Alcatel Lucent Application Partner Program
Inter-Working Report

Partner: Ascom
Application type: IP-DECT Solution for hospital and senior care segment
Application name: IP-DECT 7.0.5
Alcatel-Lucent Platform: OmniPCX Office™

The product and version listed have been tested with the Alcatel-Lucent Communication Server and the version specified hereinafter. The tests concern only the inter-working between the Application Partner product and the Alcatel-Lucent Communication platforms. The inter-working report is valid until the Application Partner issues a new version of such product (incorporating new features or functionality), or until Alcatel-Lucent issues a new version of such Alcatel-Lucent product (incorporating new features or functionality), whichever first occurs.

ALCATEL-LUCENT MAKES NO REPRESENTATIONS, WARRANTIES OR CONDITIONS WITH RESPECT TO THE APPLICATION PARTNER PRODUCT. WITHOUT LIMITING THE GENERALITY OF THE FOREGOING, ALCATEL-LUCENT HEREBY EXPRESSLY DISCLAIMS ANY AND ALL REPRESENTATIONS, WARRANTIES OR CONDITIONS OF ANY NATURE WHATSOEVER AS TO THE APPLICATION PARTNER PRODUCT INCLUDING WITHOUT LIMITATION THE IMPLIED WARRANTIES OF MERCHANTABILITY, NON INFRINGEMENT OR FITNESS FOR A PARTICULAR PURPOSE AND ALCATEL-LUCENT FURTHER SHALL HAVE NO LIABILITY TO APPLICATION PARTNER OR ANY OTHER PARTY ARISING FROM OR RELATED IN ANY MANNER TO THIS CERTIFICATE.

Certification Overview

Date of the certification	February 2014

Alcatel-Lucent's representative	Olivier Koenig
AAPP member representative	Peter Astrand

Alcatel-Lucent Communication Platform	OmniPCX Office
Alcatel-Lucent Communication Platform release	R920 / 035.002
AAPP member application release	IP-DECT R7.0.5
Application Category	DECT / Wi-Fi
	Healthcare dedicated hardware
	Mobility

Author(s): Olivier Koenig
Reviewer(s): Denis Lienhart, Rachid Himmi, Peter Astrand, Gert Wallin

Test results

☑ Passed ☐ Refused ☐ Postponed

☐ Passed with restrictions

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

IWR validity extension

None

AAPP Member Contact Information

Contact name:	Gert Wallin
Title:	Certification Program Manager
Address:	Grimbodalen 2 Box 8783
Zip Code: **City**:	402 76 Göteborg
Country:	SWEDEN
Phone: **Fax**: **Mobile Phone**:	+46 (0) 31 55 93 00 +46 (0) 31 55 20 31
Web site: **Email address**:	http://www.ascom.com gert.wallin@ascom.se

TABLE OF CONTENTS

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.

Note 1: This interworking report does not cover mass provisioning and/or remote device management of the partner device.

Note 2: This interworking report does not cover specific DECT coverage and/or multi-base station and/or multi-site scenarios including roaming/handover.

2 Validity of the Inter-Working 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 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 Inter-Working 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").

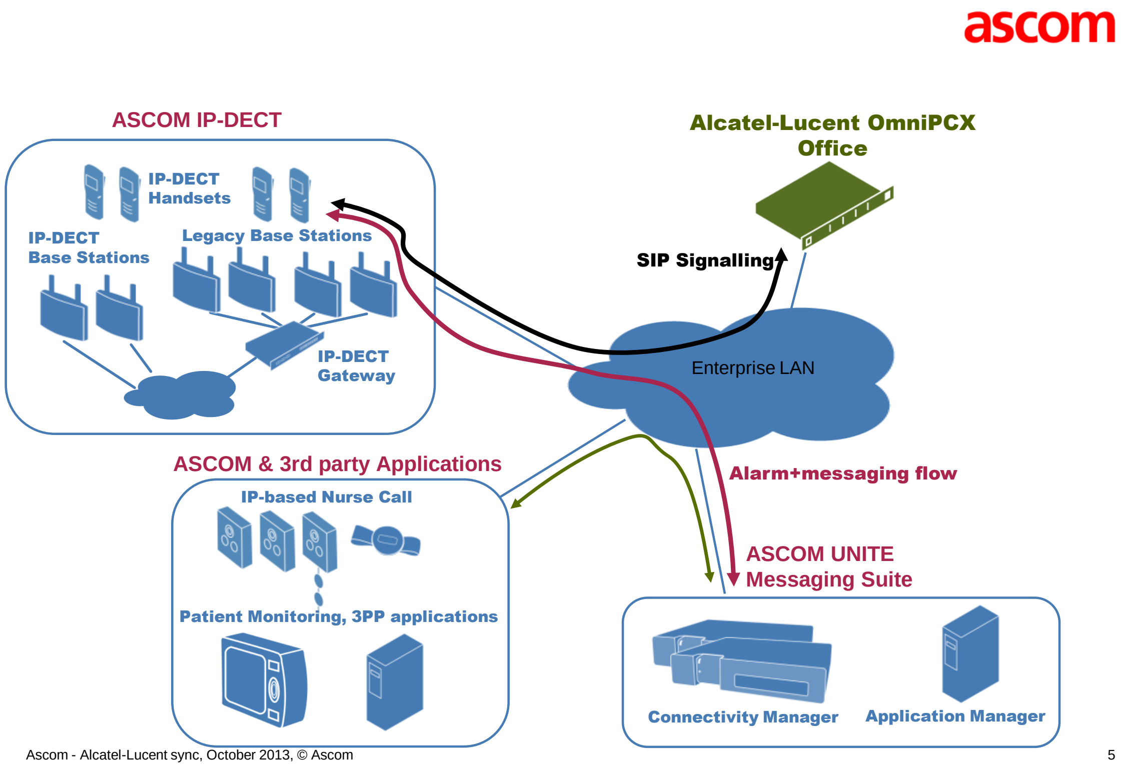
4 Application information

Application type:	SIP IP-DECT
Application commercial name:	IP-DECT d81, d62, d41 handsets for hospital and senior care segment
Application version:	7.0.5
Interface type :	SIP

Brief application description:

The Ascom IP-DECT infrastructure and handsets integrates smoothly with the Alcatel-Lucent OmniPCX Office as an excellent mobility solution for several verticals. Combined with the Ascom Nurse Call, Patient Monitoring and Unite Messaging Suite, it develops into an excellent solution specifically for the smaller hospital and senior care establishments refer to the system view below. The solution has the capability to provide primarily a secure and safe communication environment for the patient, but also be efficient and cost-effective for the caregiver staff.

The application consists of IP-DECT base stations and associated Ascom handsets. IP- DECT base stations are linked to OXO via SIP protocol. All telephony features as provided by SIP Endpoint Level of Service are available to the handsets. The Ascom & 3rd party applications and the Ascom Unite Messaging Suite complete the solution.



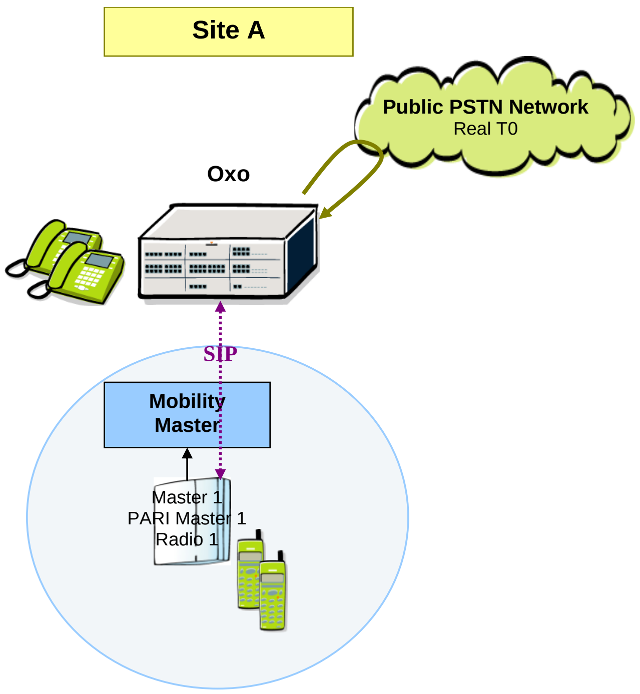
The Ascom handsets which are supported by the solution are the following:

Ascom d81

Ascom d62

Ascom d41

5 Test Environment



5.1 Hardware configuration

Alcatel-Lucent Communication Platform:

- OmniPCX Office Rack
- PowerCPU
- Release: R920 / 035.002
- OMC: R920 / 23.001

Setup Details:

Setup Information OXO	
OXO 1 IP address	**192.168.92.246**
Voicemail No	**500**
Attendant No	**9**
OXO Extension Details used for test	
IP Touch numbers	**100, 101 & 141**
IP-DECT Dir numbers	**138, 139 and 140**

5.2 Software configuration

- **Alcatel-Lucent Communication Platform**: OmniPCX Office R920/035.002
- **Partner Application**: IP-DECT 7.0.5 + Ascom handsets

Note: Ascom handsets are 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

- None

6.3 Summary of limitations

- **None**

6.4 Notes – remarks

- **Note 1: Test Ch 8.3 - Test Case 5:**
  - By default, the IP-DECT "Call waiting" feature is "Off". In this case, an incoming call to a busy IP-DECT set will generate a "486 Busy" message to OXO. If the caller is another SIP device, Oxo will generate a "500 Internal server error" on the second call leg. → eSR/1-157783293.
  - If the "Call waiting" feature is turned "on", then the set is able to manage a second call (and the caller will get the "free" RBT, thus the "Busy" state can be managed by the OXO (via the subscriber feature rights)

- **Note 2: Test Ch 8.3 - Test Case 7:**
  - OXO generates a "480 temporarily not available" in case of a DND feature activation.

- **Note 3: Test Ch 8.4 - Test Case 12a:**
  - The length of the internal number plan is configured in the IP-DECT .
  - So if we configure a 3 digits num plan in OXO, and there is an incoming call with secrecy ("xxxx" = 4 digits) → The call is considered as being "External".

- **Note 4:** IP-DECT SIP phones are 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 the and the expected result
N/A: when checked, means the test case is not applicable in the scope of the application
OK: when checked, means the test case performs as expected
NOK: when checked, means the test case has failed. In that case, describe in the field "Comment" the reason for the failure and the reference number of the issue either on 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 x138, 139 & 140 to register with the OXO IP address Check the registration on your sets and the display	☐	☒	☐	
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 138, 139 & 140 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 xxx 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 xxx 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 xxx 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 IP-DECT 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 IP-DECT Call from SIP 138 to IP Touch 141 Check that the call is established in G711 A-law. Check audio quality Call from IP Touch 141 to SIP 138 Check that the call is established in G711 A-law. Check audio quality	☐	☒	☐	
2	Select G729 as 1st codec in IP-DECT Call from SIP 138 to IP Touch 141 Check that the call is established in G729 Check audio quality Call from IP Touch 141 to SIP 138 Check that the call is established in G729 Check audio quality	☐	☒	☐	
3	Select G723 as 1st codec in IP-DECT Check that the call is established in G723 Check audio quality Call from IP Touch xxx to SIP xxx Check that the call is established in G723 Check audio quality	☐	☒	☐	
4	**Configure** 138 **to use VAD** **Configure IP Touch** 141 **NOT to use VAD** Call from SIP x138 to IP Touch 141 Check that the call is established in G711 A-law. Check audio quality **Configure SIP** 138 **to use VAD** **Configure IP Touch** 141 **to use VAD** Redo the same tests. **Configure SIP**138 **NOT to use VAD** **Configure IP Touch** 141 **to use VAD** Redo the same tests	☐	☒	☐	
5	**In OXO enable codec pass through for SIP phones.** **Call from SIP** xxx **to SIP** xxx Check that the call is established using G.729 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 expert doc.

Note: dialing will be based on direct dialing number.

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Call to a local user** With SIP Phone 138 call the IP Touch 141. Check that 141 is ringing. Take the call and check ring back tone audio and display.	☐	☒	☐	
2	**Call to local user with no answer** With SIP Phone 138 call the IP Touch 141. And never take the call. Check time out (if any) and display. Note that 141 don't have a Voice Mail	☐	☒	☐	141 is ringing until 138 releases the call – No Time out.
3	**Call to another SIP set** With the SIP phone 138 call the other SIP Phone 140 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 138 call a wrong number Check the ring back tone and display	☐	☒	☐	
5	**Call to busy user** **(SIP: "486 Busy Here")** With the SIP phone 138 call IP Touch 141, take the call and don't hang up. With other SIP phone139 call 138 which is busy Check the ring back tone and display	☐	☒	☐	**See note 1 / ch 6.4**
6	**Call to user in "Out of Service" state** **(SIP: "480 Temporarily Unavailable")** With the SIP phone xxx call the IP Touch xxx which is in "Out of Service State" Check the display and ring back tone	☐	☒	☐	
7	**Call to user in "Do not Disturb" (DND) state** **(SIP: "480 Temporarily not available")** Dial DNDon the IP Touch 141 in order to enable the DND. Wait for acknowledgement ring back tone from OXO. With the SIP phone 138 call 141. Check ring back tone and display.	☐	☒	☐	**See note 2 / ch 6.4**

Test Case Id	Test Case	N/A	OK	NOK	Comment
8	**Call to local user, immediate forward (CFU). (SIP: “181 Forwarded”)(1)** On IP Touch 141, activate the CFU. Wait for acknowledgement ring back tone from OXO. With the SIP phone 138 call the 141. Check that x100 is ringing and the display. Take the call check audio and hung up.	☐	☒	☐	
9	**Call to local user, forward on no reply (CFNR). (1)** On IP Touch 141 configure with OMC the CFNR using dynamic routing to 100. With 138 call the 141. Check that 100 is ringing but don’t take the call and wait the time out. Time out is defined in xxx dynamic routing of Timer 1. After time out check that 100 is ringing and take the call. Check the audio and display.	☐	☒	☐	NO COLP on 138
10	**Call to local user, forward on busy (CFB). (1)** On IP Touch xxx dial the xxx(xxx+<target MCDU number>) to activate the CFB. Wait for acknowledgement ring back tone from OXO. With SIP phone xxx call xxx and take the call to make it busy. With other SIP phone xxx call xxx. Check that xxx is ringing and take the call. Check the audio and display. Dial xxx on xxx for forward cancellation.	☒	☒	☐	NA
11	**Call to external number (Check ring back tone, called party display)** With SIP set 138 dial 0 (0 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 138 call IP Touch 141. Take the call on 141 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.

OXO prefixes are mandatory for several tests of this section. For more information refer to the Expert Doc.

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Local call to free SIP terminal** Local: with IP Touch 141 call SIP set 138. Check that 138is ringing and take the call Check ring back tone and called party display. Check ring back tone and called party display.	☐	☒	☐	
2	**Local call to busy SIP terminal** Local: With SIP set 138 call other SIP set 139 and take the call to make it busy, don't hang up. With IP Touch 141 call 138 which is busy Check the ring back tone and display. Network: With SIP set xxx call SIP set xxx and take the call to make it busy, don't hang up. With xxx call xxx which is busy Check ring back tone and called party display.	☐	☒	☐	**See note 1 / ch 6.4**
3A	**Local call to "unplugged" SIP terminal** Local: Switch off the 138 SIP set and call it with IP Touch 141. Check the ring back tone and display Check the ring back tone and display	☐	☒	☐	Call is routed to VMU
3B	**Local call to "unplugged" SIP terminal** Local: Remove battery of the 138 SIP set and call it with IP Touch 141. Check the ring back tone and display Check the ring back tone and display	☐	☒	☐	After 12s AP sends 603_decline
4A	**Local call to SIP terminal in Do Not Disturb (DND) mode** **By local feature if applicable:** Local: Enable DND on SIP set 139 and call it with IP Touch 141 Check the tone and display in the IP Touch phone Cancel the DND on 139	☐	☒	☐	SIP sends Busy (not DND)
4B	**By system feature** Local: Enable DND on SIP set using the 793 prefix.. Wait for acknowledgement ring back tone from OXO. With IP Touch 141 call 138 Check the ring back tone and display Cancel the DND on xxx using 790 prefix.	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
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 138 enable CFU to IP Touch 100 With SIP set 139 call 138. Check that 100 is ringing. Take the call and check audio and display. Disable CFU on 138. Disable CFU on 138.	☐	☒	☐	
5B	**By system feature:** **By local feature if applicable:** <u>Local</u>: On SIP set 138 enable CFU to IP Touch 100 With SIP set 139 call 138. Check that 100 is ringing. Take the call and check audio and display. Disable CFU on 138. Disable CFU on 138.	☐	☒	☐	
6A	**Local SIP call to SIP terminal in immediate forward on busy (CFU) to local number:** **By local feature if applicable:** <u>Local</u>: On SIP Set 138 enable CFU to SIP Set 139.With SIP set 140 call 138. Check that 139 is ringing. Take the call and check audio and display. Disable CFU on 138.	☐	☒	☐	
6B	**Local SIP call to SIP terminal in immediate forward on busy (CFU) to external number:** **By local feature if applicable:** <u>Local</u>: On SIP Set 138 enable CFU to Ext. With SIP set 140 call 138. Check that Ext is ringing. Take the call and check audio and display. Take the call and check audio and display. Disable CFU on 138.	☐	☒	☐	
6C	**By system feature:** <u>Local</u>: On SIP Set 138 enable CFU to SIP Set 139 using 791 prefix (0+ <target MCDU number>). Wait for acknowledgement ring back tone from OXO. With SIP set 140 call 138. Check that 139 is “externally” ringing. Take the call and check audio and display. Disable CFU on 138 using 790 prefix.	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
7A	**Local SIP call to SIP terminal in immediate forward (CFU) to another SIP user** **By local feature if applicable:** <u>Local</u>: On SIP set 138 enable CFU to SIP set 139 With 141 call 138. Check that 139 is ringing. Take the call and check audio and display. Disable CFU on 138.	☐	☒	☐	
7B	**By system feature:** <u>Local</u>: On SIP Set 138 (which has no feature right) enable CFU to external using 791 prefix (791+ <target MCDU number>). Veridy that the feature is refused.	☐	☒	☐	IP Set displays "Temporary failure"
7C	**By system feature:** <u>Local</u>: On SIP Set 138 enable CFU to external using 791 prefix (791+ <target MCDU number>). Wait for acknowledgement ring back tone from OXO. With SIP Set 139 call 138. Check that Ext is ringing. Take the call and check audio and display. Disable CFU on 138 using 790 prefix.	☐	☒	☐	
8A	**Local SIP call to SIP terminal in immediate forward on busy (CFB) to local number:** **By local feature if applicable:** <u>Local</u>: On SIP Set 138 enable CFU to SIP Set 139.With SIP set 140 call 138. Check that 138 is ringing. Take the call and check audio and display. <u>Local</u>: On SIP Set 138 call 141 and stay in conv. With SIP set 140 call 138. Check that 139 is ringing. Take the call and check audio and display. Disable CFU on 138.	☐	☒	☐	
8B	**Local SIP call to SIP terminal in immediate forward on busy (CFB) to external number:** **By local feature if applicable:** <u>Local</u>: On SIP Set 138 enable CFU to Ext. With SIP set 140 call 138. Check that 138 is ringing. Take the call and check audio and display. <u>Local</u>: On SIP Set 138 call 141 and stay in conv. With SIP set 140 call 138. Check that Ext is ringing (and indicating 138's ect CLIP). Take the call and check audio and display. Disable CFU on 138.	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
8C	**By system feature:** <u>Local</u>: On SIP Set 138 enable CFU to Ext (via 792 prefix). With SIP set 140 call 138. Check that 138 is ringing. Take the call and check audio and display. <u>Local</u>: On SIP Set 138 call 141 and stay in conv. With SIP set 140 call 138. Check that Ext is ringing (and indicating 138's ect CLIP). Take the call and check audio and display. Disable CFU on 138.	☐	☒	☐	
9A	**Local call to SIP terminal in "forward on no reply" (CFNR)** **By local feature if applicable** On SIP Set 138 enable CFNR to IP Touch 141 With SIP Set 139 call 138. Check that 138 is ringing and don't take the call, wait for time out (about 15 seconds). After time out expiration the 141 is ringing, take the call and check audio and display.	☐	☒	☐	
9B	**By system feature:**	☐	☒	☐	CFNR 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 138 call other SIP Set 139 (multiline set) to make it busy, take the call and don't hang up. With IP Touch 141 call 138(on 138 camp-on feature is enabled). Check the Call waiting or ring back tones and display	☐	☒	☐	
11	**External call to SIP terminal.** Check that external call back number is shown correctly: With SIP Set 138 dial 0 + 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.	☐	☒	☐	
12A	**Calling Line Identity Restriction (CLIR): Local call to SIP terminal.** On IP Touch 141 enable mask Identity and call SIP Set 138 in order to hide 141 identity. Check that 138 is ringing, take the call and check that 141 identity is hidden.	☐	☒	☐	**See note 3 / ch 6.4**
12B	**Calling Line Identity Restriction (CLIR): Local call from SIP terminal.** On SIP set 138 enable mask Identity and call SIP Set 139 in order to hide 138 identity. Check that 139 is ringing, take the call and check that 138 identity is hidden.	☐	☒	☐	Must be done through PBX feature Right configuration "Identity Secrecy"

Test Case Id	Test Case	N/A	OK	NOK	Comment
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.	☐	☒	☐	
15	SIP set is part of a sequential hunt group **(1)**. Call to hunt group. Check call/release. With IP Touch 141 call the sequential hunt group MCDU number 501 Check that 138 is ringing Take the call and don't hang up. And with IP Touch 100 call the sequential hunt group MCDU number 501 Check that 139 is ringing Take the call and don't hang up. And with SIP Set 140 call the sequential hunt group MCDU number 501 Check that the group is busy	☐	☒	☐	
16	SIP set is part of a cyclic hunt group **(2)**. Call to hunt group. Check call/release. With IP Touch 141 call the cyclic hunt group MCDU number 502 Check that 138 is ringing Take the call and hang up. And with 141 call the cyclic hunt group MCDU number 502 Check that 139 is ringing Take the call and hang up. And with SIP Set 138 call the cyclic hunt group MCDU number 502 Check that 140 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 141 call IP Touch 100 which is in MultiSet with SIP Set138. Check that 100 and 138 both ringing. Take the call from 100 and check that 138 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 expert doc.

Test Case Id	Test Case	N/A	OK	NOK	Comment
1A	**Hold and resume with local feature** (if applicable) With 138 call 139 take the call, check audio and display. With 138 put 139 on hold check tones and display on both and resume the call. Keep this call for the next test.	☐	☒	☐	
1B	**Enquiry call to another local user** (if applicable) Distant user is put on hold with local feature With 138 call 140 and take the call. 139 will be put on hold when making second call to 140 Put 140 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 138 switch between 139 and 140 lines. Check the tones and display on sets on hold state. Keep these two calls for the next test.	☐	☒	☐	
1D	There will be two active calls from the previous test case execution. In that release the first call. Keep second call. Hang up 140 and only 138 and 139 are in call Check that 138 & 139 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 device
3	**Three party conferences initiated from OXO set** With 141 call 138, take the call and don't release it. With 141 call 139, take the call and don't release it too. With 141 start a conference. Check that 141, 138 and 139 are in conference. Check audio and display.	☐	☒	☐	SIP device 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 xxx call xxx take the call and don't release it. With xxx call xxx, take the call and don't release it too. With xxx start a conference by the local feature Check that xxx, xxx and xxx are in conference. Check audio and display.	☒	☐	☐	NA
4B	**Three party conferences initiated from SIP set with OXO feature.** With xxx call xxx take the call and don't release it. With xxx call xxx, take the call and don't release it too. With xxx start a conference by the OXO suffix code.	☒	☐	☐	Conference feature is not supported within call server for SIP devices with OXO suffix
5	**Meet Me conference** With 138 call the Meet me Conference bridge dialing prefix and follow instruction to open the bride. With 139 join the conference bridge by dialing prefix and enter access code. With Ext join the conference bridge by dialing prefix and enter access code. Check that 138, 139 and Ext 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 is 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.

Semi-Attended Transfer

Test	Action			Result	Comment
	A	**B**	**C**		
	Transferee	**Transferor**	**Transfer Target**		
1	OXO	SIP	OXO	OK	ok
2	Ext Call	SIP	OXO	OK	ok
3	Ext Call	SIP	Ext Call	OK	Ok but no colp both ways
4	SIP	SIP	SIP	OK	Ok (no RBT on Transferee)
5	SIP	OXO	OXO	OK	ok
6	Ext Call	OXO	SIP	OK	Ok but no colp on transferee
7	SIP	OXO	SIP	OK	ok

Attended Transfer

Test	Action			Result	Comment
	A	B	C		
	Transferee	Transferor	Transfer Target		
1	OXO	SIP	OXO	OK	ok
2	Ext Call	SIP	OXO	OK	Ok but no colp on transferee
3	Ext Call	SIP	Ext Call	OK	Ok but no colp on both ways
4	SIP	SIP	SIP	OK	ok
5	SIP	OXO	OXO	OK	ok
6	Ext Call	OXO	SIP	OK	Ok but no colp both ways
7	SIP	OXO	SIP	OK	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 138 dial "9" (attendant call prefix) Check audio and display	☐	☒	☐	
2	**2nd incoming call while in conversation with attendant** While SIP set 138 is in conversation with the attendant, from IP Touch 141 call 138 Answer the call and check audio and display	☐	☒	☐	
3	**SIP set call to attendant, attendant transfers to OXO set, semi-attended** From SIP set 138 dial "9" (attendant call prefix) and answer. Attendant transfer semi-attended to IP Touch 141 Answer the call and check audio and display	☐	☒	☐	
4	**SIP set call to attendant, attendant transfers to OXO set, attended** From SIP set 138 dial "9" (attendant call prefix) and answer Attendant transfer attended to IP Touch 141 Check audio and display	☐	☒	☐	
5	**OXO set calls to attendant, attendant transfers to SIP set, attended** From IP Touch 141 dial "9" (attendant call prefix) and answer Attendant transfer attended to SIP set 138 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 138 and transfer attended Check audio and display	☐	☒	☐	
7	**SIP set call to attendant, attendant transfers to External** From SIP set 138, 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 67, and this service is enabled on SIP sets 138, 139, 140 and OXO sets.

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 138 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 139 call the Voice Mail at 500. Follow the instructions in order to send a voice message in SIP set 138 boxes. Check that the MWI on 138 is activated.	☐	☒	☐	
3	Message consultation With SIP set 138 call the Voice Mail at 500(or 67). Follow the instructions in order to listen your voice message leaved during the previous test. Check that your can listen it and delete. Check that MWI display is disabled on 138 after message cancellation.	☐	☒	☐	
4	SIP call to a OXO user forwarded to Voice Mail Forward the IP Touch 141 to Voice Mail . With SIP set 138 call 141 and check that you are immediately forwarded to Voice Mail. Check that you can leave a message	☐	☒	☐	
5	OXO set call to a SIP user forwarded to Voice Mail Forward the SIP set 138 to Voice Mail by dialing 791 (791 prefix + 500) With IP Touch 141 call 138 and check that you are immediately forwarded to Voice Mail. Check that you can leave a message On 138 disable Voice Mail forwarding with 790 prefix.	☐	☒	☐	

Notes:

(1) On SIP sets, in order to enable the MWI feature, you have to configure the Voice Mail number “mailing” function of the internal numbering plan.

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

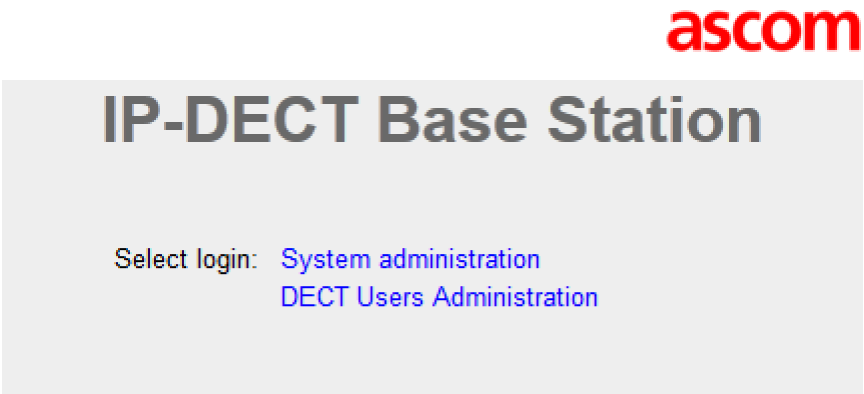
Test Case Id	Test Case	N/A	OK	NOK	Comment
1	OXO Reboot Establish an incoming ISDN call with SIP set-1. Reboot the OXO. When the OXO is up again, re-establish an incoming ISDN call with SIPset-1 and check the audio.	☐	☒	☐	
2	**Ethernet link failure** Disconnect the Ethernet link of SIP set-1. Check that 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 voice mailbox

9 Appendix A: Partner Application: configuration requirements

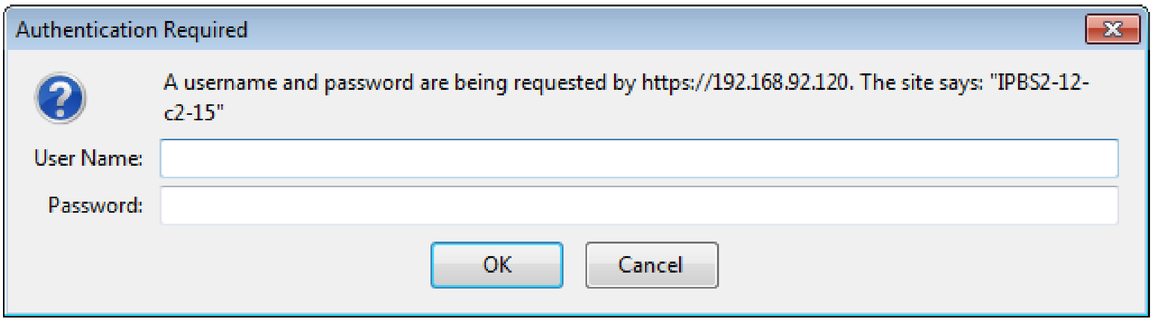
For configuration of the Ascom IP-DECT system, see the corresponding Ascom documentation.

The following screenshots show only the specific configuration parameters needed for interworking with the OmniPCX Office, and only if it differs from default configuration.

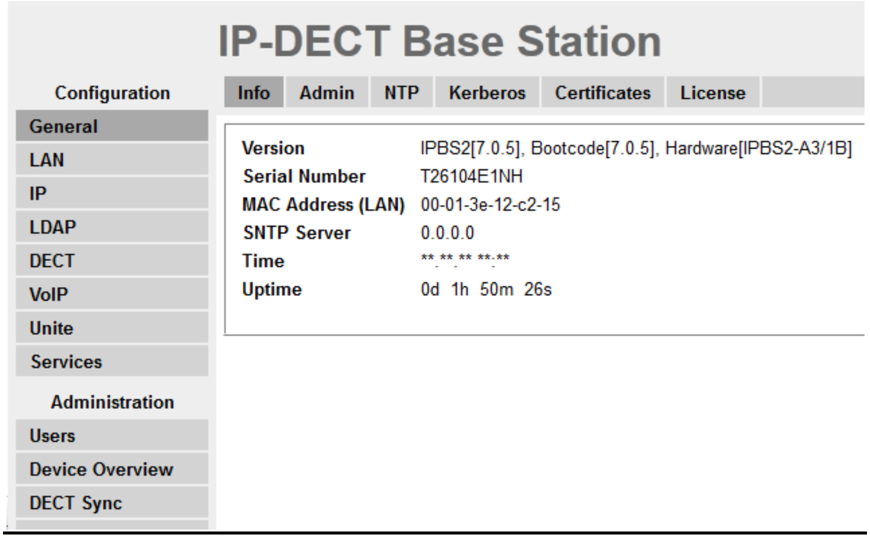
1. On the web browser , enter the IP-DECT base station IP Address (retrieved by DHCP) or Netbios name(htpp://ipbs2-xx-xx-xx 'with xx= last 6 digits of mac address')



2. Enter the administration mode

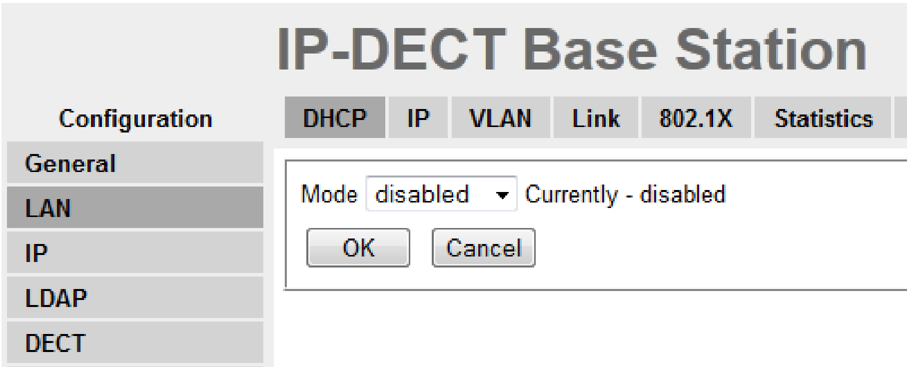


3. Enter the administrator login "**admin**"
4. Enter the administrator password ("**changeme**")
5. You will access the Homepage of the IP-DECT base station

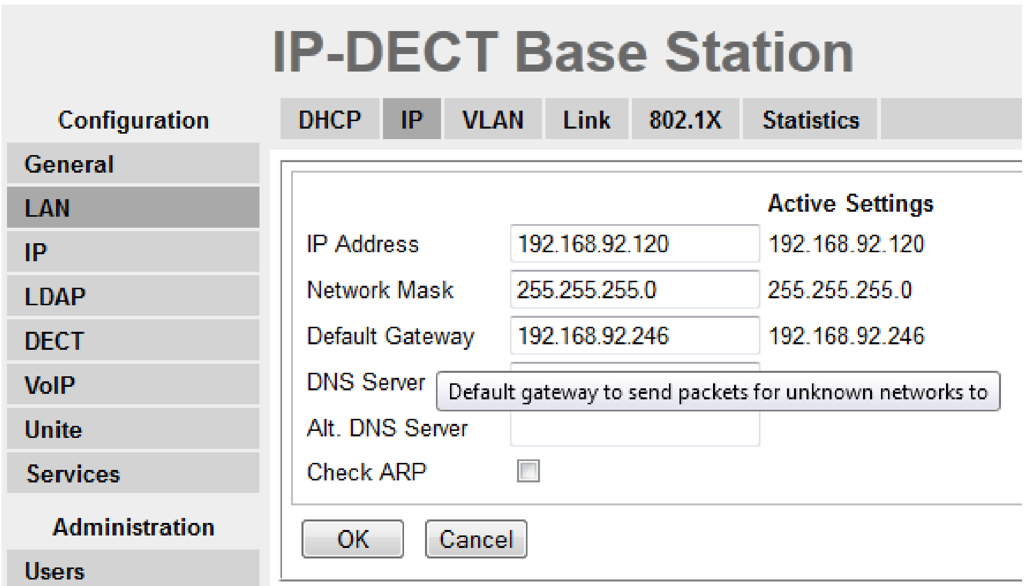


Network parameters can be configured under the “**LAN**” choice

You can either choose the **DHCP** mode

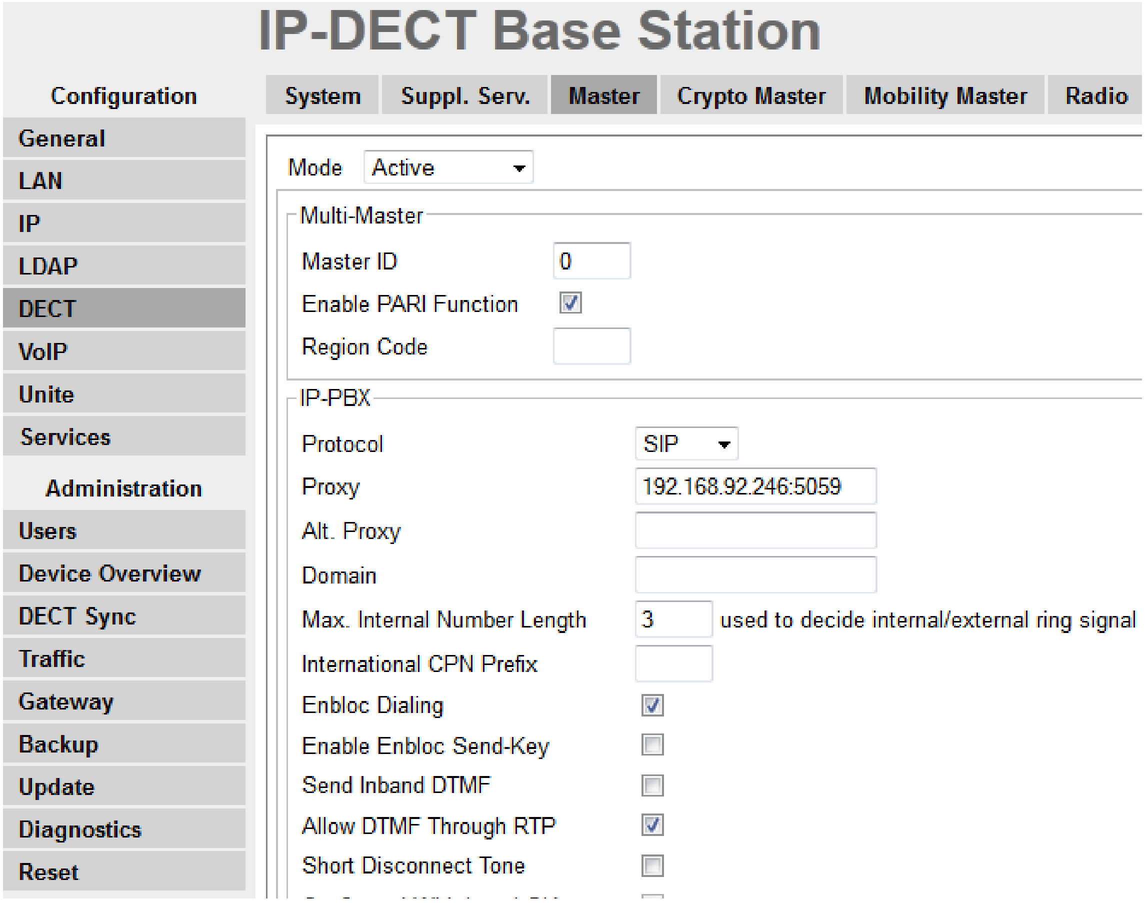


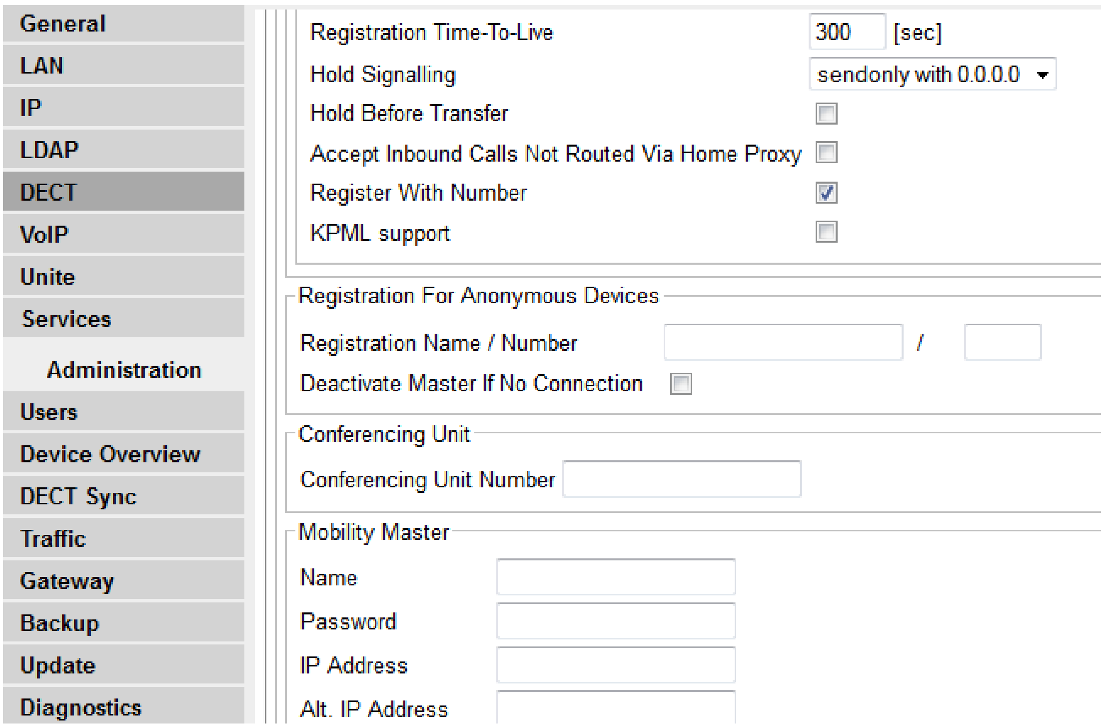
Or the **Static IP** addressing



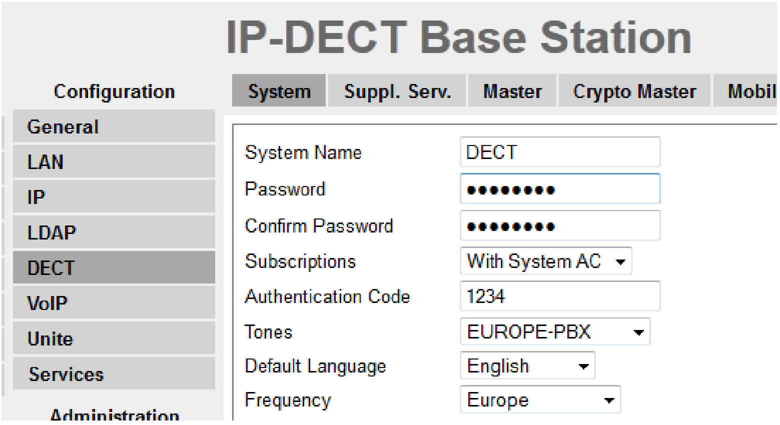
Activate the “Master” mode / restart the Base & Reconnect

Configure the parameters as described here after (**note the OXO IP address and SIP phone port**)

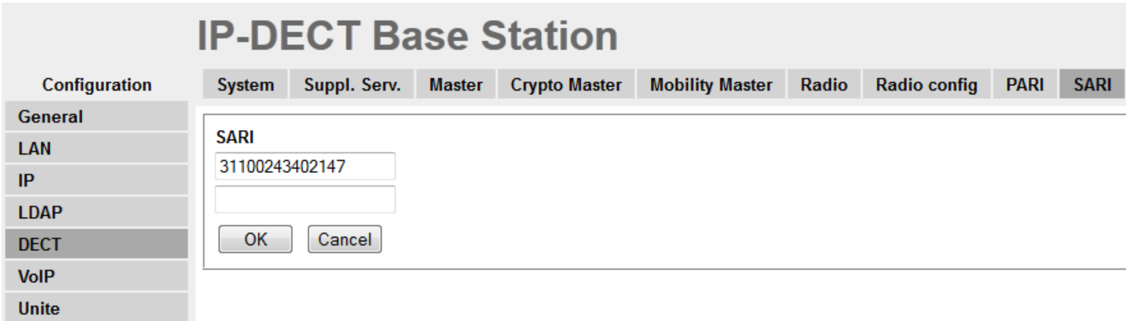




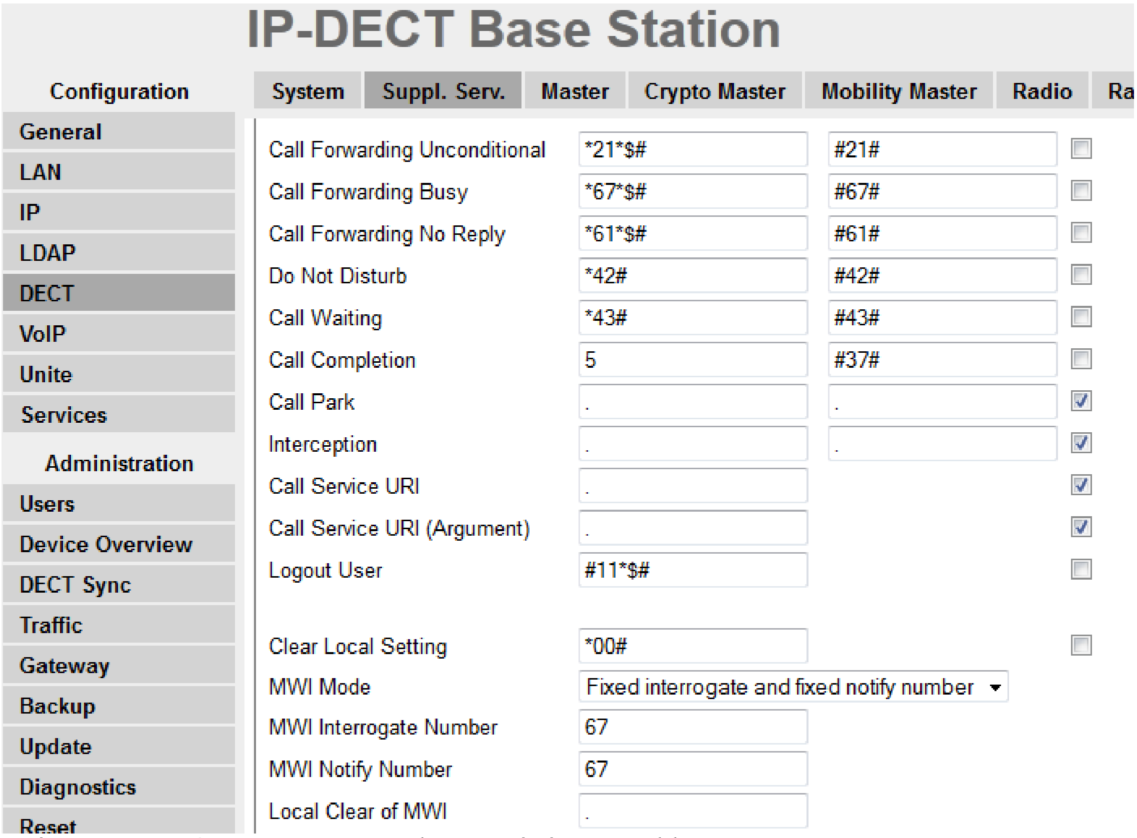
Under “**DECT**” then “**System**”, define a system password (any)



Define the System “**SARI**”

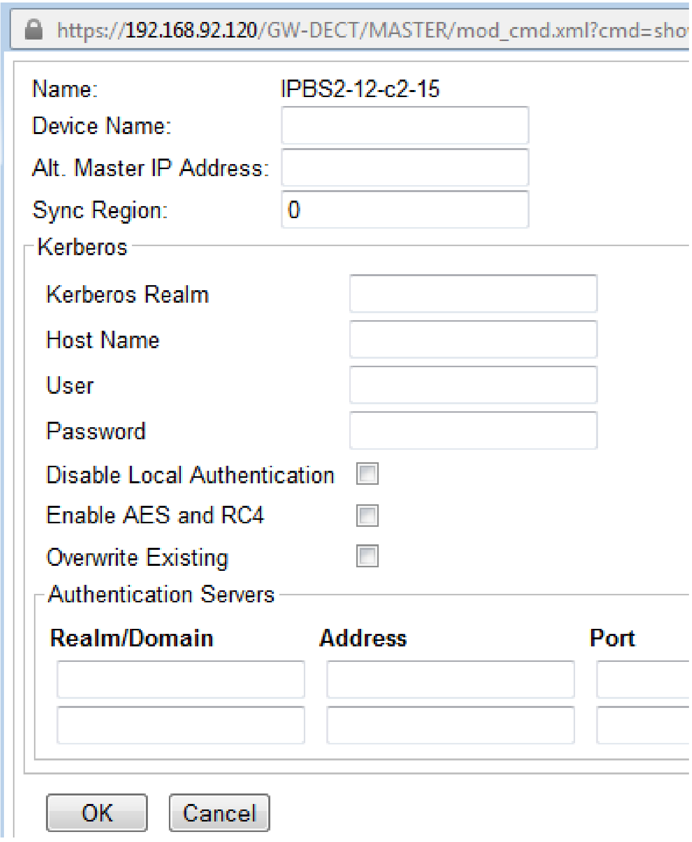


Under « **Suppl. Serv** » , configure the OXO “Mailing” code (Internal numbering Plan) for the MWI feature.

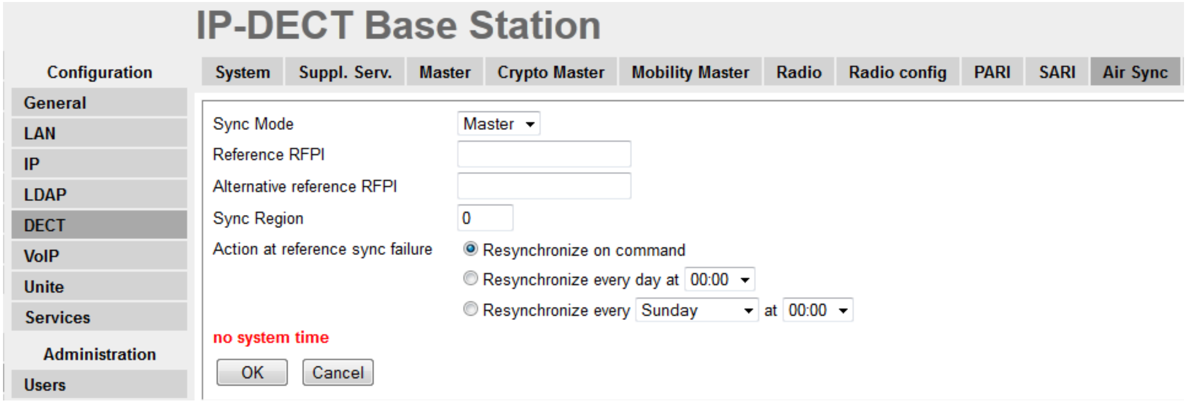


Under « **Device Overview** » , « **Radios** » , click on « **Add** »

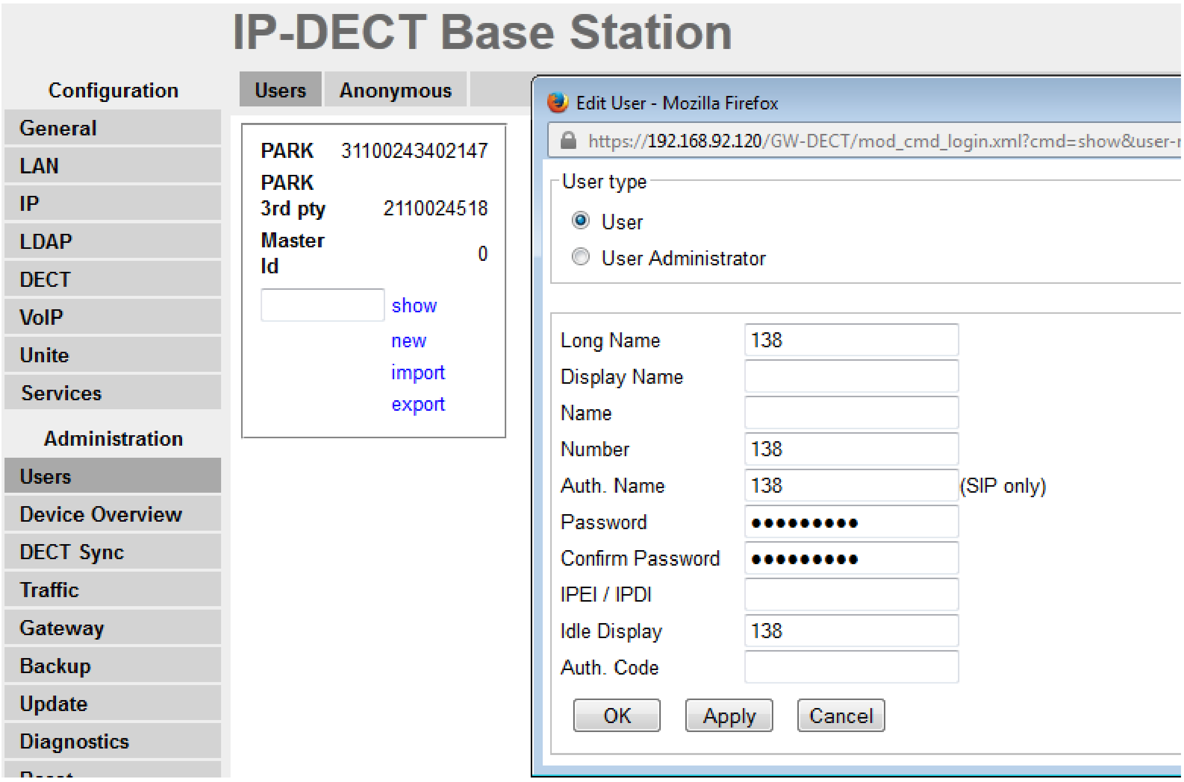
Then on « **OK** » of following screen (optional infos –refer to Ascom docs. For details)



And under « **DECT** » , « **Air Sync** », verify that Sync Mode is set to **Master** and click “OK”

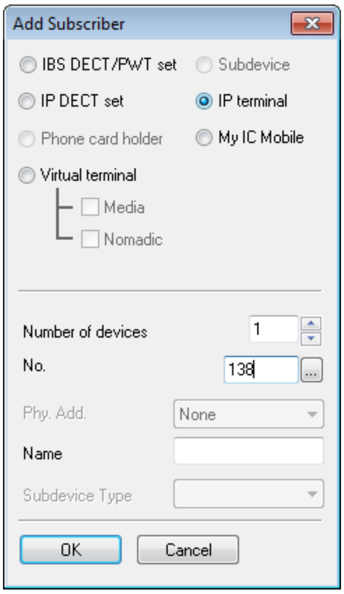


Create the IP-DECT Users (same edn and SIP password as those created in "Subscriber/Base station list" with OMC), through **"Users" then "new"**.

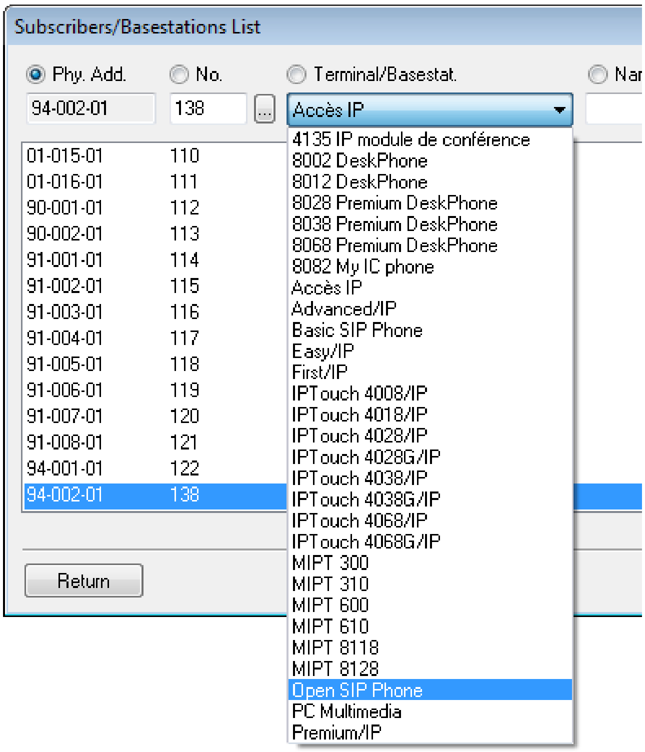


10 Appendix B: Alcatel-Lucent Communication Platform: configuration requirements

-Create an IP terminal in Subscribers.



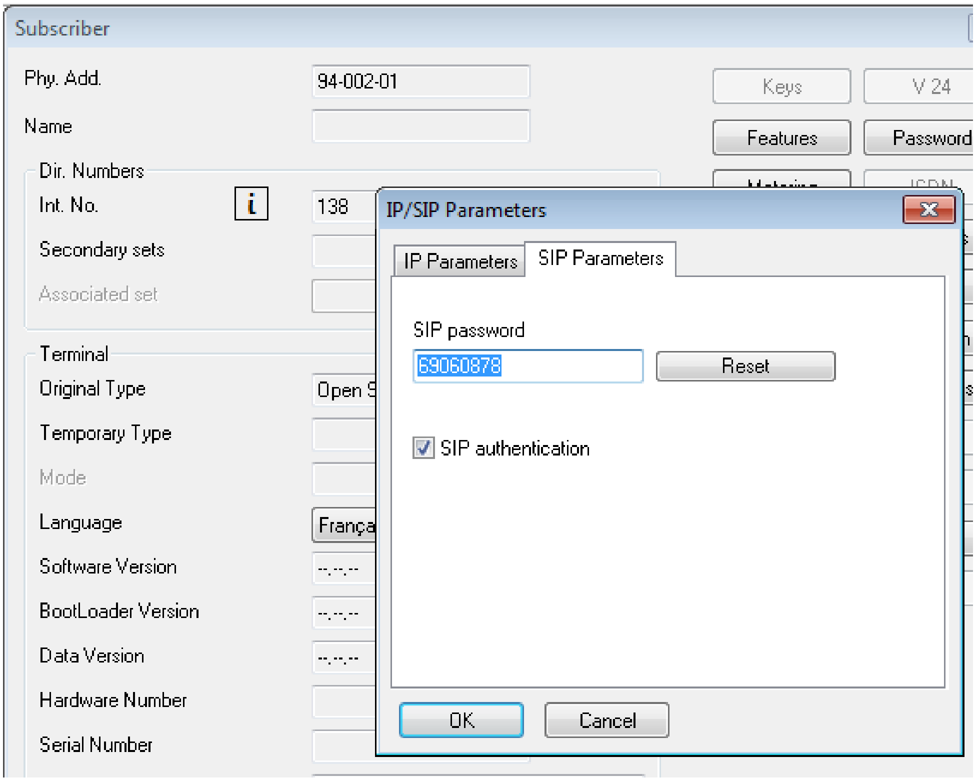
-Then modify the IP terminal to “Open SIP Phone”



-After modifying to open sip extension double click on it and click in IP/SIP button.

This will launch the IP and SIP parameters.

-Under SIP parameters make a note of the SIP password and this has to be specified under the SIP account authentication password in the IP-DECT users.



11 Appendix C: AAPP member's escalation process

The following list of contacts can be used to escalate issues regarding the Ascom IP-DECT platform:

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

12 Appendix D: AAPP program

12.1 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:

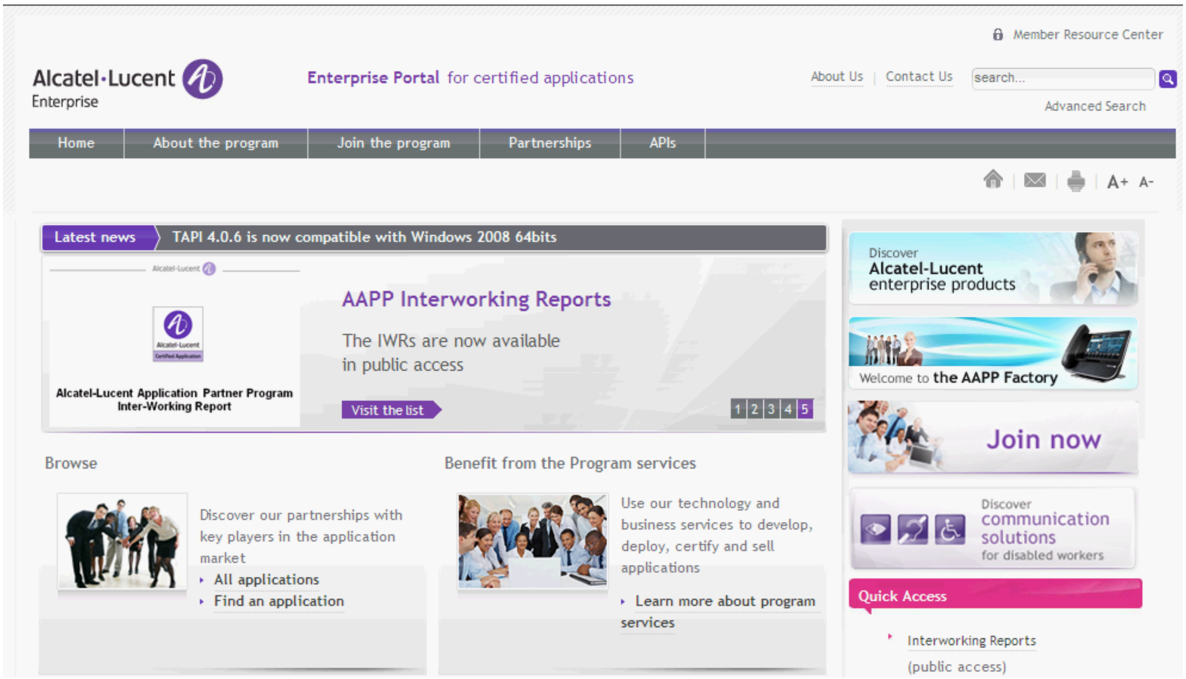
- **Provide easy interfacing for Alcatel-Lucent communication products**: Alcatel-Lucent's communication products for the enterprise market include infrastructure elements, platforms and software suites. To ensure easy integration, the AAPP provides a full array of standards-based application programming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

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



12.2 Alcatel-Lucent.com

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

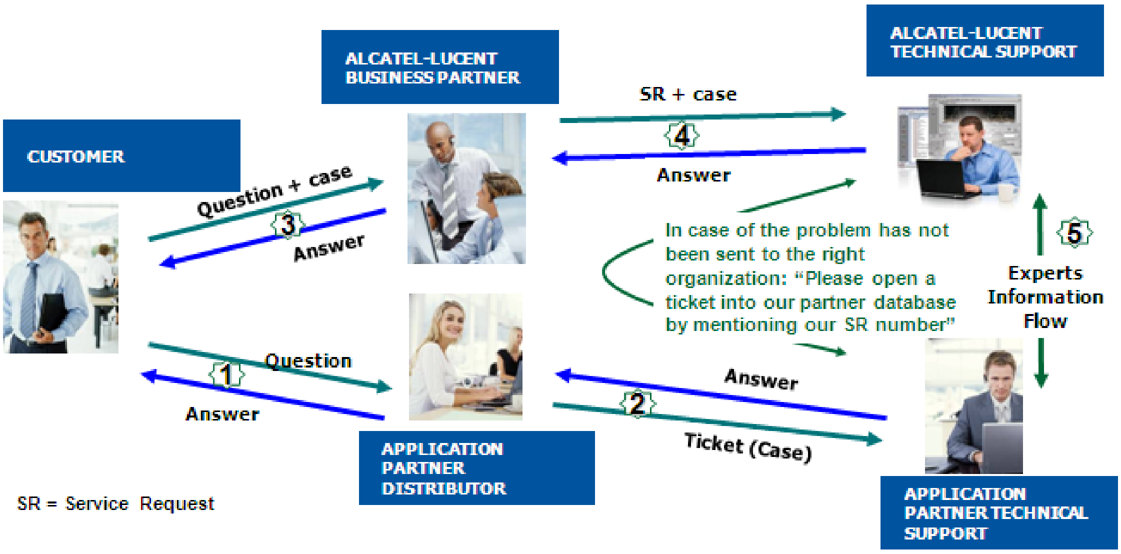
13 Appendix E: AAPP Escalation process

13.1 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

13.2 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 cannot 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 no 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
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://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.

13.3 Escalation in all other cases

These cases can cover following situations:

1. An InterWorking Report exist but is not valid (see Chap **Erreur ! Source du renvoi introuvable.** “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.

13.4 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 xxx 20 85 85
- Telephone numbers:

Alcatel-Lucent Business Partners Support Center for countries:

<table>
<tr><th>Country</th><th>Supported language</th><th>Toll free number</th></tr>
<tr><td>France</td><td rowspan="3">French</td><td rowspan="23">+800-00200100</td></tr>
<tr><td>Belgium</td></tr>
<tr><td>Luxembourg</td></tr>
<tr><td>Germany</td><td rowspan="3">German</td></tr>
<tr><td>Austria</td></tr>
<tr><td>Switzerland</td></tr>
<tr><td>United Kingdom</td><td rowspan="16">English</td></tr>
<tr><td>Italy</td></tr>
<tr><td>Australia</td></tr>
<tr><td>Denmark</td></tr>
<tr><td>Ireland</td></tr>
<tr><td>Netherlands</td></tr>
<tr><td>South Africa</td></tr>
<tr><td>Norway</td></tr>
<tr><td>Poland</td></tr>
<tr><td>Sweden</td></tr>
<tr><td>Czech Republic</td></tr>
<tr><td>Estonia</td></tr>
<tr><td>Finland</td></tr>
<tr><td>Greece</td></tr>
<tr><td>Slovakia</td></tr>
<tr><td>Portugal</td></tr>
<tr><td>Spain</td><td>Spanish</td></tr>
</table>

For other countries:

English answer :	+ 1 650 385 xxx3
French answer :	+ 1 650 385 xxx6
German answer :	+ 1 650 385 xxx7
Spanish answer :	+ 1 650 385 xxx8

END OF DOCUMENT