



## ALE Application Partner Program Inter-Working Report

**Partner: ASCOM**  
**Application type: VoWiFi handset**  
**Application name: i62**  
**Alcatel-Lucent Enterprise Platform:**  
**OmniPCX Enterprise™**

The logo for ASCOM, featuring the word 'ascom' in a bold, lowercase, red sans-serif font.

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

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## Certification overview

Date of the certification	July 2018
ALE International representative	Claire Dechristé
AAPP member representative	Ulrik Lindgren/Matthew Williams
Alcatel-Lucent Enterprise Communication Platform release	OmniPCX Enterprise
Alcatel-Lucent Enterprise Communication Platform release	OXE: R12(m1.403.16.a) OAW: 6.4.3.4 with AP205/225
AAPP member application release	i62 6.0.6
Application Category	Terminals DECT / Wi-Fi Mobility

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### Historic

Edition 1: Initial version – November 2017  
Edition 2: version 2 – July 2018

## Test results OXE R12

- Passed
  Refused
  Postponed  
 Passed with restrictions

Refer to the section 4 for a summary of the test results.

## IWR validity extension

i62 phones can be deployed as a connection user in an OpenTouch solution

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# 1 Introduction

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

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

Information contained in this document is believed to be accurate and reliable at the time of printing. However, due to ongoing product improvements and revisions, ALE International 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 Interworking Report

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

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

A new release is identified as following:

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

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

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

## 3 Limits of Technical support

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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 ALE International

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



## 4 Summary of test results

### 4.1 Summary of main functions supported

Installation and configuration of the i62 handsets through the Ascom WinPDM application (configuration also possible through remote management “over the air”)

Configuration of the i62 handsets as SIP users on the OXE registrar server

Below, telephonic features for SEPLOS (SIP Extension) supported or not by ASCOM i62 handsets:

Bloc/overlap dialing	✓
Codec	✓
Set is free	✓
Set is busy	✓
DND	✓
Out of service	✓
Interception	✓
Forward	✓
Intrusion	✓
Camp on	✓
Secret identity	✓
Call rejection	✓
Call release	✓
Hold	✓
Broker call	✓
Conference	✓
Transfer	✓
Networking	✓
Display management	✓
Multi-line	✓
Manager / Assistant	✓
Voice Mail	✓
Attendant	✓
Prefixes support	✓
Suffixes support	✓
CPU redundancy support	✓

#### 4.1.1 Wifi Infrastructure

- 8 calls are possible on a single Access Point (full voice wifi network)
- Subjective speech quality is quite good
- Stress tests for data and voice coexistence are acceptable (downstream and upstream).
- Handovers between Access Points
- Master/Local OAW configuration is supported

## 4.2 Summary of problems

- OmniPCX Enterprise **spatial redundancy** (duplicate CS on different IP sub networks) is supported but switchover is **not completely transparent** for the users:
  - During a switchover the existing call are not maintained. In dialog SIP messages are sent to the old main call server. It is not possible to update an existing call. I62 uses the IP address of its DNS cache until it registers on the new main call server.
  - A new call is not possible just after a switchover. I62 uses the IP address of its DNS cache until it registers on the new main call server. The maximum failover timer would be an REGISTER expires (180 seconds) + a sip message timeout (32s).
  - Switching between call servers only occurs after the next "REGISTER".
  - Ascom i62 v. 5.4.0 or later also switches over to the secondary call server as soon as an outgoing call fails (i.e. after a sip message timeout). The handset then polls the primary SIP proxy for availability at regular intervals.
- Callback request does not always work when using OXE suffix 5. The callback is initiated on both sets, but when the users answer, the call is cut on i62. This is followed using Ascom ticket number: [MRS-86](#) and OXE [CROXE-6430](#)
  - ➔ Fixed on OXE side since OXE M2.300.22.
  - ➔ An HF is available for OXE m2.300.20 patch

## 4.3 Summary of limitations

- SIP Keep Alive mechanism with SIP OPTIONS messages is not supported by ASCOM base station.
- When CAC is activated and the call limit is exceeded, an i62 is unable to authenticate at power up. The phone receives a "Deauth" answer from the AP. The association is possible when one call is ended on the AP (1)
- The call between the transferee and the transferor is disconnected if the transferor initiates a blind transfer to a non-existing extension.
- G723 is not supported by i62.
- Refer to OmniPCX Enterprise release R12 notes for information on general limitations for SIP/SEPLoS devices on OmniPCX Enterprise.
- OXE management of SIP gateway domain name is case sensitive. As a result, ASCOM Primary SIP proxy setting must match the string defined in SIP>SIP gateway > DNS local domain name. If it does not match, there will be issues in voice message notifications (MWI).

### Notes

(1): The best common practice is not to use TSPEC-based CAC, but rather the method were SIP traffic is intercepted and calls are rejected on a SIP level. Please refer to AP documentation for further information.

### 4.3.1 Wifi Infrastructure

- Roaming between master/local controllers: the state of RP connexion isn't exchanged between controllers. You have to open all possible RTP ports on OAW to keep the voice after the roaming.
- Voice/Data coexistence: We recommend to set the maximal number of calls per AP to 8. We have noticed sub-optimal performance in the prioritization of voice traffic when other traffic load is medium/high on the OAW controller, so the restriction of 8 calls per AP is OK.
- DFS channels: We recommend avoiding the use of DFS channels in the 5Ghz radio band.
- WMM and U-APSD: OK. QoS configuration should be carefully made. On the wired network, the RTP/RTCP packets must be configured with an 802.1d/DSCP value that matches the Voice WMM Access Category. (6 for the 802.1d, 46 for the DSCP).

- U-APSD/retries : During a call, the AP sends a downstream voice packet with EOSP=1 (that is acknowledged by the i62) then resends the same voice packet (that is not received by the phone because it is in power save mode). Increase the number of retries
- A problem was observed when mixing 802.11b and 802.11g data rates. The solution is to disable all 802.11b data rates on the SSID used.
- De-authenticated occurs during roaming when using WEP. The solution is to avoid WEP as encryption method.

#### 4.4 Notes, remarks

- OXE system features (CFU, DND...) can be used instead of i62 local features. Configuration details are given in appendix A.
- A complete interworking report about WLAN infrastructure is available on Ascom Partner Web site
- WEP and TKIP are supported on i62 phones but are not part of the IEEE 802.11n standard. 802.11n deployments should always employ security profiles based on WPA2-AES.
- Configuring an NTP device on i62 is strongly recommended
- It must be mentioned that in VoWIFI systems, voice quality is strongly linked to the RF site survey deployment on the field. The specificities of each installation as far as radio coverage is concerned should always be analysed case by case and optimised accordingly.
- Ascom abbreviations :
  - Non Conformity Report (NCR)
  - Change or feature Request (CR)

## 5 Application information

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<b>Application type:</b>	VoWiFi handset
<b>Application commercial name:</b>	i62
<b>Application version:</b>	6.0.6
<b>Interface type:</b>	SIP/IP/802.11

### Brief 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.

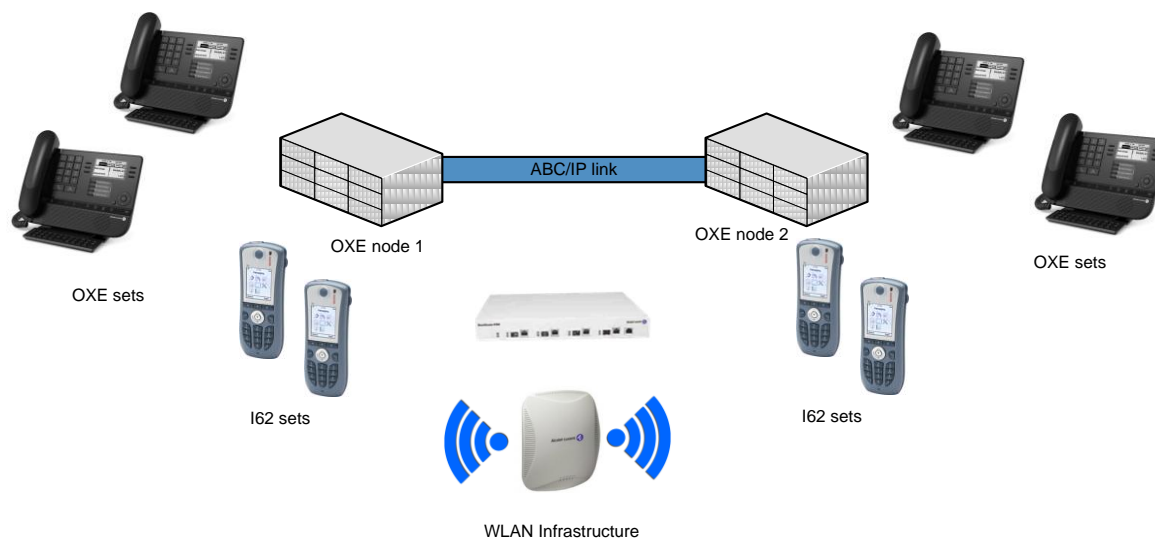


i62

## 6 Tests environment

### 6.1 General architecture

OXE / SEPLOS Test Environnement:



## 6.2 Hardware configuration:

### Node1:

- OmniPCX Enterprise common hardware.
- Duplicated call servers
- Spatial redundancy (Different IP subnetworks)

Crystal 1 :

```

+-----+-----+-----+-----+-----+-----+
| Cr | cpl | cpl type | hw type | cpl state | coupler ID |
+-----+-----+-----+-----+-----+-----+
| 1 | 0 | GD-3 | ----- | IN SERVICE | BAD PCMS CODE |
| 1 | 1 | MIX244 | ----- | IN SERVICE | BAD PCMS CODE |
| 1 | 2 | PRA T2 | ----- | IN SERVICE | BAD PCMS CODE |
| 1 | 3 | PRA T2 | ----- | IN SERVICE | BAD PCMS CODE |
| 1 | 4 | CS | ----- | CS+4645 | BAD PCMS CODE |
| 1 | 6 | PRA T2 | ----- | IN SERVICE | BAD PCMS CODE |
| 1 | 8 | CS | ----- | CS+4645 | BAD PCMS CODE |
+-----+-----+-----+-----+-----+-----+

```

### Node2:

- OmniPCX Enterprise common hardware.
- Duplicated call servers
- Spatial redundancy (Different IP subnetworks)
- Two media gateways

Crystal 0 :

```

+-----+-----+-----+-----+-----+-----+
| Cr | cpl | cpl type | hw type | cpl state | coupler ID |
+-----+-----+-----+-----+-----+-----+
| 0 | 6 | CPU_CS | ----- | IN SERVICE | BAD PCMS CODE |
| 0 | 10 | CPU_CS | ----- | IN SERVICE | BAD PCMS CODE |
+-----+-----+-----+-----+-----+-----+

```

> Display list of PCS :

```

+-----+-----+-----+
| PCS Name | Address | State |
+-----+-----+-----+
| etesting_pcs | 10.10.11.10 | INACTIVE |
+-----+-----+-----+

```

Crystal 1 :

```

+-----+-----+-----+-----+-----+-----+
| Cr | cpl | cpl type | hw type | cpl state | coupler ID |
+-----+-----+-----+-----+-----+-----+
| 1 | 0 | GD | ----- | IN SERVICE | BAD PCMS CODE |
| 1 | 1 | MIX244 | ----- | IN SERVICE | BAD PCMS CODE |
| 1 | 2 | PRA T2 | ----- | IN SERVICE | BAD PCMS CODE |
| 1 | 3 | PRA T2 | ----- | IN SERVICE | BAD PCMS CODE |
| 1 | 4 | GA | ----- | REG NOT INIT | BAD PCMS CODE |
| 1 | 5 | APPLI | ----- | LANX 16 | BAD PCMS CODE |
| 1 | 6 | PRA T2 | ----- | IN SERVICE | BAD PCMS CODE |
| 1 | 8 | CS | ----- | CS+4645 | BAD PCMS CODE |
+-----+-----+-----+-----+-----+-----+

```

Crystal 6 :

```

+-----+-----+-----+-----+-----+-----+

```

Cr	cpl	cpl type	hw type	cpl state	coupler ID
6	0	GD		IN SERVICE	BAD PCMS CODE
6	1	PRA T2		IN SERVICE	BAD PCMS CODE
6	2	PRA T2		IN SERVICE	BAD PCMS CODE
6	3	MG_IVR_Z30		IN SERVICE	BAD PCMS CODE

### 6.3 Software configuration

- **Alcatel-Lucent Enterprise Communication Platform:** OmniPCX Enterprise R12 (m1.403.16.a)

**Note:** SIP Endpoint Level of Service (SEPLoS) is used for the external sets. This means that all telephone features must be tested once by prefixes and once locally on the phone, if available.

- **WLAN infrastructure:** Alcatel OAW with AP205/225 access points. Two controllers were deployed in a Master/Local redundant configuration.
- **Application platform:** i62 6.0.6

## 7 Test Result Template

---

The results are presented as indicated in the example below:

Test	Action	Result	Comment
------	--------	--------	---------

**Test Case Id:** a feature testing may comprise multiple steps depending on its complexity. Each step has to be completed successfully in order to conform to the test.

**Test Case:** describes the test case with the detail of the main steps to be executed the and the expected result

**N/A:** when checked, means the test case is not applicable in the scope of the application

**OK:** when checked, means the test case performs as expected

**NOK:** when checked, means the test case has failed. In that case, describe in the field "Comment" the reason for the failure and the reference number of the issue either on ALE International side or on AAPP member side

**Comment:** to be filled in with any relevant comment. Mandatory in case a test has failed especially the reference number of the issue.



## 8 Omni PCX Enterprise - SEPLoS Test Results

### 8.1 Connectivity and Setup

#### 8.1.1 Test Objectives

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

#### 8.1.2 Test Results

Test	Action	Result	Comment
1	I62 configuration using Ascom WinPDM application.	OK	
2	DHCP registration	OK	
3	NTP registration / The SIP phone is configured to retrieve the date and time from the OXE internal NTP server and external NTP server Check the phone retrieves the information and displays it.	OK	
4	SIP set registration to OXE in static IP addressing mode (without authentication) / The phone is configured to use IP static parameters	OK	
5			
5	SIP registration, using DNS The phone is configured to use a domain name as registrar / proxy server address. The DNS IP address is the OXE CPUs addresses	OK	On i62, do not declare primary SIP proxy IP addresses (0.0.0.0), only SIP proxy ID. A SIP proxy ID can be a FQDN. DNS servers can either be statically defined or updated dynamically through DHCP.
6	SIP registration, using DNS The phone is configured to use a domain name as registrar / proxy server address. The DNS IP address is a third party DNS server	OK	On i62, do not declare primary SIP proxy IP addresses (0.0.0.0), only SIP proxy ID. A SIP proxy ID can be a FQDN. DNS servers can either be statically defined or updated dynamically through DHCP.
7	Support of "423 Interval Too Brief" for registration or subscribe (1)	OK	
8	SIP registration with authentication: Turn on SIP Digest authentication, specify realm on OXE, and specify user name and password on SIP client. / SIP digest authentication is activated on OXE and phone side. Check also that outgoing call is authenticated.	OK	

#### Notes:

- (1) On the SIP client, specify a default registration period inferior to that of OXE SIP registrar. OXE will reject with error "423 Interval Too Brief". Check that SIP set increases registration period accordingly.

## 8.2 Outgoing Calls

### 8.2.1 Test Objectives

The calls are generated to several users belonging to the same network.

Called party can be in different states: free, busy, out of service, do not disturb, etc.

Calls to data devices are refused.

Points to be checked: tones, voice during the conversation, display (on caller and called party), hang-up phase.

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

### 8.2.2 Test Results

Test	Action	Result	Comment
1	Call to a local user	OK	
2	Call to a local user with overlap dialing: <i>Dial a part of the number, wait and continue.</i>	OK	
3	Call to a local user with overlap dialing, timeout: <i>Dial a part of the number, wait and stop.</i>	NA	i62 only supports enbloc dialing
4	Call to a local user with overlap dialing, release: <i>Dial a part of the number, wait and release the call.</i>	NA	i62 only supports enbloc dialing
5	Call to local user with no answer.	OK	
6	Call to another SIP set / <i>The SIP phone under test calls another SIP phone. Check the display and audio during all steps (dialing, ring back tone, conversation, release).</i>	OK	
7	Call to wrong number (SIP: "404 Not Found")	OK	Displays "Invalid Number"
8	Call to busy user (SIP: "486 Busy Here") <i>Check busy tone.</i>	OK	
9	Call to user in "Out of Service" state (SIP: "480 Temporarily Unavailable")	OK	Display "No response"
10	Call to user in "Do not Disturb" state (SIP: "603 Decline" or "486 User Busy") / <i>Dial "42+password", wait for acknowledgement ring back tone from OXE</i>	OK	Busy tones are played. The user has to end the call.
11	Call to local user, immediate forward (CFU). (SIP: "302 Moved Temporarily")(1) / <i>Dial "51 + target extension", wait for acknowledgement ring back tone from OXE / Dial "41" for forward cancellation</i>	OK	Name and number are updated when the call is answered
12	Call to local user, forward on no reply (CFNR). (1) / <i>Dial "53 + target extension", wait for acknowledgement ring back tone from OXE / Dial "41" for forward cancellation</i>	OK	Name and number are updated when the call is answered

13	Call to local user, forward on busy (CFB). (1) / Dial "54 + target extension", wait for acknowledgement ring back tone from OXE / Dial "41" for forward cancellation	OK	Name and number are updated when the call is answered
14	Call to a local user without proxy Authentication	OK	By changing OXE proxy settings
15	Call within same IP domain. SIP set in domain A (intra-domain=without compression). Call to OXE set in domain A (intra-domain=without compression). Check order of codecs in SDP list. Check chosen codec. We expect G.711.	OK	See note (2)
16	Call to another IP domain. SIP set in domain A (extra-domain=with compression). Call to OXE set in domain B (extra-domain=with compression). Check order of codecs in SDP list. Check chosen codec. We expect G723 or G.729	OK	See note (2) G723 is not supported by i62
17	Call to external number (via T2 loopback) (Check ring back tone, called party display)	OK	ISDN trunk name is displayed on i62. i62 displays P-asserted-identity which contains the trunk name
18	SIP session timer expiration: Check if call is maintained or released after the session timer has expired.	OK	

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.

(2) For IP domain tests, the following setup is used:

```
| number | 0 | 1 |
| type   | NIPR | IP_R |
| allowed | ffff | 1 |
| used   | 0 | 0 |
| IPP Intr | G711 | G711 |
| IPP Extr | G729 | G729 |
```

Partner SIP set is in domain 1.

## 8.3 Incoming Calls

### 8.3.1 Test Objectives

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

### 8.3.2 Test Results

Test	Action	Result	Comment
1	Local /network call to free SIP terminal / Check ring back tone, called party display	OK	
2	Local/network call to busy SIP terminal / Call a set which is already in conversation	OK	
3	Local/network call to battery-removed SIP terminal	OK	i62 unregisters correctly before being turned off (EXPIRES=0)
4	Local/network call to SIP terminal in <b>Do Not Disturb (DND) mode</b>		
4A	By local feature if applicable	OK	
4B	By system feature (SEPLoS) Dial "42 + user password" wait for acknowledgement ring back tone from OXE	OK	We recommend to use the OXE system feature, see appendix A and B for more details about the configuration
5	Local/network/SIP call to SIP terminal in <b>immediate forward (CFU) to local user:</b>		
5A	By local feature if applicable	OK	
5B	By system feature (SEPLoS) Dial "51 + target extension" wait for acknowledgement ring back tone from OXE / Dial "41" for forward cancellation	OK	We recommend to use the OXE system feature, see appendix A and B for more details about the configuration
6	Local/network/SIP call to SIP terminal in <b>immediate forward (CFU) to network number:</b>		
6A	By local feature if applicable	OK	
6B	By system feature (SEPLoS) / Dial "51 + target extension" wait for acknowledgement ring back tone from OXE / Dial "41" for forward cancellation	OK	We recommend to use the OXE system feature, see appendix A and B for more details about the configuration
7	Local/network/SIP call to SIP terminal in <b>immediate forward (CFU) to another SIP user</b>		
7A	By local feature if applicable	OK	

7B	By system feature (SEPLoS) Dial "51 + target extension" wait for acknowledgement ring back tone from OXE / Dial "41" for forward cancellation	OK	We recommend to use the OXE system feature, see appendix A and B for more details about the configuration
8	Local call to SIP terminal in "forward on busy" (CFB) state:		
8A	By local feature if applicable	NT	Not possible to test this as OXE sends a 183 Session in Progress with reason set to "busy" to the calling set
8B	By system feature (SEPLoS) Dial "52 + target extension" wait for acknowledgement ring back tone from OXE / Dial "41" for forward cancellation	OK	We recommend to use the OXE system feature, see appendix A and B for more details about the configuration
9	Local call to SIP terminal in "forward on no reply" (CFNR)	OK	
9A	By local feature if applicable	OK	
9B	By system feature (SEPLoS) Dial "53 + target extension" wait for acknowledgement ring back tone from OXE / Dial "41" for forward cancellation	OK	We recommend to use the OXE system feature, see appendix A and B for more details about the configuration  Missed call displayed
10	Call to busy user, Call waiting. (Camp-on), local feature With one i62 (SIPset-1) call an OXE set (OXEset-1) to make it busy, take the call and don't hang up. OXEset-1 is a multiline set.  With another i62 SIPset-2 calls OXEset-1 and on SIPset-2 use the camp-on feature. Check the tones and display on SIPset-2.	OK	
11	External call to SIP terminal. Check that external call back number is shown correctly.	OK	
12	Secret identity : Local call to SIP terminal. / Dial "409 + extension". Check that caller id is not shown.	OK	See appendix A and B for configuration details  "Withheld" is displayed on i62
13	Display: Call to free SIP terminal from user with a name containing non-ASCII characters. Check caller display.	OK	
14	Display: Call to free SIP terminal from user with a UTF-8 name containing non-ASCII characters. Check caller display.	OK	
15	SIP set is part of a sequential hunt group. Call to hunt group. Check call/release.	OK	
16	SIP set is part of a cyclic hunt group. Call to hunt group. Check call/release.	OK	

17	SIP set is declared as a twin set (tandem). Call to main set and see if twin set rings. Take call with twin set.	OK	Call Missed notification is presented on i62 set if the call is answered on main tandem set.
18	<p>Call Pick Up.          With SIPset-1 call SIPset-2.          While SIPset-2 is ringing, pick-up the call from SIPset-3 by dialing the pick-up prefix + SIPset-2 ("55"+number of target set)</p> <p>Check audio and display.</p>	OK	See appendix A and B for configuration details

## 8.4 Features during Conversation

### 8.4.1 Test Objectives

Features during conversation between local user and SIP user must be checked.  
Check that right tones are generated on the SIP phone.

### 8.4.2 Test Results

Test	Action	Result	Comment
1	Hold and resume (both directions) (Check tones)	OK	There was no "hold" display on the set placed on hold, there is only hold tone/music
2	Second call to another local user. Distant user is put on hold.	OK	
3	Broker request (toggle back and forth between both lines, local feature if applicable)	OK	
4	Release first call. Keep second call.	OK	
5	Call park: - Call between SIP set and OXE set. - Put your call on hold. - New call: Dial the prefix for call parking ("402"+number). Now call can be hung up. Later call can be retrieved by calling the park prefix "402".	OK	
6	Send/receive DTMF	OK	
7	Meet Me conference (prefix: 509 + conference number)	OK	
8	Three party conference initiated from OXE set. Released by OXE set. / Dial extension target + suffix "3".	OK	
9	Three party conference initiated from SIP set (local feature). Released by SIP set.	OK	The other parties can be OXE sets, i62 or external parties
10	Call back on free or busy set from SIP set. The SIP set calls another set which is in conversation. Then press the call back suffix "5".	NOK	The callback is initiated on both sets, but when the users answer, the call is cut on i62. The issue is random, but most of the time it does not work. It looks like a race condition.  This is followed using Ascom ticket number: <b>MRS-86</b> and OXE <b>CROXE-6430</b>  <b>Fixed since OXE M2.300.22</b>  There is no caller display update if the set for which the call back is requested answers before the callback initiator

## 8.5 Call Transfer

### 8.5.1 Test Objectives

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

Audio, tones and display must be checked.

We use the following scenario, terminology and notation:

There are three actors in a given transfer event:

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

The scenario is:

1. A (Transferee) calls B (Transferor).
2. B (Transferor) calls C (Transfer Target). A is on hold during this phase.
3. B (Transferor) executes the transfer. B drops out of the communication. A is now in communication with C.

There are three sorts of transfers in the SIP world:

- *Unattended Transfer or Basic Transfer*: Unattended Transfer consists of the Transferor providing 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 Early Attended Transfer or Transfer on ringing* : Semi-Attended Transfer consists of the Transferor having a session in hold state with the Transferee and initiating the Transfer when the consultation call to the Target Transfer is in ringing state.
- *Attended Transfer or Transfer in conversation*: Attended Transfer consists of the Transferor having a session in hold state with the Transferee and initiating the Transfer when the session between the Transferor and Transfer target is established.

Note: Unattended transfers are not provided by OXE. This kind of transfer is only managed by the SIP set (i62)

Semi attended transfers are not provided by the SIP set (i62). This kind of transfer is only managed by OXE.

Note : Desactivate VAD to avoid one way call when connected to a GD : Ascom NCR 18050.



## 8.5.2 Test Results

In the tables below, *SIP* means a partner SIP set, *OXE* means a ALE OXE (Z/UA/IP/SIP) set.

To start an unattended transfer during a conversation, select “more>Tranfs. to new” and enter the transfer target extension number.

### 8.5.2.1 Unattended transfers

Test	Action			Result	Comment
	A Transferee	B Transferor	C Transfer Target		
1	OXE/Ext . call	SIP	OXE/Ext . call	OK	CLIP correctly updated after transfer
2	SIP	OXE	OXE/Ext . call	OK	
3	OXE/Ext . call	OXE	SIP	OK	
4	SIP	OXE	SIP	OK	
5	OXE/Ext . call	SIP	SIP	OK	CLIP correctly updated after transfer
6	SIP	SIP	OXE/Ext . call	OK	CLIP correctly updated after transfer
7	SIP	SIP	SIP	OK	CLIP correctly updated after transfer

Note: The call between the transferee and the transferor is disconnected if the transferor (Ascom device) initiates a transfer to a non existing extension.

### 8.5.2.2 Semi-attended transfers

Test	Action			Result	Comment
	A Transferee	B Transferor	C Transfer Target		
1	OXE/Ext . call	SIP	OXE/Ext . call	NA	Not provided by the SIP set (i62).
2	SIP	OXE	OXE/Ext . call	OK	CLIP correctly updated after transfer
3	OXE/Ext . call	OXE	SIP	OK	The call between the transferor and the transfer target appears as missed call in the call log after the transfer establishment.  CLIP correctly updated after transfer

4	SIP	OXE	SIP	OK	The call between the transferor and the transfer target appears as missed call in the call log after the transfer establishment.  CLIP correctly updated after transfer
5	OXE/Ext . call	SIP	SIP	NA	Not provided by the SIP set (i62).
6	SIP	SIP	OXE/Ext . call	NA	Not provided by the SIP set (i62).
7	SIP	SIP	SIP	NA	Not provided by the SIP set (i62).

### 8.5.2.3 Attended transfers

To start an attended transfer during a conversation, select “more>New call” and enter the transfer target extension number. After the conversation period with the transfer target, enter the transfer target, select “more>Transfer to [Transferee name]”

Test	Action			Result	Comment
	A Transferee	B Transferor	C Transfer Target		
1	OXE/Ext . call	SIP	OXE/Ext . call	OK	
2	SIP	OXE	OXE/Ext . call	OK	
3	OXE/Ext . call	OXE	SIP	OK	
4	SIP	OXE	SIP	OK	
5	OXE/Ext . call	SIP	SIP	OK	
6	SIP	SIP	OXE/Ext . call	OK	
7	SIP	SIP	SIP	OK	

## 8.6 Attendant

### 8.6.1 Test Objectives

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

### 8.6.2 Test Results

Test	Action	Result	Comment
1	<p>SIP set call to attendant</p> <p>With SIPset-1, call attendant with attendant call prefix, attendant answers.</p> <p>Check ringing back tone, display and audio.</p>	OK	
2	<p>Call to attendant (using attendant call prefix "***8"). Second internal incoming call while in conversation with attendant.</p> <p><i>Expected result : second call refused</i></p>	OK	<p>If 2<sup>nd</sup> call is accepted on i62, "488 Not Acceptable here" is sent by OXE</p> <p>"Hold rejected" shown correctly on display.</p> <p>First call is muted. It can be retrieved correctly by pressing unmute.</p>
3	<p>2nd incoming call to SIP set while in conversation with attendant.</p> <p>SIPset-1 being in conversation with the attendant.</p> <p>Make an ISDN incoming call to SIPset-1 and try to answer the call.</p> <p>SIPset-1 should not be able to take the second incoming call. Check that the PSTN incoming call is properly managed (the PSTN user should receive a busy tone). Check that SIPset-1 stays in conversation with the attendant station (SIPset-1 can display an error message when it tries to put the attendant station on hold).</p>	OK	<p>The external call is not presented by OXE to i62</p> <p>If 2<sup>nd</sup> call is accepted on i62, "488 Not Acceptable here" is sent by OXE</p> <p>"Hold rejected" shown correctly on display.</p> <p>First call is muted. It can be retrieved correctly by pressing unmute.</p>

<p><b>4</b></p>	<p>SIP set call to attendant, attendant transfers to OXE set, semi-attended</p> <p>With SIPset-1, call attendant with prefix 9 (attendant call prefix), attendant answers.</p> <p>From the attendant, call OXEset-1 and transfer semi-attended.</p> <p>Answer the call and check audio and display.</p> <p>Redo the test with Nwkset-1.</p>	<p>OK</p>	
<p><b>5</b></p>	<p>SIP set call to attendant, attendant transfers to OXE set, attended</p> <p>With SIPset-1, call attendant with prefix 9 (attendant call prefix), attendant answers.</p> <p>From the attendant, call OXEset-1 and transfer attended.</p> <p>Check audio and display.</p> <p>Redo the test with Nwkset-1.</p>	<p>OK</p>	
<p><b>6</b></p>	<p>OXE set calls to attendant, attendant transfers to SIP set, semi-attended</p> <p>With OXEset-1, call attendant with prefix 9 (attendant call prefix), attendant answers.</p> <p>From the attendant, call SIPset-1 and transfer semi-attended.</p> <p>Answer the call and check audio and display.</p>	<p>OK</p>	
<p><b>7</b></p>	<p>OXE set calls to attendant (using attendant call prefix), attendant transfers to SIP set, attended</p> <p>With OXEset-1, call attendant with prefix 9 (attendant call prefix), attendant answers.</p> <p>From the attendant, call SIPset-1 and transfer attended.</p> <p>Check audio and display.</p>	<p>OK</p>	

<p><b>8</b></p>	<p>External ISDN Call to attendant, attendant transfers to SIP set, attended</p> <p>ISDN call to the attendant.</p> <p>From the attendant, call SIPset-1 and transfer attended</p> <p>Check audio and display</p>	<p>OK</p>	
<p><b>9</b></p>	<p>SIP set call to attendant, attendant transfers to External, semi-attended</p> <p>From SIPset-1, call attendant with prefix 9 (attendant call prefix), attendant answers.</p> <p>From the attendant, call an external ISDN destination and transfer semi-attended</p> <p>Once in conversation, check audio and display.</p>	<p>OK</p>	

## 8.7 Voice Mail

### 8.7.1 Test Objectives

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

### 8.7.2 Test Results

Test	Action	Result	Comment
1	A Voice Mail message for the SIP subscriber is generated. Check that MWI is activated. / Dial prefix "570" + target extension or target extension + suffix "8"	OK	Tested on another OXE system, same R12 version (etesting3.etesting.lab)
2	Message consultation	OK	
3	Password modification	OK	
4	SIP call to a OXE user forwarded to Voice Mail / Dial "51 + VoiceMail extension", wait for acknowledgement ring back tone from OXE / Dial "41" for forward cancellation	OK	
5	OXE call to a SIP user forwarded to Voice Mail / Dial "51 + VoiceMail extension", wait for acknowledgement ring back tone from OXE / Dial "41" for forward cancellation	OK	

## 8.8 Duplication and Robustness

### 8.8.1 Test Objectives

Check how the system will react in case of a CPU reboot, switchover or link failure etc. The test system is configured with spatial redundancy (duplicate call servers on two different IP subnetworks).

Spatial redundancy can be configured in two ways:

“Alternate Proxy method”: Specify both CS MAIN addresses as primary and alternative proxy respectively. Requires that on non availability of primary proxy, secondary proxy is used. Requires ability to send outgoing call and accept incoming calls from secondary proxy just after the switchover. In dialog, it is not possible to update a call with this redundancy method (put on hold, transfer....)

“DNS method”: Do not specify a proxy address, only SIP domain. Specify the CS MAIN addresses as first and second DNS server, respectively. Requires that (at least on non availability of current proxy) a new DNS request is issued for every message. Only MAIN CS will respond. Requires ability to send outgoing call and to accept incoming calls from secondary CS when it becomes new MAIN.

For each configuration, check:

Can new outgoing calls be made immediately after switchover?

Are existing calls maintained after switchover?

Are incoming calls (from new MAIN CS) accepted immediately after switchover?

Can existing call be modified (transfer, hang-up, etc.) after switchover?

Check if a session that has been started before switchover is maintained after switchover, i.e. does the new MAIN CS send session updates and is this accepted by the client?

### 8.8.2 Test Results

Test	Action	Result	Comment
1	Spatial redundancy, using “Alternate Proxy method”: Switchover to standby call server. Check that an existing calls are maintained.	OK	Does not work in 6.0.1 if using IP addresses in SIP proxies method Fixed in 6.0.4 or later  SIP RE-INVITE after a session timer exceed coming from the new main call server are accepted on i62
2	Spatial redundancy, using “Alternate Proxy method”: Switchover to standby call server. Check that an outgoing call can be done just after the switchover.	OK but	Switches between call servers only occur after “Register”. Before this registration, it is not possible to evolve an existing call (place on hold, transfer...) and to place a new call. Receiving call is possible.
3	Spatial redundancy, using “DNS method”, OXE used as DNS server or using a delegation server: Switchover to standby call server. Check that an existing call can be updated (put on hold, transfer...).	OK but	Switches between call servers only occur after “Register”. Before this registration, it is not possible to evolve an existing call (place on hold, transfer...) and to place/receive a new call.. I62 uses the IP address of its DNS cache until it registers on the new main call server. Ascom CR18091

4	Spatial redundancy, using “DNS method”, OXE used as DNS server or using a delegation server: Switchover to standby call server. Check that an outgoing call can be done just after the switchover.	OK but	Switches between call servers only occur after “Register”. Before this registration, it is not possible to evolve an existing call (place on hold, transfer...) and to place/receive a new call. I62 uses the IP address of its DNS cache until it registers on the new main call server. Ascom CR18091
5	Switchover to Passive Communication Server (PCS). (IP link to main/stand-by call servers down)	OK but	Switches between call servers only occur after “Register”. Before this registration, it is not possible to evolve an existing call (place on hold, transfer...) and to place/receive a new call. On i62 it is only possible to configure two proxies. In case of spatial redundancy deployment, proxy 1 should be configured with OXE FQDN.
6	SIP device reboot. Check that calls are possible as soon as device has come back to service.	OK	
7	Temporary Link down with the PBX	OK	

Notes:

In order to have acceptable switchover time the keep alive mechanism (SIP Options) can be used but this feature is unavailable on Ascom side. This is an Ascom limitation.



## 9 WiFi Infrastructure Testing

### 9.1 Security - i62 association

#### 9.1.1 Cipherring modes testing

The aim is to verify that i62 phones can associate to an AP with the following security:

- None
- WPA-PSK / TKIP
- WPA2-PSK / AES

Then a basic call is made.

Test description: Configure the SSID and the phone with the tested security. The phone is powered on and a basic call is performed.

Test results:

Test	Result	Comment
Associate with the no security. Make a basic call.	OK	
Associate with the WPA-PSK/TKIP security. Make a basic call.	OK	
Associate with the WPA2-PSK/AES security. Make a basic call.	OK	

#### 9.1.2 Omni Access ALG SIP firewall

The aim is to verify the compatibility of the Omni Access Wireless switch SIP firewall ALG and the i62 phones.

Test description:

The OAW is configured to give the i62 phones a role with the following security policies:

Id	Source	Destination	Service	Action
1	Any	Any	Dhcp	Permit
2	User	CS(proxy SIP)	Svc-sip-udp	Permit
3	CS(proxy Sip)	User	Svc-sip-udp	Permit
4	User	NTP server	Svc-ntp	Permit
5	Any	Any	Icmp	Permit

The phone is powered on. It should be ready to do or receive a call. It is correctly registered on the Sip registrar. Incoming and outgoing calls are performed.

Verify that the firewall ALG SIP opens the RTP and RTCP ports correctly and that the voice data path session is labeled with the "V" (like Voice).

Test results:

Test	Result	Comment
ALG SIP. Power on the phone. The phone must be registered and ready to do receive a call.	OK	
ALG SIP. Basic calls are done.	OK	
ALG SIP. Advanced telephony services are done (transfer, forward...)	OK	

## 9.2 Voice quality

### 9.2.1 Speech quality

**Description:** This test aims at checking the voice quality between 2 set and one of them is a SIP set

- Make a call from a SIP set of <<<System A>>> to another set of <<<System B>>>. Check the subjective quality of the voice. Voice quality is OK
- Same test in reverse direction : voice quality is OK

Test	Result	Comment
Make a call from a SIP set (in Active mode) to another set. Check the subjective quality of the voice on both directions.	OK	
Make a call from one set to the SIP set (in Active mode). Check the subjective quality of the voice on both directions.	OK	
Make a call from a SIP set (in U-APSD mode) to another set. Check the subjective quality of the voice on both directions.	OK	
Make a call from one set to the SIP set (in U-APSD mode). Check the subjective quality of the voice on both directions.	OK	

## 9.3 Quality of Service

### 9.3.1 Number of i62 on call per Access Point

The aim is to validate that we can put at least 8 i62 phones on call per AP and that these calls are stable for at least 1 hour.

Test description:

Several phones are associated with a single AP. 6 sets are on call.  
Add new phones and try to make call: 8 phones.

Parameters:

- OXE: Framing 20ms, codec G.711 A or  $\mu$
- i62: max 20 dBm
- AP: 20 dBm
- The access was configured to support 802.11b/g/n (2.4 Ghz) and 802.11a/n (5Ghz).

Test results:

Tests were performed with 12 i62 phones.

Test	Result	Comment
Max_on_call_WPA2-PSK/AES (in Active mode) 2.4 Ghz (802.11 b/g/n) and 5 Ghz (802.11 a/n)	OK	8 i62 are in call during 20 minutes.
Max_on_call_WPA2-PSK/AES (in U-APSD mode) 2.4 Ghz (802.11 b/g/n) and 5 Ghz (802.11 a/n)	OK	8 i62 are in call during 20 minutes.
12 hour call between two SIP sets 2.4 Ghz (802.11 b/g/n) and 5 Ghz (802.11 a/n)	OK	
12 hour call between an i62 and the GD (UA set). 2.4 Ghz (802.11 b/g/n) and 5 Ghz (802.11 a/n)	OK	

### 9.3.2 Voice/Data coexistence

The aim is to validate the voice/data coexistence.

#### Test description:

Two laptops are associated to the same AP as phones and generate data traffic. Iperf is used to generate the data traffic and determines the available data throughput.

This test is a stress test as Iperf will try to use the maximal bandwidth.

Outgoing and incoming data traffic are performed in 2 tests. Phones are putting on call progressively up to 8 phones (4 calls).

#### Parameters:

- The laptops and the phones are at first connected to the 802.11b/g/n radio and then on the 802.11a/n radio.
- OXE: Framing 30ms, codec G.711
- Iperf generates TCP traffic. Data traffic is encrypted with WPA2-PSK.

#### Test results:

Test	Result	Comment
Voice_data_coexistence_WPA2-PSK /AES _upstream 2.4 Ghz (802.11 b/g/n)	OK	
Voice_data_coexistence_WPA2-PSK/ AES _downstream 2.4 Ghz (802.11 b/g/n)	OK	
Voice_data_coexistence_WPA2-PSK /AES _upstream 5 Ghz (802.11 a/n)	OK	
Voice_data_coexistence_WPA2-PSK/ AES _downstream 5 Ghz (802.11 a/n)	OK	

During the test, only g clients were present. The data traffic is TCP traffic

### 9.3.3 Roaming

The aim is to determine the subjective voice disturbance during a roaming.

This test will be performed in:

- WPA2-PSK / AES on 2.4 GHz (802.11 b/g/n)
- WPA2-PSK / AES on 5 GHz (802.11 a/n)

#### Test description

2 APs are on the same channel and connected to the same OAW or on a Master/local configuration. Several handovers are performed between these 2 APs. We check the subjective voice disturbance.

Test results:

Test	Result	Comment
Handover in call between 2 APs connected to the same OAW. Security: WPA2-PSK/AES on 2.4 GHz (802.11 b/g/n)	OK	No heard loss of voice
Roaming in call between 2 APs connected to different OAWs. Controllers are declared as Master/Local. Security: WPA2-PSK/AES on 2.4 GHz (802.11 b/g/n)	OK_but	RTP port range must be opened on OAW firewall. If not, the RTP flows are dropped on the OAW.
Handover in call between 2 APs connected to the same OAW. Security: WPA2-PSK/AES on 5 GHz (802.11 a/n)	OK	No heard loss of voice
Roaming in call between 2 APs connected to different OAWs. Controllers are declared as Master/Local. Security: WPA2-PSK/AES on 5 GHz (802.11 a/n)	OK_but	RTP port range must be opened on OAW firewall. If not, the RTP flows are dropped on the OAW.

The time displays in the comment column is the time interval between the last voice packet sent/received on the AP1 and the first voice packet sent/received on AP2.

### 9.3.4 Call Admission Control

Test description:

On the Omni Access wireless switch, we set the number of allowed calls per AP to 2. 2 phones are on call. Initiate a new call (incoming and outgoing) and check the behavior.

With a phone on call, try to roam to a full AP.

Test parameters:

- The QoS configuration for the test is:
  - call-admission-control
  - sip-call-capacity 2
  - call-handoff-reservation 0

The tested configuration is:

- TSPEC “disable” on the i62 phones and OAW controller / “Disconnected Extra Call” unchecked on the OAW QoS configuration

Test results:

Test	Result	Comment
Access point is full. Power on an i62 phone.	NOK	The phone receives a “Deauth” answer from the AP. The association is possible when one call is ended on the AP.
Access point is full. Call from an idle i62.	OK	The call is not accepted. The set receives an “User busy” message from the controller.
Access point is full. An idle i62 receives a call.	OK	The call is not accepted. OXE receives an “User busy” message from the controller.

### 9.3.5 WMM and U-APSD

On the OXE PABX, specify a DSCP value 46 for voice.

Wireless captures are performed and analyzed to see WMM and U-APSD information.

“show” command are used on the OAW switch to check some wmm information about the client and the AP.

#### Test results:

Regarding the Voice/Data convergence test, voice packets are prioritized against data packets (see “Voice/Data coexistence” tests results).

Test	Result	Comment
WMM: downstream voice packets are classified into the Access Category Voice	OK	
U-APSD: U-APSD seems working correctly	OK	

Depending on the DSCP/802.1d values, the downstream voice packets are classified into the matching WMM Access Category.

For example, a RTP/RTCP packets sent by an IPTouch with:

- a DSCP=46 (in decimal) will be classified into the Voice Access Category.
- A DSCP=0 will be classified into the Best Effort Access Category.

The i62 phones use U-APSD for the Voice Access Category.

**Downstream RTP/RTCP packets must be classified into the Voice Access Category. So, a correct DSCP value must be used. These pre-requisites are satisfied if you set the DSCP=46 in the OXE PBX qos configuration.**

## 9.4 Defence

### 9.4.1 Omni Access VRRP redundancy

For these tests, two OAW are configured with VRRP redundancy. The aim is to check the phone behavior when the master vrrp OAW fails and the backup one becomes master.

#### Test description:

Two OAW are configured with vrrp redundancy. The AP has the vrrp virtual address defined as lms-ip.

The i62 phone is on call with an IP Touch or another i62 phone. We disconnect the master vrrp OAW. The expected behavior is that the call is broken (there is no firewall table exchange between the OAW) but that the system is operational.

#### Test parameters:

Security: WPA2-PSK, WPA2

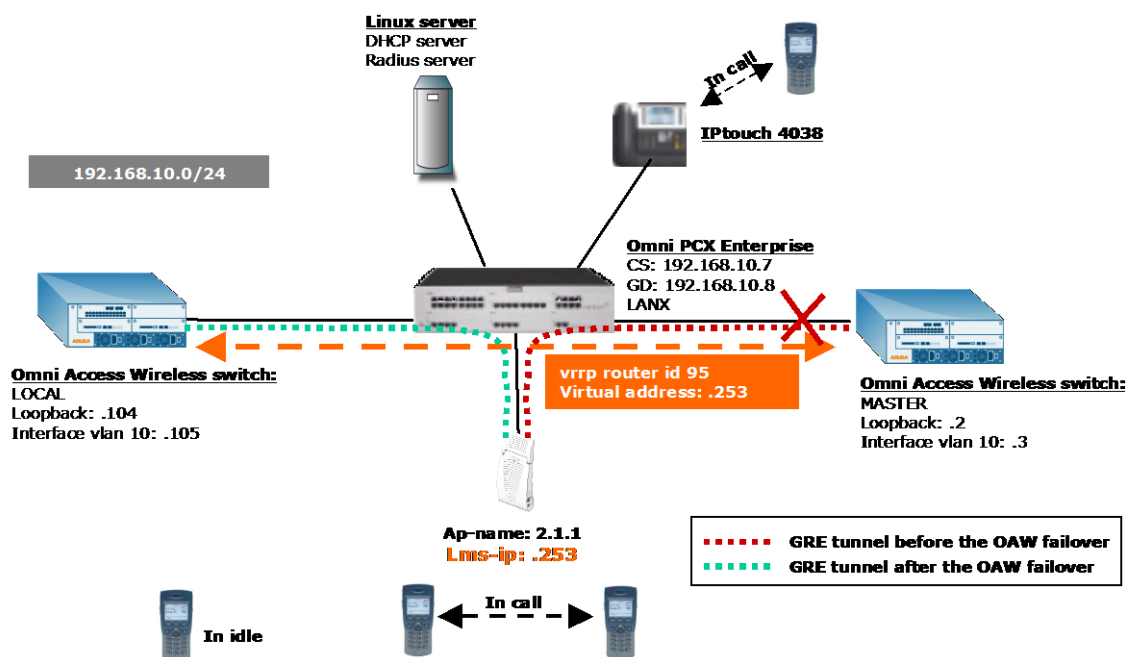
No mobility between both OAW controllers.

#### Tests results:

The following diagram describes the test platform. A vrrp router is configured on both OAW. The AP establishes its GRE tunnel to the virtual ip address.

For the test, a call between 2 i62 and a call between an i62 and an IPTouch are performed. An other i62 is in idle mode. The Ethernet link of the MASTER switch (that is the MASTER vrrp for the router id) is disconnected. The LOCAL becomes the new MASTER vrrp and the AP establishes its GRE tunnel to the LOCAL switch.

We check the state of both calls and if the i62 phones are operational.



Test	Result	Comment
Defense: i62 - IPTouch call, WPA2-PSK	OK	See remarks below
Defense: i62 - i62 call, WPA2-PSK	OK	See remarks below

After the failover, the RTP/RTCP flows are denied as there is no firewall session table replication. The i62 phones still continue to send RTP/RTCP traffic (they stay active and “connected” is displayed on the screen) even if they don’t receive RTP/RTCP packets; **they don’t hang up automatically**. The user has to hang up.

After that, they are ready to receive or initiate a new call.  
“IP mobility” must be disabled otherwise an IP mobility tunnel is created between both OAW when the failed OAW becomes the new active; this results in an instable situation.



## 10 Appendix A : AAPP member's Application description

---

- CONFIGURATION :

The WinPDM PC tool application is used to configure Ascom i62 sets. WinPDM application manages the set through PDM cradle, a specific charger with an USB connection.

Minimum PC environment for WinPDM is Xp win SP2 and Java 6.

Recover the i62 set binaries file (.bin) and i62 definition file (.def file).

Recover the WinPDM software for i62.

After WinPDM installation, connect the cradle on PC and place i62 mobile on charger with its battery. Switch on the mobile. Launch the application, WinPDM should recognise the mobile on. If not, you have to run the setup for the usb driver.

If i62 isn't recognised on base, you have to add the i62 definition file .def for WinPDM. (file > definition file > add)

Once detected, accept to store data in local database to edit parameters when mobile off line.

- SOFTWARE UPGRADE :

Select 'Devices' tab and double click in the list on device to upgrade. Import binaries ..pkg file and accept. Download is pending.

Moreover, there is a centralized device management "over the air" option. This procedure allows an update of i62 using the WLAN network. Please contact Ascom Wireless for further details.

- SETUP A TEMPLATE FOR A SITE

Select 'Template' tab and click 'new' to add and give a name to your site.

An example template is attached to this interworking report.

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

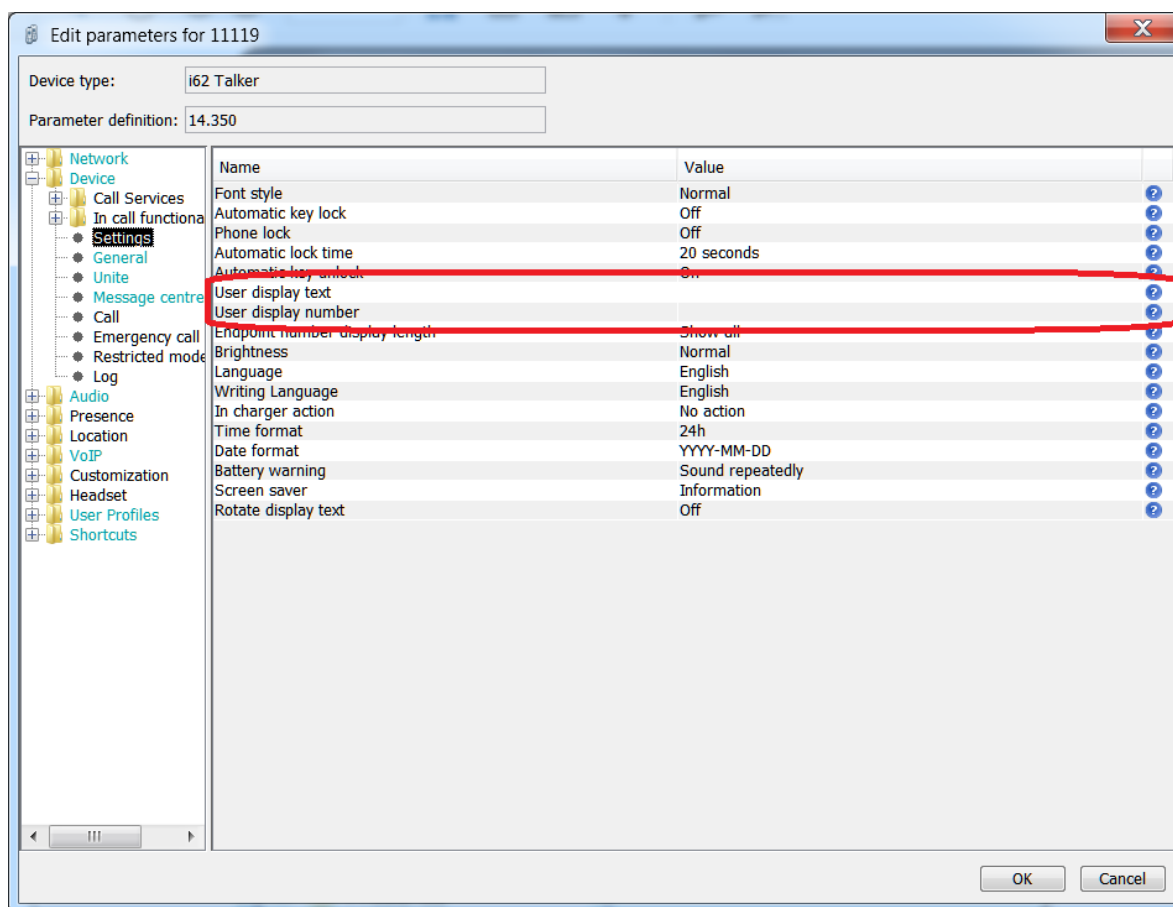
## 11.1 Network parameters

Device type: i62 Talker  
Parameter definition: 14.350

Name	Value
Network name	INTOP
DHCP mode	On
802.11 protocol	802.11a/n
SSID	INTOP5GHZ
Security mode	WPA-PSK & WPA2-PSK
WPA-PSK passphrase	*****
Voice power save mode	U-APSD
802.11a/n channels	UNII-1, UNII-2
Advanced: 802.11 channels	
World mode regulatory domain	World mode (802.11d)
Transmission power	Automatic
IP DSCP for voice	0x2E (46) - Expedited Forwarding
IP DSCP for signaling	0x1A (26) - Assured Forwarding 31
TSPEC Call Admission Control	Off
Transmit gratuitous ARP	Yes
Deauthenticate on roam	No
Roaming methodology	802.11 roaming
Maximum transfer unit	1400
Aruba 800 controller compability	No
Check IP connectivity after roaming	No

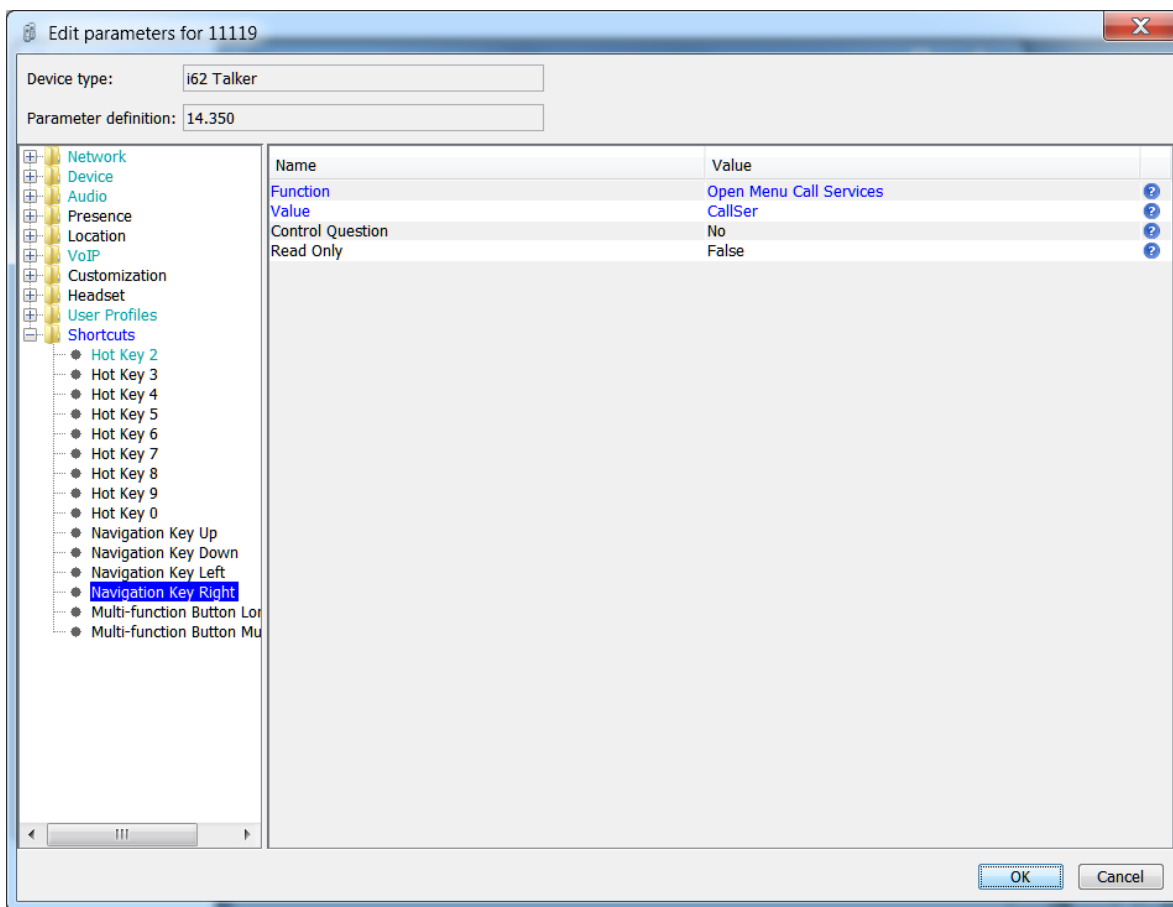
WPA2 PSK security. Reg. domain is 802.11.

## 11.2 General settings



The administrator can specify a user display text and an endpoint ID (can be used to hide the twin set extension/name in case of association with a twin set).

### 11.3 OXE feature code softkeys configuration



Edit parameters for 11119

Device type: i62 Talker

Parameter definition: 14.350

- Network
- Device
  - Call Services
    - General service 1**
    - General service 2
    - General service 3
    - General service 4
    - General service 5
    - General service 6
    - General service 7
    - General service 8
    - General service 9
    - General service 1
    - General service 1
    - General service 1
    - General service 1
    - General service 1
    - General service 1
  - In call functionality
    - Settings
      - General
      - Unite
      - Message centre
    - Call
    - Emergency call Num
    - Restricted mode
    - Log
  - Audio
  - Presence
  - Location
  - VoIP
  - Customization
  - Headset

Name	Value
Name	DND (personal code)
Data	42\U
Auto disconnect	Yes

OK Cancel

Edit parameters for 11119

Device type: i62 Talker

Parameter definition: 14.350

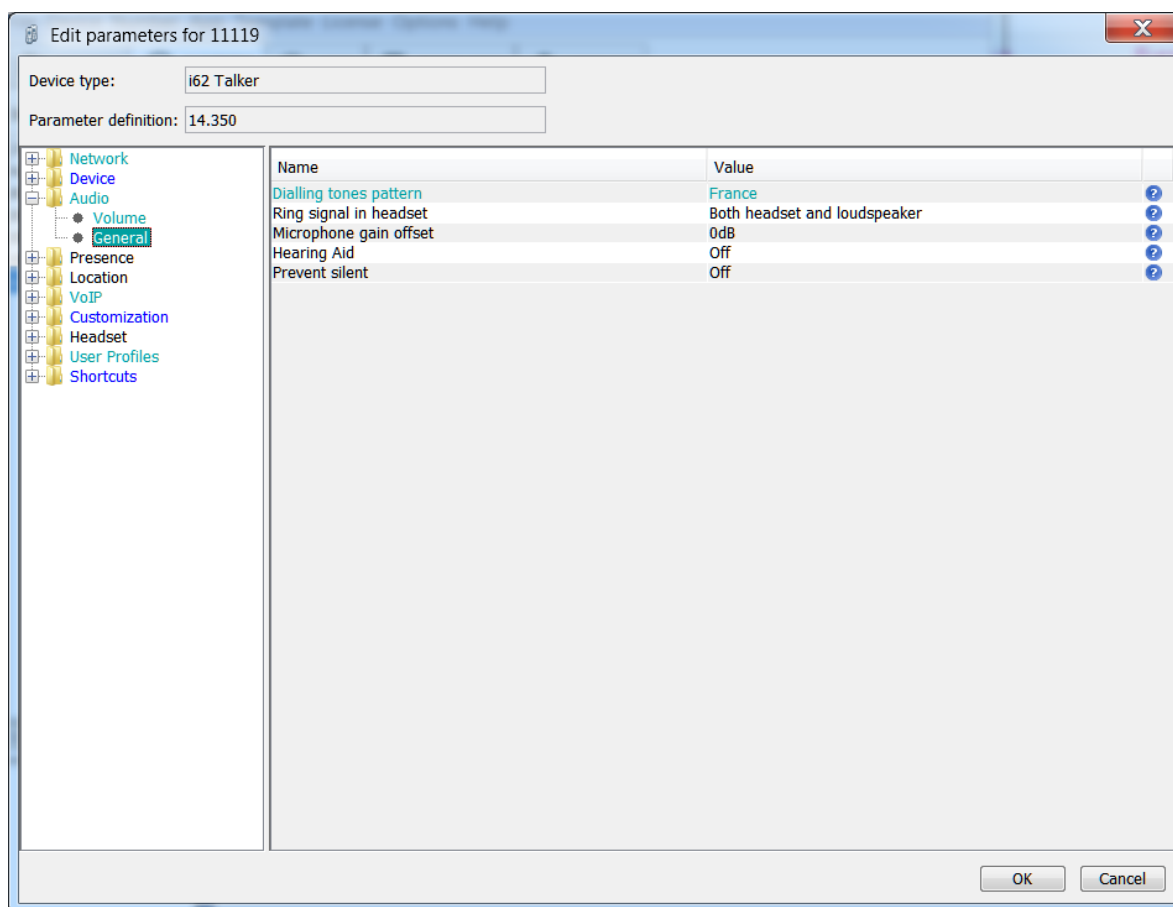
- Network
- Device
- Audio
- Presence
- Location
- VoIP
- Customization
  - Visibility**
- Headset
- User Profiles
- Shortcuts

Name	Value
Connections/Headset	Show
Connections/Network	Show
Connections/In Charger	Show
Calls	Show
Calls/Call services	Show
Calls/Call services/Divert calls	Hide
Calls/Call services/Do not disturb	Hide
Contacts	Show
Contacts/Central Phonebook	Show
Shortcuts/Soft keys	Show
Shortcuts/Hot keys	Show
Shortcuts/Navigation keys	Show
Shortcuts/Multifunction button	Show
Profiles	Show
Profiles/Normal	Show
Profiles/Profile 1	Show
Profiles/Profile 2	Show
Profiles/Profile 3	Show
Profiles/Profile 4	Show
Settings/Sounds & Alerts	Show
Settings/Sounds & Alerts/Volume	Show
Settings/Sounds & Alerts/Ring signals	Show
Settings/Sounds & Alerts/Vibrating alert	Show
Settings/Sounds & Alerts/Key sound	Show
Settings/Sounds & Alerts/Battery warning	Show
Settings/Locks	Show
Settings/Display	Show
Settings/Time & Date	Show
Settings/Answering	Show
Settings/Language	Show
Settings/Owner ID	Show
Settings/Device Info	Show

OK Cancel

This configuration allows hiding local set features. The administrator can create a call service corresponding to an OXE feature as presented in the previous captures for Do Not Disturb feature.

## 11.4 Tones configuration



Adapt the dialling tones pattern to your OXE location.

## 11.5 SIP configuration

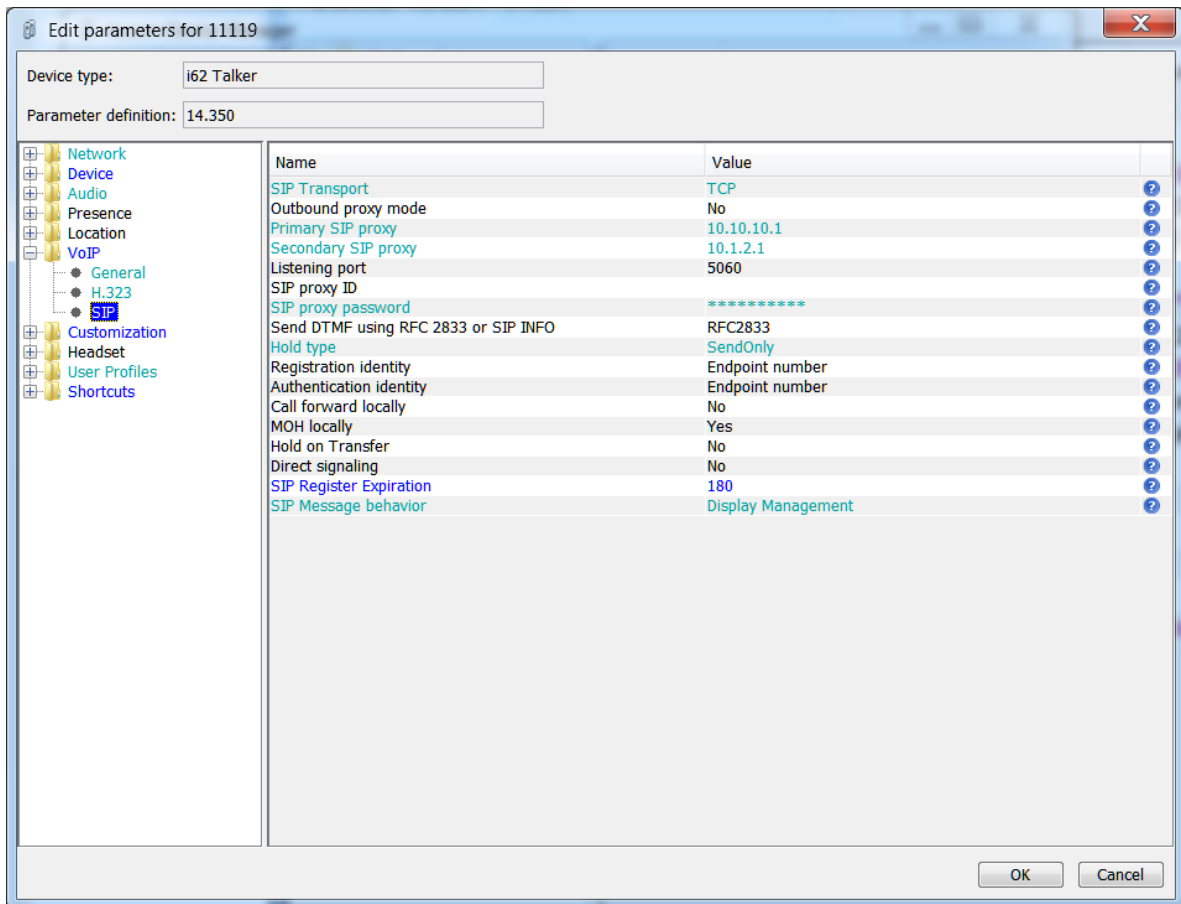
Device type:

Parameter definition:

Name	Value
Replace Call Rejected with User Busy	No
ICE negotiation	No
VoIP protocol	SIP
Codec configuration	G.711 A-law
Codec packetization time configuration	20
Offer Secure RTP	No
Internal call number length	5
Endpoint number	11119
Endpoint ID	

OK Cancel





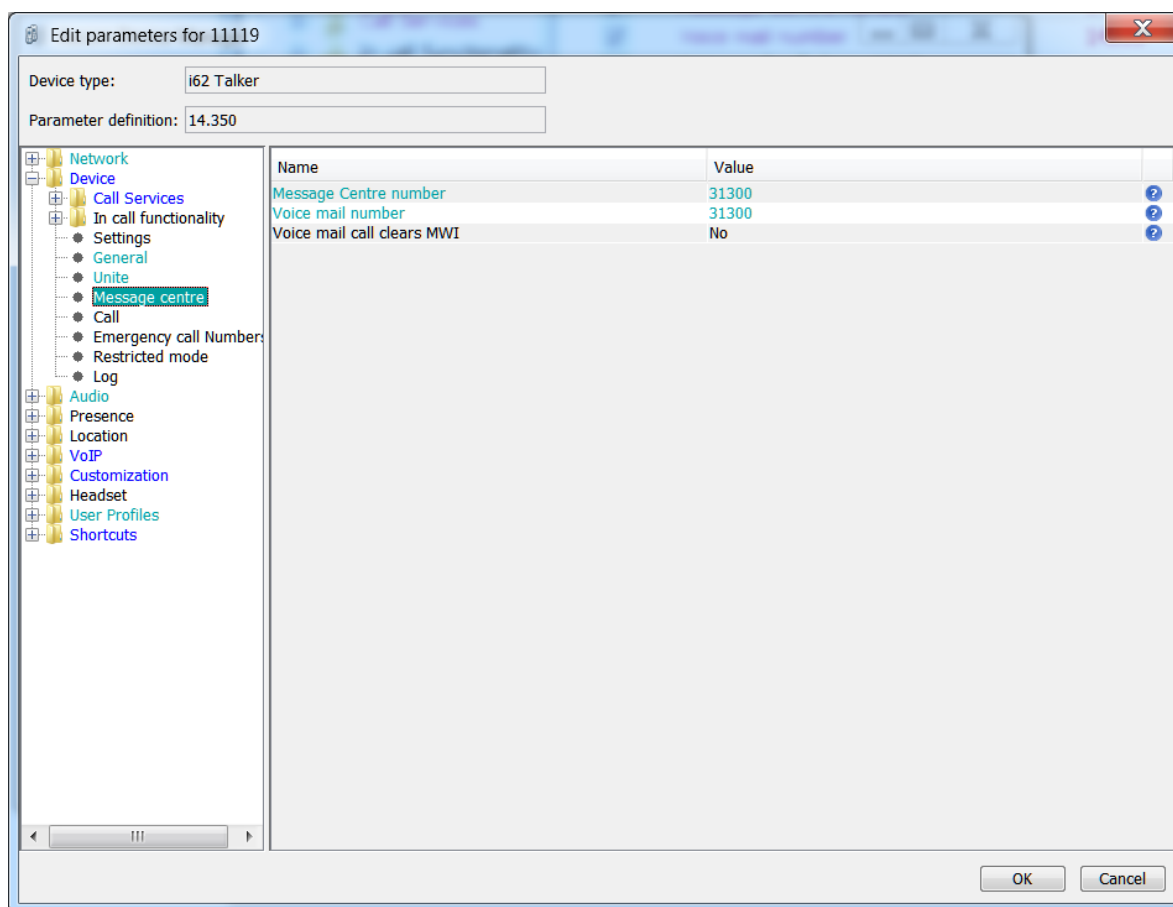
Here is the example with OXE CPU ip addresses. SIP registration should be configured with 180 seconds, REGISTER must not be used like a Keep-Alive mechanism.

Redundancy tests with DNS method have also been done with following settings:

Primary SIP proxy with etesting.xmlforum.com

SIP proxy ID with etesting.xmlforum.com

## 11.6 Voice mail configuration



Specify the voice mail number to activate voice mail speed dial and message notifications.

- SET UP A NEW NUMBER IN THE SITE

Select 'Number' tab and click 'new' to add a number. Select the related template.  
You can write a range of numbers.

Note: Edit a number doesn't modify the template

- ERASE A NUMBER / DEVICE

Go to a 'device' menu > factory reset and then associate with number. You can unplug the device to see the screen "Remotely updated"

#### Display information on handset:

**"No system"**: i62 cannot find the wireless infrastructure with settings matching those configured in the handset. Check the VoWLAN ESSID, the security settings, the regulatory domain settings and the channels used. Then correct system (A, B, C or D) setting if wrong selected.

**"No access"**: Once i62 is associated to the VoWLAN, it could encounter network issues. Check first that the handset has an IP address by entering the "System information" screen and correct with appropriate IP configuration method used (DHCP server or not). Check either the gatekeeper and IMS addresses. Try to ping the default gateway from another wireless client. Try to ping the IMS from another wireless client. Check the SIP proxy address and then authentication method used, i62 does not support Digest authentication method.

**"Voice only"**: means the handset is configured to use both a gatekeeper and an IMS/IP-WiFi, but the set has lost contact with the gatekeeper.

**"Messaging only"**: means the handset is configured to use both a gatekeeper and an IMS/IP-WiFi, but the set has lost contact with the IMS/IP-WiFi.


## 12 Appendix C: Alcatel-Lucent Enterprise Communication Platform: configuration requirements

List of prefixes and suffixes defined on OmniPCX TSS lab system. These prefixes can be entered in the call services menu (on WinPDM Device>Call Services) to be used by the end customer via a speed dial button on the i62 (on i62 menu>Calls>Call services):

dir	mean
400	Set_In/Out_of_service
401	Recordable_Voice_Guides
402	Park_Call/Retrieve
403	Charging_meter_readout
404	Associated_Set_No_Modif
405	Password_modification
406	Redial_last_number
407	Night_service_answering
408	Contrast_programmation
409	Secret/Identity
41	Forward_cancellation
42	Do_not_disturb
43	Voice_Mail
44	Canc_auto_call_back_on_busy
45	PadLock
46	Consult_Call_back_list
470	Waiting_call_consultation
471	Business_account_code
472	Consult_Messages
473	Paging_call_answer
474	Language
480	Set_group_entry
481	Set_group_exit
482	Switch_off_Message_LED
483	Mask_Remote_Calling_Identity
484	Cancel_Remote_forward
485	Overfl_busy_to_assoc_set
486	Overf_busy/no_repl_assoc_set
487	Recording_Conversation
490	Ubiquity_Mobile_Programming
491:493	Ubiquity_Services_Pfx
495	Ubiquity_Assistant
500	Last_Caller_Call_back
501	Remote_forward
502	Overflow_on_associated_set
503	Cancel_Overfl_on_assoc_set
504	Protection_against_beeps
505	Substitution
506	Wake_up/appointment_remind
507	Cancel_Wake_up
508	Forward_cancel_by_destinat
509	Meet_me_Conference
51	Immediate_forward
52	Immediate_forward_on_busy
53	Forward_on_no_reply
54	Forward_on_busy_or_no_reply
55	Direct_call_pick_up
56	Group_call_pick_up
570	Voice_Mail_Deposit
580	Tone_test
581	Personal_directory_Progr
582	Personal_Directory_Use

583	Force_type_identification_pfx	
584	Suite_Wakeup	
585	Suite_Wakeup_Cancel	
586	Suite_Dont_Disturb	
587	Room_status_management	
588	Mini_bar	
589	Direct_Paging_Call	
591	Pabx_address_in_DPNSS	
599	Professional_trunk_seize	
899	Pabx_address_in_DPNSS	
9	Attendant_Call	
*	DTMF_End_to_End_Dialling	
#	Speed_call_to_associated_set	

162 are declared as SEPLOS SIP users (SIP extension):

	 13113 ASCOM
<b>Common attributes</b>	
<i>Hierarchy</i>	/ ALE AAPP / etesting1
<b>User type</b>	OXE
Salutation	
<b>Last name</b>	13113
First name	ASCOM
<b>User ID</b>	ASCOM 13113
8770 password	
Email address	
<b>OXE attributes</b>	
OXE Meta-Profile	
<b>OXE ID</b>	etesting1
<b>OXE directory number</b>	13113
Device type	SIP extension
OXE profile	
Key Profiles	None
SIP password	*****
OXE mailbox directory number	31300
Cost center	
Applications	None

OXE SIP Gateway management:

<b>Instance (reserved)</b>	1
SIP Subnetwork	5
SIP Trunk Group	1
IP Address	10.10.10.1
Machine name - Host	etesting
SIP Proxy Port Number	5060
SIP Subscribe Min Duration	1800
SIP Subscribe Max Duration	86400
Session Timer	180
Min Session Timer	120
Session Timer Method	RE_INVITE
DNS local domain name	XmiForum.com
DNS type	DNS A
SIP DNS1 IP Address	10.1.2.15
SIP DNS2 IP Address	
SDP in 18x	<input checked="" type="checkbox"/>
Cac SIP-SIP	<input type="checkbox"/>
INFO method for remote extension	<input type="checkbox"/>
Dynamic Payload type for DTMF	97
Overflow Licenses Threshold	80

OXE SIP Proxy management:

<b>Instance (reserved)</b>	1
SIP initial time-out	500
SIP timer T2	4000
Dns Timer overflow	100000
Timer TLS	30
Recursive search	<input type="checkbox"/>
Minimal authentication method	SIP Digest
Authentication realm	alcatel
Only authenticated incoming calls	<input type="checkbox"/>
Framework Period	3
Framework Nb Message By Period	25
Framework Quarantine Period	1800
TCP when long messages	<input checked="" type="checkbox"/>
Retransmission number for INVITE	3
Degraded mode Time To Live	1800
User Agent Identifier	%

OXE SIP Registrar management:

<b>Instance (reserved)</b>	1
SIP Min Expiration Date	180
SIP Max Expiration Date	86400

SIP Extension's Classes of service must be managed as below:

<b>Phone COS</b>	0
Display UTF-8	YES
Display call server information	YES
Keep Alive	NO
Send NOTIFY instead of MESSAGE	NO
Tandem Call Identification	YES



## OAW configuration

A complete interworking report about WLAN infrastructure is available on Ascom Partner Web site. An example of OAW configuration is attached to this Inter working.

## 13 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 CH	Juerg Schaffner	<a href="mailto:juerg.schaffner@ascom.com">juerg.schaffner@ascom.com</a>
Ascom NO	Håkon Storm	<a href="mailto:Hakon.Storm@ascom.com">Hakon.Storm@ascom.com</a>
Ascom, NL	Hans van Duijne	<a href="mailto:Hans.vanDuijne@ascom.com">Hans.vanDuijne@ascom.com</a>
Ascom, UK	Richard Turner	<a href="mailto:Richard.Turner@ascom.com">Richard.Turner@ascom.com</a>
Ascom, US	Nathan Wilson	<a href="mailto:Nathan.Wilson@ascom.com">Nathan.Wilson@ascom.com</a>
Ascom France, FR	Ludovic Petit	<a href="mailto:Ludovic.PETIT@ascom.com">Ludovic.PETIT@ascom.com</a>
Ascom Danmark, DK	Frans Richter Christensen	<a href="mailto:Frans.RichterChristensen@ascom.com">Frans.RichterChristensen@ascom.com</a>
Ascom Germany GmbH, DE	Torsten Wolf	<a href="mailto:torsten.wolf@ascom.com">torsten.wolf@ascom.com</a>
Ascom, BE	Peter Moens	<a href="mailto:Peter.moens@ascom.com">Peter.moens@ascom.com</a>
Ascom Austria, AT	Torsten Wolf	<a href="mailto:torsten.wolf@ascom.com">torsten.wolf@ascom.com</a>
Ascom Sverige, SE	Niclas Holmblad	<a href="mailto:Niclas.Holmblad@ascom.com">Niclas.Holmblad@ascom.com</a>
Ascom, IT	Tiziano Pigozzi	<a href="mailto:tiziano.pigozzi@ascom.com">tiziano.pigozzi@ascom.com</a>
International	Marko Savinainen	<a href="mailto:Marko.savinainen@ascom.com">Marko.savinainen@ascom.com</a>

## 14 Appendix E: AAPP program

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### 14.1 Alcatel-Lucent Application Partner Program (AAPP)

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

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

- **Provide easy interfacing for Alcatel-Lucent Enterprise communication products:** Alcatel-Lucent Enterprise's communication products for the enterprise market include infrastructure elements, platforms and software suites. To ensure easy integration, the AAPP provides a full array of standards-based application programming interfaces and fully-documented proprietary interfaces. Together, these enable third-party applications to benefit fully from the potential of Alcatel-Lucent Enterprise products.
- **Test and verify a comprehensive range of third-party applications:** to ensure proper inter-working, ALE International tests and verifies selected third-party applications that complement its portfolio. Successful candidates, which are labelled Alcatel-Lucent Enterprise Compliant Application, come from every area of voice and data communications.

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

## Web site

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

The screenshot displays the Alcatel-Lucent Enterprise Portal. At the top, the Alcatel-Lucent logo and 'Enterprise Portal for certified applications' are visible. A navigation menu includes 'Home', 'About the program', 'Join the program', 'Partnerships', and 'APIs'. A search bar and 'Member Resource Center' link are also present. The main content area features a 'Latest news' banner for TAPI 4.0.6 compatibility. Below this, a large section titled 'AAPP Interworking Reports' states that IWRs are now available in public access, with a 'Visit the list' button and a pagination control showing '1 | 2 | 3 | 4 | 5'. To the left, a 'Browse' section offers links for 'All applications' and 'Find an application'. To the right, a 'Benefit from the Program services' section provides a link to 'Learn more about program services'. A sidebar on the right contains promotional banners for 'Alcatel-Lucent enterprise products', 'the AAPP Factory', and 'Join now', along with a 'Quick Access' section highlighting 'Interworking Reports (public access)'.

## 14.2 Enterprise.Alcatel-Lucent.com

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

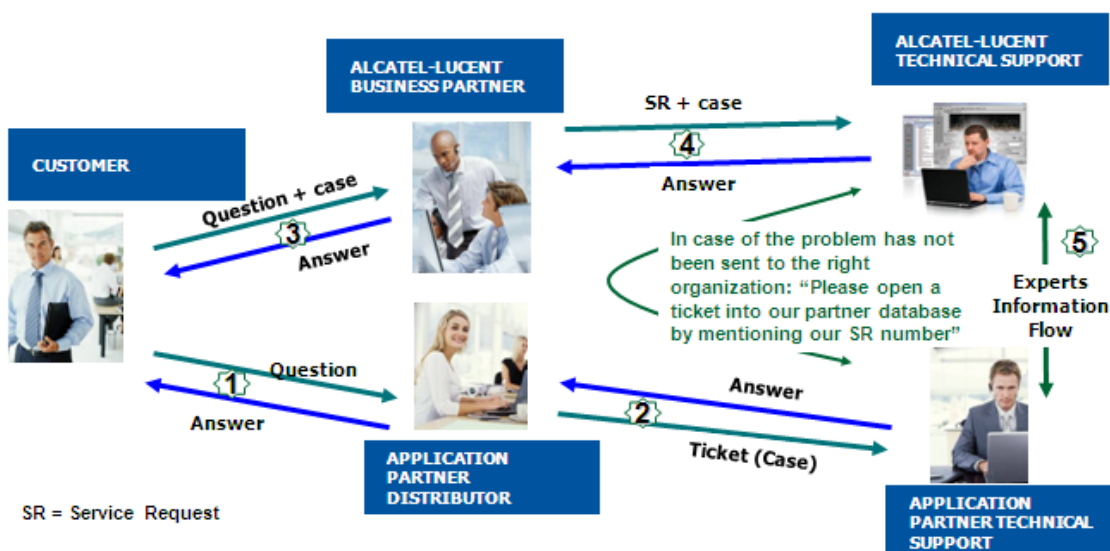
# 15 Appendix F: AAPP Escalation process

## 15.1 Introduction

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

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

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



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

## 15.2 Escalation in case of a valid Inter-Working Report

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

This defines the scope of what has been certified.

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

Case 1: the responsibility can be established 100% on ALE International side.

In that case, the problem must be escalated by the ALE Business Partner to the ALE International Support Center using the standard process: open a ticket (eService Request – eSR)

Case 2: the responsibility can be established 100% on Application Partner side.

In that case, the problem must be escalated directly to the Application Partner by opening a ticket through the Partner Hotline. In general, the process to be applied for the Application Partner is described in the IWR.

Case 3: the responsibility can not be established.

In that case the following process applies:

- The Application Partner shall be contacted first by the Business Partner (responsible for the application, see figure in previous page) for an analysis of the problem.
- The ALE International Business Partner will escalate the problem to the ALE International Support Center only if the Application Partner has demonstrated with traces a problem on the ALE International side or if the Application Partner (not the Business Partner) needs the involvement of ALE International

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

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

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

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

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

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

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

## 15.3 Escalation in all other cases

These cases can cover following situations:

1. An InterWorking Report exist but is not valid (see Chap 2 “Validity of the Interworking Report”)
2. The 3<sup>rd</sup> 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 3<sup>rd</sup> party company is NOT referenced as AAPP participant

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

## 15.4 Technical support access

The ALE International **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 ALE International Business Partners Web site (if registered Alcatel-Lucent Enterprise Business Partners): <https://businessportal.alcatel-lucent.com> click under "Contact us" the *eService Request* link
- e-mail: [Ebg\\_Global\\_Supportcenter@al-enterprise.com](mailto:Ebg_Global_Supportcenter@al-enterprise.com)
- Fax number: +33(0)3 69 20 85 85
- Telephone numbers:

ALE International Business Partners Support Center for countries:

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

For other countries:

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

END OF DOCUMENT