extension display is not updated when OXO users initiate the conference

Test Case Id	Test Case	N/A	OK	NOK	Comment
4A	**Three party conferences initiated from SIP set with local feature** (if applicable) With 127 call 124, take the call and don't release it. With 127 call 123, take the call and don't release it too. With 127 start a conference by the local feature Check that 124,123 and 127 are in conference. Check audio and display.	☐	☒	☐	Display on IP Touch sets doesn't show the second user in the conference initiated on the SIP extension.
4B	**Three party conferences initiated from SIP set with OXO feature**	☒	☐	☐	Conference feature is not supported within call server for SIP device.
5	**Meet Me conference** With 134 call the Meet me Conference bridge dialing prefix 68 and follow instruction to open the bride. With 127 join the conference bridge by dialing prefix 69 and enter access code. With 124 join the conference bridge by dialing prefix 69 and enter access code. Check that 124, 127 and 134 are in conference.	☐	☒	☐	

8.6 Call Transfer

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 OmniPCX Office.

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.

Test	Action			Result	Comment
	A	B	C		
	Transferee	Transferor	Transfer Target		
1	OXO	SIP	OXO	OK	
2	Ext Call	SIP	OXO	OK	
3	Ext Call	SIP	Ext Call	OK	
4	SIP	SIP	SIP	OK	
5	SIP	OXO	OXO	OK	
6	Ext Call	OXO	SIP	OK	
7	SIP	OXO	SIP	OK	

8.7 Attendant

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

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**SIP set call to attendant** From SIP set 127 dial “9” (attendant call prefix) Check audio and display	☐	☒	☐	Display on 127 shows internal number/name of Attendant extension
2	**2nd incoming call while in conversation with attendant** While SIP set 127 is in conversation with the attendant, from IP Touch 123 call 127 Answer the call and check audio and display	☐	☒	☐	
3	**SIP set call to attendant, attendant transfers to OXO set, semi-attended** From SIP set 127 dial “9” (attendant call prefix) and answer. Attendant transfer semi-attended to IP Touch 123 Answer the call and check audio and display	☐	☒	☐	
4	**SIP set call to attendant, attendant transfers to OXO set, attended** From SIP set 127 dial “9” (attendant call prefix) and answer Attendant transfer attended to IP Touch 123 Check audio and display	☐	☒	☐	
5	**OXO set calls to attendant, attendant transfers to SIP set, attended** From IP Touch 123 dial “9” (attendant call prefix) and answer Attendant transfer attended to SIP set 127 Check audio and display	☐	☒	☐	
6	**External ISDN Call to attendant, attendant transfers to SIP set, attended** ISDN incoming call to the attendant. From the attendant call SIP set 127 and transfer attended Check audio and display	☐	☒	☐	
7	**SIP set call to attendant, attendant transfers to External** From SIP set 127, dial “9” (attendant call prefix) and answer From the attendant, call an external ISDN destination and transfer semi-attended Answer and check audio and display.	☐	☒	☐	

8.8 Voice Mail

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

The default Voice Mail number is 500, and this service is enabled on SIP sets 127, 134 and OXO 124.

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

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	Password modification With SIP set 134 call the Voice Mail at 500 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.	☐	☒	☐	
2	Message display activation, MWI (1): With SIP set 127 call the Voice Mail at 500. Follow the instructions in order to send a voice message in SIP set 134 boxes. Check that the MWI on 134 is activated.	☐	☒	☐	MWI is only available from OXO version 920.041.001
3	Message consultation With SIP set 134 call the Voice Mail at 500. Follow the instructions in order to listen your voice message leaved during the previous test. Check that you can listen it and delete. Check that MWI display is disabled on 134 after message cancellation.	☐	☒	☐	
4	SIP call to a OXO user forwarded to Voice Mail Forward the IP Touch 124 to Voice Mail by dialing *61500 (*61 prefix + <Voice Mail number>). With SIP set 134 call 124 and check that you are immediately forwarded to Voice Mail. Check that you can leave a message On 124 disable Voice Mail forwarding with *60 prefix.	☐	☒	☐	
5	OXO set call to a SIP user forwarded to Voice Mail Forward the SIP set 134 to Voice Mail by dialing *61500 (*61 prefix + <Voice Mail number>). With IP Touch 124 call 134 and check that you are immediately forwarded to Voice Mail. Check that you can leave a message On 134 disable Voice Mail forwarding with *60 prefix.	☐	☒	☐	

Notes:
(1) On SIP sets, in order to enable the MWI feature, you have to configure the Voice Mail number.

8.9 Defence

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

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**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-1 and check the audio.	☐	☒	☐	
2	**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.	☐	☒	☐	ISDN call is rerouted to attendant after register time-out

9 Appendix A: AAPP member's Application Description

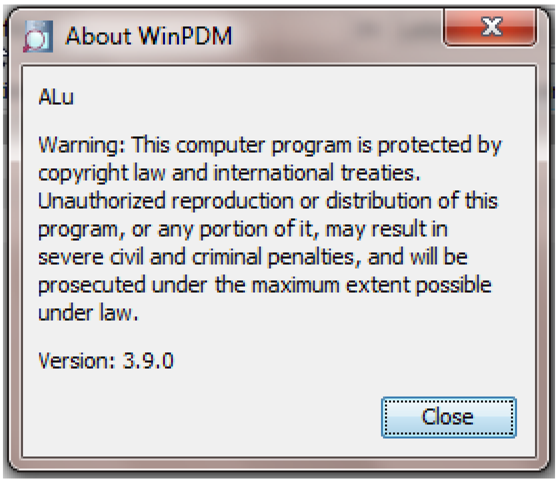
The Ascom i62 VoWiFi handset delivers wireless telephony service and integrates smoothly in the Alcatel-Lucent OXE/OAW environment through the standardized SIP and WiFi endpoint interfaces. All telephony features provided by Alcatel-Lucent OXE SEPLOS (SIP Endpoint Level of Service) are available to the Ascom i62 handsets, including Broker call, Transfer, Conference, Call back, Waiting, Forwards, in intra-node and inter-nodes communications. The solution also provides attendant calls and access to the OXE Voice mail boxes and DTMF dialing for navigation.

The Alcatel-Lucent OAW infrastructure and Ascom i62 forms a high-performance mobility solution with excellent speech quality and campus-wide handover while maintaining high network security through the use of WPA2 and 802.1X port-based authentication.

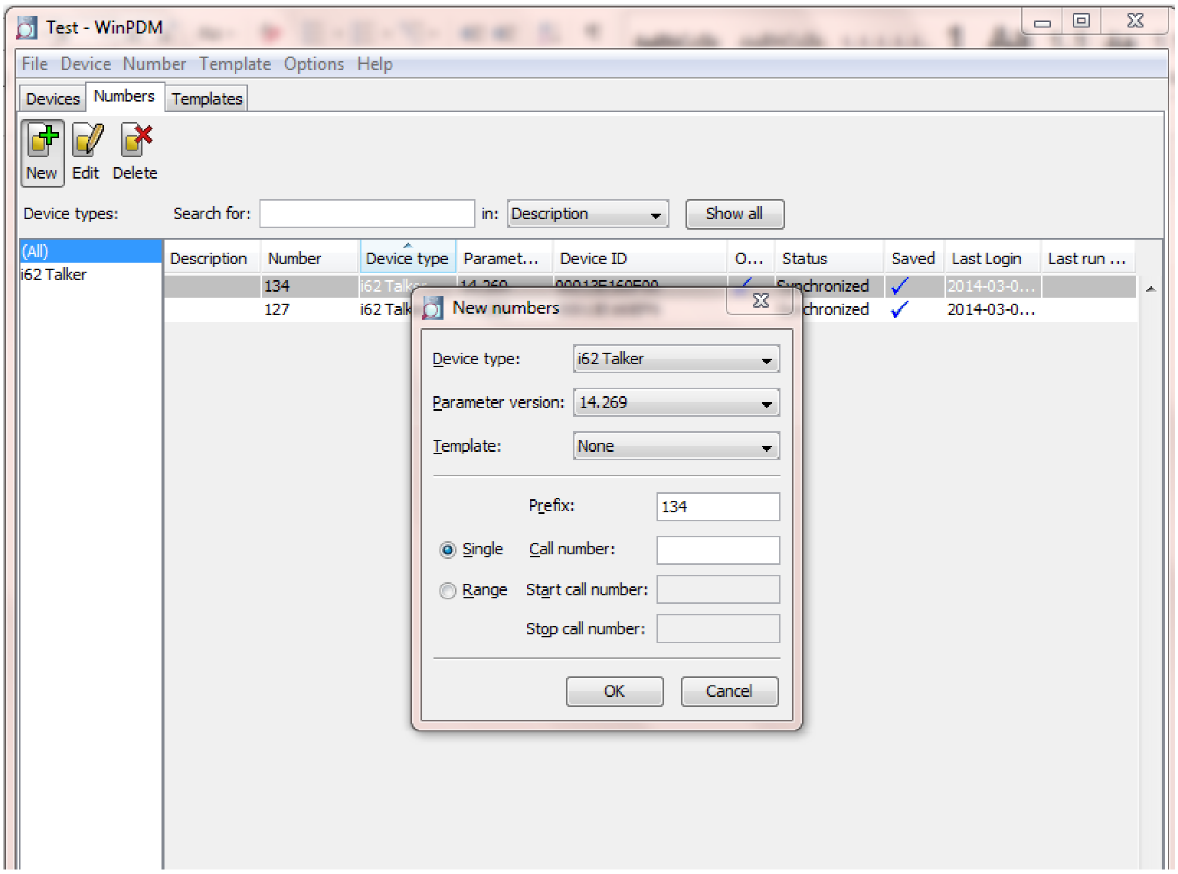
10 Appendix B: Configuration requirements of the AAPP member's application

Ascom i62:

Ascom requires Win PDM software for configuration.

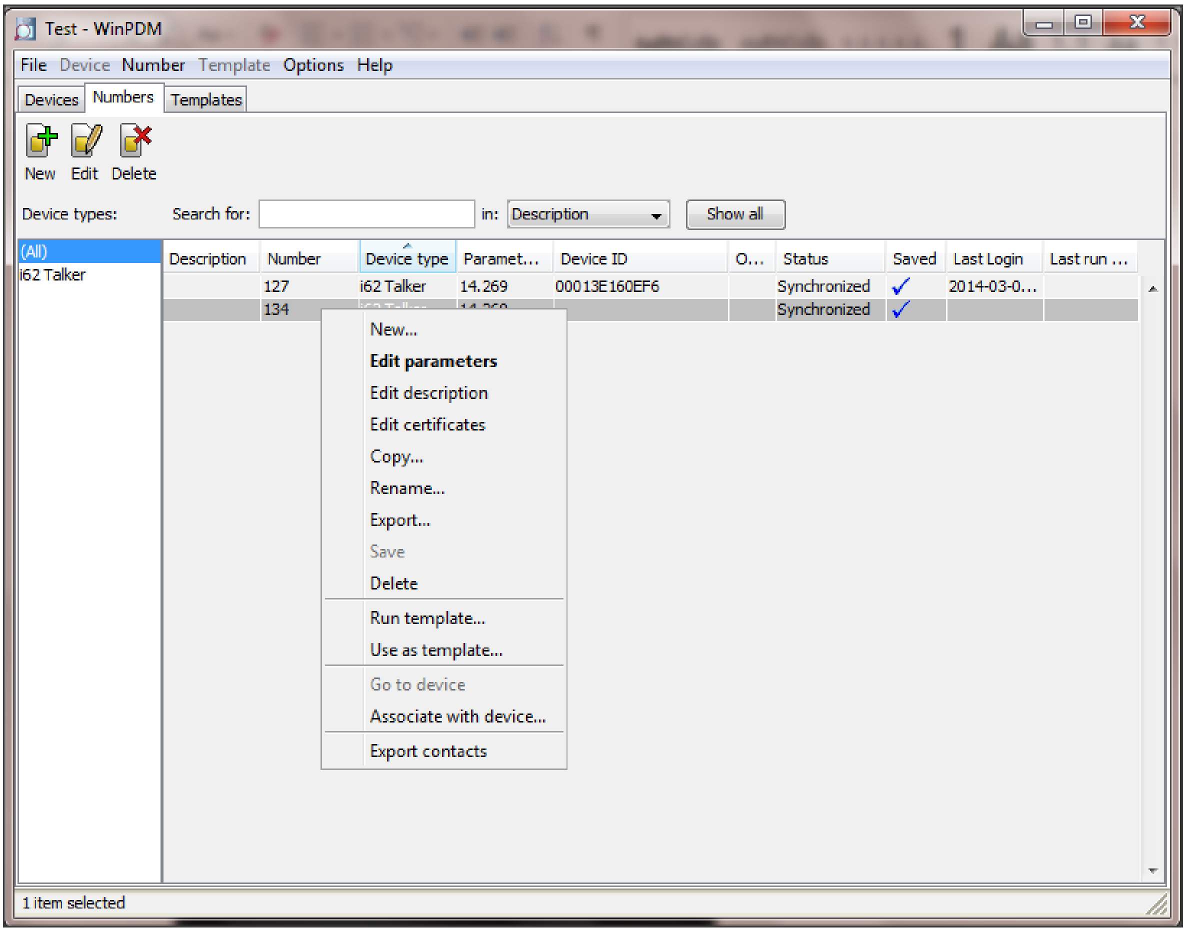


To configure the device we need to create a number in the WIN PDM software as below.

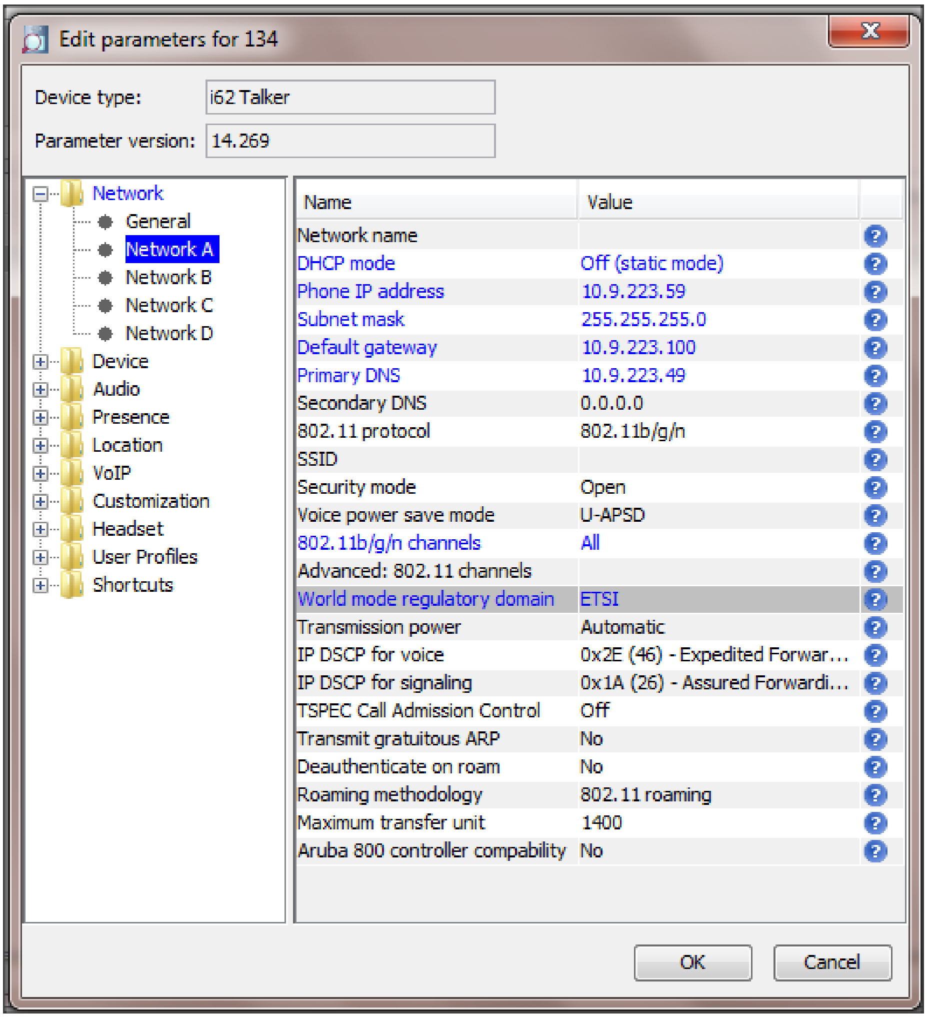


After creation of a number the SIP parameters of the extension have to be configured.

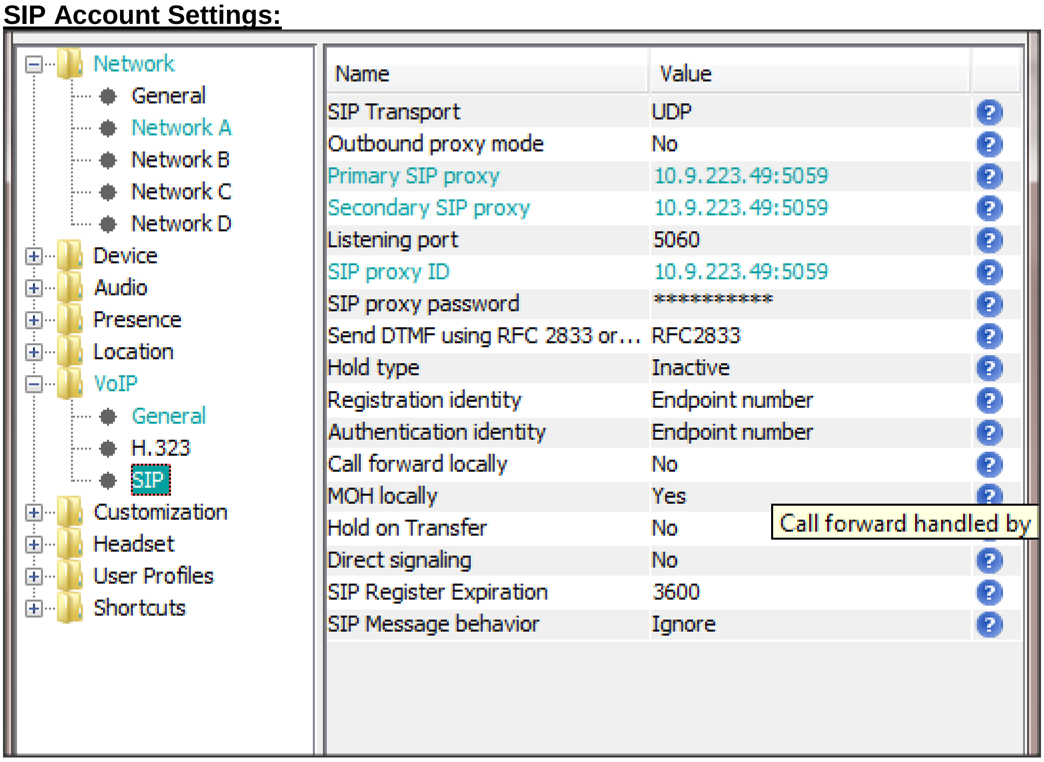
The extension must have parameters of the extension created in the OXO.



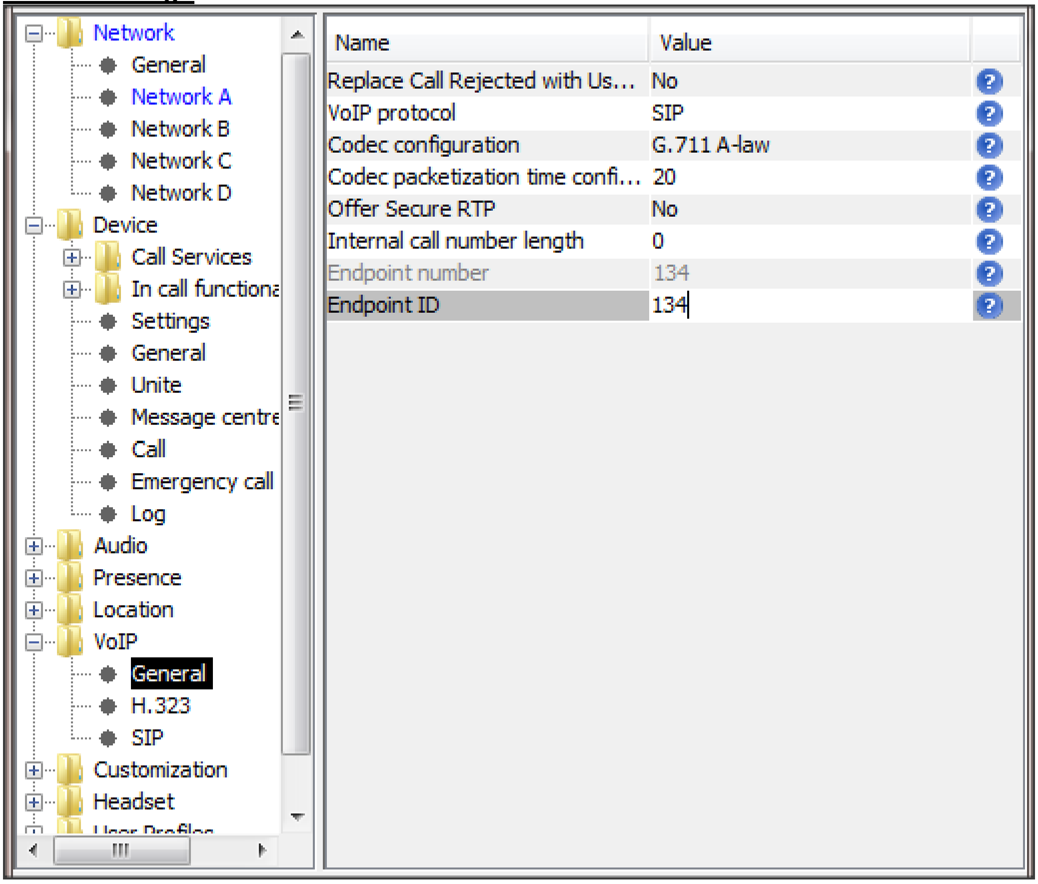
Network settings



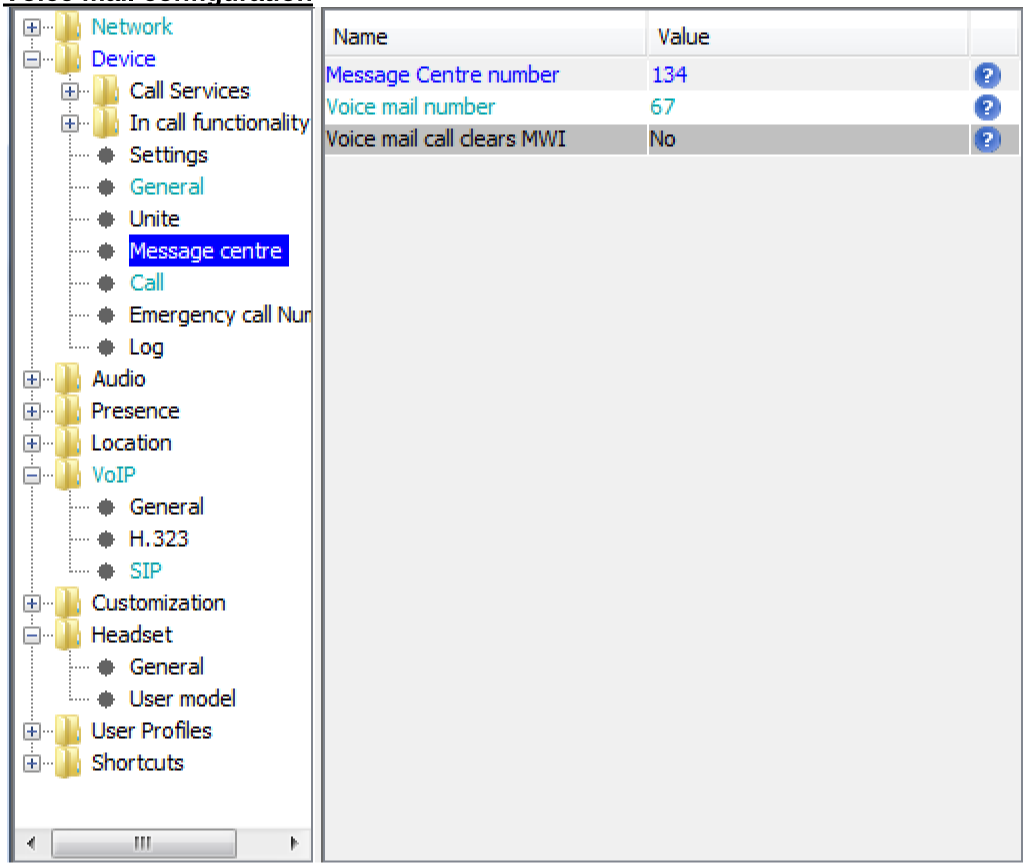
SIP Account Settings:



Codec settings

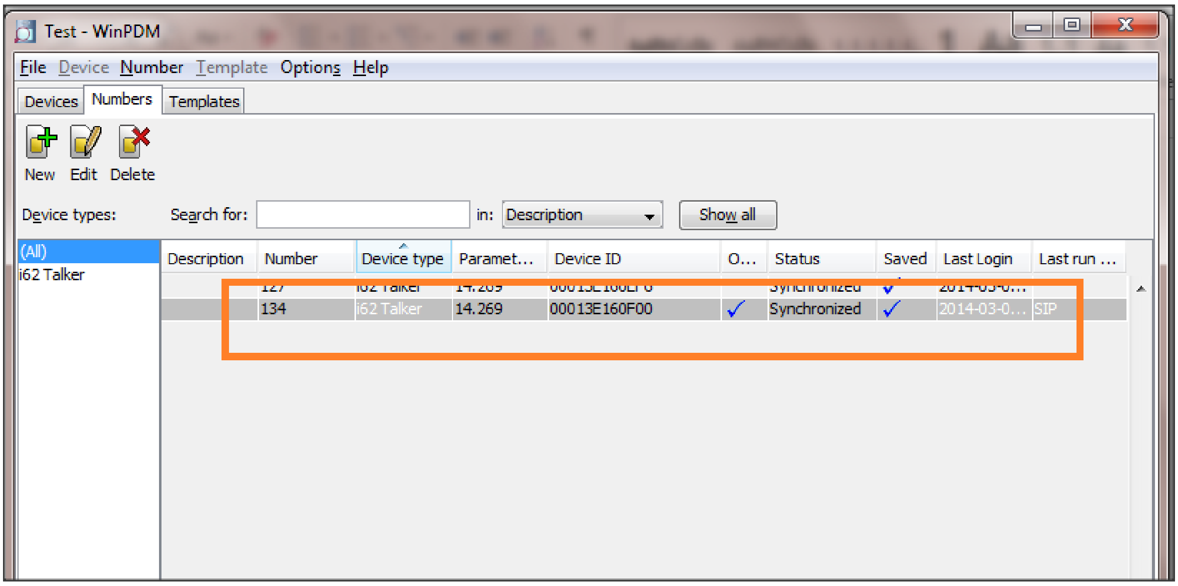


Voice mail configuration

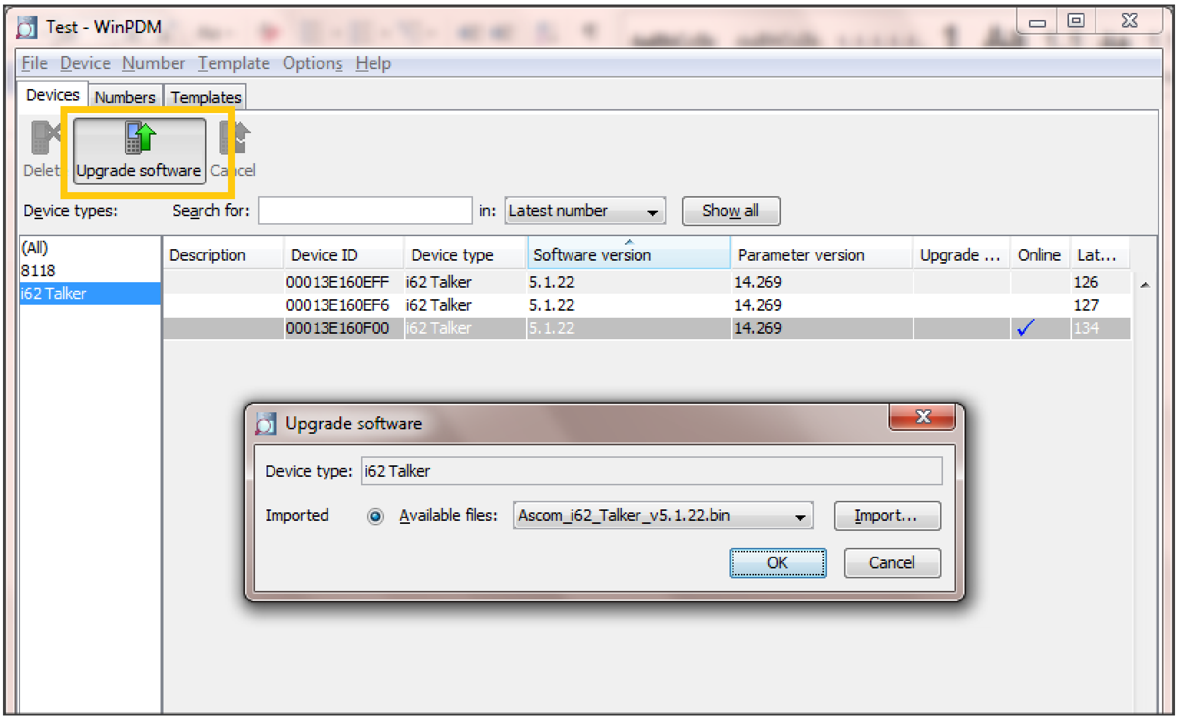


134 is the SIP OXO extension number, 67 is the “Mailing” prefix
After configuring the parameters insert the Ascom device in the USB dongle connected to the PC and in the displayed menu we need to associate the created number to the device to apply the settings to the device.

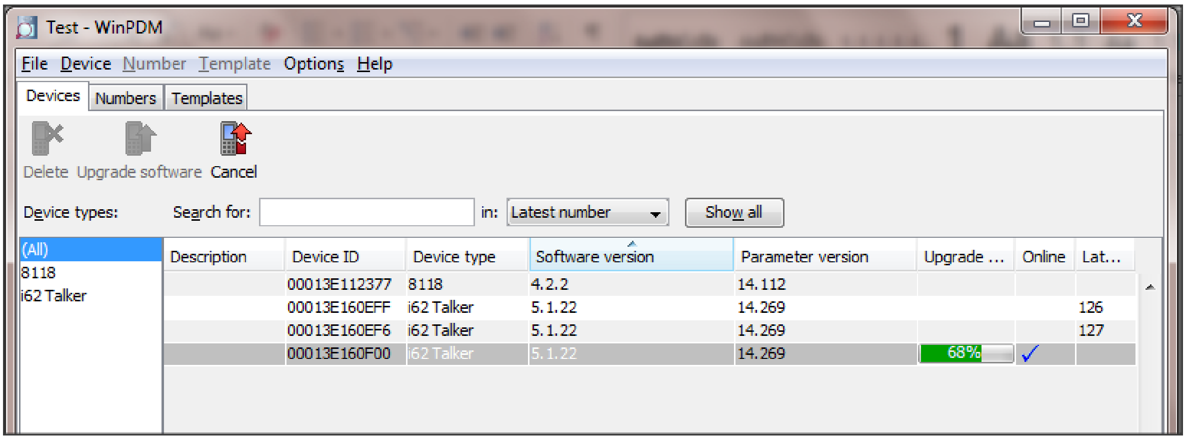
After associating to the number the appropriate Device ID should appear next the number in the table shown below. The device will reboot and come in service if the setting are correct.



Software upgrade



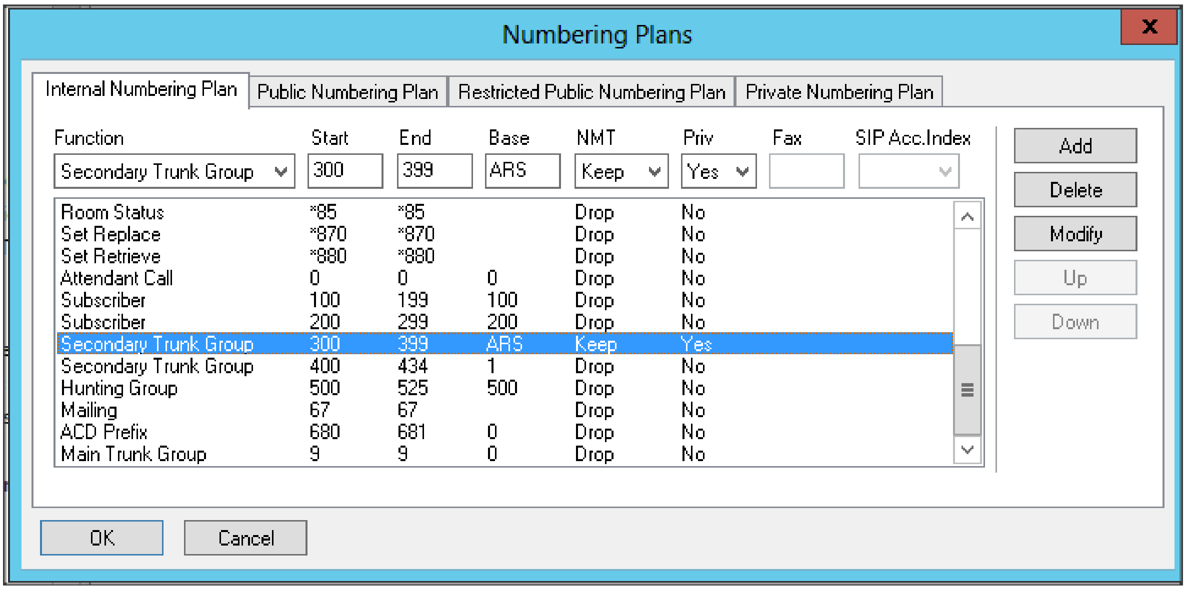
Place the device in dongle select upgrade in the menu and flashing process will start and status can be seen as below.



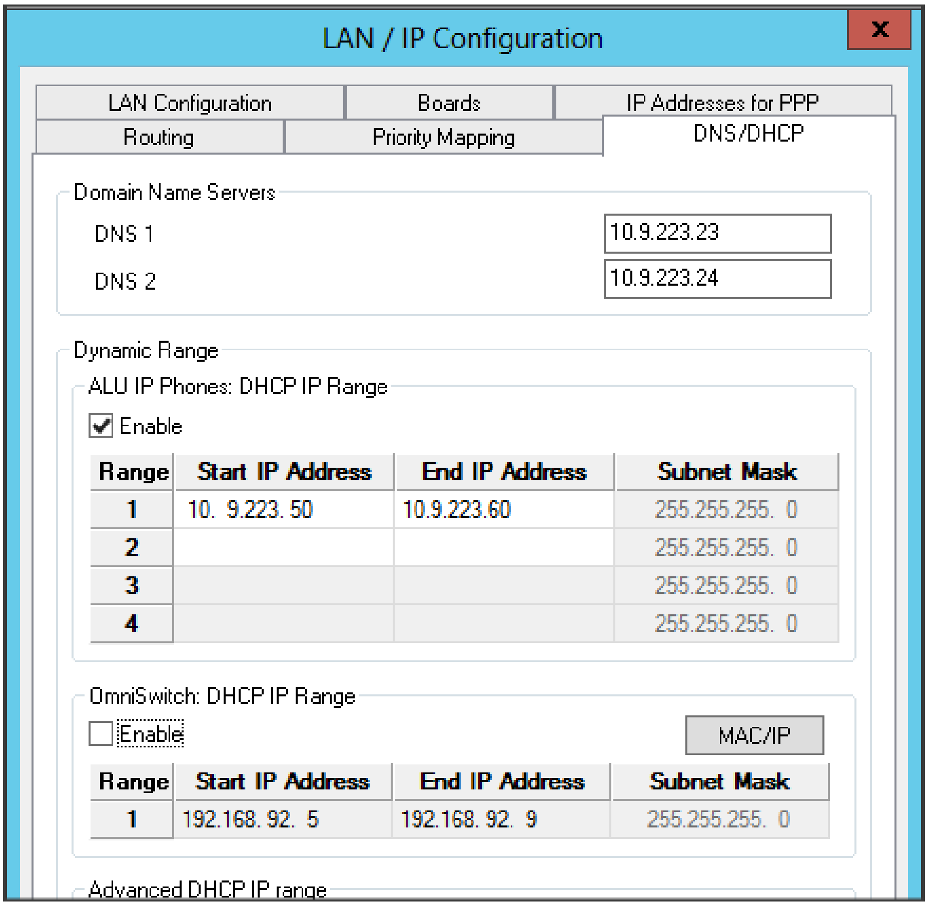
11 Appendix C: Alcatel-Lucent Communication Platform: configuration Requirements

OXO Configuration

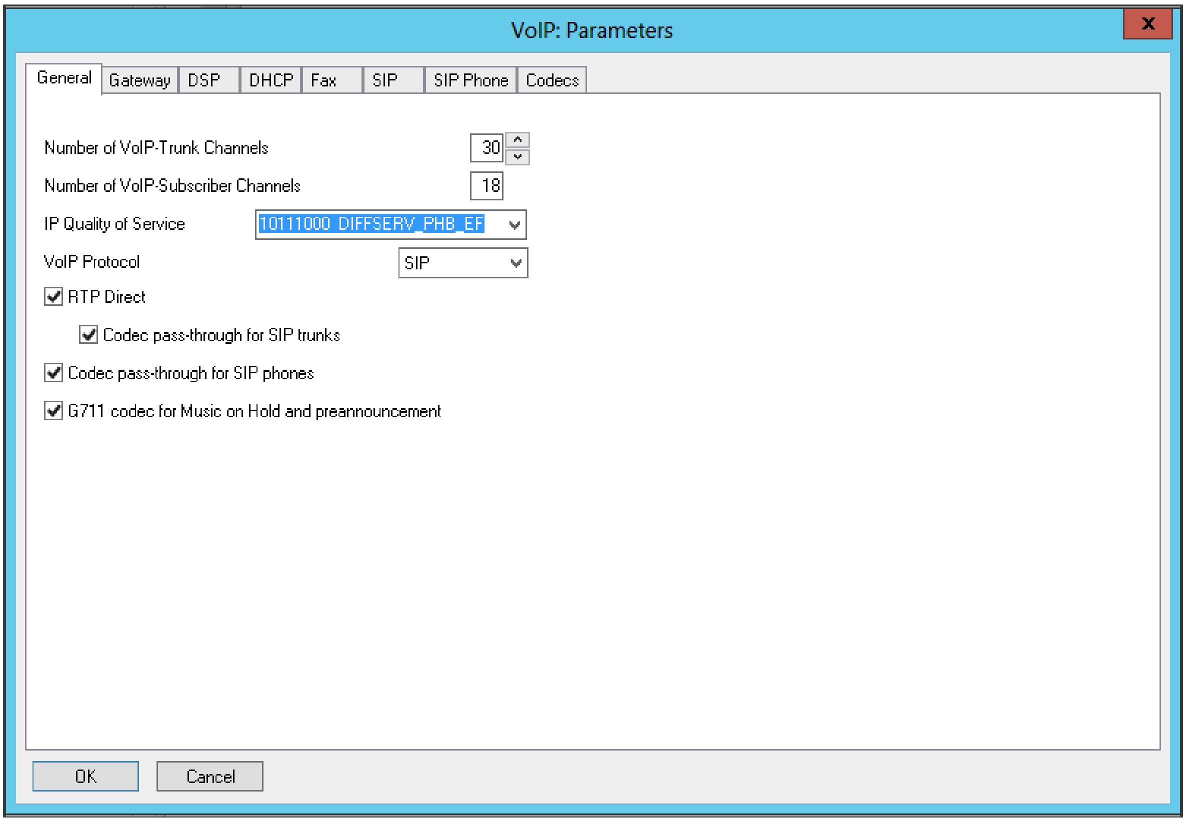
1. Dialing Plan

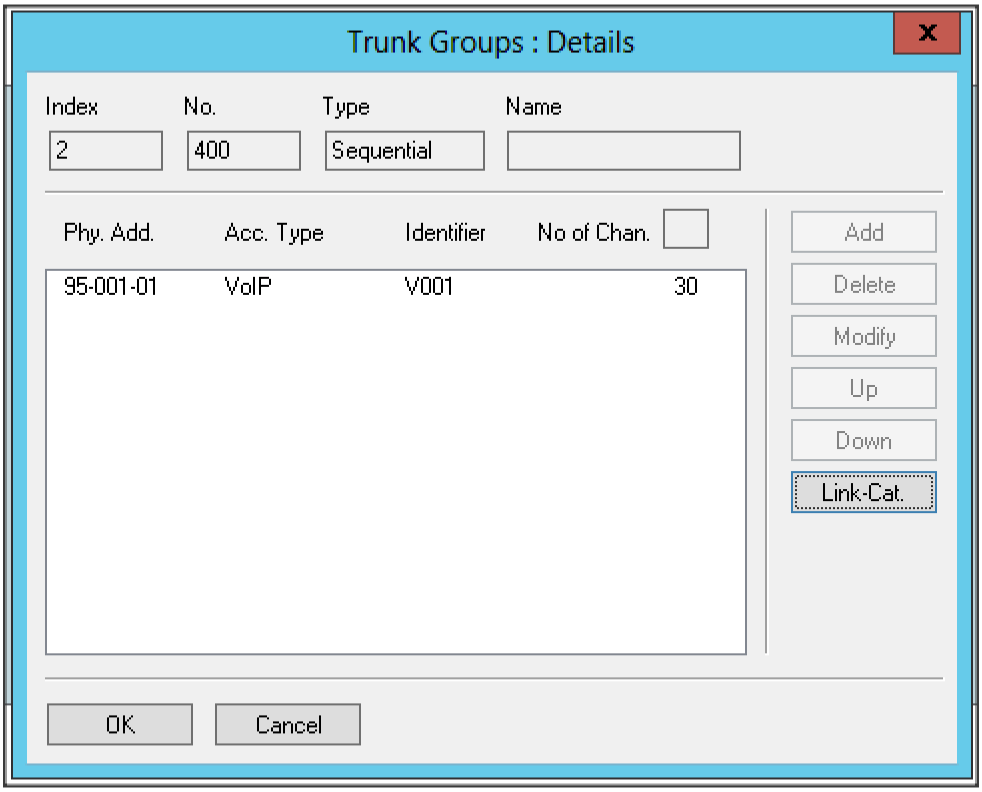


2. DNS/DHCP Configuration

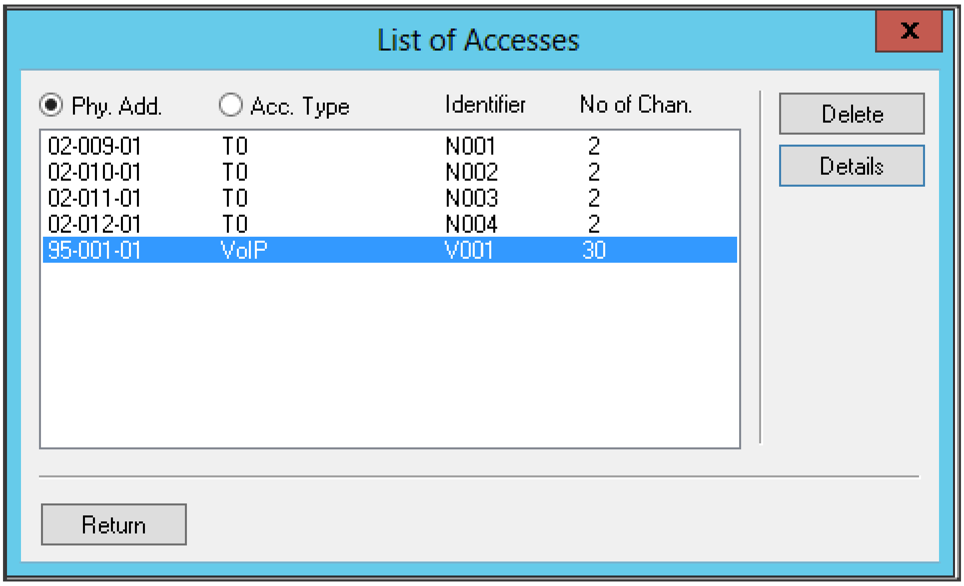


3. Trunk Configuration

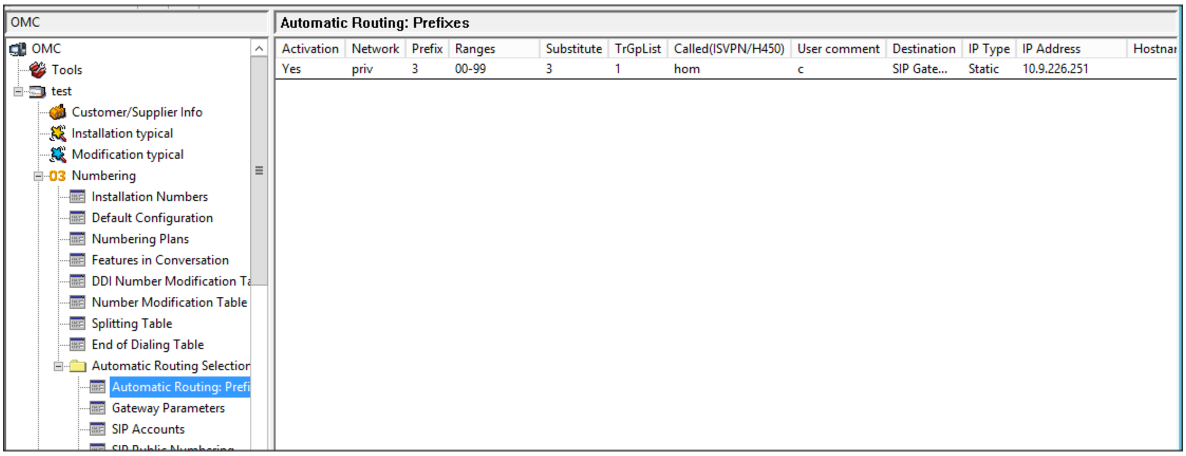




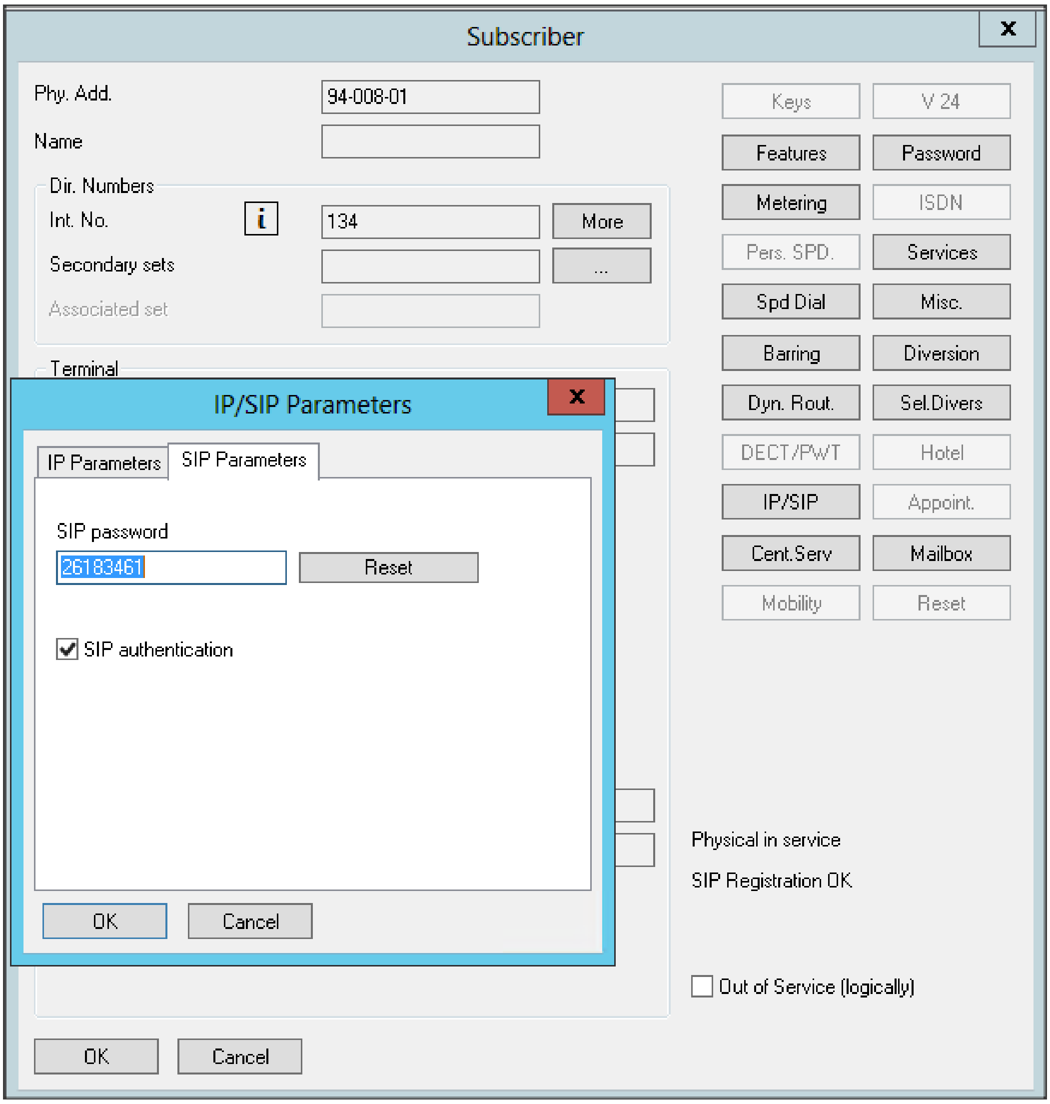
4. Trunk Access



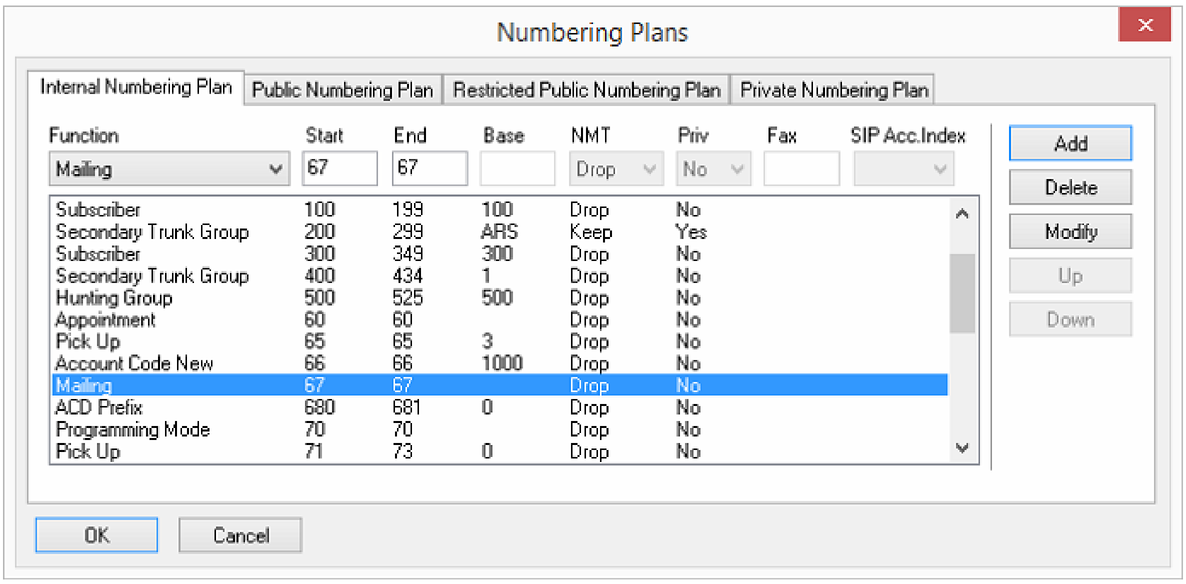
5. ARS Configuration



6. SIP Set Configuration



7. Lists of OXO prefixes used in tests



12 Appendix D: AAPP member's Escalation Process

The following list of contacts can be used to escalate possible issues according to the country:

Company/Country	Technical Manager/ Service Manager	e-mail
Ascom AG, Wireless Solutions, CH	Christoph Gsell	christoph.gsell@ascom.ch
Ascom Tateco AS, NO	Morten S. Pettersen	Morten.Pettersen@ascom.no
Ascom Nira BV, NL	Kees Voorwinden	Kees.Voorwinden@ascom.nl
Ascom Nira BV, NL	Jacques Koring	Jacques.Koring@ascom.nl
Ascom Tele-Nova Ltd, UK	Adrian Davenport	Adrian.Davenport@ascomtelenova.co.uk
Ascom Wireless Solutions Inc., USA	Tim Overstreet	Tim.Overstreet@ascomwireless.com
Ascom France, FR	Jose Rodrigues	jose.rodrigues@ascom.fr
Ascom Danmark, DK	Jaap Bootsman	Jaap.bootsman@ascom.dk
Ascom Germany GmbH, DE	Hermann Füg	Hermann.Fueg@ascom.de
Ascom NV/SA, BE	Kees Voorwinden	Kees.Voorwinden@ascom.nl
Ascom Austria, AT	Bernhard Muller	Bernhard.muller@ascom.com
Ascom Sverige, SE	Charlotta Nordelöf	Charlotta.nordelöf@ascom.se
Exhibo SpA, IT	Domenico Pirillo	domenico.pirillo@exhibo.it
International	Marko Savinainen	marko.savinainen@ascom.se

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's product family.
The program provides tools and support for developing, verifying and promoting compliant third-party applications that complement Alcatel-Lucent's product family. Alcatel-Lucent facilitates market access for compliant applications.

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

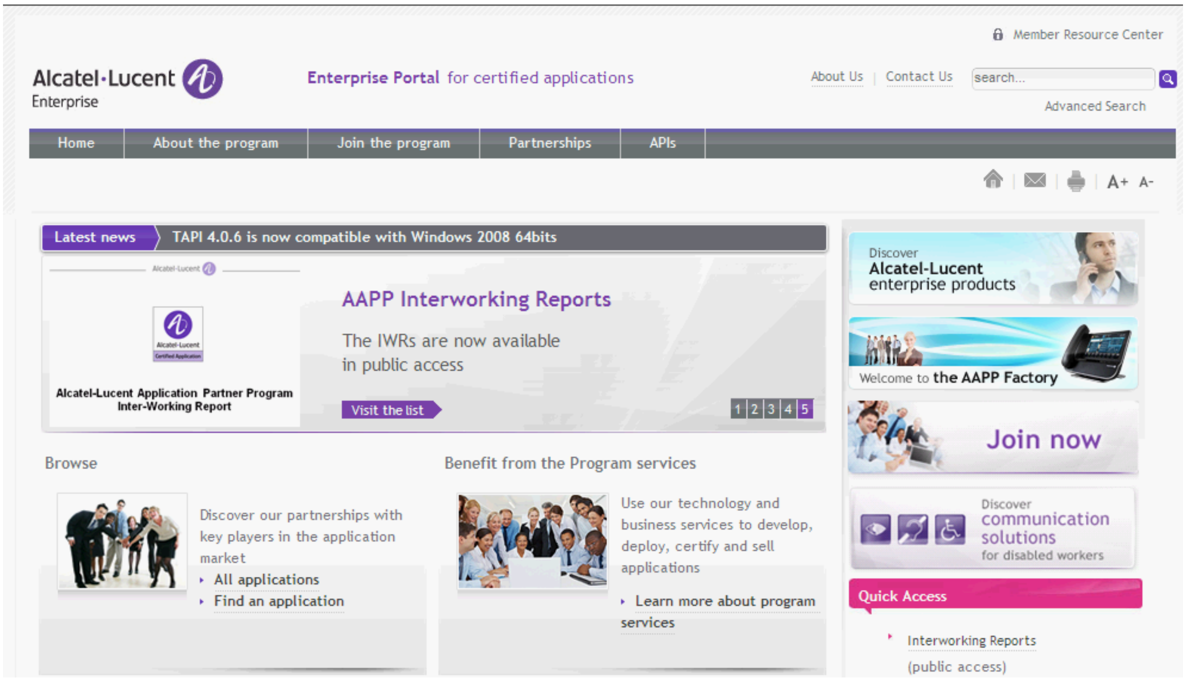
- **Provide easy interfacing for Alcatel-Lucent communication products**: Alcatel-Lucent's communication products for the enterprise market include infrastructure elements, platforms and software suites. To ensure easy integration, the AAPP provides a full array of standards-based application programming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

- **Test and verify a comprehensive range of third-party applications**: to ensure proper inter-working, Alcatel-Lucent tests and verifies selected third-party applications that complement its portfolio. Successful candidates, which are labelled Alcatel-Lucent 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.

13.2 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
http://applicationpartner.alcatel-lucent.com



13.3 Alcatel-Lucent.com

You can access the Alcatel-Lucent website at this URL: http://www.Alcatel-Lucent.com/

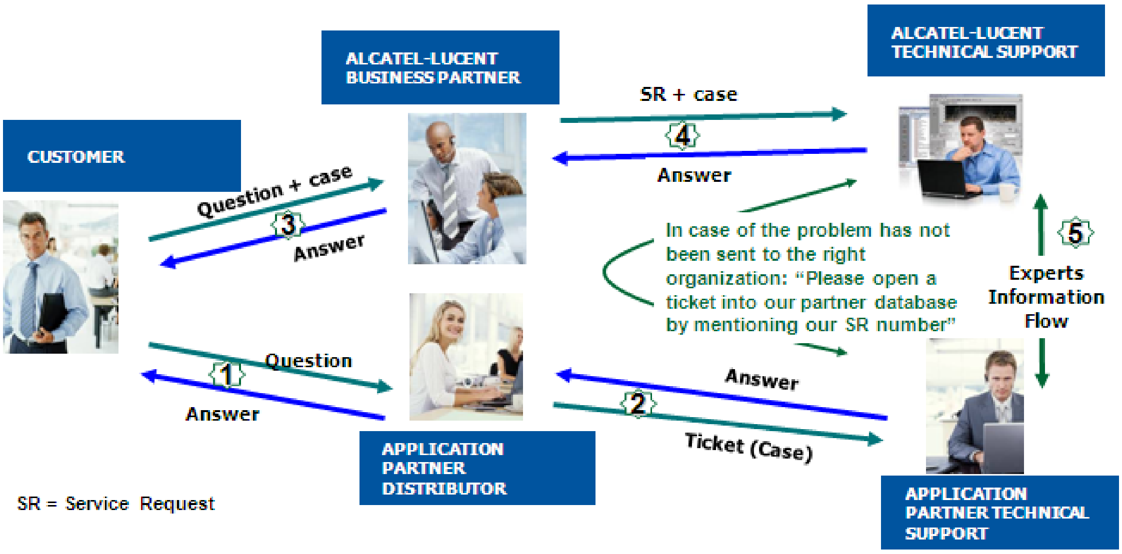
14 Appendix F: AAPP Escalation process

14.1 Introduction

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

The principle is that Alcatel-Lucent 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, Alcatel-Lucent and the Application Partner are engaged as following:



(*) The Application Partner Business Partner can be a Third-Party company or the Alcatel-Lucent 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, Alcatel-Lucent and the Application Partner, are engaged:

Case 1: the responsibility can be established 100% on Alcatel-Lucent side.
In that case, the problem must be escalated by the ALU Business Partner to the Alcatel-Lucent 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 Alcatel-Lucent Business Partner will escalate the problem to the Alcatel-Lucent Support Center only if the Application Partner has demonstrated with traces a problem on the Alcatel-Lucent side or if the Application Partner (not the Business Partner) needs the involvement of Alcatel-Lucent.

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

Alcatel-Lucent 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, Alcatel-Lucent offers the "On Demand Diagnostic" service where Alcatel-Lucent will provide 8 hours assistance against payment.

IMPORTANT NOTE 1: The possibility to configure the Alcatel-Lucent PBX with ACTIS quotation tool in order to interwork with an external application is not
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://private.applicationpartner.alcatel-lucent.com) or Enterprise Business Portal (Url: Enterprise Business Portal) web sites.

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

14.3 Escalation in all other cases

These cases can cover following situations:

1. An InterWorking Report exist but is not valid (see Chap 2 “Validity of an Interworking Report”)

2. The 3rd party company is referenced as AAPP participant but there is no official InterWorking Report (no IWR published on the Enterprise Business Portal for Business Partners or on the Alcatel-Lucent Application Partner web site) ,

3. The 3rd party company is NOT referenced as AAPP participant

In all these cases, Alcatel-Lucent offers the “On Demand Diagnostic” service where Alcatel-Lucent will provide 8 hours assistance against payment.

14.4 Technical support access

The Alcatel-Lucent **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): http://applicationpartner.alcatel-lucent.com
- e-Support from the Alcatel-Lucent Business Partners Web site (if registered Alcatel-Lucent Business Partners): https://businessportal.alcatel-lucent.com click under “Let us help you” the *eService Request* link
- e-mail: Ebg_Global_Supportcenter@alcatel-lucent.com
- Fax number: +33(0)3 69 20 85 85
- Telephone numbers:

Alcatel-Lucent 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

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