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Certified Application

Application Partner Program
Inter-Working Report

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

ascom

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

Certification overview

Date of the certification	February 2014

Alcatel-Lucent's representative	Residori Florian
AAPP member representative	Peter Åstrand

Alcatel-Lucent Communication Platform	OmniPCX Enterprise
Alcatel-Lucent Communication Platform release	OXE : R11.0 (k1.400.30f) OAW: 6.3.1.2 with AP105/12x
AAPP member application release	i62 5.1.22
Application Category	Terminals DECT / Wi-Fi Mobility

Author(s):   Residori Florian
Reviewer(s):   Lienhart Denis, Ascom/Gert Wallin

Historic
Edition 1: Recertification with Ascom release i62 5.1.22, OXE R11.0, OAW 6.3.1.2– *February 2014*

Test results OXE 11

☐ Passed   ☐ Refused   ☐ Postponed

☑ Passed with restrictions

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

IWR validity extension

None

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1 Introduction

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

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

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

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

- the Application Partner portal (https://applicationpartner.alcatel-lucent.com) with free access.

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

2 Validity of the Interworking Report

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

This inter-working report is valid unless specified until the AAPP member issues a new major release of such product (incorporating new features or functionalities), or until 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 Interworking report becomes automatically obsolete when the mentioned product releases are end of life.

3 Limits of Technical support

Technical support will be provided only in case of a <u>valid Interworking Report</u> (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 Summary of test results

4.1 Summary of main functions supported

4.1.1 OXE / SEPLOS

- 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
- Registration
- Prefixes are supported (mainly all SEPLOS features)
- Basic communications (internal and external, intra-node, inter-nodes)
- Supports telephonic features (Broker call, Transfer, Conference, Call back, Waiting, Forwards, in intra-node and inter-nodes communications).
- Attendant calls
- Access to the OXE Voice mail boxes and DTMF dialling for navigation
- Codec choice

4.1.2 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. Ascom CR18091
    - 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 (900 seconds)+ a sip message timeout (32s).
    - Switches between call servers only occur after "REGISTER".

4.3 Summary of limitations

4.3.1 OXE / SEPLOS

- SIP Keep Alive mechanism with SIP OTIONS 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 form the AP. The association is possible when one call is ended on the AP.
- 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.

4.3.2 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) than 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
- Only AP105/12x Access Points are covered by this IWR.
- 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.
- 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 :
  - o Non Conformity Report (NCR)
  - o Change or feature Request (CR)

5 Application information

Application type: VoWiFi handset

Application commercial name: i62

Application version: 5.1.22

Interface type: 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:
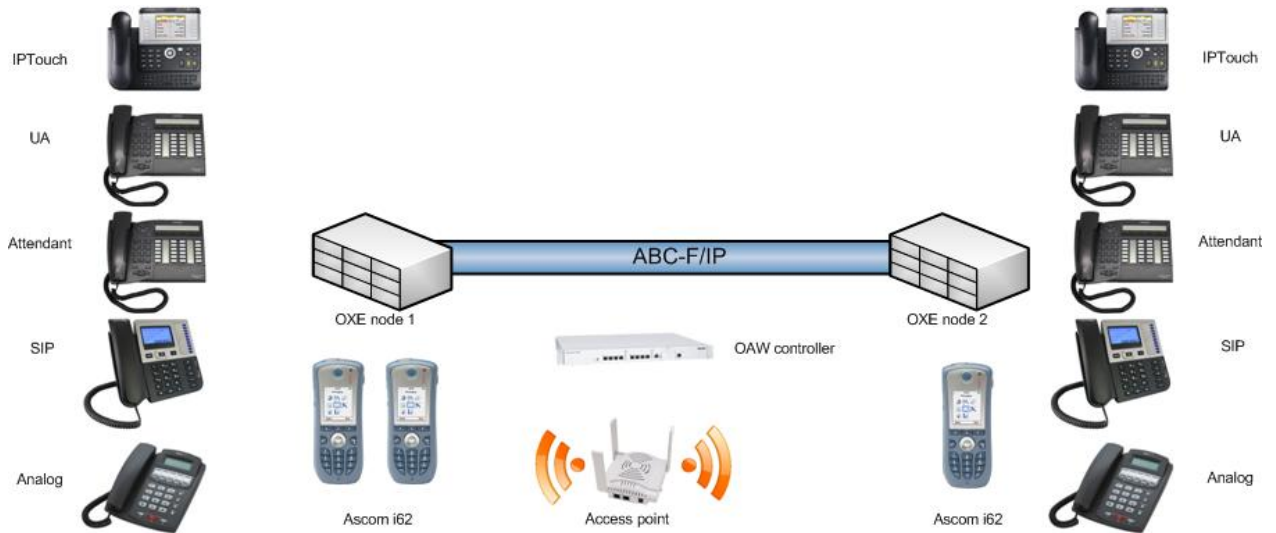
6.2 Hardware configuration:

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 Communication Platform**: OmniPCX Enterprise R11 (k1.400.30f)

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 AP105/12x access points. Two controllers were deployed in a Master/Local redundant configuration.

- **Application platform**: i62 5.1.22.

7 Test Result Template

The results are presented as indicated in the example below:

Test	Action	Result	Comment

Test/Step: Number of the test/step. A test may comprise multiple steps depending on its complexity. Each step has to be completed successfully in order to conform to the test. Step 0 when present represents the initial state for all the following steps.

Action: describes which action to realize in order to set-up the conditions of the test.

Result: describes the result of the test from an external point of view.
> **OK** for positive result.
> **NOK** for negative result. In the latter case, the Comment column describes as precisely as possible the problem.
> **N/A** if this test is not applicable to this application. (Use Comment column to describe why).
> **NTY** apply for "Not Tested Yet" tests.

Comment: This column has to be filled in when a problem occurs during the test and if any additional restriction applies or information has to be communicated. It must contain a high level evaluation of the localization of the responsibility: Alcatel-Lucent or the Partner.

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 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	SIP registration, using DNS (without authentication) / 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 (without authentication) / 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: Dial a part of the number, wait and continue.	OK	
3	Call to a local user with overlap dialing, timeout: Dial a part of the number, wait and stop.	NA	No timeout.
4	Call to a local user with overlap dialing, release: Dial a part of the number, wait and release the call.	OK	
5	Call to local user with no answer.	OK	
6	Call to another SIP set / The SIP phone under test calls another SIP phone. Check the display and audio during all steps (dialing, ring back tone, conversation, release).	OK	
7	Call to wrong number (SIP: "404 Not Found")	OK	Displays "Invalid Number"
8	Call to idle user, decline this call (SIP: "603 Decline")	OK	Displays "Call failed" Note : OXE usually implements coverage for callee (ie forwarded to attendant, VM ask for a callback…).
9	Call to busy user (SIP: "486 Busy Here") Check busy tone.	OK	
10	Call to user in "Out of Service" state (SIP: "480 Temporarily Unavailable")	OK	Display "No response"
11	Call to user in "Do not Disturb" state (SIP: "603 Decline" or "486 User Busy") / Dial "42+password", wait for acknowledgement ring back tone from OXE	OK	Busy Ringtone is played
12	Call to local user, immediate forward (CFU). (SIP: "302 Moved Temporarily")(1) / Dial "51 + target extension", wait for acknowledgement ring back tone from OXE / Dial "41" for forward cancellation	OK	

13	Call to local user, forward on no reply (CFNR). (1) / Dial "53 + target extension", wait for acknowledgement ring back tone from OXE / Dial "41" for forward cancellation	OK	
14	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	
15	Call to a local user with proxy Authentication	OK	
16	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)
17	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
18	Call to external number (via T2 loopback) (Check ring back tone, called party display)	OK	
19	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  |   3  |
|    type | NIPR | IP_R |
| allowed | ffff |    1 |
|    used |    0 |    0 |
| IPP Intr| G711 | G711 |
| IPP Extr| G729 | G729 |
```

Partner SIP set is in domain 3.

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 unplugged SIP terminal	OK	
4	Local/network call to SIP terminal in Do Not Disturb (DND) mode		
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 immediate forward (CFU) to local user:		
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 immediate forward (CFU) to network number:		
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 immediate forward (CFU) to another SIP user		
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	OK	
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)		
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
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
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	

18	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 Check audio and display.	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 prefix+number again.	OK	
6	Send/receive DTMF	OK	
7	Meet Me conference	OK	Post-dial issue (i62 was not able to dial the prefix, then the target extension while connected to the gateway driver) corrected with i62 maintenance release 2.5.7
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".	OK	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 proprietary OXE (Z/UA/IP) 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	
2	SIP	OXE	OXE/Ext. call	NA	Not provided by OXE
3	OXE/Ext. call	OXE	SIP	NA	Not provided by OXE
4	SIP	OXE	SIP	NA	Not provided by OXE
5	OXE/Ext. call	SIP	SIP	OK	
6	SIP	SIP	OXE/Ext. call	OK	
7	SIP	SIP	SIP	OK	

Note: The call between the transferee and the transferor is disconnected if the transferor 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	
3	OXE/Ext. call	OXE	SIP	OK	
4	SIP	OXE	SIP	OK	
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 converstaion 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	SIP set call to attendant With SIPset-1, call attendant with prefix 9 (attendant call prefix), attendant answers. Check ringing back tone, display and audio.	OK	
2	2nd incoming call to SIP set while in conversation with attendant. SIPset-1 being in conversation with the attendant. Make an ISDN incoming call to SIPset-1 and try to answer the call. 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-1stays in conversation with the attendant station (SIPset-1 can display an error message when it tries to put the attendant station on hold).	OK	
3	SIP set call to attendant, attendant transfers to OXE set, semi-attended With SIPset-1, call attendant with prefix 9 (attendant call prefix), attendant answers. From the attendant, call OXEset-1 and transfer semi-attended. Answer the call and check audio and display. Redo the test with NwkSIPset-1.	OK	

4	SIP set call to attendant, attendant transfers to OXE set, attended With SIPset-1, call attendant with prefix 9 (attendant call prefix), attendant answers. From the attendant, call OXEset-1 and transfer attended. Check audio and display. Redo the test with NwkSIPset-1.	OK	
5	OXE set calls to attendant, attendant transfers to SIP set, semi-attended With OXEset-1, call attendant with prefix 9 (attendant call prefix), attendant answers. From the attendant, call SIPset-1 and transfer semi-attended. Answer the call and check audio and display.	OK	
6	OXE set calls to attendant (using attendant call prefix), attendant transfers to SIP set, attended With OXEset-1, call attendant with prefix 9 (attendant call prefix), attendant answers. From the attendant, call SIPset-1 and transfer attended. Check audio and display.	OK	
7	External ISDN Call to attendant, attendant transfers to SIP set, attended ISDN call to the attendant. From the attendant, call SIPset-1 and transfer attended Check audio and display	OK	
8	SIP set call to attendant, attendant transfers to External, semi-attended From SIPset-1, call attendant with prefix 9 (attendant call prefix), attendant answers. From the attendant, call an external ISDN destination and transfer semi-attended Once in conversation, check audio and display.	OK	

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	
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	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 sould 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 Ciphering 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 µ
- 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:

    o call-admission-control

    o sip-call-capacity 2

    o 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 form 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.
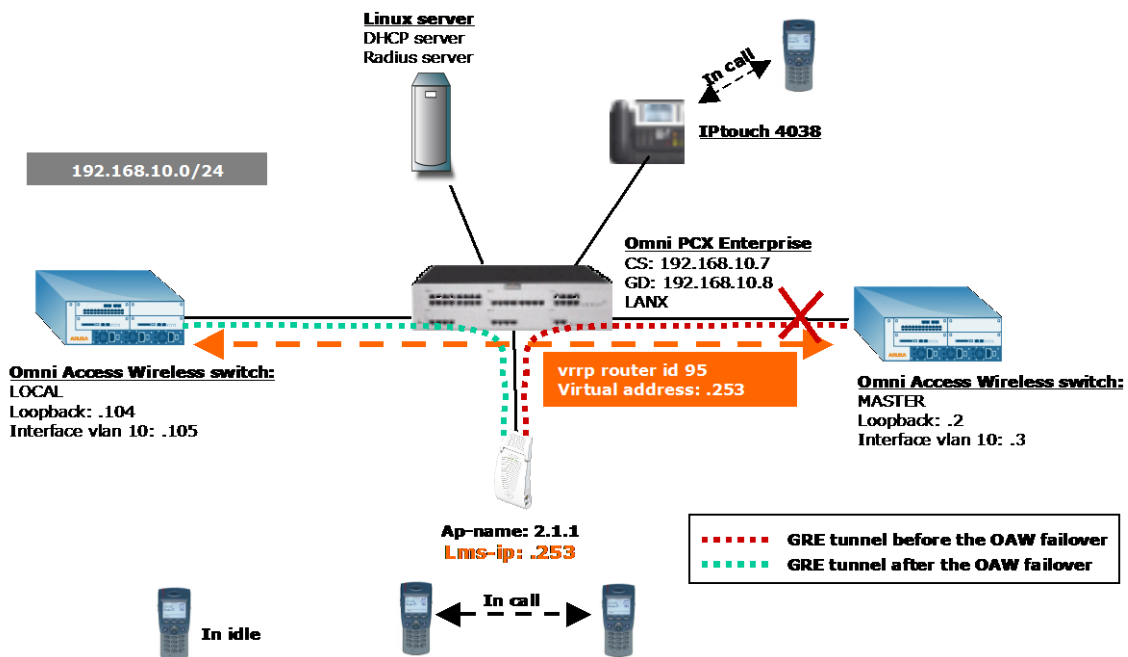
Test		Result	Comment

| 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 : Application description

    o   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.

    o   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.

    o   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 attaced to this interworking report.

10.1 Network parameters
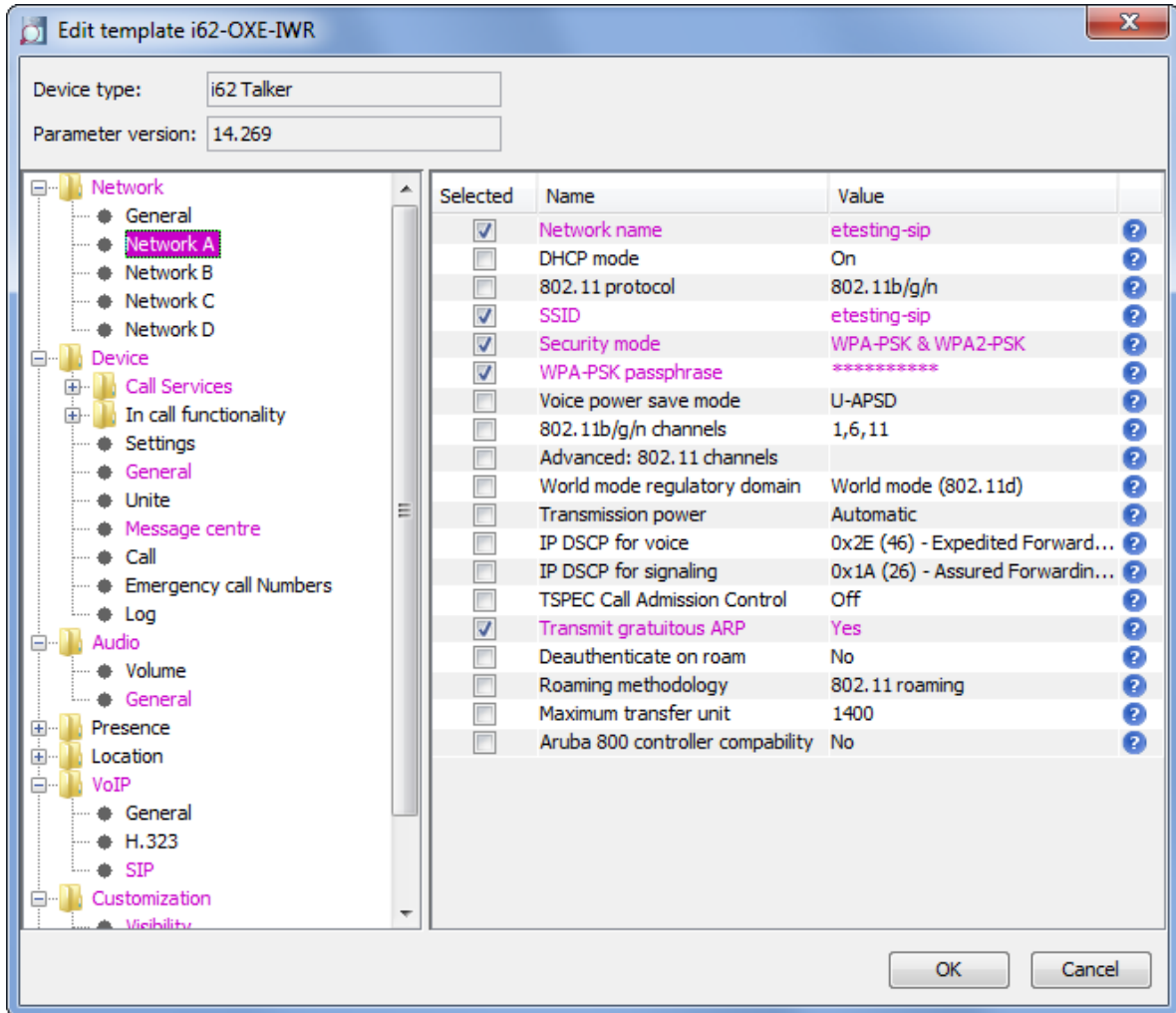
Here system A sets up an access to etesting-sip VoWLAN with WPA2 PSK security. Reg. domain is 802.11.

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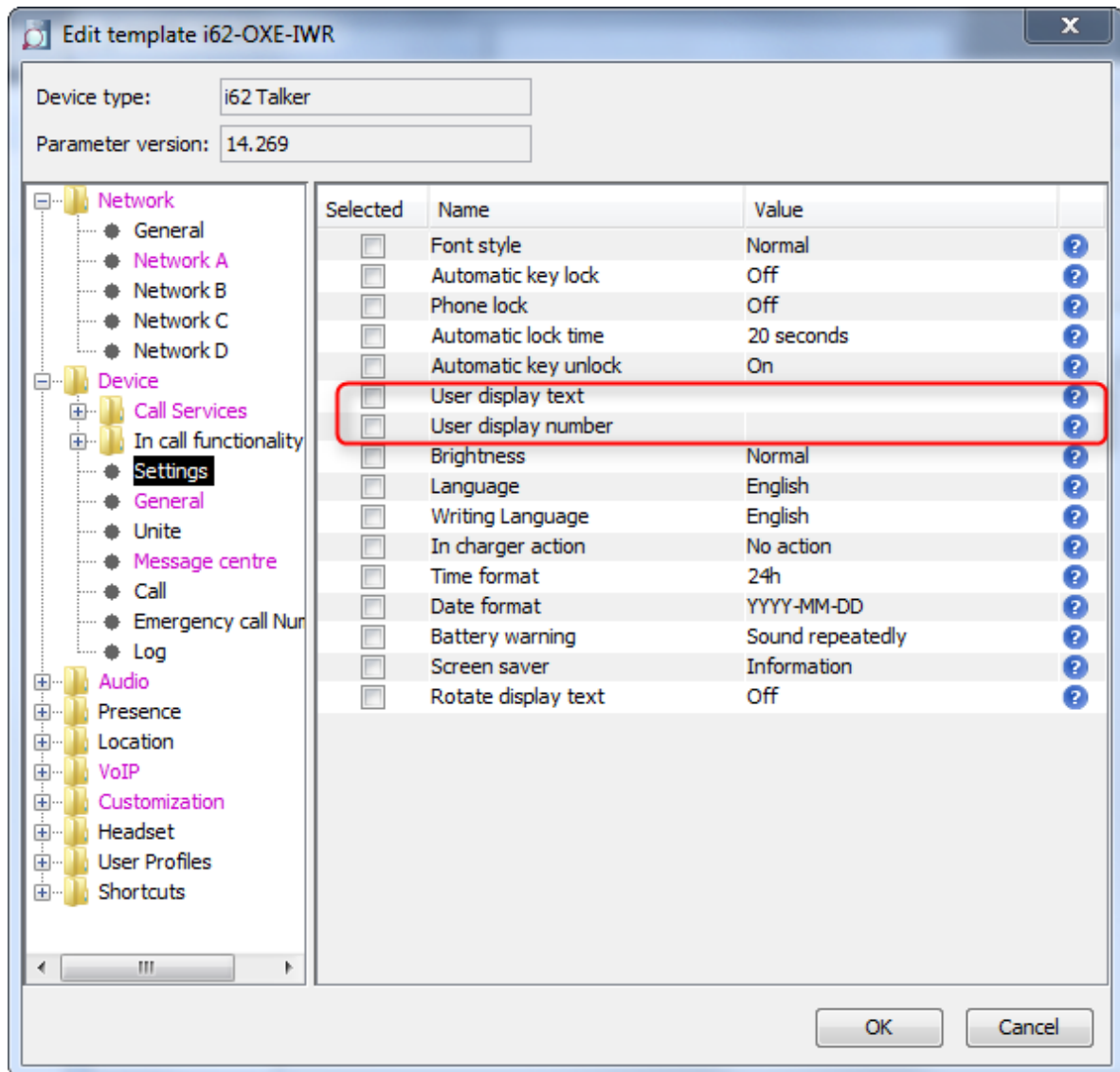
The administrator can specify a user display text and an endpoint ID (can be used to hide the twin set extension/name in case of association with a twin set).

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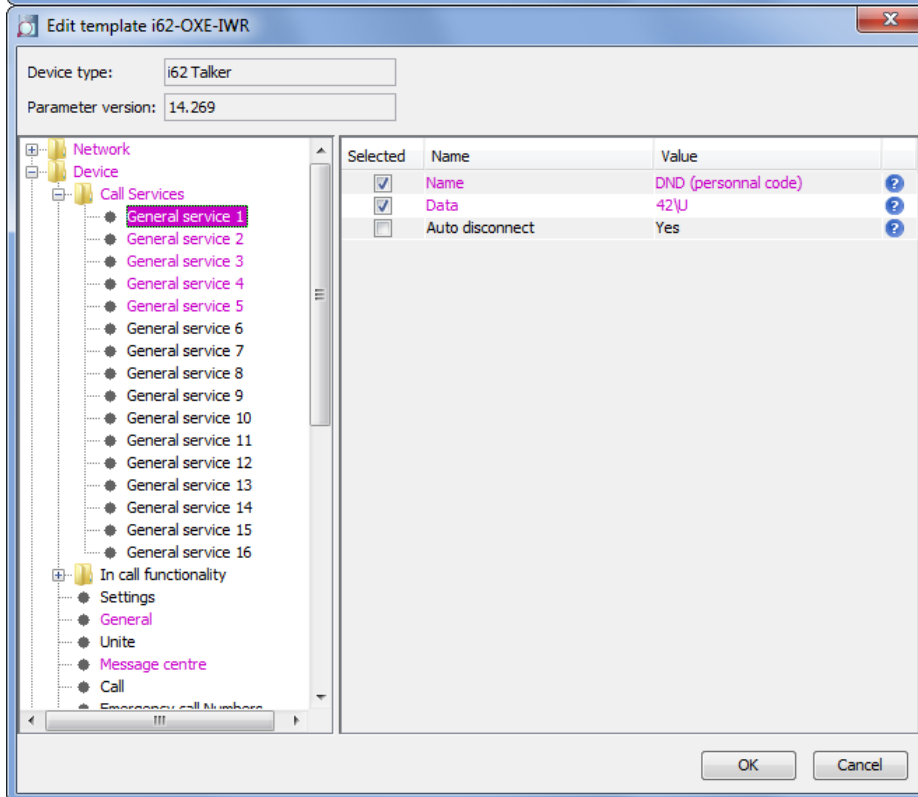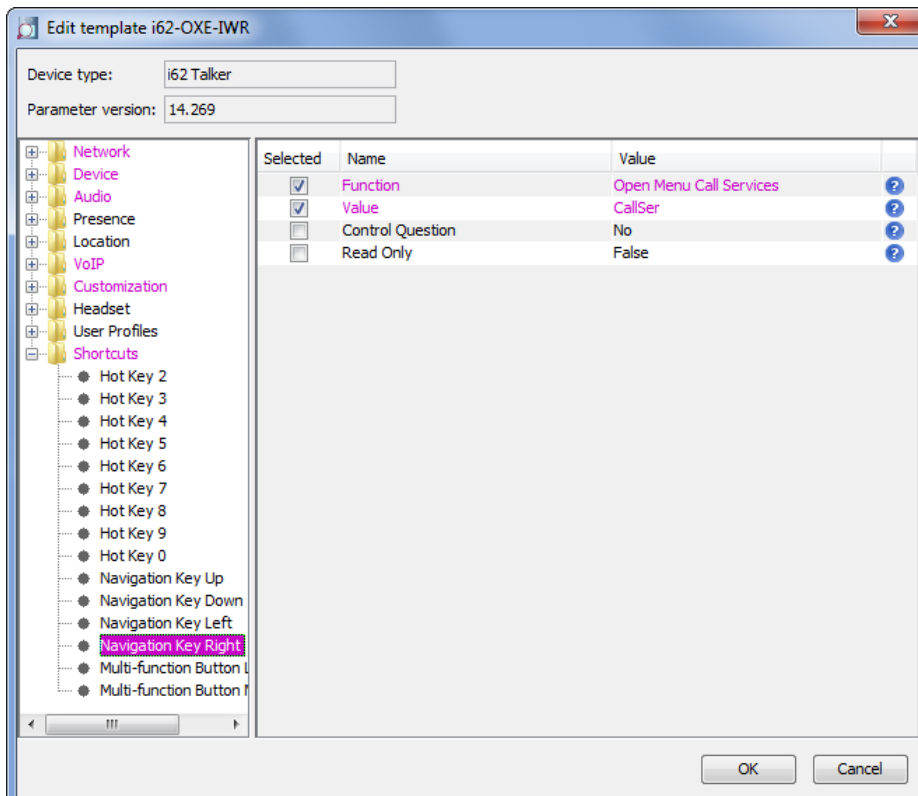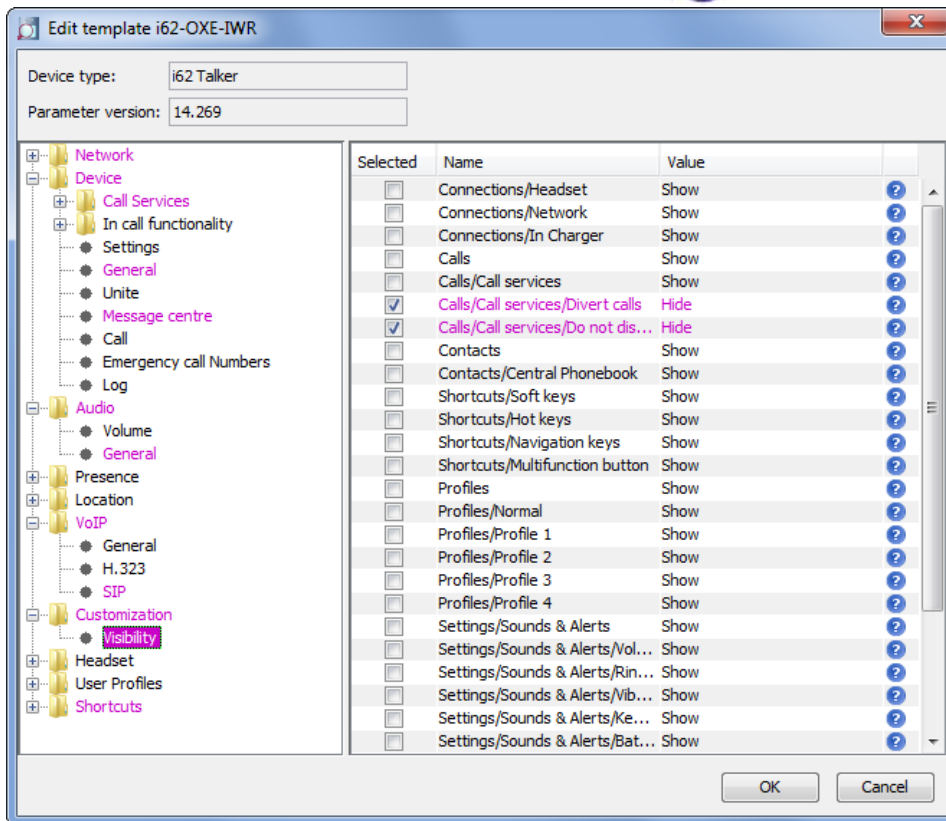
This configuration allows hiding local set features. The administrator can create a call service corresponding to an OXE feature as presented in the previous captures for Do Not Disturb feature.

10.4 Tones configuration
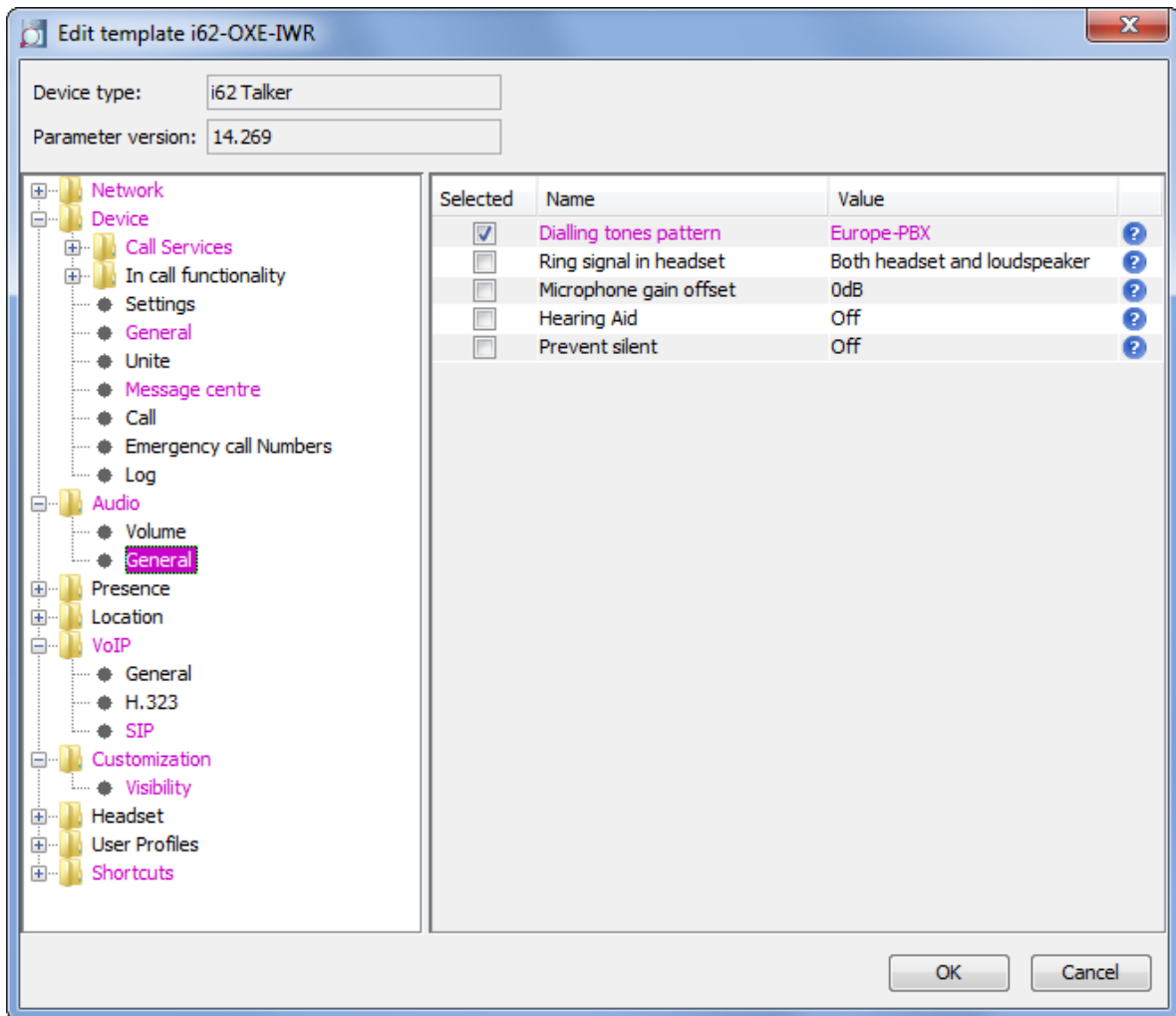
Addapt the dialling tones patern to your OXE location.

10.5 SIP configuration
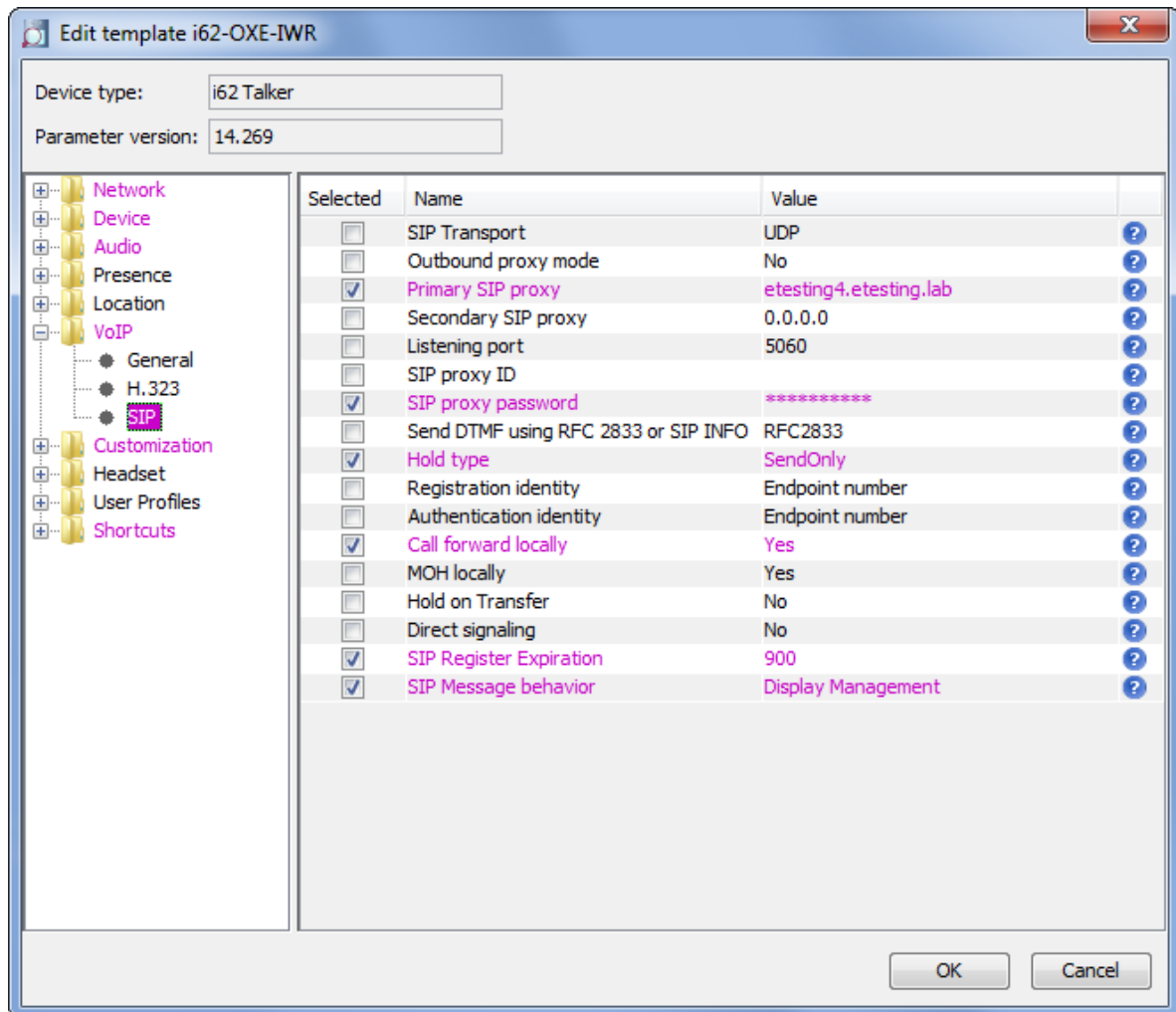
Here the settings of the SIP configuration. It is the OXE node+domain name. SIP registration should be configured with 900 seconds, REGISTER must not be used like a Keep-Alive mechanism.

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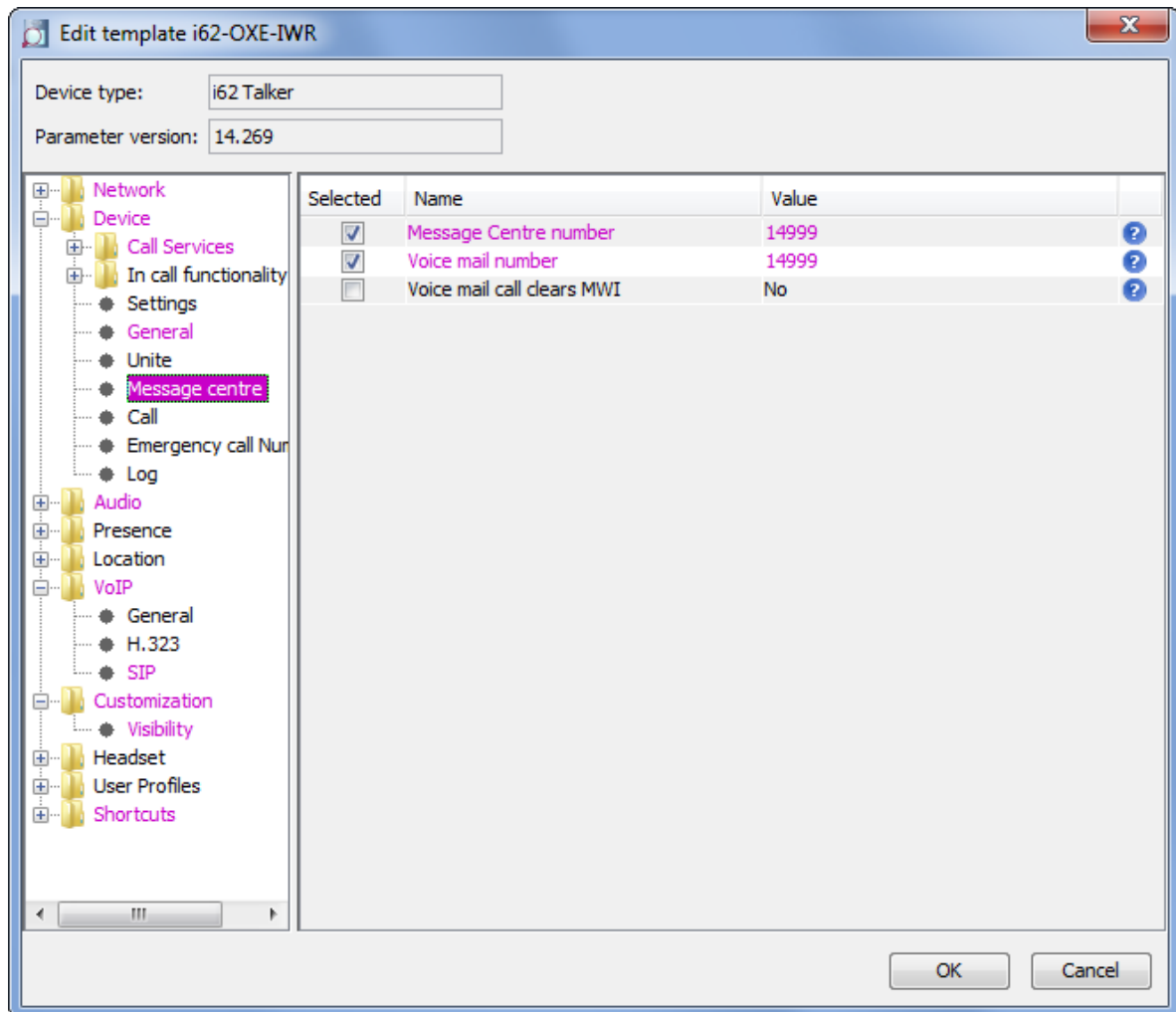
Specify the voice mail number to activate voice mail speed dial and message notifications.

    o   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

    o   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.

11 Appendix B: Alcatel-Lucent 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                     |
```

I62 are declared as SEPLOS SIP users (SIP extension):

```
lqReview/Modify: Usersqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqq
x
x              Node Number (reserved) : 4
x                    Directory Number : 14107
x
x                      Directory name : ASCOM1█
x                Directory First Name : 14107
x                 UTF-8 Directory Name : ------------------------
x          UTF-8 Directory First Name : ------------------------
x                       Location Node : 4
x                       Shelf Address : 255
x                       Board Address : 255
x                   Equipment Address : 255
x                            Set Type + SIP extension
x                       Entity Number : 1
x                        Set Function + Default
x                        Profile Name : ---------------
x                        Key Profiles + None
x                   Domain Identifier : 0
x                         Language ID : 2
x
x                         Secret Code : ****
x                             Confirm : ****
x
x                 Associated Set No. : 14107
x                      Cost Center ID : 255
x                    Cost Center Name : ----------
x                        Charging COS + Justified
x                  Public Network COS  : 0
x              External Forwarding COS : 255
x           Tel.Facility Category Id : 0
x                      Connection COS : 0
x
```

OXE SIP Gateway management:

```
lqReview/Modify: SIP Gatewayqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqq
x
x              Node Number (reserved) : 4
x                   Instance (reserved) : 1
x                   Instance (reserved) : 1
x
x                      SIP Subnetwork : 1
x                     SIP Trunk Group : 1
x                          IP Address : 10.1.12.1
x                 Machine name - Host : etesting4
x              SIP Proxy Port Number : 5060
x        SIP Subscribe Min Duration : 900
x        SIP Subscribe Max Duration : 1800
x                      Session Timer : 180
x                  Min Session Timer : 90
x              Session Timer Method + UPDATE
x              DNS local domain name : etesting.lab
x                           DNS type + DNS A
x              SIP DNS1 IP Address : 10.1.2.15
x              SIP DNS2 IP Address : ------------------
x                        SDP in 18x + True
x                      Cac SIP-SIP + True
x   INFO method for remote extension + False
x      Dynamic Payload type for DTMF : 101
x
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqq
```

Master and Standby base stations must be managed on Trusted IP Address List:

```
lq[ 2 ] Instances: Trusted IP Addressesqk
x                                        x
x ->  10.1.18.44                         x
x     10.1.18.78                         x
x                                        x
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqj
```

OXE SIP Proxy management:

```
lqReview/Modify: SIP Proxyqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqq
x
x            Node Number (reserved) : 4
x                Instance (reserved) : 1
x                Instance (reserved) : 1
x
x            SIP initial time-out : 500
x                    SIP timer T2 : 4000
x              Dns Timer overflow : 5000
x                      Timer TLS : 30
x                Recursive search + False
x      Minimal authentication method + SIP None
x              Authentication realm : etesting4
x Only authenticated incoming calls + False
x                Framework Period : 3
x      Framework Nb Message By Period : 25
x        Framework Quarantine Period : 1800
x            TCP when long messages + False
x    Retransmission number for INVITE : 3
x
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqq
```

OXE SIP Registrar management:

```
lqReview/Modify: SIP Registrarqqqqqqqqqqqqqqqqqqqqqqqqk
x                                                    x
x            Node Number (reserved) : 4              x
x                Instance (reserved) : 1             x
x                Instance (reserved) : 1             x
x                                                    x
x        SIP Min Expiration Date : 300              x
x        SIP Max Expiration Date : 6000             x
x                                                    x
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqj
```

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

```
lqReview/Modify: Phone classes of serviceqqqqqqqqqqqk
x                                                    x
x            Node Number (reserved) : 4              x
x                Instance (reserved) : 1             x
x                        Phone COS : 0               x
x                                                    x
x                    Display UTF-8 + YES             x
x    Display call server information + YES           x
x                      Keep Alive + NO               x
x    Send NOTIFY instead of MESSAGE + NO             x
x        Tandem Call Identification + YES            x
x                                                    x
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqj
```

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.

12 Appendix C: Partner escalation process

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

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

13 Appendix D: AAPP program

13.1 Alcatel·Lucent Application Partner Program (AAPP)

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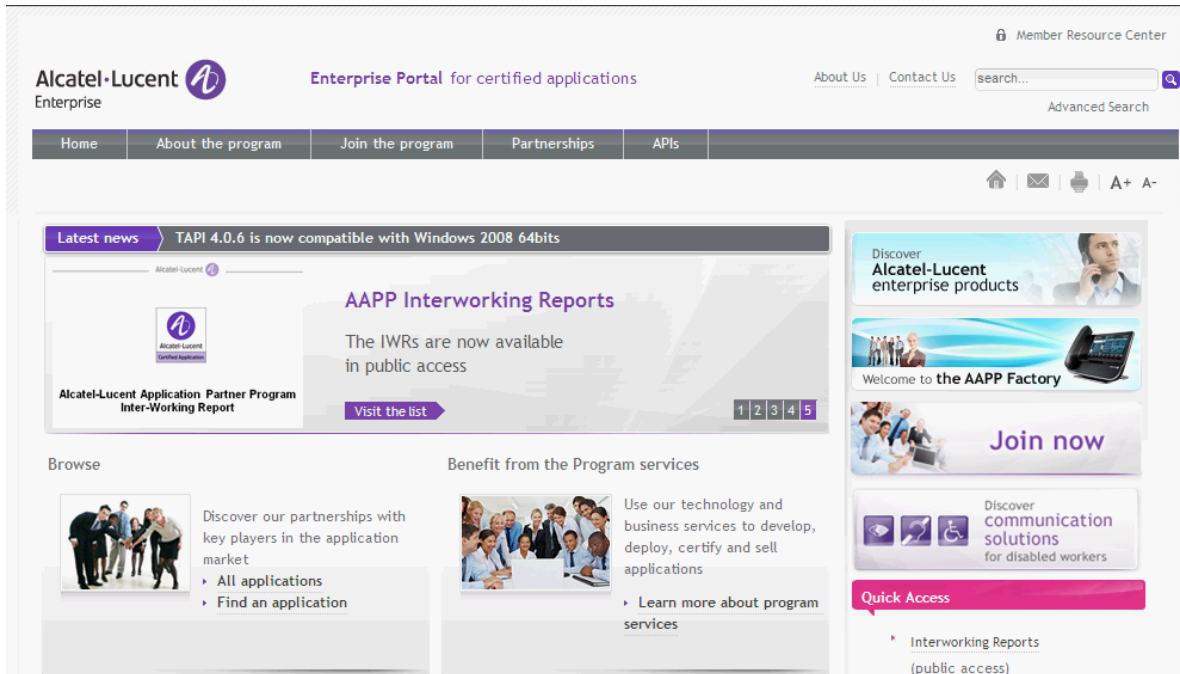
The Alcatel-Lucent Application Partner Program (AAPP) has two main objectives:

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

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

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

Web site

The Application Partner Portal is a website dedicated to the AAPP program and where the InterWorking Reports can be consulted. Its access is free at
http://applicationpartner.alcatel-lucent.com

13.2 Alcatel-Lucent.com

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

14 Appendix E: AAPP Escalation process

14.1 Introduction

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

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

In case technical support is granted, Alcatel-Lucent and the Application Partner, are engaged as following:
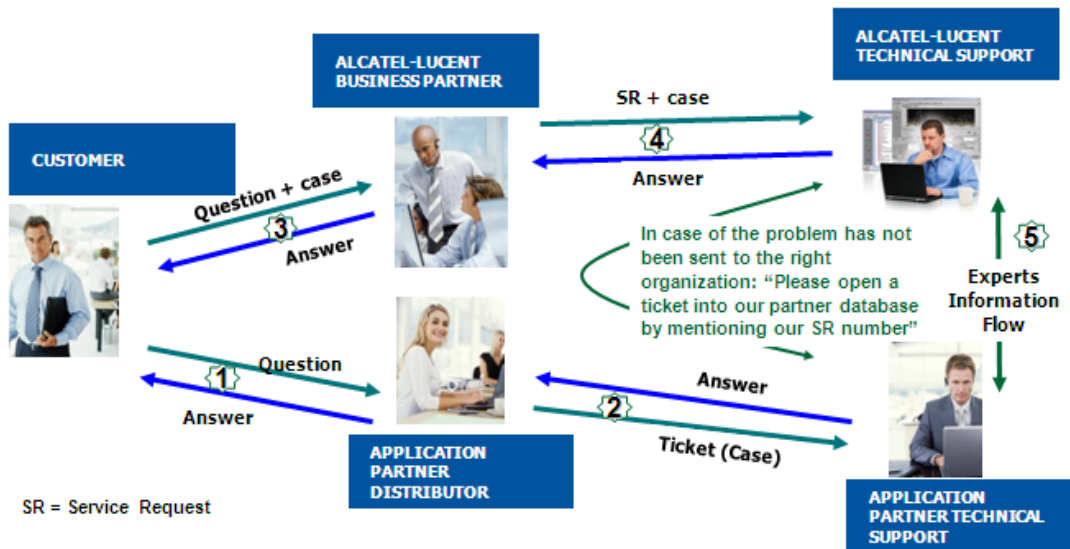
(*) The Application Partner Business Partner can be a Third-Party company or the Alcatel-Lucent Business Partner itself

---

14.2 Escalation in case of a valid Inter-Working Report

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

This defines the scope of what has been certified.

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

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

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

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

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

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

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

Alcatel-Lucent reserves the right to close the case opened on his side if the investigations made on the Application Partner side are insufficient or do 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.

14.3 Escalation in all other cases

These cases can cover following situations:

1. An Interworking Report exist but is not valid (see Chapter 2 "Validity of an Interworking Report")

2. The 3rd party company is referenced as <u>AAPP participant</u> 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 <u>AAPP participant</u>

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

14.4 Technical Support access

The Alcatel-Lucent **Support Center** is open 24 hours a day; 7 days a week:

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

Alcatel-Lucent Business Partners Support Center for countries:

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

For other countries:

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

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