Alcatel-Lucent Application Partner Program
Inter-Working Report

Partner: Ascom

Application type: IP-DECT Solution

Application name: IP-DECT

Alcatel-Lucent Platform: OmniPCX Enterprise™

Certification overview

Date of certification	October 2013

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Alcatel-Lucent Communication Platform	OmniPCX Enterprise
Alcatel-Lucent Communication Platform Release	R11.0 (K1.400.25e)
AAPP member application version	R6 (6.1.3)
Application Category	DECT / Wi-Fi Mobility

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Revision History

Edition 1: creation of the document – *2013-10-09*
Edition 1.1: update to mention R6.1.3 for IP-DECT version -Dec 2013

Test results

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

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

2 Validity of the 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 valid Interworking Report (see chapter 2 "Validity of the Interworking Report") and in the scope of the features which have been certified. That scope is defined by the Interworking report via the tests cases which have been performed, the conditions and the perimeter of the testing as well as the observed limitations. All this being documented in the IWR. The certification does not verify the functional achievement of the AAPP member's application as well as it does not cover load capacity checks, race conditions and generally speaking any real customer's site conditions.

Any possible issue will require first to be addressed and analyzed by the AAPP member before being escalated to Alcatel-Lucent.

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

For more details, please refer to Appendix F "AAPP Escalation Process".

3.1 Case of additional Third party applications

In case at a customer site an additional third party application NOT provided by Alcatel-Lucent is included in the solution between the certified Alcatel-Lucent and AAPP member products such as a Session Border Controller or a firewall for example, Alcatel-Lucent will consider that situation as to that where no IWR exists. Alcatel-Lucent will handle this situation accordingly (for more details, please refer to Appendix F "AAPP Escalation Process").

4 Summary of test results

4.1 Summary of main functions supported

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

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	x
Networking	✓
Display management	✓
Multi-line	✓
Manager / Assistant	✓
Voice Mail	✓
Attendant	✓
Prefixes support	✓
Suffixes support	✓
CPU redundancy support	x

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: IPBS users have no access to the system for up to the registration period after switchover (900 seconds).Reminder: REGISTER must not be used like a Keep-Alive mechanism

- When semi-attended transfer is performed, Ring Back Tone is not heard ate the transferee side, communication between transferee and transfer target is established only when transfer target answers

4.3 Summary of limitations

- Keep Alive mechanism is not supported by ASCOM base station.

- Refer to OmniPCX Enterprise release R11 notes for information on general limitations for SIP/SEPLOS devices on OmniPCX Enterprise.

- Spatial redundancy and PCS cannot be used at the same time because of configuration limitation on Ascom IP-DECT system.

- The Ascom IP-DECT system transparently manages media establishment and redirection when users roam or hand-over between different base stations, including roaming between different sites. However from an OXE point of view, the users are still seen with the IP addresses of their Master base station. This may cause **Call Admission Control (CAC) and/or voice coding issues**, when IP domains with restricted coding or CAC are managed. See section 8.10 for details.

- Initiation of a **three-party conference** is not possible from a DECT handset.

- DECT handsets cannot be part of a **parallel hunt** group and cannot use **barge-in** because these features are not supported for multi-line subscribers.

- **Use of prefixes (SEPLOS features): the MESSAGE notification is not supported by Ascom. Thus neither visual nor acoustic feedback** is provided for prefix activation by system feature. The user cannot see in what state its set is in (Forward, DnD, etc.).

- It is not possible to create **Assistant keys** on SIP set. Thus the Assistant features are limited. The SIP set **can not be Manager Set**. Manager/Assistant features have only be tested on a local node.

4.4 System Limits

Ascom IPBS/IPBL:

- **Max 1000 users per IPBS** Master base station. (500 SIP/TLS otherwise).
- **Max 1000 users per IPBL**. (500 SIP/TLS otherwise).
  Multiple sites; Multiple masters:
- 1,000 IP-DECT base station radios per master.
- 20,000 users/PARI master.
- 100 masters/mobility master.
- 100,000 users/mobility master.
- 1,000 IPBS radios/PARI master or 240 IPBL w/ radios/PARI master or a mix of IPBS/IPBL.

OmniPCX Enterprise R11:

- **Max 5000 SIP users per node**.

4.5 Notes, remarks

The interworking tests only cover the Ascom DECT base stations and handsets. Support for Alcatel-Lucent or other third party vendor DECT handsets has not been evaluated.

5 Application information

Application type:	DECT/IP Solution. Ascom DECT handsets and Ascom IP-DECT base stations linked to OXE via SIP/IP/Ethernet.
Application commercial name:	IP-DECT R6
Application version:	IPBS[6.1.3], Bootcode[6.2.3], Hardware[IPBS2]
Interface type:	SIP/IP/Ethernet

Brief application description:

The application consists of DECT base stations and associated Ascom handsets. DECT base stations are linked to OXE via SIP protocol. All telephony features as provided by SEPLOS (SIP Endpoint Level of Service) are available to the handsets.

6 Tests environment

6.1 General architecture

The tests are performed on the Alcatel-Lucent TSS Applications International platform in the following environment:

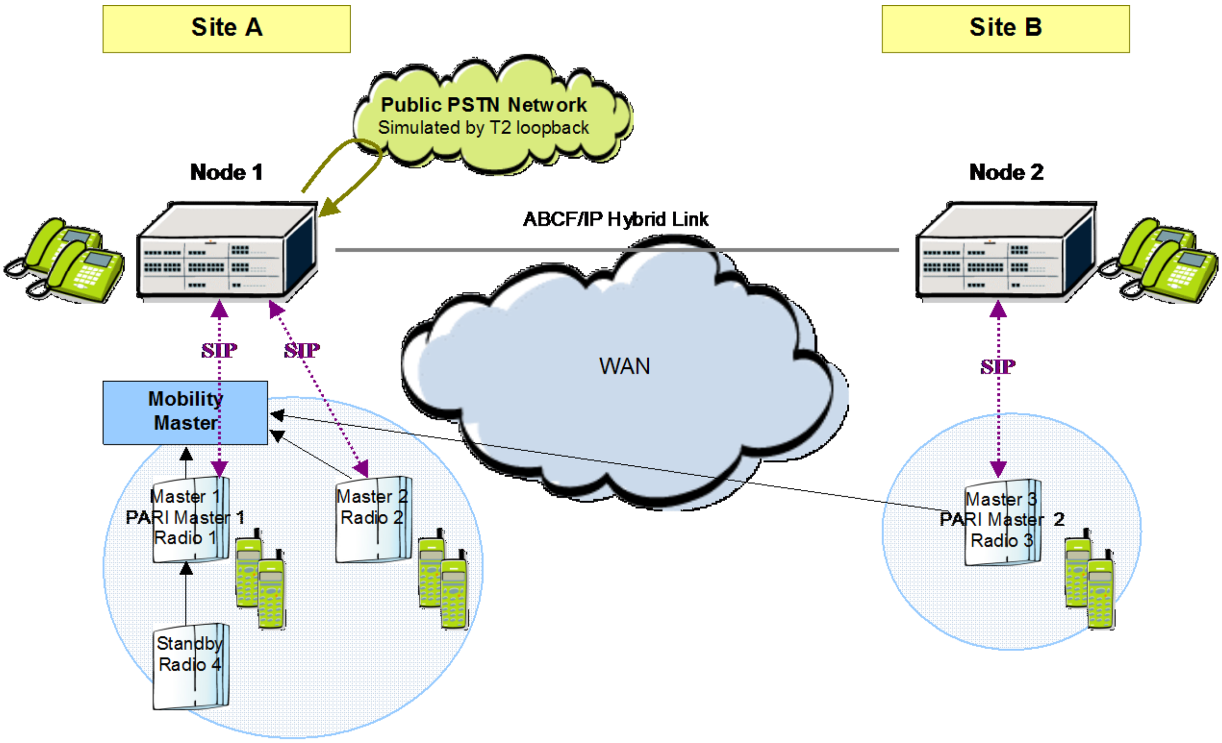


Figure 2 **Test System Architecture**

6.2 Hardware configuration

6.2.1 **Alcatel-Lucent Communication Platform**:

Node1 (node name: node0003):
- OmniPCX Enterprise common hardware
- Passive Call Server
- Spatial redundancy (Different IP subnetworks)
- Two media gateways:

```
Cristal 0 :
   +-------------------------------------------------------------------------+
   | Cr | cpl| cpl type    | hw type    |  cpl  state   |  coupler ID      |
   |----|----|------------|------------|---------------|------------------|
   |  0 |  6 | App. Server|---------- |   IN SERVICE |   BAD PCMS CODE |
   |  0 | 10 | App. Server|---------- |   IN SERVICE |   BAD PCMS CODE |
   +-------------------------------------------------------------------------+

Display list of PCS  :
+-------------------------+------------------+--------------------+
|         PCS Name        |      Address     |        State       |
|-------------------------|------------------|--------------------|
|                  PCS01  |   10.21.251.201     |       INACTIVE    |
|-------------------------|------------------|--------------------|

Cristal 1 :
   +-------------------------------------------------------------------------+
   | Cr | cpl| cpl type    | hw type    |  cpl  state   |  coupler ID      |
   |----|----|------------|------------|---------------|------------------|
   |  1 |  0 |          GD|---------- |   IN SERVICE |   BAD PCMS CODE |
   |  1 |  1 |      UAI 16|---------- |   IN SERVICE |   BAD PCMS CODE |
   +-------------------------------------------------------------------------+

Cristal 2 :
   +-------------------------------------------------------------------------+
   | Cr | cpl| cpl type    | hw type    |  cpl  state   |  coupler ID      |
   |----|----|------------|------------|---------------|------------------|
   |  4 |  0 |          GD|---------- |   IN SERVICE |   BAD PCMS CODE |
   |  4 |  1 |      UAI 16|---------- |   IN SERVICE |   BAD PCMS CODE |
   |  4 |  2 |      PRA T2|---------- |   IN SERVICE |   BAD PCMS CODE |
   +-------------------------------------------------------------------------+
```

Node2 (node name: node0001):
- OmniPCX Enterprise crystal hardware (4400) voice hub
- Single CPU
- One INTIP:

```
Cristal 0 :
   +-------------------------------------------------------------------------+
   | Cr | cpl| cpl type    | hw type    |  cpl  state   |  coupler ID      |
   |----|----|------------|------------|---------------|------------------|
   |  0 |  1 |        CPU7|---------- |   IN SERVICE |   BAD PCMS CODE |
   |  0 |  3 |        UA32|---------- |   IN SERVICE |   BAD PCMS CODE |
   |  0 |  4 |      INTIPA|     INT-IP |   IN SERVICE |   BAD PCMS CODE |
   +-------------------------------------------------------------------------+
```

6.2.2 **Ascom platform**:

- 3x Ascom IPBS2 base stations

Ascom DECT Handsets (**release 4.2.2**):
- 1x Ascom d62-Messenger DECT handset
- 1x Ascom d41-Advanced DECT handset

- 1x Ascom d81-Protector DECT handset

6.3 Software configuration

6.3.1 Alcatel-Lucent Communication Platform:

Version: OmniPCX Enterprise R11.0 K1.400.25e

Note: Handsets are declared as SIP extensions (SIP Endpoint Level of Service - SEPLOS). This means that all telephone features must be tested once by prefixes and once locally on the phone, if available.

Recommended configuration:

Under SIP > SIP Gateway
- Subscribe Min Duration = **900**
- SIP Subscribe Max Duration = **1800**
- Session Timer = **1800**
- Min Session Timer = **900**
- Session Timer Method = **UPDATE**
- SDP in 18x = **TRUE**
- CAC SIP-SIP = **TRUE**
- Dynamic Payload Type for DTMF = **101**

Under SIP > SIP Registrar
- SIP Min Expiration Date = **900**
- SIP Max Expiration Date = **1800**

Under SIP > Trusted IP Addresses
- Manage here the MASTER/STANDBY IP Addresses

Under SIP Extensions > Phone classes of service
- Display Call Server information = **YES**
- Keep Alive = **False**
- Send NOTIFY instead of MESSAGE = **NO**

6.3.2 Ascom platform:

Version: IPBS[6.1.3], Bootcode[6.2.3], Hardware[IPBS2]

Required configuration:

Under IPBS > DECT > Master:
- Enbloc Dialing must be ***enabled***.
- Enable Enbloc Send-Key must be ***enabled***.
- Send Inband DTMF must be ***disabled***.
- Allow DTMF Through RTP must be ***enabled***.
- Register With Number must be ***enabled***.

Recommended configuration:

Under IPBS > Suppl. Serv.:
- MWI Mode should be ***Fixed interrogate and fixed notify number***
- MWI Interrogate Number should contain ***Voicemail Directory Number***
- MWI Notify Number should contain ***Voicemail Directory Number***

Under IPBS > DECT > Master:
- Registration time-to-live [sec]=**900**.
- Hold Signaling=**sendonly**

Under IPBS > DECT > System:

- Coder, Frame (ms).

Here, the same values as configured on OXE should be chosen, typically 20ms

Under IPBS > VOIP:

- Session Timer [sec] = **900**

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.
- **NA** if this test is not applicable to this application. (Use Comment column to describe why)

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 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	Provisioning	Not Tested	Via web interface on base stations. User declaration on Master base station. Possibility to use LDAP. See partner's documentation.
2	DHCP registration (with OXE internal DHCP server)	Not Tested	
3	NTP registration (with OXE internal NTP server)	OK	Time and date is displayed on the handsets (d62, d41 and d81). Only one NTP IP@ can be specified => issue in case of CPU switchover in spatial redundancy
4A	SIP registration, using OXE MAIN IP addresse(s) (without authentication)	OK	
4B	SIP registration, using DNS (without authentication)	OK	Both type DNS A and DNS SRV request are supported.
5	Support of "423 Interval Too Brief" (1)	OK	By default, *OXE>SIP>SIP Registrar>Min Expires*=900s and *IPBS>VoIP>SIP>Registration time-to-live [sec]*=300s. So "423 Interval Too Brief" will be triggered by default. Values are configurable on both sides.
6	SIP registration with authentication: Turn on SIP Digest authentication, specify realm on OXE, and specify user name and password on SIP client.	OK	See note (2)

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.

(2) SIP user name and password are configured in `IPBS>Administration>Users`. It must match with `OXE>SIP Authentication user name/password`. If not matched, the set displays "**PBX Out of service**".

On `IPBS > DECT > Master > Domain`: **`node0003`** (realm name configured on SIP > SIP Proxy)

On `IPBS > DECT > Master > Register with Number:` **`TRUE`**

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 (Check ring back tone, called party display)	OK	Tone and display OK
2	Call to a local user with overlap dialling: Dial a part of the number, wait and continue.	OK	Only OK if Enbloc Dialing=No. Dialled “13”, then press dial button, then dialled anothers three “0”. => Calls set 13000.
3	Call to a local user with overlap dialling, timeout: Dial a part of the number, wait and stop.	NOK	No timeout.
4	Call to a local user with overlap dialling, release: Dial a part of the number, wait and release the call.	OK	
5	Call to local user with no answer. Check timeout.	OK	No timeout.
6	Call to another SIP set (Check ring back tone, called party display)	OK	Tone played by OXE and display OK When call is answered, RTP flow doesn’t transit through PBX anymore
7	Call to wrong number (SIP: “404 Not Found”)	OK	Display: “Vacant”, short tones during 30 sec and then the phone hangs up.
8	Call rejected by call handling (SIP: “183 Progress/487 Request Terminated”) e.g. max number of calls reached etc.	OK	Test done with CAC=0 on IP Domain 1 and second test with Compressor=0 on GD. For both, OXE generates 480 Temporarily Not Available
9	Call to busy user (SIP: “486 Busy Here”) Check busy tone.	OK	183 Sessions Progress (reason header: busy) generated by OXE, busy tone is connected, reason header is not supported on ASCOM side 486 Busy Here generated when we call a Busy ASCOM DECT Short tones, “Hung up” after 30 secs.
10	Call to user in “Out of Service” state (SIP: “480 Temporarily Unavailable”)	OK	Display: “Not reachable”, short tones during 30 secs and the phone hung up.
11	Call to user in “Do not Disturb” state	OK	OXE responds “183 Session Progress”, reason header “Do not disturb”.. Released tone is played. “Hung up” after 15 secs.

12	Call to local user, immediate forward (CFU). (SIP: "302 Moved Temporarily")(1)	OK	Display OK
13	Call to local user, forward on no reply (CFNR). (1)	OK	Ring back tone is heard before and after forward
14	Call to local user, forward on busy (CFB). (1)	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).	OK	Check order of codecs in SDP list. Check chosen codec. We expect G.711 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).	OK	Check order of codecs in SDP list. Check chosen codec. We expect G.729 See note (2)
18	Call to external number (via T2 loopback) (Check ring back tone, called party display)	OK	Ring back tone OK. Display the ISDN Trunk Name (PAI on 200 OK)
19	Check generation of accounting ticket for external call	Not tested	Not tested.
20	SIP session timer expiration: Check if call is maintained or released after the session timer has expired See note (3)	OK	Ok for both UPDATE and Re-INVITE methods.
21	Set lock/unlock. Dial the padlock prefix ("45"). Check that you can't dial other prefixes than unlock. To unlock, dial padlock prefix ("45") + personal password.	OK	
22	Use of abbreviated numbers (Speed dialing) for both internal and external numbers.	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  |    2  |
|     type | NIPR  | IP_R  | IP_R  |
|  allowed | ffff  | ffff  | ffff  |
|     used |     0 |     0 |     0 |

|  RIP Intr| G711  | G711  | G711  |
|  RIP Extr| G711  | G729  | G711  |
|  IPP Intr| G711  | G711  | G711  |
|  IPP Extr| G711  | G729  | G711  |
|  G722 Int|    NO |    NO |    NO |
|  G722 Ext|    NO |    NO |    NO |
```

Partner SIP set is in domain 1.

Tested:
IPBS>DECT>System>Coder=G711A, non exclusive.

SIP domain 1 calls OXE domain 1: G.711, G.729, G.723 proposed, G.711 chosen. (OK)
SIP domain 1 calls OXE domain 2: G.711, G.729, G.723 proposed, G.729 chosen. (OK)
When called, IPBS will choose the first (in order of appearance) coder in the received SDP list it can handle, independently of the preferred coder set in *IPBS>DECT>System>Coder* (and if *Exclusive* is not set).

When calling, IPBS sends preferred coder (*IPBS>DECT>System>Coder*) first in SDP list.
OXE chooses first codec that is acceptable, starting from proposed list.

(3) We used the following setting for the test:
OXE>SIP>SIP Gateway:

```
Session Timer : 180
Min Session Timer : 90
Session Timer Method + RE_INVITE
```

Then, wait more than 180 seconds to see if call is released.

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/OK	
2	Local/network call to busy SIP terminal	OK/OK	ASCOM sends 486 Busy Here. (OK) OXE immediately displays "Please go on hook" on calling set without Busy indication.
3	Local/network call to unplugged SIP terminal	OK/OK	Busy ring tone heard, 603 decline sends by ASCOM
4	Local/network call to SIP terminal in Do Not Disturb (DND) mode:		
4A	By local feature	OK	SIP set display "busy" call from OXE set display "go on hook". 486 User Busy generated by ASCOM
4B	By system feature (SEPLOS) (prefix "42"+ user password)	OK	
5	Local/network/SIP call to SIP terminal in immediate forward (CFU) to local user:		
5A	By local feature	OK	
5B	By system feature (SEPLOS) (prefix "51"+number/"41")	OK	
6	Local/network/SIP call to SIP terminal in immediate forward (CFU) to network number:		
6A	By local feature	OK	
6B	By system feature (SEPLOS) (prefix "51"+number/"41")	OK	
7	Local/network/SIP call to SIP terminal in immediate forward (CFU) to another SIP user		
7A	By local feature	OK	
7B	By system feature (SEPLOS) (prefix "51"+number/"41")	OK	
8	Local call to SIP terminal in "forward on busy" (CFB) state:		
8A	By local feature	OK	
8B	By system feature (SEPLOS) (prefix "52"+number/"41")	OK	
9	Local call to SIP terminal in "forward on no reply" (CFNR)		

9A	By local feature	OK	Forward information is not displayed
9B	By system feature (SEPLOS) (prefix “53”+number/”41”)	OK	
10	Call to busy user, Call waiting. (Camp-on)	OK	
11	External call to SIP terminal. Check that external call back number is shown correctly.	OK	
12	Identity secrecy/CLIR: Local call to SIP terminal. Check that caller id is not shown.	OK	Display shows “External call” and a line of asterisks.
13	Display: Call to free SIP terminal from user with a name containing non-ASCII characters. Check caller display.	OK	Tested Ok for Latin-1 characters
14	Display: Call to free SIP terminal from user with a UTF-8 name containing non-ASCII characters. Check caller display.	OK	Tested Ok for Latin-1 characters
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 part of a parallel hunt group. Call to hunt group. Check call/release.	NA	By default, SEPLOS sets are multiline. Parallel hunt groups are not supported for multiline sets.
18	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	When calling twin set directly, name of main set is displayed on caller.
18.2	Same as 18. Then transfer to main set. (R + 4)	OK	Ok, for both unattended and attended transfer. Possibility to transfer via R+4, or Menu or Hand up
19	Call Pick-up (Supervision): A call from OXE set to another OXE set is picked up from a SIP set by dialling the call pick-prefix (“55”+number of target set)	OK	
20	Call Pick-up (Supervision): A call from SIP set to another SIP set is picked up from a OXE set by dialling the call pick-prefix (“55”+number of target set)	OK	

Note: MESSAGE notification is not used/displayed by Ascom. Thus neither visual nor accoustic feedback is provided for prefix activation by system feature. The user cannot see in what state its set is in (Forward, DnD, etc.).
Note: Only Notify Events: message-summary is supported by ASCOM

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	Press R to put on hold, and press R again to resume.
2	Second call to another local user. Distant user is put on hold.	OK	Press R + second call number. Enbloc dialing parameter must be enabled on IPBS.
3	Broker request (toggle back and forth between both lines, local feature)	OK	R + "2" to switch between participants
4	Release first call. Keep second call.	OK	R + "1" to finish the current call
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	R + "402" + number.
6	Send/receive DTMF	OK	DTMF are sent as RFC4733. Need to have *IPBS>DECT>Master>Allow DTMF through RTP* **enabled**.
7	Three party conference initiated from OXE set (suffix "3"). Released by OXE set.	OK	
8	Three party conference initiated from SIP set (local feature). Released by SIP set.	NA	Feature not available.
9	Barge-in (Intrusion) to SIP set. The SIP set is in conversation with another set. A third set calls the SIP set and wants to barge-in.	NA	Feature not available
10	Barge-in (Intrusion) from SIP set. The SIP set calls another set which is in conversation. Then press the barge-in suffix "4".	OK	
11	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	
12	Busy Camp-on from SIP set. The SIP set calls another set which is in conversation. Then press the camp-on suffix "6".	OK	

13	Voice mail deposit from SIP set. The SIP set calls another set. Then press the message deposit suffix “8”.	OK	
14	Meet-me conference: Participate in ongoing meet-me conference from SIP set (prefix “509”+number+code). Released by SIP set.	OK	

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.

There are three sorts of transfers in the SIP world:

- ***Unattended Transfer*** or *Basic Transfer*: The Transferor provides the Transfer Target's contact to the Transferee. The Transferee attempts to establish a session using that contact and reports the results of that attempt to the Transferor.
  Note: Unattended Transfer is not provided by the OXE

- ***Semi-Attended Transfer*** or *Early Attended Transfer* or *Transfer on ringing*:
  1. A (Transferee) calls B (Transferor).
  2. B (Transferor) calls C (Transfer Target). A is on hold during this phase. C is in ringing state (does not pick up the call).
  3. B executes the transfer. B drops out of the communication. A is now in contact with C, in ringing state. When C picks up the call it is in conversation with A.

- ***Attended Transfer*** or *Consultative Transfer* or *Transfer in conversation*:
  1. A (Transferee) calls B (Transferor).
  2. B (Transferor) calls C (Transfer Target). A is on hold during this phase. C picks up the call and goes in conversation with B.
  3. B executes the transfer. B drops out of the communication. A is now in conversation with C.

8.5.2 Test Results

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

Semi-Attended Transfer (on Ringing)
Semi-Attended transfer procedure for Ascom handsets: Press “R” + destination number. Wait until ringing. Then hangup.

Test	Action			Result	Comment
	A **Transferee**	B **Transferor**	C **Transfer Target**		
1	SIP	OXE	OXE/Ext Call	OK/OK	Transferee display not updated after transfer to external
2	OXE/Ext Call	SIP	OXE	OK, but/OK, But	No Ring Back Tone heard by transferee after the transfer completionbecause REFER is generated only at the reception of the 200 OK SDP.
3	OXE/Ext Call	OXE	SIP	OK/OK	
4	OXE / Ext call	SIP	SIP	OK, but/OK, But	No Ring Back Tone heard by transferee after the transfer completionbecause REFER is generated only at the reception of the 200 OK SDP.
5	SIP	OXE	SIP	OK	
6	SIP	SIP	OXE/Ext Call	OK, but/OK, But	No Ring Back Tone heard by transferee after the transfer completionbecause REFER is generated only at the reception of the 200 OK SDP.
7	SIP	SIP	SIP	OK, but	No Ring Back Tone heard by transferee after the transfer completionbecause REFER is generated only at the reception of the 200 OK SDP.

Note: only Attended Transfer is applied on ASCOM Base station. At the reception of the 180 Ringing, call leg is not cancelled when transfer is completed. Refer request is generated only when 200 OK SDP (target answer) is received. Result is: Ring Back Tone is not heard, only Music On Hold which is cut at the target answer.

Attended Transfer (in Conversation)
Attended transfer procedure for Ascom handsets: Press “R” + destination number. Wait until call establishment (off-hook). Then hangup.

Test	Action			Result	Comment
	A **Transferee**	B **Transferor**	C **Transfer Target**		
8	SIP	OXE	OXE/Ext Call	OK/OK	Transferee display not updated after transfer to external
9	OXE/Ext Call	SIP	OXE	OK/OK	
10	OXE/Ext Call	OXE	SIP	OK/OK	
11	OXE / Ext call	SIP	SIP	OK/OK	
12	SIP	OXE	SIP	OK	
13	SIP	SIP	OXE/Ext Call	OK/OK	Transferee display not updated after transfer to external
14	SIP	SIP	SIP	OK	No display names, only numbers for both transferee and target after transfer and off hook.

Note: Transfer from Ascom set is done by pressing R + targetm then dial button then hang up for transfer. (System option `No Transfer on Hangup` = **disabled**).

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.

Recommended configuration:
`Attendants > Attendants sets > Tone presence`: **TRUE**

8.6.2 Test Results

Test	Action	Result	Comment
1	Call to attendant (using attendant call prefix “9”)	OK	
2	Call to attendant (using attendant call prefix “9”), attendant transfers to OXE set / Ext Call, unattended.	OK/OK	Display not updated after transfer to external.
3	Call to attendant (using attendant call prefix “9”), attendant transfers to OXE set / Ext Call, attended.	OK/OK	Display not updated after transfer to external.
4	OXE set / Ext. calls to attendant (using attendant call prefix “9”), attendant transfers to SIP set, attended.	OK/OK	
5	Call to attendant (using attendant call prefix “9”). Second incoming call while in conversation with attendant.	OK	Second call is refused as expected.

8.7 Manager/Assistant

8.7.1 Test Objectives

Created Manager/Assistant configuration:
Manager set: IP Touch 4068
Assistant set: Ascom d62/d41
Created assistant call key.

8.7.2 Test Results

Test	Action	Result	Comment
1	From manager set(OXE), call assistant(SIP) via assistant call key	OK	

Note: It is not possible to create Assistant keys on SIP set.
Thus the Assistant features are limited.
The SIP set can not be Manager Set.

8.8 Voice Mail

8.8.1 Test Objectives

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

8.8.2 Test Results

Test	Action	Result	Comment
1	A Voice Mail message for the SIP subscriber is generated. Check that MWI is activated.	OK	
2	Message consultation	OK	
3	Password modification	OK	
4	SIP call to a OXE user forwarded to Voice Mail	OK	
5	OXE call to a SIP user forwarded to Voice Mail	OK	Local feature OK / OXE system feature OK

8.9 Duplication and Robustness

8.9.1 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 sub networks).

For each configuration, check:

- Can new outgoing calls be made immediately after switchover?
- Are incoming calls (from new MAIN CS) accepted immediately after switchover?
- Are existing calls maintained after switchover?
- Can existing calls be modified (transfer, hang-up, etc.) immediately 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?

8.9.2 Test Results

Test	Action	Result	Comment
1	Spatial redundancy via alternate proxy method Define two SIP proxies: Primary = CS A (Proxy), Secondary = CS B (Alternate proxy method) (if no delegation).	OK, but	Domain value must not be filled if the authentication with realm is not used. In such case, only Proxy and Alternate Proxy filled must be filled with both nodes names After a switchover, the next call can be established after the next registration period, up to 900 seconds.
2	Spatial redundancyvia DNS method Configure the FQDN on the proxy field only (if delegation)	Not Tested	
3	Switchover to Passive Call Server (PCS). (IP link to main/stdby call servers down)	OK	
4	SIP device reboot. Check that calls are possible as soon as device has come back to service.	OK	
5	Temporary Link down with the PBX	OK	Display “PBX Out of service”

Notes:
Spatial redundancy using DNS method does NOT work.
DNS method configuration:
DNS1 = CS A
DNS2 = CS B
Proxy = nodename.domain

The consequence is that you **cannot use both, *spatial redundancy* and *PCS* at the same time**.

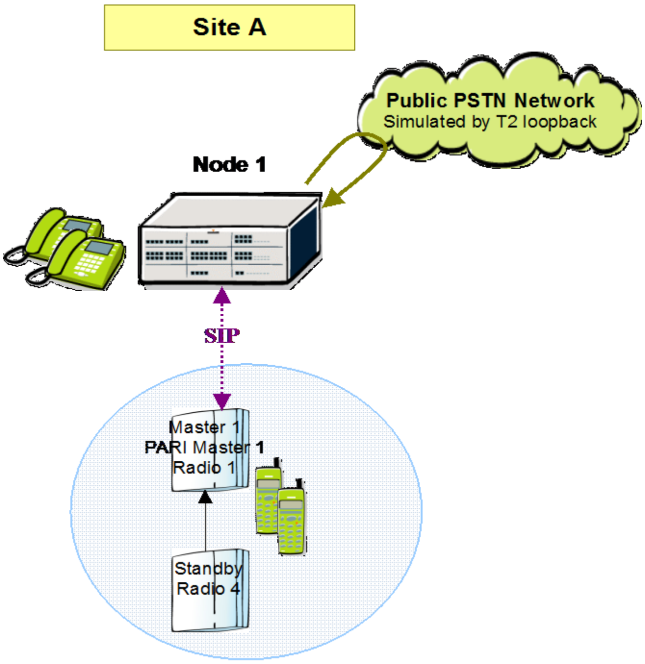
In order to have acceptable switchover time the keep alive mechanism (SIP Options) can be used but this feature is unavailable on Ascom side. This is an Ascom limitation.

8.10 DECT Multi-Master/Multi-Site

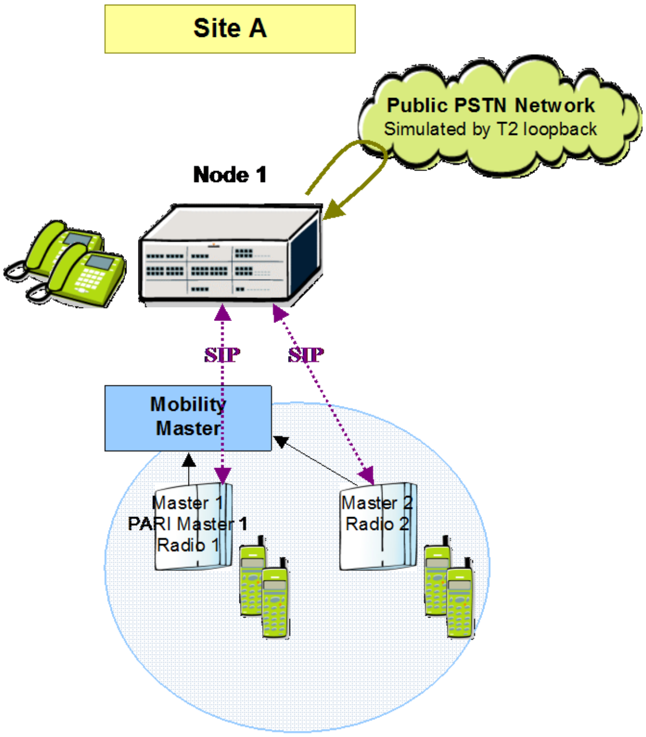
8.10.1 Test Objectives

Test different combinations of sites and IP DECT base station topologies.
Each time:
Test basic call.
Test roaming and/or handover during conversation (if applicable).
Test failover (if applicable).

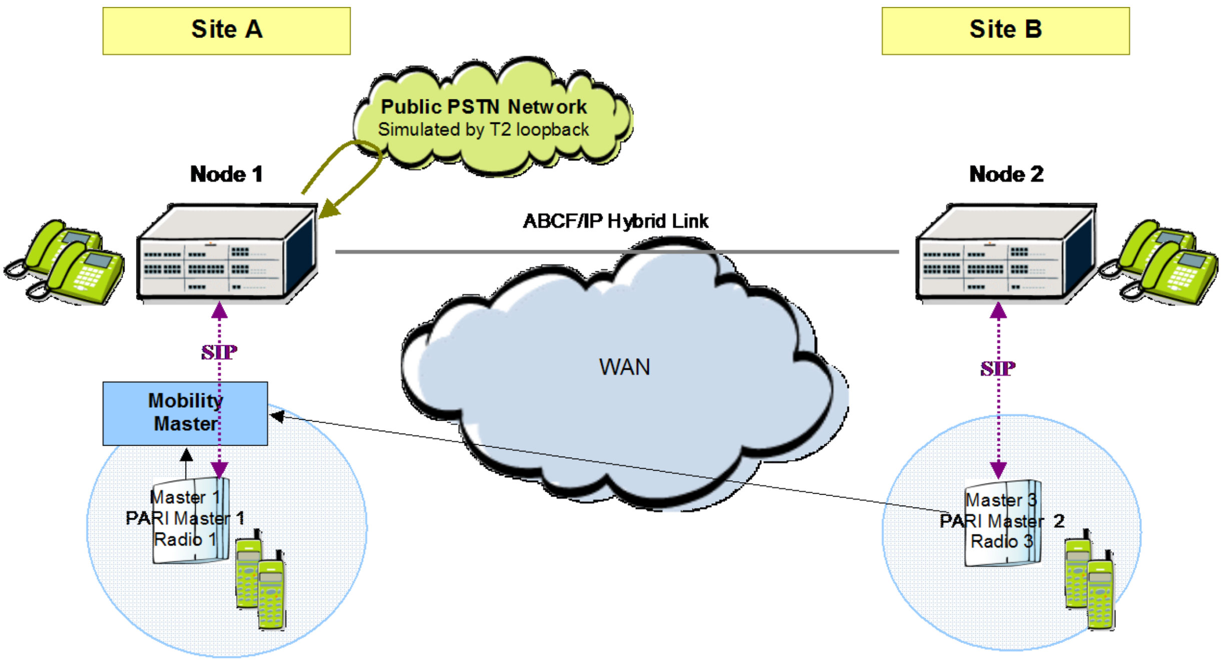
Single site, single master.
Standby master. Two radios.
No mobility master.



Single site, multi-master.
Using mobility master on IPBS-1.



Multi-site, multi-master.
Using mobility master on IPBS-1.

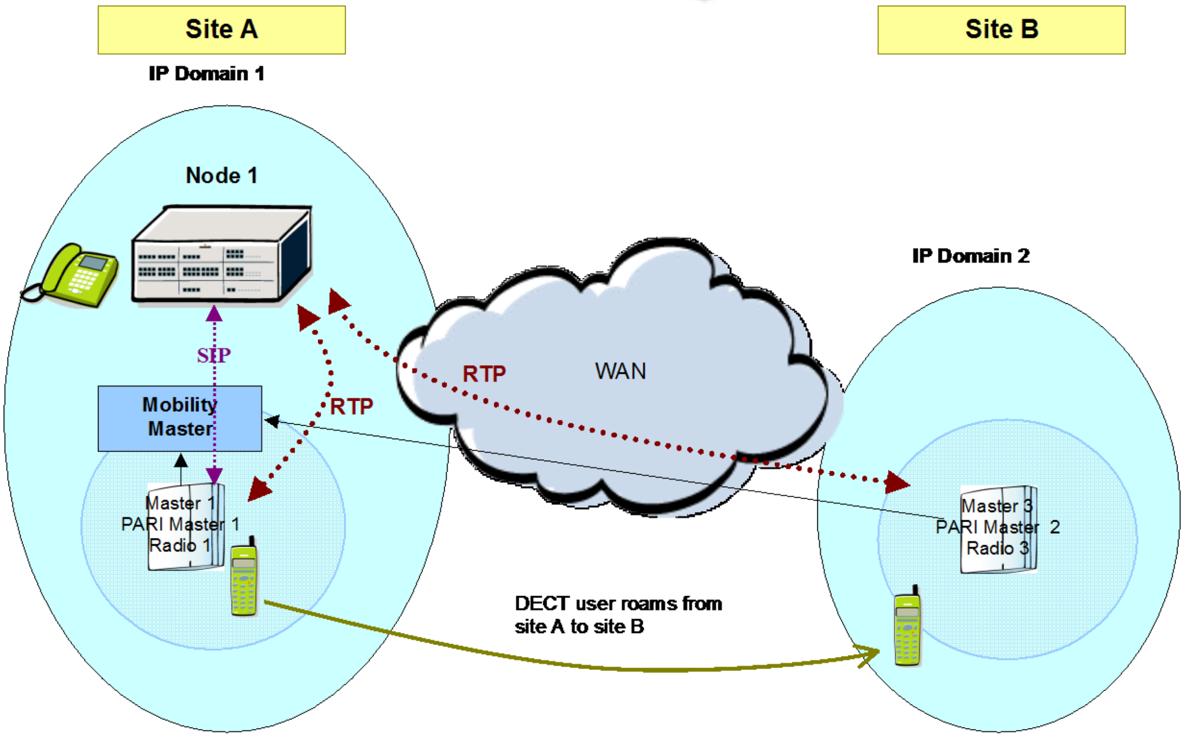


8.10.2 Test Results

Test	Action	Result	Comment
	Single site, single master. No mobility master.		
1	Handover between DECT base stations while in conversation	OK	Transparent for the PBX since RTP flows are maintained on the initial base station and bounce off to following base station. Warning: CAC might not be respected when handing over or roaming to a base station of a different IP domain. (1)
2	IPBS Master failure (physical down). Switchover to IPBS Standby Master.	OK	Works, but switch from Master to Standby can take up to 1 to 2 minutes, with intermittent network loss and reconnection of DECT handsets. Switch back to Master is much quicker. Calls are cut off.
	Single site, multi-master. Using mobility master.		
3	Call from set on Master 1/Radio 1 to node 1 local user	OK	
4	Call from set on Master 2/Radio 1 to node 1 local user	OK	Potential CAC problems. See note (1).
5	Call from set on Master 2/Radio 2 to set on Master 1/Radio 1. Handover from Radio 2 to Radio 1.	OK	We end up with a double RTP flow from Radio 1 to Radio 2 and back (one-way)!
	Multi-site, multi-master. Using mobility master.		
6	Call from set declared on Master 3 to OXE node 2 local user.	OK	Potential CAC problems. See note (1).
7	Roam a set declared on Master 3 to site A. Call to node 1 local user.	OK	Potential CAC problems. See note (1).
8	Roam a set declared on Master 3 to site A. Call to node 2 local user.	OK	Potential CAC problems. See note (1).
9	Roam a set declared on Master 3 to site A. Call to node 1 SIP user.	OK	Potential CAC problems. See note (1).

Notes:
(1) IPBS uses the following operation principle: SIP signaling is always performed by the Master IPBS where the user is declared. Media channels (RTP) are established to/from the base station where the handset is located. This means that IP addresses of signaling and media endpoints are different if handset is located on a base station different from its Master.
OXE uses the IP address of declared SIP extension (signaling address) to determine IP domain association and therefore CAC and codec choice.

Consider the following figure:
A DECT handset is declared on Site A, IPBS Master 1.
Signaling of this handset always goes to Node 1, coming from IPBS Master 1.
RTP will be established with Radio 1, which has the same IP address.
Correct CAC and coding algorithm can be applied.
If the handset moves to site B, signaling still goes from Master 1 to Node 1, but RTP will be established with Radio 3, with a different IP address.
As signaling address is used to check CAC/codec, user is still seen in IP domain 1 although RTP will be established to IP domain 2.=> Wrong CAC and codec will be applied!

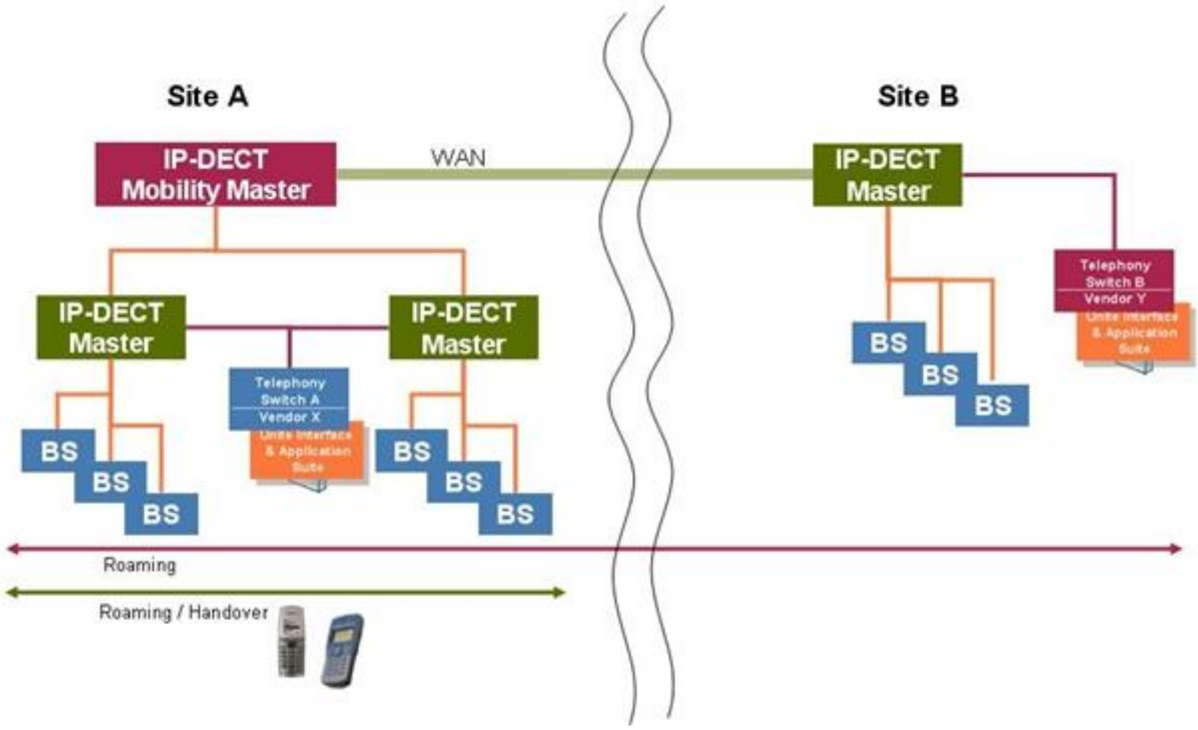

Site A
Site B
IP Domain 1
Node 1
IP Domain 2
SIP
RTP
RTP
WAN
Mobility Master
Master 1
PARI Master 1
Radio 1
Master 3
PARI Master 2
Radio 3
DECT user roams from site A to site B

9 Appendix A: AAPP member's application description

The IP-DECT system from Ascom combines the VoIP world with traditional wireless DECT in an innovative package. One unique advantage is that you can have both packet data and high-quality voice connections on the same network and look forward to superb quality of service in addition to excellent messaging capabilities in a secure radio environment.

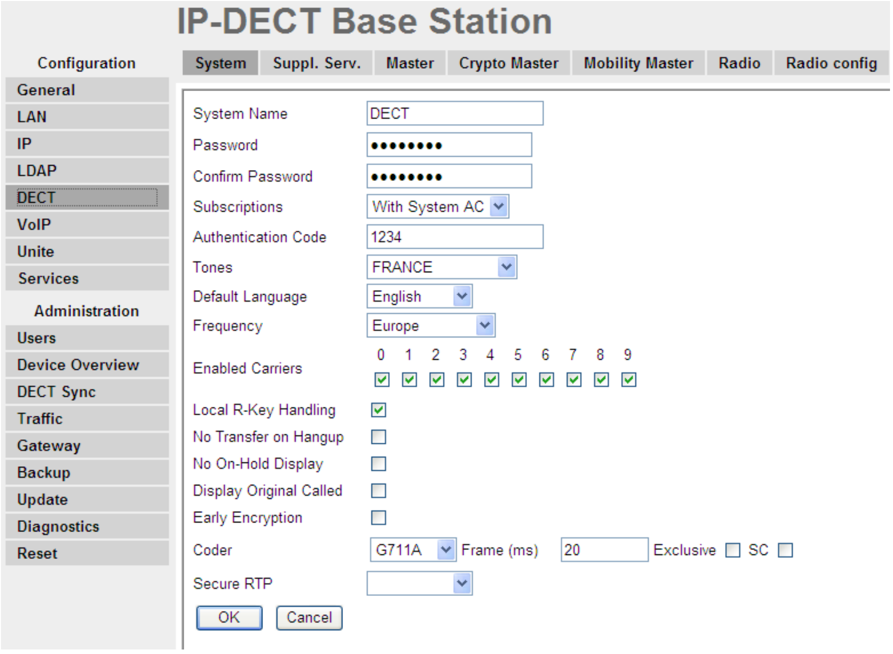
The multi-master concept was introduced with IP-DECT release 3 (R3), and enables a completely new principle on how to build large systems and gives the opportunity to balance the load between different IP-PBX's

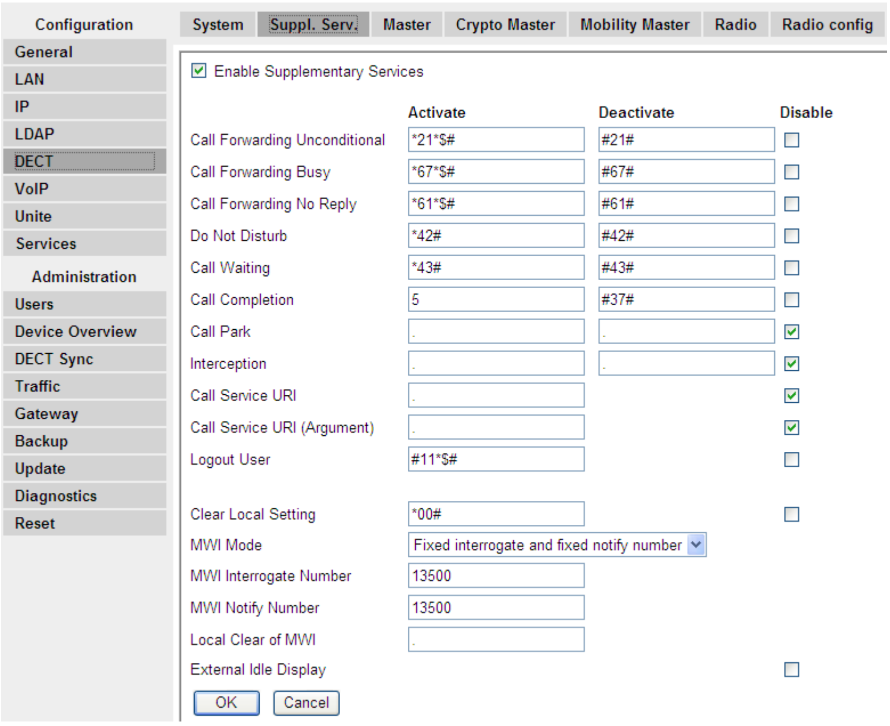
The picture below illustrates a typical multi-Master system:

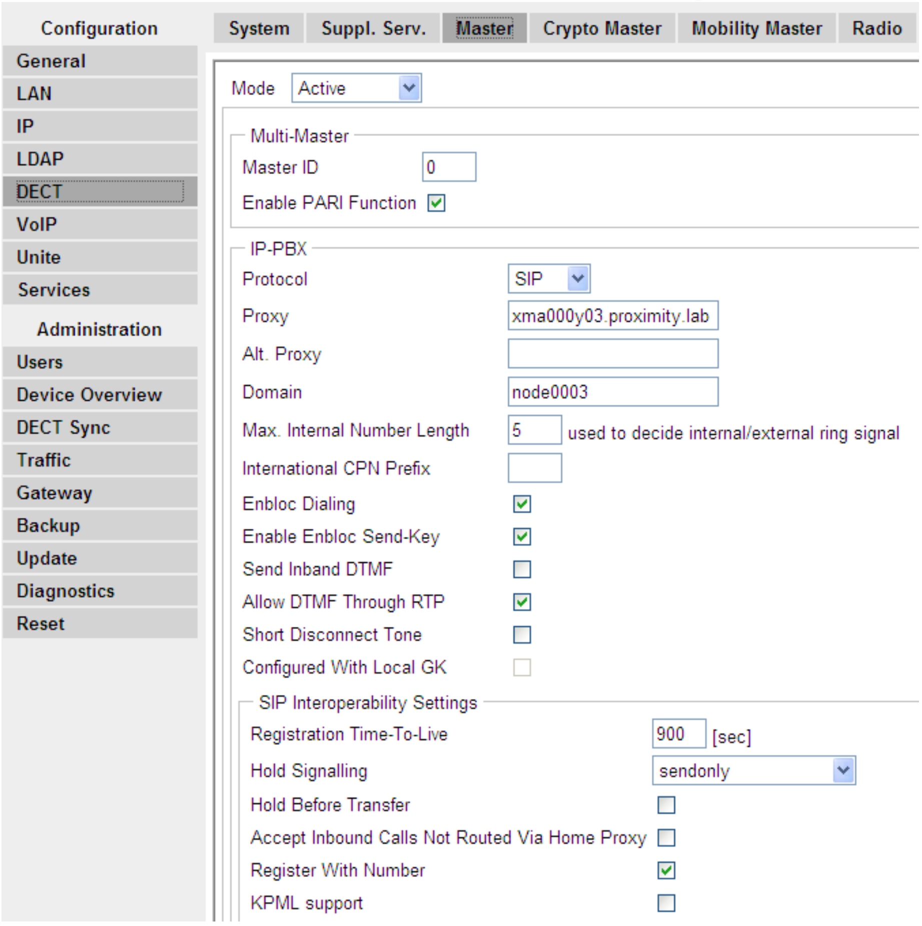


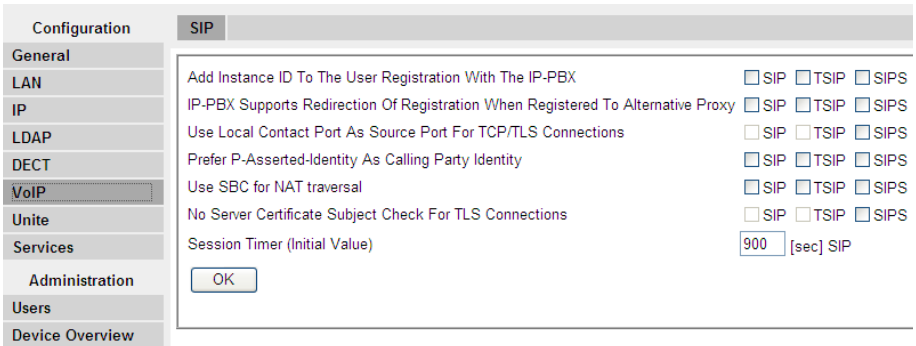
For configuration of the Ascom IP-DECT system, see the corresponding Ascom documentation.

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






Configuration
General
LAN
IP
LDAP
DECT
VoIP
Unite
Services
Administration
Users
Device Overview
DECT Sync
Traffic
Gateway
Backup
Update
Diagnostics
Reset
System
Suppl. Serv.
Master
Crypto Master
Mobility Master
Radio
Mode
Active
Multi-Master
Master ID
0
Enable PARI Function
IP-PBX
Protocol
SIP
Proxy
xma000y03.proximity.lab
Alt. Proxy
Domain
node0003
Max. Internal Number Length
5
used to decide internal/external ring signal
International CPN Prefix
Enbloc Dialing
Enable Enbloc Send-Key
Send Inband DTMF
Allow DTMF Through RTP
Short Disconnect Tone
Configured With Local GK
SIP Interoperability Settings
Registration Time-To-Live
900
[sec]
Hold Signalling
sendonly
Hold Before Transfer
Accept Inbound Calls Not Routed Via Home Proxy
Register With Number
KPML support


Configuration
General
LAN
IP
LDAP
DECT
VoIP
Unite
Services
Administration
Users
Device Overview
SIP
Add Instance ID To The User Registration With The IP-PBX
SIP TSIP SIPS
IP-PBX Supports Redirection Of Registration When Registered To Alternative Proxy
SIP TSIP SIPS
Use Local Contact Port As Source Port For TCP/TLS Connections
SIP TSIP SIPS
Prefer P-Asserted-Identity As Calling Party Identity
SIP TSIP SIPS
Use SBC for NAT traversal
SIP TSIP SIPS
No Server Certificate Subject Check For TLS Connections
SIP TSIP SIPS
Session Timer (Initial Value)
900
[sec] SIP
OK

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

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

```
┌─Review/Modify: Users──────────────────────────────────────────────────┐
│                                                                        │
│             Node Number (reserved) : 1                                 │
│                   Directory Number : 13040                             │
│                                                                        │
│                     Directory name : ASCOM                             │
│               Directory First Name :                                   │
│               UTF-8 Directory Name : ---------------------------------------
│         UTF-8 Directory First Name : ---------------------------------------
│                      Location Node : 1                                 │
│                      Shelf Address : 255                               │
│                      Board Address : 255                               │
│                  Equipment Address : 255                               │
│                           Set Type + SIP extension                     │
│                      Entity Number : 1                                 │
│                       Set Function + Default                           │
│                       Profile Name : ---------------                   │
│                       Key Profiles + None                              │
│                  Domain Identifier : 0                                 │
│                        Language ID : 1                                 │
│                                                                        │
│                        Secret Code : ****                              │
│                                                                        │
└────────────────────────────────────────────────────────────────────────┘
```

OXE SIP Gateway management:

```
+-Review/Modify: SIP Gateway---------------------------------------------------------------+
 ¦                                                                                        ¦
 ¦              Node Number (reserved) : 103                                              ¦
 ¦                 Instance (reserved) : 1                                                ¦
 ¦                 Instance (reserved) : 1                                                ¦
 ¦                                                                                        ¦
 ¦                      SIP Subnetwork : 15                                               ¦
 ¦                     SIP Trunk Group : 26                                               ¦
 ¦                          IP Address : 10.21.5.3                                        ¦
 ¦                 Machine name - Host : node0003                                         ¦
 ¦               SIP Proxy Port Number : 5060                                             ¦
 ¦          SIP Subscribe Min Duration : 900                                              ¦
 ¦          SIP Subscribe Max Duration : 1800                                             ¦
 ¦                       Session Timer : 1800                                             ¦
 ¦                   Min Session Timer : 900                                              ¦
 ¦                Session Timer Method + RE_INVITE                                        ¦
 ¦               DNS local domain name : proximity.lab                                    ¦
 ¦                            DNS type + DNS A                                            ¦
 ¦                 SIP DNS1 IP Address : -----------------------------------------------  ¦
 ¦                 SIP DNS2 IP Address : -----------------------------------------------  ¦
 ¦                          SDP in 18x + True                                             ¦
 ¦                         Cac SIP-SIP + True                                             ¦
 ¦    INFO method for remote extension + False                                            ¦
 ¦       Dynamic Payload type for DTMF : 101                                              ¦
 ¦                                                                                        ¦
 +----------------------------------------------------------------------------------------+
```

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

```
+-----------------------------------------------------------------------+
|                      Trusted IP Address List                          |
+-----------------------------------------------------------------------+
Trusted IP Address  1   : 10.20.32.5   (8440 Voicemail)
Trusted IP Address  3   : 10.21.30.59   (ASCOM base station 1)
Trusted IP Address  4   : 10.21.30.61   (ASCOM base station 2)
```

OXE SIP Proxy management:

```
+-Review/Modify: SIP Proxy---------------------------------------------------------+
¦                                                                                  ¦
¦            Node Number (reserved) : 103                                          ¦
¦               Instance (reserved) : 1                                            ¦
¦               Instance (reserved) : 1                                            ¦
¦                                                                                  ¦
¦              SIP initial time-out : 500                                          ¦
¦                      SIP timer T2 : 4000                                         ¦
¦                Dns Timer overflow : 5000                                         ¦
¦                         Timer TLS : 30                                           ¦
¦                  Recursive search + False                                        ¦
¦       Minimal authentication method + SIP None                                   ¦
¦              Authentication realm : ------------------------------------------   ¦
¦ Only authenticated incoming calls + True
¦                  Framework Period : 3
¦     Framework Nb Message By Period : 25
¦        Framework Quarantine Period : 1800
¦             TCP when long messages + True
¦  Retransmission number for INVITE : 3
¦
¦
+----------------------------------------------------------------------------------+
```

OXE SIP Registrar management:

```
+-Review/Modify: SIP Registrar-----------------------------------------------------+
¦                                                                                  ¦
¦            Node Number (reserved) : 103                                          ¦
¦               Instance (reserved) : 1                                            ¦
¦               Instance (reserved) : 1                                            ¦
¦                                                                                  ¦
¦              SIP Min Expiration Date : 900
¦              SIP Max Expiration Date : 1800
¦
+----------------------------------------------------------------------------------+
```

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

```
+-Review/Modify: Phone classes of service------------+
¦                                                    ¦
¦            Node Number (reserved) : 103            ¦
¦               Instance (reserved) : 1              ¦
¦                         Phone COS : 0              ¦
¦                                                    ¦
¦                     Display UTF-8 + YES            ¦
¦    Display call server information + YES           ¦
¦                        Keep Alive + NO             ¦
¦     Send NOTIFY instead of MESSAGE + NO            ¦
¦         Tandem Call Identification + YES           ¦
¦                                                    ¦
+----------------------------------------------------+
```

Software locks:
177: Total number of SIP users (including SIP devices and extensions).
345: Number of SIP extensions users (SEPLOS).

Suffix Plan (Default):
1 - Broker Call
2 - Consultation Call
3 - Three-Party Conference
4 - Barge-in (Intrusion)
5 - Callback On Free Or Busy Set
6 - Busy Camp-on
7 - Paging Request
8 - Voice Mail Deposit
* - DTMF end-to-end dialing

Prefix plan on OmniPCX TSS lab system (default):

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

11 Appendix C: AAPP member's escalation process

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

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

12 Appendix D: AAPP program

12.1 Alcatel-Lucent Application Partner Program (AAPP)

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

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

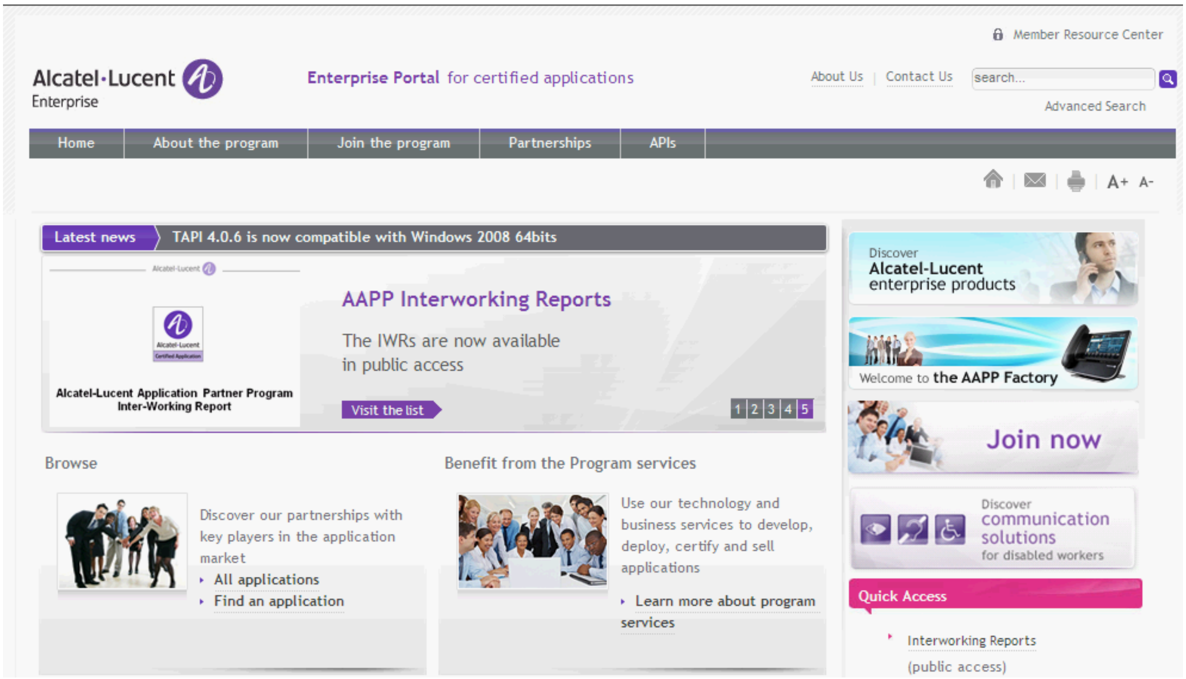
- **Provide easy interfacing for Alcatel-Lucent communication products**: Alcatel-Lucent's communication products for the enterprise market include infrastructure elements, platforms and software suites. To ensure easy integration, the AAPP provides a full array of standards-based application programming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

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

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

Web site

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



12.2 Alcatel-Lucent.com

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

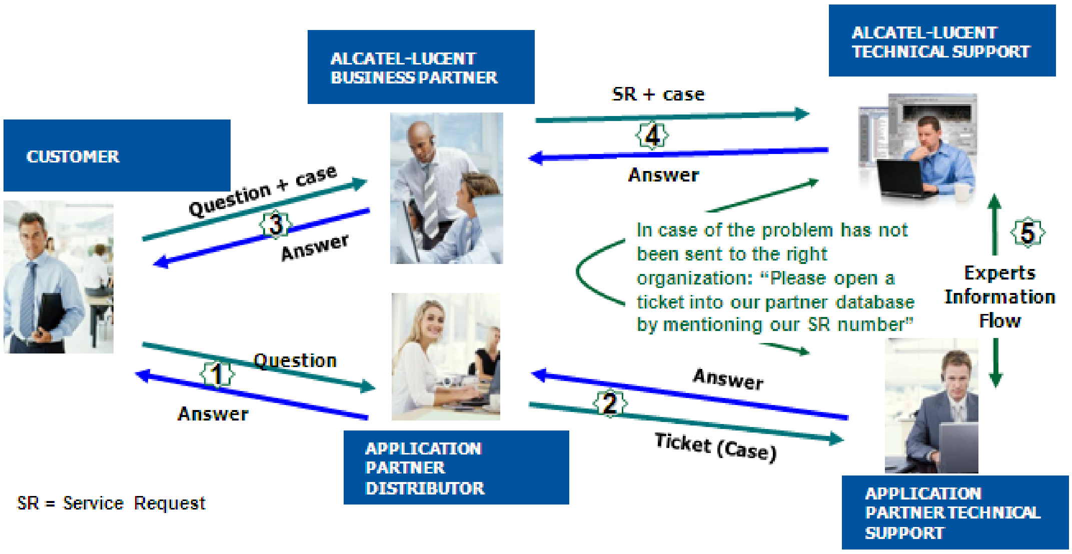
13 Appendix E: AAPP Escalation process

13.1 Introduction

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

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

In case technical support is granted, Alcatel-Lucent and the Application Partner, are engaged as following:



(*) The Application Partner Business Partner can be a Third-Party company or the Alcatel-Lucent Business Partner itself

13.2 Escalation in case of a valid Inter-Working Report

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

This defines the scope of what has been certified.

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

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

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

Case 3: the responsibility 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.

13.3 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 AAPP participant but there is no official Interworking Report (no IWR published on the Enterprise Business Portal for Business Partners or on the Alcatel-Lucent Application Partner web site) ,

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

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

13.4 Technical Support access

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

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

Alcatel-Lucent Business Partners Support Center for countries:

<table>
<tr><th>Country</th><th>Supported language</th><th>Toll free number</th></tr>
<tr><td>France</td><td rowspan="3">French</td><td rowspan="24">+800-00200100</td></tr>
<tr><td>Belgium</td></tr>
<tr><td>Luxembourg</td></tr>
<tr><td>Germany</td><td rowspan="3">German</td></tr>
<tr><td>Austria</td></tr>
<tr><td>Switzerland</td></tr>
<tr><td>United Kingdom</td><td rowspan="17">English</td></tr>
<tr><td>Italy</td></tr>
<tr><td>Australia</td></tr>
<tr><td>Denmark</td></tr>
<tr><td>Ireland</td></tr>
<tr><td>Netherlands</td></tr>
<tr><td>South Africa</td></tr>
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<tr><td>Poland</td></tr>
<tr><td>Sweden</td></tr>
<tr><td>Czech Republic</td></tr>
<tr><td>Estonia</td></tr>
<tr><td>Finland</td></tr>
<tr><td>Greece</td></tr>
<tr><td>Slovakia</td></tr>
<tr><td>Portugal</td></tr>
<tr><td>Spain</td><td>Spanish</td></tr>
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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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