Developer and Solution Partner Program
Inter-Working Report

Partner: **ASCOM**
Solution name: **IP DECT**
Alcatel-Lucent Enterprise Platform:
OmniPCX Enterprise

October 2020

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Disclaimer

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

While efforts were made to verify the completeness and accuracy of the information contained in this documentation, this document is provided "as is".

In the interest of continued product development, ALE International reserves the right to make improvements to this documentation and the products it describes at any time, without notice or obligation.

Document history

Revision	Date	Author	Details
1	October 2020	Thierry Chevert	Creation

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Tests Overview

Date	**October 2020**
ALE representative	**Thierry Chevert**
Partner representative	**Matthew Williams**
ALE platform	**OmniPCX Enterprise**
ALE release	**R12.4**
Partner solution	**IP DECT**
Partner release	**11.1.5**
Solution categories	**DECT handset**

Tests results

☑ Passed ☐ Passed with restriction ☐ Postponed ☐ Refused

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

IWR validity extension

None

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Table of contents

Table of contents

Table of contents

Chapter 1 INTRODUCTION

1.1 Definition

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

It certifies proper inter-working with the DSPP member's solution.

Information contained in this document is believed to be accurate and reliable at the time of printing. However, due to ongoing product improvements and revisions, ALE 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 Interworking Reports corner (access is restricted to Business Partners and DSPP members)

1.2 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 DSPP member issues a new major release of such product (incorporating new features or functionalities), or until ALE issues a new major release of such Alcatel-Lucent Enterprise product (incorporating new features or functionalities), whichever first occurs.

A new release is identified as following:

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

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

Note 1: *The InterWorking report becomes automatically obsolete when the mentioned product releases are end of life.*

Note 2: The renewal of the interoperability test (certification) is under the responsibility of the partner

Note 3: ALE usually generate a major release every 18 or 24 months. Therefore the IWR is implicitly valid for two year after the publication.

1.3 Limit of the technical support

For certified DSPP solutions, Technical support will be provided within the scope of the features which have been certified in the InterWorking report. The scope is defined by the InterWorking report via the tests cases which have been performed, the conditions and the perimeter of the testing and identified limitations. All those details are documented in the IWR. The Business Partner must verify an InterWorking Report (see above “Validity of the InterWorking Report) is valid and that the deployment follows all recommendations and prerequisites described in the InterWorking Report.

The certification does not verify the functional achievement of the DSPP member’s solution as well as it does not cover load capacity checks, race conditions and generally speaking any real customer's site conditions.

Access to technical support by the ALE Business Partner requires a valid ALE maintenance contract

For details on all cases (3rd party application certified or not, request outside the scope of this IWR, etc.), please refer to Appendix “DSPP Escalation Process”.

1.3.1 Case of additional Third-party applications

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

Chapter

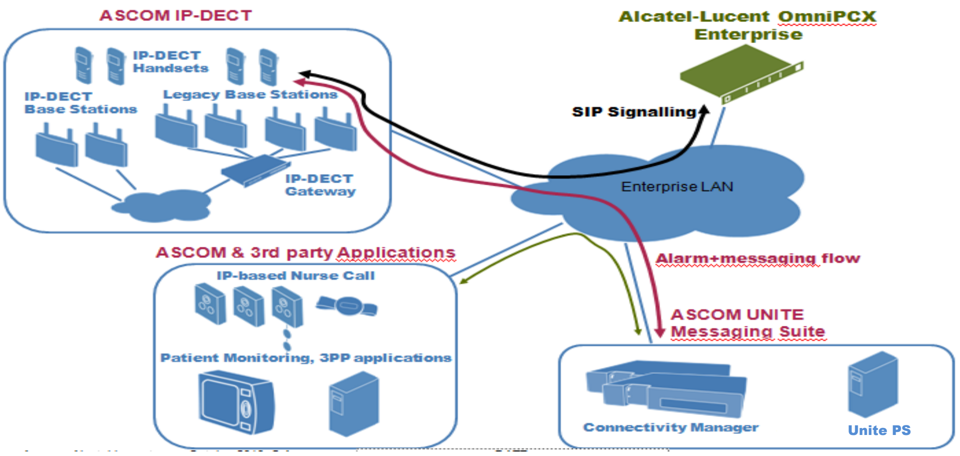
2

SOLUTION INFORMATION

Solution name	**ASCOM IP-DECT R10**
Solution version	**11.1.5**
Interface/API	**IP - SIP**
Interface/API version if relevant	

Brief Solution description:

The Ascom IP-DECT infrastructure and handsets integrates smoothly with the Alcatel-Lucent OmniPCX Office as an excellent mobility solution for several verticals. Combined with the Ascom Nurse Call, Patient Monitoring and Unite Messaging Suite, it develops into an excellent solution specifically for the smaller hospital and senior care establishments refer to the system view below. The solution has the capability to provide primarily a secure and safe communication environment for the patient, but also be efficient and cost-effective for the caregiver staff.
The application consists of IP-DECT base stations and associated Ascom handsets. IP-DECT base stations are linked to OXE via SIP protocol. All telephony features as provided by SIP Endpoint Level of Service are available to the handsets. The Ascom & 3rd party applications and the Ascom Unite Messaging Suite complete the solution.



Chapter 2 SOLUTION INFORMATION

The Ascom IP-DECT access points which are supported by the solution are the following:

IPBS2/IPBS3

IPBS1 (DECT radio only)

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

Ascom d81

Ascom d81ex

Ascom d62

Ascom d41

Chapter 2 SOLUTION INFORMATION

Important:

With the introduction of software version 9.1.x, Ascom IPBS1 has only radio functionality.

From the latest release notes: ***Downgrade/Upgrade concerns***

Background:
Due to lack of available flash space for new firmware/boot on IPBS1, we needed to remove reserved space for persistent data in order to make more space available.

Solution:
This means that the central software components are no longer supported on IPBS1. The IPBS1 is now only able to host the DECT Radio component. All IPBS1's in a system using any other functionality than DECT Radio component (i.e. Master, Mobility Master, Crypto Master, Kerberos server, Central Phonebook, Gateway) need to be replaced/swapped with IPBS2/IPVM/IPBL1 before upgrading to 9.1.X. If central software components are enabled on an IPBS1 there is a risk that there's already too little space in the flash to be able to upload the 9.1.X firmware. In that case a factory reset is needed to resolve the issue.

Chapter

3

TEST ENVIRONMENT

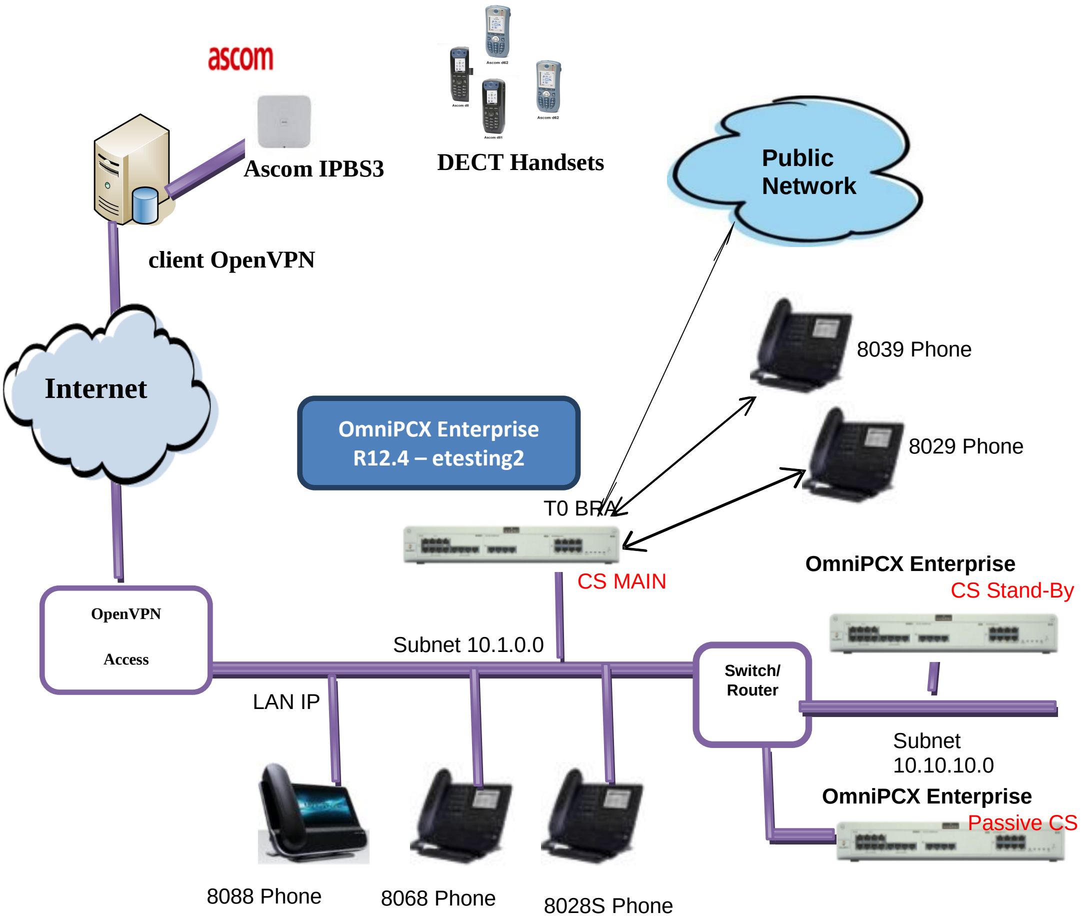


Figure 1 **Test environment**

3.1 Hardware configuration

3.1.1 Alcatel-Lucent Enterprise Communication Platform:

- Virtualized OXE CPU
- Spatial redundancy (Different IP subnetworks) – 2 CS-V
- One media gateway (GD3)
- One media gateway (GD2) with Passive CS (CS2)
- Various IP Touch Premium DeskPhones

3.1.2 Ascom platform:

- *1x Ascom IPBS3 base station : Mobility Master*
- *1x Ascom IPBS3 base station : PARI Master*
- *2x Ascom IPBS3 base station : Master User*
- *1x Ascom IPBS3 base station : Radio*
- 1x Ascom d62-Messenger DECT handset (release 4.3.6)
- 2x Ascom d63-Messenger DECT handset (release 2.10.2)
- 1x Ascom d41-Advanced DECT handset (release 4.3.6)
- 1x Ascom d43-Advanced DECT handset (release 2.0.8)
- 4x Ascom d81-Protector DECT handset (release 4.12.1)

3.2 Software configuration

3.2.1 Alcatel-Lucent Communication Platform:

Version: OmniPCX Enterprise R12.4 – M5.204

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.

1.1.2 Ascom platform:

Version: IPBS[11.1.5], Bootcode[11.1.5], Hardware[IPBS2 and IPBS3]

3.3 System Limits

Ascom IPBS/IPBL/IPVM:

- **Max 1000 users per IPBS/IPBL** Master base station. (500 SIP/TLS users).
  Max 4000 users per IPVM (Virtual Master appliance; 4000 SIP/TLS users)
  Multiple masters supported when more than 1000 users required, can be deployed at multiple sites;
- 2,047 IP-DECT base station radio per PARI master.
- 20,000 users/PARI master.
- 100 masters/mobility master.
- 100,000 users/mobility master.

OmniPCX Enterprise R12:

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

3.4 Summary of main functions supported

Below, telephonic features for SEPLOS (SIP Extension) supported or not by ASCOM base station and Dect sets.
OK: feature works well, **NOK**: issue while testing the feature, **OKBut**: it dosen't work completely, **NA**: not applicable or not tested

Features	Status	Comments
Bloc/overlap dialing	NA	
Codec	OK	
Set is free	OK	
Set is busy	OK	
DND	OK	
Out of service	OK	
Interception	OK	
Forward	OK	
Intrusion	OK	
Camp on	OK	
Secret identity	OK	
Call rejection	OK	
Call release	OK	
Hold	OK	
Broker call	OK	
Conference	OKBut	Dect handsets do not provide conferencing feature.
Transfer	OK	
Display management	OK	
Multi-line	OK	
Tandem, twin sets	OK	
Hunting Group	OK	
Voice Mail	OK	
Attendant	OK	
Prefixes support	OK	
Suffixes support	OK	
CPU redundancy support	OKBut	Alternate IP address proxy and SRV record but not Delegation.
PCS support	OKBut	Only in alternate proxy mode.

3.5 Summary of problems

- None

3.6 Summary of limitations

- OmniPCX Enterprise **spatial redundancy** (duplicate CS on different IP sub networks) and Passive Call Server are supported only with alternate proxy, The DNS method does not work as desired, see chapter 4.8 for further information.

- Voicemail: DECT handset does not show the number of messages left in voicemail. It only takes the first line of the OXE SIP NOTIFY message into account.

- On most DECT handsets, display management layout can be modified to accommodate additional rows and characters. Please refer to the handset's configuration manual for more information. By default, the DECT handset only display 2 lines of 12 characters each. In some cases, the strings sent by OXE in SIP MESSAGE are not fully displayed but are cut. There are otherwise 2 possible workarounds:
  - workaround on IPBS >supplement services> "external idle display" parameter: can be checked to remove the display of the display
  - workaround on OXE: sends NOTIFY instead of SIP MESSAGE in SIP Extension's Classes of service

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

- SIP Keep Alive mechanism with SIP OPTIONS messages is not supported by ASCOM base station

- "488 Not Acceptable here" messages are not managed correctly on DECT handsets. No warning on the screen, call is not stopped. This is a known issue on ASCOM side.

- SIP TLS not supported natively by the OXE for SIP extensions.

- G722.2 (AWR-WB) is not managed on OXE SIP extension devices. The codec is removed from the SDP list if it is proposed by the SIP set.

- There was an issue on UTF8 name sent by OXE with " (double quote) that was not correctly sent to SIP extensions. The single quote does work fine. See chapter 4.3, test 14 for more information.

3.7 Notes

- Configuring an NTP server on IP-DECT base station is **strongly** recommended

- The interworking tests only involve Ascom DECT base stations and handsets. Support for other third-party vendor DECT handsets has not been evaluated.

- OXE management of SIP gateway domain name is case sensitive. As a result, ASCOM Primary SIP proxy setting must match the string defined in SIP>SIP gateway > DNS local domain name. If it does not match, there will be issues in voice message notifications (MWI).

- A missed call is shown on DECT handset for some forward and twin set scenarios. However it is possible to switch off the display of the missed call popup by configuring the DECT handset parameter “show missed calls dialog window” to No. This is administered ideally from the Ascom Device Manager.

- OmniPCX Enterprise **spatial redundancy** (duplicate CS on different IP sub networks) is supported but CPU switchover is **not completely transparent** for the IP DECT users:
    - During a switchover the existing call is maintained, but in dialog SIP messages are sent to the old main call server. It is not possible to update an existing call. Dect sets use the IP address of its DNS cache until it registers on the new main call server. DNS answer TTL=0 is not taken into account.
    - A new call is not possible just after a switchover. Dect sets use the IP address of its DNS cache until it registers on the new main call server. The maximum failover timer would be a REGISTER expires (180 seconds) + a sip message timeout (32s).
    - Switchover to redundant call server or PCS only occurs after next “REGISTER”.

Chapter

4 TESTS RESULT

4.1 Template

The results are presented as indicated in the example below:

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Test case 1** • Action • Expected result	☐	☒	☐	
2	**Test case 2** • Action • Expected result	☐	☒	☐	The application waits for PBX timer or phone set hangs up
3	**Test case 3** • Action • Expected result	☒	☐	☐	Relevant only if the CTI interface is a direct CSTA link
4	**Test case 4** • Action • Expected result	☐	☐	☒	No indication, no error message
…	…	☐	☐	☐	

Test Case Id: a feature testing may comprise multiple steps depending on its complexity. Each step has to be completed successfully in order to conform to the test.
Test Case: describes the test case with the detail of the main steps to be executed the <u>and the expected result</u>
N/A: when checked, means the test case is not applicable in the scope of the application
OK: when checked, means the test case performs as expected
NOK: when checked, means the test case has failed. In that case, <u>describe in the field “Comment” the reason for the failure and the reference number of the issue either on ALE side or on partner side</u>
Comment: to be filled in with any relevant comment. Mandatory in case a test has failed especially the reference number of the issue.

4.2 Connectivity and Setup

4.2.1 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).

4.2.2 Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Provisioning** *Expected result: users created*	☐	☒	☐	Tested via web interface on base station. User declaration on Master base station.
2	**DHCP registration** (with OXE internal DHCP server) *Expected result: IPBS2 retrieves its IP address via OXE DHCP*	☒	☐	☐	Mirrored base stations are configured statically.
3	**NTP registration** (with OXE internal NTP server and external NTP) *Expected result : correct date and time on DECT handset*	☐	☒	☐	Time and date is displayed on the handsets (d81 and d63). The NTP configuration applied on Master Base Station for remaining tests.
4A	**SIP registration, using OXE MAIN IP addresses** (without authentication) *Expected result : SIP account with DECT handset number registered*	☐	☒	☐	OXE send 200-OK immediately and no 401-Unauthorized when Authentication is on.
4B	**SIP registration, using DNS** (without authentication) *Expected result : SIP account with DECT handset number registered*	☐	☒	☐	Both DNS A and DNS SRV requests are supported.
5	**SIP registration with authentication** Turn on SIP Digest authentication, specify realm on OXE, and specify username and password on SIP client.	☐	☒	☐	All SIP Handsets register OK. All DECT Handsets registered with FQDN address of Master Base Station when switched on.

	Expected result : SIP registration is authenticated				See Note (1) and (2).
6	Support of "423 Interval Too Brief" (1) *Expected result : SIP registration is performed based on OXE min interval*	☐	☒	☐	Expires set to 300s on Master Base Station. Register negotiates 300s, while subscribe will use 600s as specified by OXE. See Note (3).

Notes:
(1) 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: **etesting2.etesting.lab** (realm name configured on SIP > SIP Proxy)
On IPBS > DECT > Master > Register with Number: **TRUE**

(2) In the rest of the document, all the tests are done with SIP authentication set to SIP digest, as explained in OXE configuration section **Error! Reference source not found.**

(3) 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.

```
(102)etesting2-a> sipregister
sipregister h       To get help menu.
*************************************************
Dump local registrar base
-------------------------------------------------
Address of record : 12221
contact : sip:12221@10.1.2.244:2051, TCP, 282 s
-------------------------------------------------
Address of record : 12222
contact : sip:12222@10.1.2.244:2050, TCP, 112 s
-------------------------------------------------
Address of record : 12102
contact : sip:12102@10.1.2.91:53632, udp, 1744 s
-------------------------------------------------
Address of record : 12220
contact : sip:12220@10.1.2.244:2049, TCP, 149 s
-------------------------------------------------
Address of record : 12138
contact : sip:12138@10.1.18.108, udp, 3335 s
*************************************************
******   registred user number : 5
*************************************************
(102)etesting2-a>
```

4.3 Outgoing Calls

4.3.1 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.

4.3.2 Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Call to a local user** *Expected result : Ring back tone played, correct number displayed*	☐	☒	☐	The display on the handset is updated when the called user answers.
2	**Call to a local user with overlap dialing** Dial a part of the number, wait and continue. *Expected result : local call is performed correctly*	☒	☐	☐	Not possible to do overlap dialing on DECT handsets d81 and d63
3	**Call to a local user with overlap dialing, timeout.** Dial a part of the number, wait and stop. *Expected result : the call is released automatically*	☒	☐	☐	Not possible to do overlap dialing on DECT handsets d81 and d63 No timeout (PBX controlled).
4	**Call to a local user with overlap dialing, release** Dial a part of the number, wait and release the call. *Expected result : the call is correctly released*	☒	☐	☐	Not possible to do overlap dialing on DECT handsets d81 and d63
5	**Call to local user with no answer** Check timeout. *Expected result : if available, the call is stopped after a*	☐	☒	☐	Handsets d81 and d63 keep on ringing. No timeout (PBX controlled)

	timeout				
6	**Call to another SIP set** *Expected result : Ring back tone played, correct number displayed*	☐	☒	☐	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"** *Expected result : The call is released, an error message is displayed on the handset*	☐	☒	☐	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"** *Expected result : The call is released after playing a voice guide*	☐	☒	☐	OXE phones could not decline a call. Tested by calling another SIP Set. Correct handling for dect handsets.
9	**Call to busy OXE user** *Expected result : The call is released after playing a voice guide –*	☐	☒	☐	We hear busy tones, then call is released after 30s.
10	**Call to user in "Out of Service" state** SIP: "480 Temporarily Unavailable" *Expected result : The call is released automatically*	☐	☒	☐	Display: "Not reachable", short tones during 30 secs and the phone hung up.
11	**Call to user in "Do not Disturb" state** *Expected result : The call is released after playing a voice guide* *OXE prefix 42/Cancel prefix 42*	☐	☒	☐	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) *Expected result : The call is started with the forward target,*	☐	☒	☐	See note (1)

	the display is updated on the handset with the forward target details CFU prefix “51”/Cancel “41”.				
13	**Call to local user, forward on no reply (CFNR)** *Expected result : The call is started with the forward target, the display is updated on the handset with the forward target details* CFNR prefix “53”/Cancel “41”.	☐	☒	☐	Forwarded after ringing for 15s. See note (1)
14	**Call to local user, forward on busy (CFB)** *Expected result : The call is started with the forward target, the display is updated on the handset with the forward target details* CFNR prefix “52”/Cancel “41”.	☐	☒	☐	See note (1)
15	**Call to a local user without proxy Authentication**	☐	☒	☐	Note: in the rest of the document, SIP Digest authentication is activated
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). *Expected result : call is established using G711 codec*	☐	☒	☐	G711A negotiated when calling from Ascom DECT to OXE Phone. SIP Sets and OXE phone are in same IP domain 0. 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). *Expected result : call is established using G729*	☐	☒	☐	SIP Set is still in IP domain 0 and OXE phone 12014 is in domain 1. See note (2)
18	**Call to external number** *Expected result : public call is established, ring back tone is*	☐	☒	☐	*Ring back tone OK.* *Display the ISDN Trunk Name (PAI on 200 OK).*

	played on the handset				
19	**SIP session timer expiration** Check if call is maintained or released after the session timer has expired See note (3) *Expected result : call is running after the session expiration*	☐	☒	☐	Several hour calls have been tested OK
20	**Set lock/unlock** Dial the padlock prefix (“45”). To unlock, dial padlock prefix (“45”) + personal password. *Expected result : dial other prefixes than unlock is not allowed*	☐	☒	☐	Display of set shows “Set is locked”
21	**Use of abbreviated numbers** (Speed dialing) for both internal and external numbers. *Expected result : dial using abbreviated numbers is available*	☐	☒	☐	Softkey / hotkeys can be configured on phones.

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.

(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.

Additional informations:

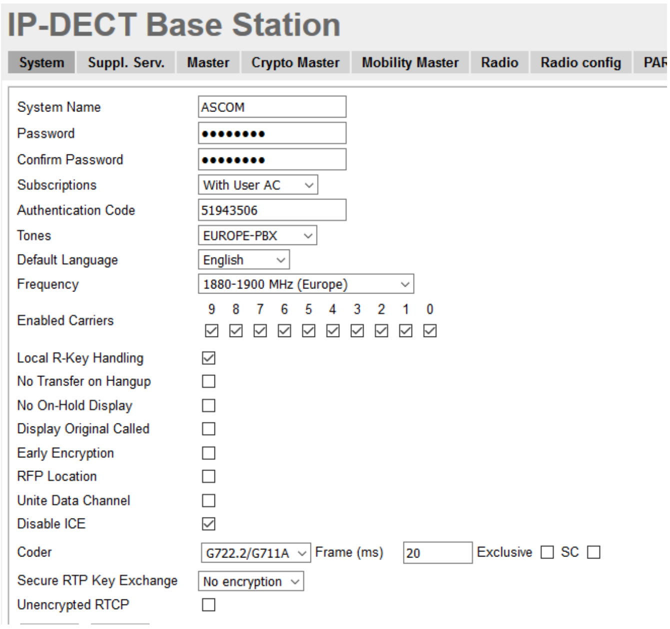
```
(102)etesting2-a> cnx dom

Tue Nov  3 19:37:54 CET 2020


                        IP Domain

   domain type : IP_R-> IP_REMOTE   NIPR-> NO_IP_REMOTE
```

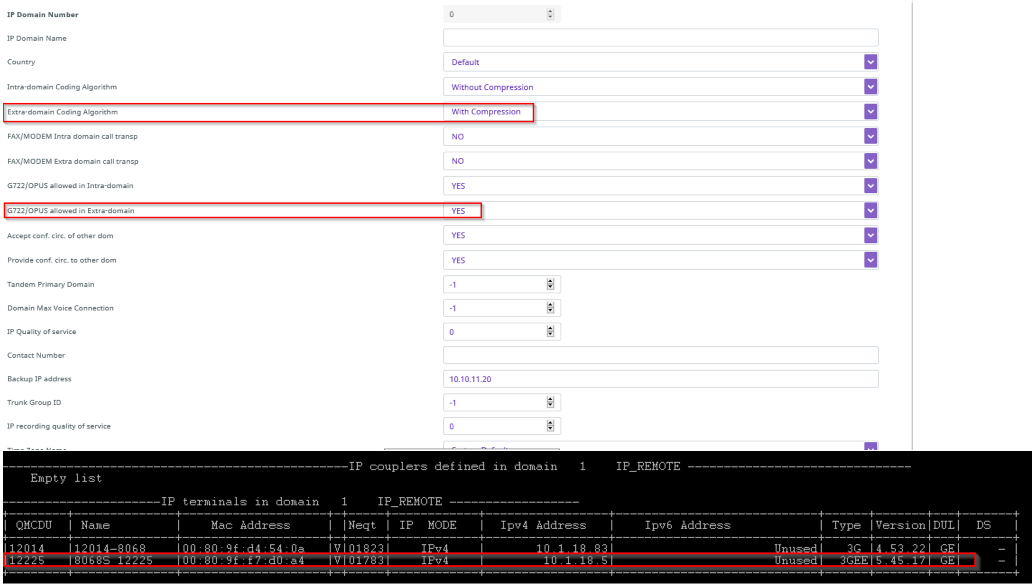
```
|   number |   0  |   1  |  11  |
|     type | NIPR | IP_R | IP_R |
|  allowed | ffff | ffff | ffff |
|     used |    0 |    0 |    0 |

| RIP Intr| G711 | G711 | G711 |
| RIP Extr| G729 | G711 | G711 |
| IPP Intr| G711 | G711 | G711 |
| IPP Extr| G729 | G711 | G711 |
| WB Int|  YES |  YES |   NO |
| WB Ext|  YES |  YES |   NO |
| UseOthCC|  YES |  YES |  YES |
| ProvidCC|  YES |  YES |  YES |

| cac over|    0 |    0 |    0 |
| comp alw|   47 |    0 |   24 |
| comp use|    0 |    0 |    0 |
| comp fre|   45 |    0 |   21 |
| comp out|    2 |    0 |    3 |
| comp ovr|    0 |    0 |    0 |
| Tand PDm|   -1 |   -1 |   -1 |
| Tand CAC|   -1 |   -1 |   -1 |
cnx [ cfg | obj | cr | load | WORD_# ]


Domstat command and option 9: display all domain devices

----------------------SIP devices in domain   0  DEFAULT_DOM -------------------
    Empty list

----------------------SIP extension terminals in domain   0  DEFAULT_DOM -------
+--------+-----+----------------+---------------+-----+
| QMCDU  |Neqt |      Name      |  Ip Address   |State|
+--------+-----+----------------+---------------+-----+
|12138   |03620|8088SIPExten 121|010.001.018.108| ES  |
|12220   |03625|Etesting asc 122|010.001.002.244| ES  |
|12221   |03626|Etesting asc 122|010.001.002.244| ES  |
|12222   |03627|Etesting asc 122|010.001.002.244| ES  |
+--------+-----+----------------+---------------+-----+

No. of SIP extension terminals connected to the domain 0 is: 4


----------------------IP Dect terminals in domain 0 DEFAULT_DOM --------------
    Empty list

        If the State shows
        ES: means the set is in service
        HS: means the set is out of service


=========================================================================================================
=======================

----------------------IP terminals in domain   1    IP_REMOTE ------------------
+--------+--------------+--------------------+-+-----+------------+-----------------+------------------
----------+------+-------+---+-------+
| QMCDU  | Name         |    Mac Address     | |Neqt | IP  MODE   |  Ipv4 Address   |    Ipv6 Address
| Type  |Version|DUL|  DS    |
+--------+--------------+--------------------+-+-----+------------+-----------------+------------------
----------+------+-------+---+-------+
|12014   |12014-8068    |00:80:9f:d4:54:0a   |V|01823|    IPv4    |       10.1.18.83|
Unused|   3G |4.53.22| GE|     - |
+--------+--------------+--------------------+-+-----+------------+-----------------+------------------
----------+------+-------+---+-------+

        Total number of IP Phones : 1


(102)etesting2-a> compvisu eqt all

Tue Nov  3 19:38:05 CET 2020
```

```
+===============================================================================+
|                              C O M P V I S U                                  |
+===============================================================================+
| neqt                :                     1823 <--> 3625                      |
| (neqt_it)           :                   (1823) <--> (3625)                    |
| coupler type        :                UA_FICTIF <--> UA_FICTIF                 |
| (cr-cpl-term)       :                  (2-0-0) <--> (2-1-0)                   |
| type term           :      IPT 8068, mcdu=12014 <--> SIP, mcdu=12220          |
| infocomp            :             IP:Profile #2 <--> IP:Profile #2            |
| nbcomp / comp type  :    - / LIOE_IP-G729 (43) <--> - / LIOE_IP_NOT-G729 (44)|
|-------------------------------------------------------------------------------|
| neqt                :                     3629 <-->                 AUX_ZERO |
| (neqt_it)           :                   (3629) <-->                           |
| coupler type        :                UA_FICTIF <-->                           |
| (cr-cpl-term)       :                  (2-1-0) <-->                           |
| type term           :              OP IPT 8068 <-->                           |
| nbcomp / comp type  :                 - / 0x00 <-->                           |
+===============================================================================+
(102)etesting2-a>
```

Tested:
IPBS>DECT>System>Coder=G711A, non exclusive.
SIP domain 0 calls OXE domain 0: G.711, G.729, G.723 proposed, G.711 chosen. (OK)
SIP domain 0 calls OXE domain 1: G.711, G.729, G.723 proposed, G.729 chosen. (OK)

Note: If G722.2 (AMR-WB) is proposed by ASCOM set, it is removed from the SDP list by OXE as it is not supported on OXE SIP extensions.
Ascom IPDECT sends AMR-WB to OXE that is supported by DECT handsets d43 and d63.





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.

4.4 Incoming Calls

4.4.1 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.

Call waiting needs to be configured to have 2 lines on the handset.

4.4.2 Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	Local /network/external call to free SIP terminal *Expected result : Ring back tone played, correct number displayed*	☐	☒	☐	External calls: DECT handset uses another type of ring than for local calls.

2	Local/network call to busy SIP terminal *Expected result : Call is disconnected*	☐	☒	☐	486 Busy Here generated when we call a Busy DECT handset Short tones, “Hung up” after 5 secs.
3	Local/network call to unplugged SIP terminal *Expected result : Call is disconnected*	☐	☒	☐	480 Temporarily not available 2 behaviors: - IP DECT has been switched off: REGISTER with expires 0 is sent “Out of service” is displayed on OXE phone and the call is disconnected OXE cancels call after no response to INVITE (timeout approx. 10 seconds) - Battery has been removed: no REGISTER with expires 0 sent, OXE tries a few times to reach the phone, then hangs up.
4	Local/network call to SIP terminal in Do Not Disturb (DND) mode:	☐	☒	☐	
4A	By local feature (prefix “*42#”/cancel “#42#”) *Expected result : DND activated*	☐	☒	☐	486 Busy Here generated when we call a Busy ASCOM DECT Short tones, “Hung up” after 5 secs.
4B	By system feature (SEPLOS) (prefix “42”+ user password) *Expected result : DND activated*	☐	☒	☐	Message “Do Not Disturb” is displayed on DECT handset See Note (1)
5	Local/network/SIP call to SIP terminal in immediate forward (CFU) to local user:	☐	☒	☐	
5A	By local feature (prefix “*21*<extn>#”/cancel “#21#”) *Expected result : call forwarded to forward target*	☐	☒	☐	
5B	By system feature (SEPLOS) (prefix “51”+number/”41”) *Expected result : call forwarded to forward target*	☐	☒	☐	Message “Immediate forward to extension number” is displayed on DECT handset See Note (1)
6	Local/network/SIP call to SIP terminal in immediate forward (CFU) to network number:				

6A	By local feature *Expected result : call forwarded to forward target*	☐	☒	☐	
6B	By system feature (SEPLOS) (prefix “51”+number/”41”) *Expected result : call forwarded to forward target*	☐	☒	☐	Message “Immediate forward to extension number” is displayed on DECT handset See Note (1)
7	Local/network/SIP call to SIP terminal in immediate forward (CFU) to another SIP user				
7A	By local feature *Expected result : call forwarded to forward target*	☐	☒	☐	
7B	By system feature (SEPLOS) (prefix “51”+number/”41”) *Expected result : call forwarded to forward target*	☐	☒	☐	Message “Immediate forward to extension number” is displayed on DECT handset See Note (1)
8	**Local call to SIP terminal in “forward on busy” (CFB) state**				
8A	**By local feature** (prefix “*67*<extn>#”/cancel “#67#”) *Expected result : call forwarded to forward target*	☐	☒	☐	
8B	**By system feature (SEPLOS)** (prefix “52”+number/”41”) *Expected result : call forwarded to forward target*	☐	☒	☐	By default, a SIP extension user on OXE is configured with two lines. To reach the busy state on the DECT handset, you need to activate the local Call waiting feature to present the second call on the handset Message “forward on busy to extension number” is displayed on DECT handset See Note (1)
9	**Local call to SIP terminal in “forward on no reply” (CFNR)**				
9A	**By local feature** (prefix “*61*<extn>#”/cancel “#61#”) *Expected result : call forwarded to forward target*	☐	☒	☐	Forwarded after 15s.

9B	**By system feature (SEPLOS)** (prefix “53”+number/”41”) *Expected result : call forwarded to forward target*	☐	☒	☐	Message “forward on no reply to extension number” is displayed on DECT handset See Note (1)
10	**Call to busy user, Call waiting (Camp-on)** *Expected result : call waiting on the busy set*	☐	☒	☐	
11	**External call to SIP terminal** *Expected result: external call back number is shown correctly.*	☐	☒	☐	Display of dect phone: External call 388612014
12	**Identity secrecy/CLIR: Local call to SIP terminal** Dial “409 + extension”. Check that caller id is not shown. *Expected result :.caller id is not presented*	☐	☒	☐	Display on Called party: Identity Secrecy
13	**Display: Call to free SIP terminal from user with a name containing non-ASCII characters** *Expected result : caller display is correct*	☐	☒	☐	Tested Ok for Latin-1 characters Non ASCII characters could not be managed in the directory name except $ and * but not accented characters.
14	**Display: Call to free SIP terminal from user with a UTF-8 name containing non-ASCII characters** *Expected result : caller display is correct*	☐	☒	☐	*Caller user UTF8 directory name was set as: àéè&$£µ*ù%. This name display is ok on Bria 4 and ok on DECT handsets except µ that is displayed with strange ? in a black lozenge.* *Issue with “ (double quotes) characters that prevent call on Bria4 and give strange behavior on Dect handsets ‘On Dect the display shows External Call and the ring is like external. The OXE caller party is released after 10 seconds. When call is released on caller then the dect phone still rings waiting for action. Same behavior with d41,*

					d62 and d81). It works fine with ' (single quotes).
15	**SIP set is part of a sequential hunt group** Call to hunt group. Check call/release. *Expected result : call / release OK*	☐	☒	☐	Hunt group number was 12136. Both SIP sets are free, calls goes always to first element of group.
16	**SIP set is part of a cyclic hunt group** Call to hunt group. *Expected result : call / release OK*	☐	☒	☐	Both SIP sets are free, calls goes alternatively to each element of group.
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. *Expected result : answers call from the twin set is working, answers call on the deskphone stops ringing the handset*	☐	☒	☐	*Main set in tandem was 12014 and tandem directory number was 12220 SIP Set.* *When calling twin set directly, name of main set is displayed on caller.* *Missed call is shown on DECT handset if the call is answered on main set.* *See Note (2).*
18	**Same as 17. Then transfer to main set. (hang up)** *Expected result : call transferred to the deskphone*	☐	☒	☐	*On SIPset twin do R12014 then caller is on-hold and call ring on main.Do R4 to execute transfer. Works with attended and semi-attended transfers.*
19	**Call to OXE phone Pick-up by SIP Set (Supervision)** A call from OXE set to another OXE set is picked up from a SIP set by dialing the call pick-prefix ("55"+number of target set) *Expected result : call pick up on the ascom handset*	☐	☒	☐	*Call to SIP set and pick-up tested from other SIP set and OXE phone.*
20	**Call to SIP Set Pick-up by OXE phone (Supervision)**	☐	☒	☐	

	A call from SIP set to another SIP set is picked up from a OXE set by dialing the call pick-prefix (“55”+number of target set) *Expected result : call pick up on the ascom handset*				

Note (1): DECT handset can only display 2 lines of 12 characters each. In some cases, the strings sent by OXE in SIP MESSAGE are not fully displayed, but are cut.

There are 2 possible workarounds:

- workaround on IPBS >supplement services> "external idle display" parameter : can be checked to remove the display of the display
- workaround on OXE: sends NOTIFY instead of SIP MESSAGE in SIP Extension’s Classes of service

NOTE: On most DECT handsets, display management layout can be modified to accommodate additional rows and characters. Please refer to the handset's configuration manual for more information.

Note (2): It is possible to switch off the display of the missed call popup by configuring DECT handset parameter “show missed calls dialog window” to No. This is administered ideally from the Ascom Device Manager.

4.5 Features during Conversation

4.5.1 Test Objectives

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

"Call waiting" parameter must be enabled on IPBS to activate the multi line capability.

1.1.3 Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Hold and resume (both directions)** Check tones *Expected result : place a call on hold, then resume it*	☐	☒	☐	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** *Expected result : second call established*	☐	☒	☐	Press R + second call number.
3	**Broker request** (toggle back and forth between both lines, local feature) *Expected result : audio switched between call 1 and call 2, display is correctly updated*	☐	☒	☐	*R + "2" to switch between participants* *(first switch fails)*
4	**Release first call. Keep second call** *Expected result : second call still established*	☐	☒	☐	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. *Expected result : call retrieved*	☐	☒	☐	*R then "402" + number of parking and got voice prompt to accept.* *To retrieve dial 402 + number of parking.*

6	**Send/receive DTMF** *Expected result : possibility to send DTMF*	☐	☒	☐	DTMF are sent as RFC2833. Need to have *IPBS>DECT>Master>Allow DTMF through RTP* **enabled**. Tested while calling voice mail 12999.
7	**Three party conference initiated from OXE set** For IPTouch there are dynamic keys to set conference and for others (analog and SIP) there is suffix 3. Released by OXE set. *Expected result : conference established*	☐	☒	☐	
8	**Three party conference initiated from SIP set** (local feature). Released by SIP set. *Expected result : conference established*	☒	☐	☐	Feature not available.
9	**Barge-in (Intrusion) from SIP set** The SIP set calls another set which is in conversation. Then press the barge-in suffix “4”. *Expected result : call intrusion established*	☐	☒	☐	Have to remove all Barge-in protection in the Phone feature facility.
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”. *Expected result : call back configured*	☐	☒	☐	
11	**Busy Camp-on from SIP set** The SIP set calls another set which is in conversation. Then press the camp-on suffix “6”. *Expected result : hold music listen on handset during the camp on*	☒	☐	☐	

	period				
12	**Voice mail deposit from SIP set** The SIP set calls another set. Then press the message deposit suffix “8”. *Expected result : reach the user mailbox after dialing the prefix*	☐	☒	☐	OXE user is fully busy and diverted on busy to voice mail.
13	**Meet-me conference** Participate in ongoing meet-me conference from SIP set (prefix “509”+number+code). Released by SIP set. *Expected result : reach the meet-me conference after dialing the prefix*	☐	☒	☐	Tested with 4 participants, on SIP Set dial 509 then got prompt to enter personal code and dial 1234 for instance.

4.6 Call Transfer

4.6.1 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 for OXE set

- ***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.

4.6.2 Test Results

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

For each test, we have verified that CLIP is correctly updated after the transfer.

Unattended Transfer (blind, transfer before ringing)
Unattended transfer procedure for Ascom handsets: RR + Phone number (quickly), then press hash or wait for the timeout.

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**	On OXE phone, dial directly the number on keypad while ringing.
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**	Missed call is seen on the transfer target and must be acknowledged in call list to clear. See Note 1 (all of the above same as before).

Note (1): It is possible to switch off the display of the missed call popup by configuring DECT handset parameter “show missed calls dialog window” to No. This is ideally administered from Ascom Device Manager.

Note (2): At the end of some transfer cases, there is no RTP direct between the DECT handsets. Though the SIP devices are in the same IP domain, OXE adds the IP of the GD board as the IP connection info to transcode one leg in G711 and the other in G729. Codec renegotiation not managed by OXE for SEPLOS sets after transfer.

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

Test	Action			Result	Comment
	A **Transferee**	B **Transferor**	C **Transfer Target**		
1	SIP	OXE	OXE/Ext Call	**OK**	
2	OXE/Ext Call	SIP	OXE	**OK**	
3	OXE/Ext Call	OXE	SIP	**OK**	Missed call is seen on the transfer target and must be acknowledged. See Note (1) (same as before).
4	OXE / Ext call	SIP	SIP	**OK**	CLIP not updated on C party until answer.
5	SIP	OXE	SIP	**OK**	CLIP updated on C upon transfer. Missed call is seen on the transferor and must be acknowledged. See Note (1) (same as before)
6	SIP	SIP	OXE/Ext Call	**OK**	First SIP set remain in hold state after release by transferor and before answer by target.
7	SIP	SIP	SIP	**OK**	CLIP not updated on C party until answer. See Note (2)

Note (1): It is possible to switch off the display of the missed call popup by configuring DECT handset parameter “show missed calls dialog window” to No. This is ideally administered from Ascom Device Manager.

Note (2): At the end of some transfer cases, there is no RTP direct between the DECT handsets. Though the SIP devices are in the same IP domain, OXE adds the IP of the GD board as the IP connection info to transcode one leg in G711 and the other in G729. Codec renegotiation not managed by OXE for SEPLOS sets after transfer.

Attended Transfer (transfer in conversation)

Attended transfer procedure for Ascom handsets: Press “R” + destination number. Wait until destination number answers the call. Then hangup (System option No Transfer on Hangup = **disabled,** otherwise the end user should press R + 4 to transfer the call).

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**	See Note (2)

4.7 Attendant

4.7.1 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**

4.7.2 Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Call to attendant and release** From SIP set call attendant using attendant call prefix “9”. *Expected result : call established with the attendant station*	☐	☒	☐	
2	**Call to attendant and semi-attended transfer to OXE set** From SIP set call attendant using attendant call prefix “9”. Attendant transfers to OXE set in semi attended mode. *Expected result : call transferred from the attendant station*	☐	☒	☐	Ext call: no name update on Dect handset after transfer
3	**Call to attendant and attended transfer to OXE set** From SIP set call attendant using attendant call prefix “9”. attendant transfers to OXE set in attended mode. *Expected result : call transferred from the attendant station*	☐	☒	☐	
4	**Call to attendant and attended transfer to SIP set** OXE set / Ext. calls to attendant (using attendant call	☐	☒	☐	

	prefix “9”), attendant transfers to SIP set, attended. *Expected result : call transferred from the attendant station*				
5	**Second call to SIP set in conversation with Attendant** Call to attendant (using attendant call prefix “9”). Second incoming call while in conversation with attendant. *Expected result : second call refused*	☒	☐	☐	Second call is camp-on, according to management, but could not be retrieved as it is refused by OXE as expected. But on ASCOM set, the call is not rejected. “488 Not Acceptable here” sent but Dect handset does not show any warning on the screen, call not stopped. This is a known issue on ASCOM side.

4.8 Voice Mail

4.8.1 Test Objectives

Voice Mail notification, consultation and password modification must be checked.
MWI (Message Waiting Indication) has to be checked.
The SIP set did subscribe to voice mail service in order to get MWI notification.
As soon as voice mail service is set on SIP Set then it receive a notification to call voice mail and personalize its box.

4.8.2 Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**A Voice Mail message for the SIP subscriber is generated.** *Expected result : MWI is activated*	☐	☒	☐	See Note (1) MWI indication is displayed by message on screen and icon.
2	**Message consultation** *Expected result : message consulted via ascom handset*	☐	☒	☐	See Note (1)
3	**Message deletion** *Expected result : message is*	☐	☒	☐	See Note (1) The icon for message

	deleted via ascom handset				indication disappeared when all messages are deleted.
4	**Password modification** *Expected result : user is able to change its password dialing a new password via DTMF*	☐	☒	☐	
5	**SIP call to a OXE user forwarded to Voice Mail** *Expected result : call is forwarded to voice mail*	☐	☒	☐	
6	**OXE call to a SIP user forwarded to Voice Mail** *Expected result : call is forwarded to voice mail*	☐	☒	☐	SIP set dial 51 for imm. Forward then on prompt dial 12999 voice mail directory number.

Note (1): DECT handset does not show the number of messages left in voicemail. It only takes the first line of the OXE SIP NOTIFY message into account.

4.9 Duplication and Robustness

4.9.1 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?

4.9.2 Test Results

Test Case Id	Test Case	N/A	OK	NOK	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). *Expected result: call keeps running. It possible to evolve a n existing call (place on hold, transfer) and to place a new call*	☐	☒	☐	After a switchover, the next call can be established after the next registration period, here set to 300 seconds. Before this registration, it is not possible to evolve an existing call (place on hold, transfer…) and to place a new call. In IPDECT, DECT/Master the “Domain” should be empty to use the IP addresses.
2A	**Spatial redundancy via DNS Delegation method** Configure the FQDN on the proxy field only (if delegation) *Expected result: call keeps running. It possible to evolve a n existing call (place on hold, transfer) and to place a new call*	☒	☐	☐	Not Ok with DNS delegation as the IPDECT BS asked always for SRV DNS query.
2B	**Spatial redundancy via DNS SRV method** Configure the FQDN on the proxy field only (if delegation)	☐	☒	☐	IPDECT request to DNS for “_sip._tcp.etesting2srv.etesting.lab” (etesting2srv.etesting.lab is “Proxy” value).

	Expected result: call keeps running. It possible to evolve a n existing call (place on hold, transfer) and to place a new call				See (1) below. The SRV request was done by IPDECT and answered by DNS.
3A	**Passive Call Server (PCS) in alternate proxy method** (IP link to main/stand-by call servers down) *Expected result : It possible to place a new call after the activation of the PCS*	☐	☒	☐	Switchover to Passive Call Server only occurs after next “Register”. Before this registration, it is not possible for an existing call to evolve (place on hold, transfer…) and to place/receive a new call. It works also if main proxy is SRV and Alt.Proxy is set to PCS IP@.
3B	**Passive Call Server (PCS) in DNS SRV method** (IP link to main/stand-by call servers down) *Expected result : It possible to place a new call after the activation of the PCS*	☒	☐	☐	Does not work with SRV method. There were already the 2 CPU CS into SRV record and PCS was the third one. It did switch to PCS at place of going to second CPU even if PCS was inactive. *This is a known limitation. Ascom recommends configuring an IP address for the PCS on the base station.*
4	**SIP device reboot** Check that calls are possible as soon as device has come back to service. *Expected result : can establish a call as soon as the SIP phone is rebooted*	☐	☒	☐	
5	**Temporary Link down with the PBX** *Expected result : can establish a call as soon as the network link is re established*	☐	☒	☐	Display “PBX Out of service” or “Master out of service”

Notes:

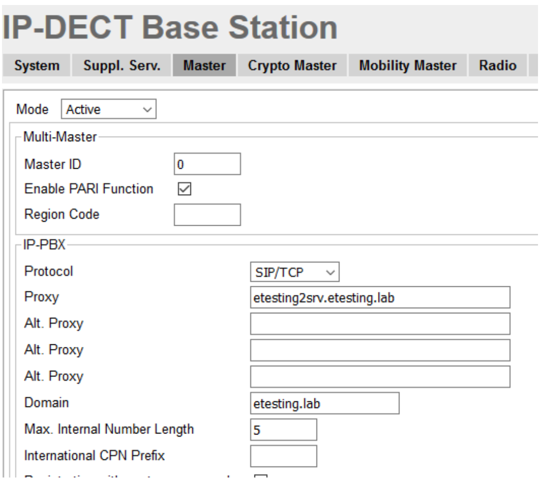
Redundancy tests were done with master and radio running on one base station only.
In order to have acceptable switchover time the keep-alive mechanism (SIP Options) can be used but this feature is unavailable on Ascom side. This is an Ascom limitation.

Chapter 4 TESTS RESULT

OmniPCX Enterprise **spatial redundancy** (duplicate CS on different IP sub networks) is supported but CPU switchover is **not completely transparent** for the IP DECT users:
During a switchover the existing call is maintained, but in dialog SIP messages are sent to the old main call server. It is not possible to update an existing call. Dect sets use the IP address of its DNS cache until it registers on the new main call server. **DNS answer TTL=0 is not taken into account by IPBS.**
A new call is not possible just after a switchover. DECT sets use the IP address of its DNS cache until it registers on the new main call server. The maximum failover timer would be a REGISTER expires (180 seconds) + a sip message timeout (32s).
Switchover to redundant call server or PCS only occurs after next “REGISTER”.

(1)

```
nslookup

> _sip._tcp.etesting2srv.etesting.lab
Server:  localhost
Address:  127.0.0.1

_sip._tcp.etesting2srv.etesting.lab     SRV service location:
          priority       = 10
          weight         = 10
          port           = 5060
          svr hostname   = etesting2-am.etesting.lab
_sip._tcp.etesting2srv.etesting.lab     SRV service location:
          priority       = 20
          weight         = 10
          port           = 5060
          svr hostname   = etesting2-bm.etesting.lab
etesting2-am.etesting.lab       internet address = 10.1.6.1
etesting2-bm.etesting.lab       internet address = 10.10.10.11
>
```



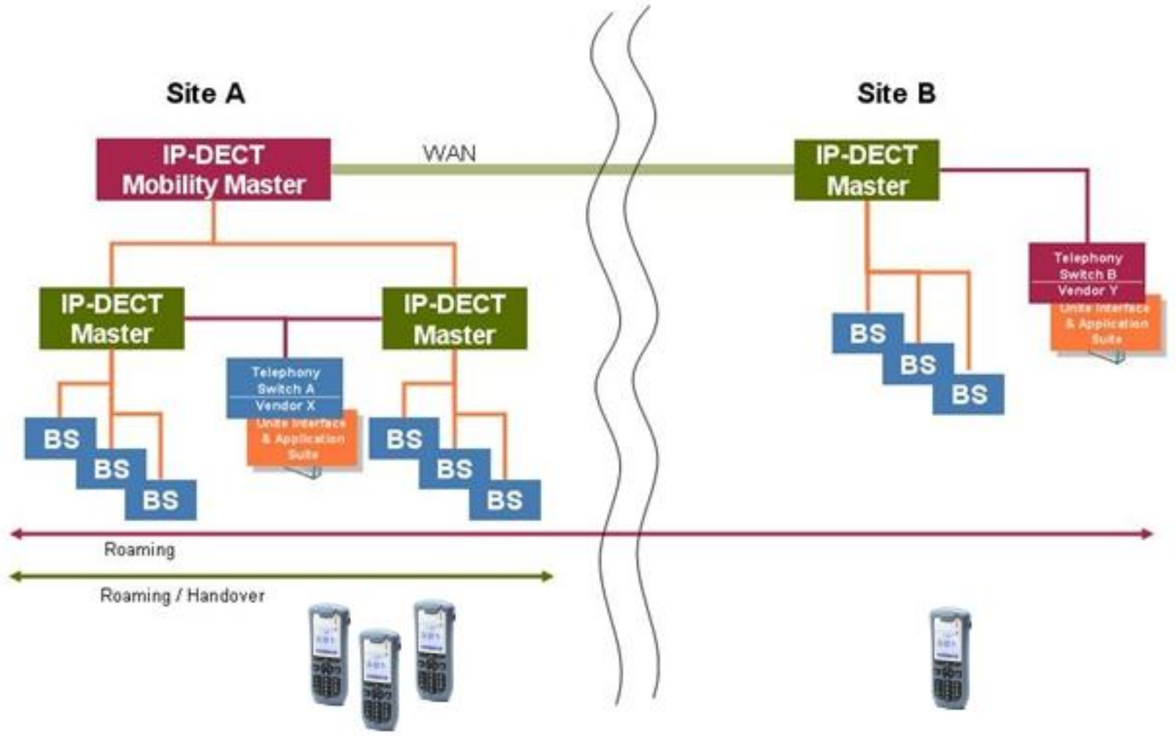
Chapter

5 Appendix A: SOLUTION 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 enables a completely new principle on how to build large systems and gives the opportunity to balance the load between different IP-PBX's

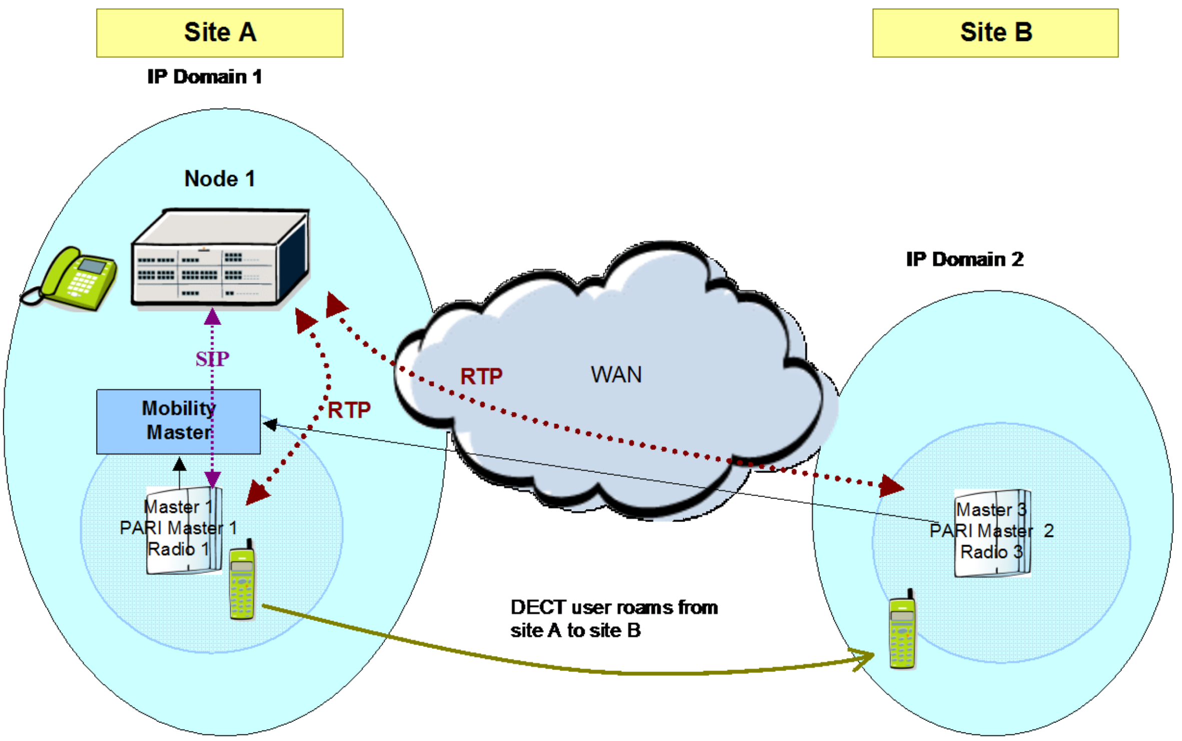
The picture below illustrates a typical multi-Master system:



5.1 DECT Multi-Master/Multi-Site

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!

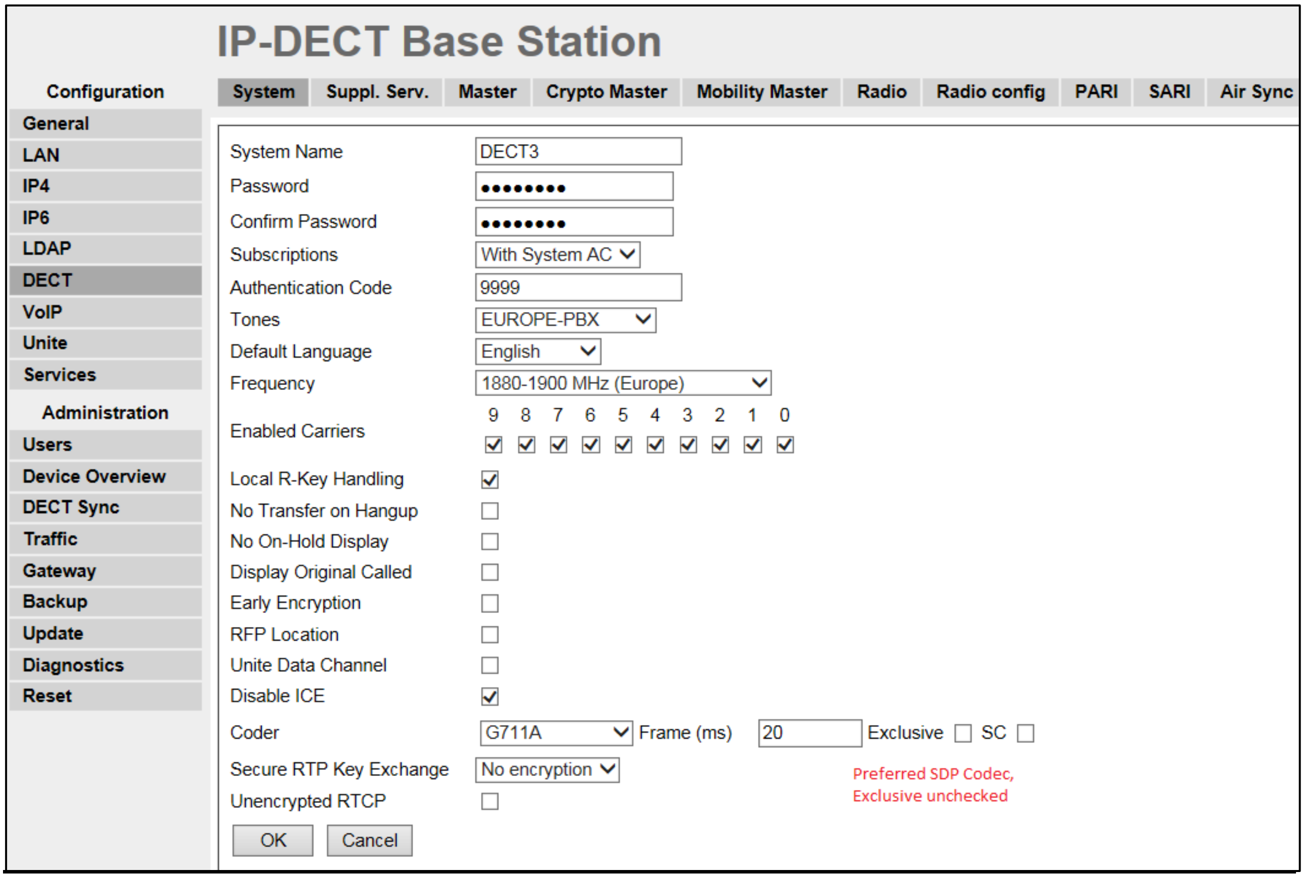


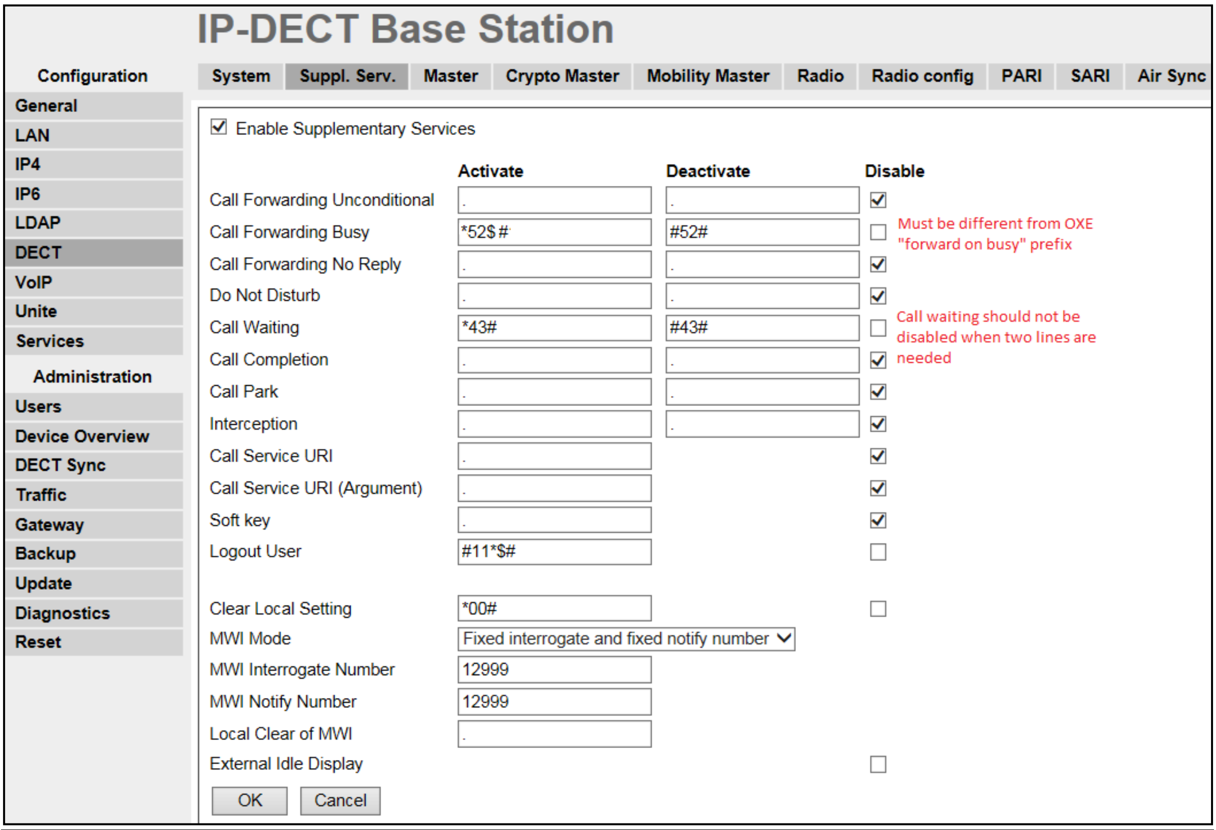
Chapter 6 Appendix B: PARTNER side CONFIGURATION

For configuration of the Ascom IP-DECT system, refer to Ascom “Installation and Operation Manual IP-DECT base station” 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.

DECT system:



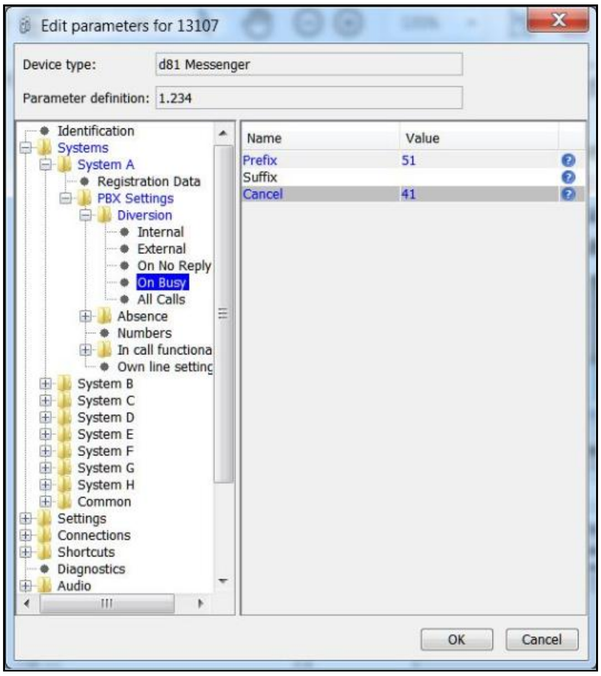


DECT Supplementary services:

“MWI Interrogate Number” and MWI Notify Number” must be configured with OXE Voicemail number

OXE external SIP set is a multiline station with 2 lines configured. If call waiting feature is not activated on DECT handset, call forwarding on busy set should be configured on Ascom system and not on OXE system to transfer a second incoming call.

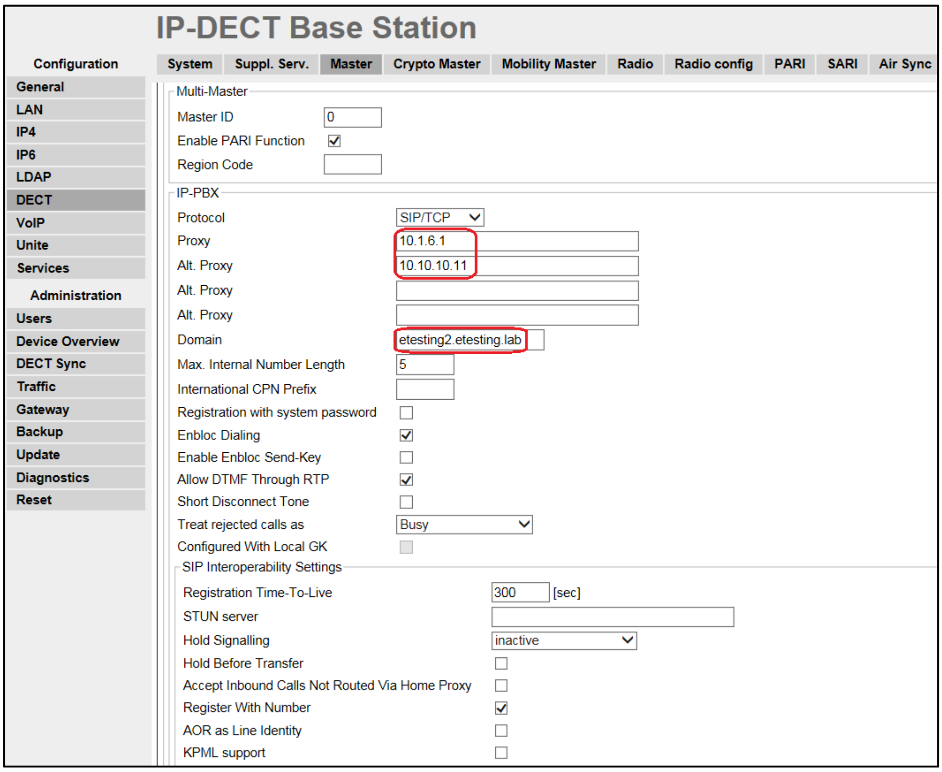
It is possible to disable the dect system local feature, and use the OXE feature instead. For such configuration, direct access buttons can be configured on DECT handset. To do so, check the “disable” button on the associated local feature. OXE system CFU prefixes (activation/cancellation) have been activated in the following example:



Note: Screenshot included as reference.

SIP configuration – alternate proxy configuration:

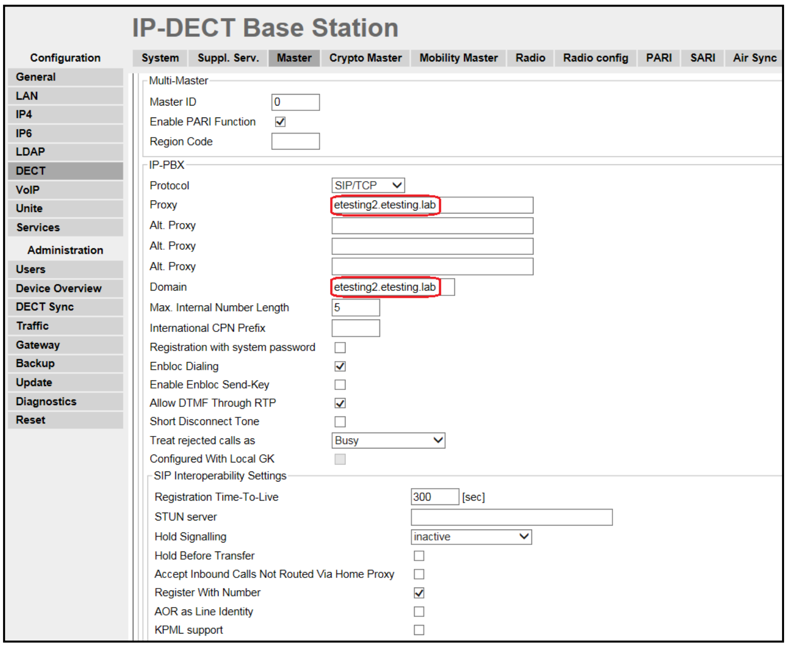
Add a third IP address for PCS.



Note: During these tests, the “Registration Time-to-Live” was set to 300 seconds.

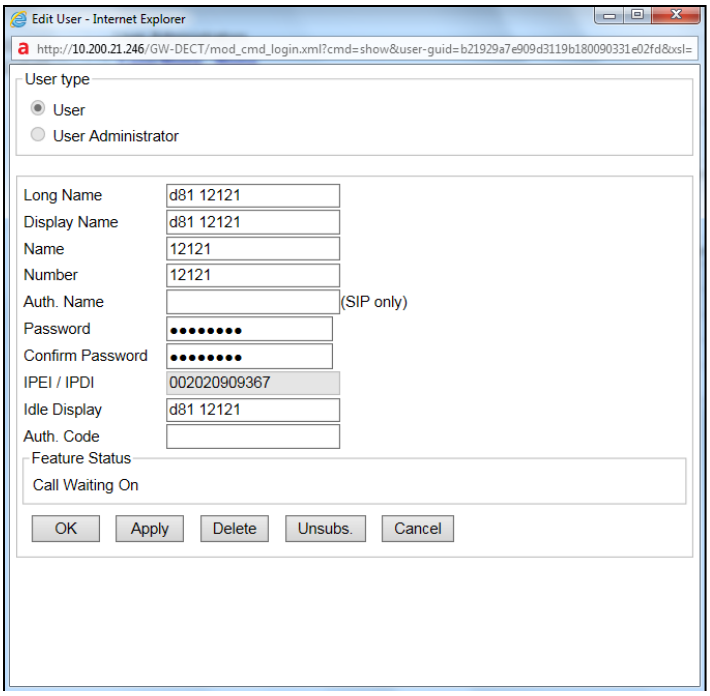
SIP configuration – FQDN configuration:

The switch-over of CS is not supported by IPBS.



Note: "Registration Time-to-Live" was set to 300 seconds.

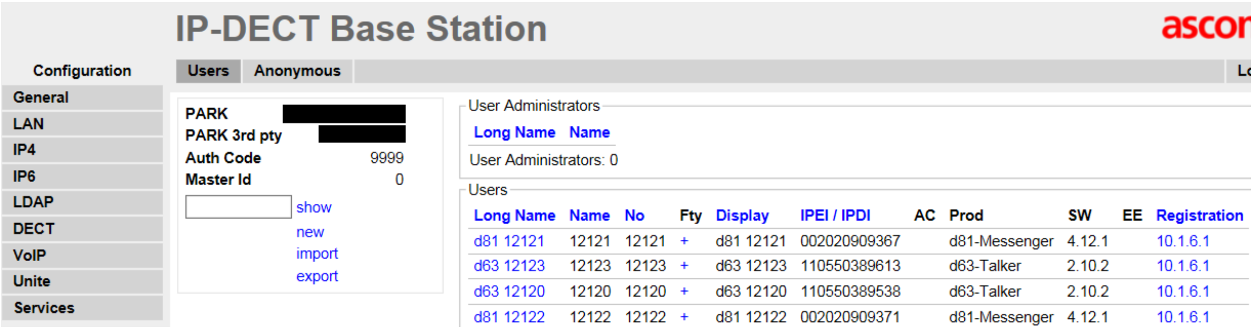
User configuration – Edit user:



Chapter 6 Appendix B: PARTNER side CONFIGURATION

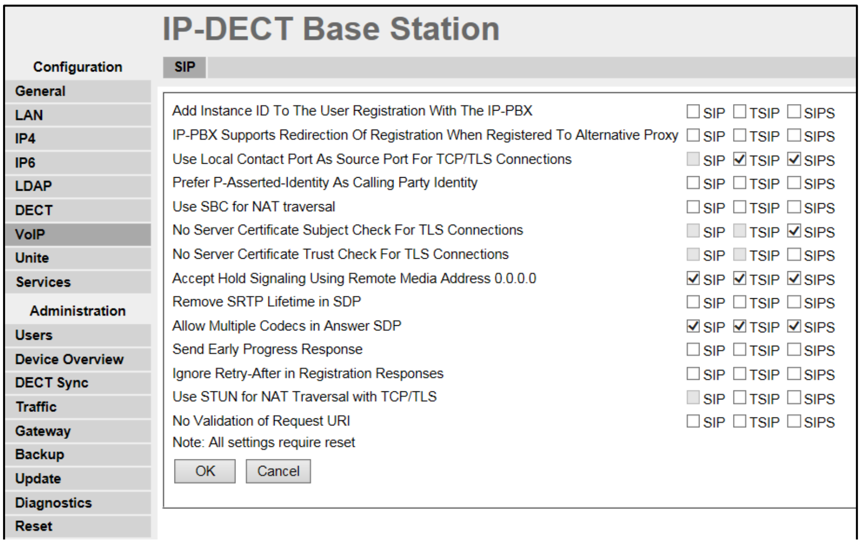
Note: Typical user configuration during these tests.

User configuration – Registration:

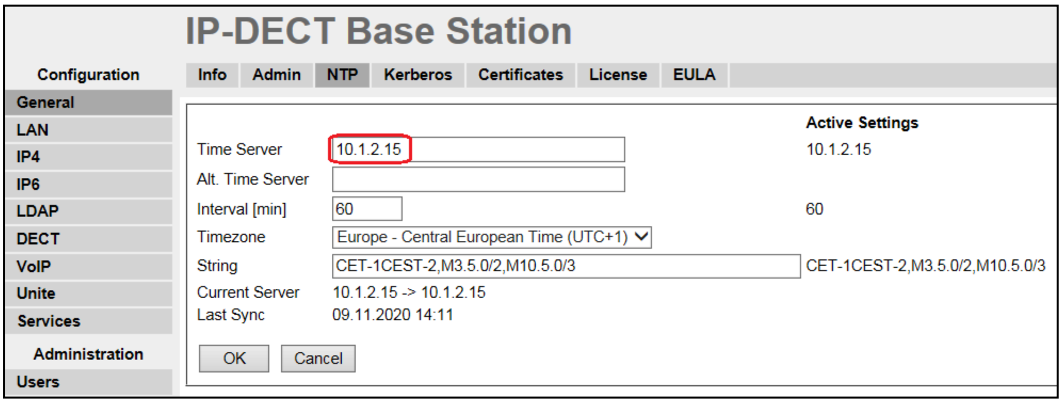


Users are registered towards OXE instance with IP address 10.1.6.1.

VOIP SIP Configuration:



NTP configuration:

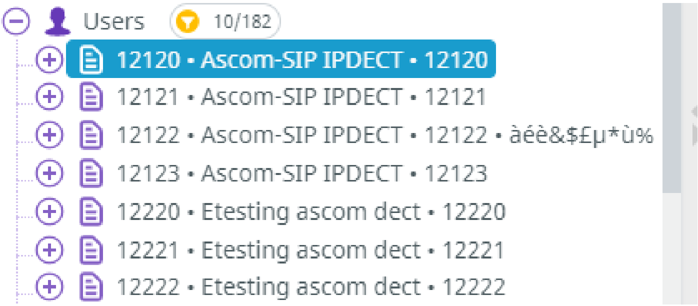


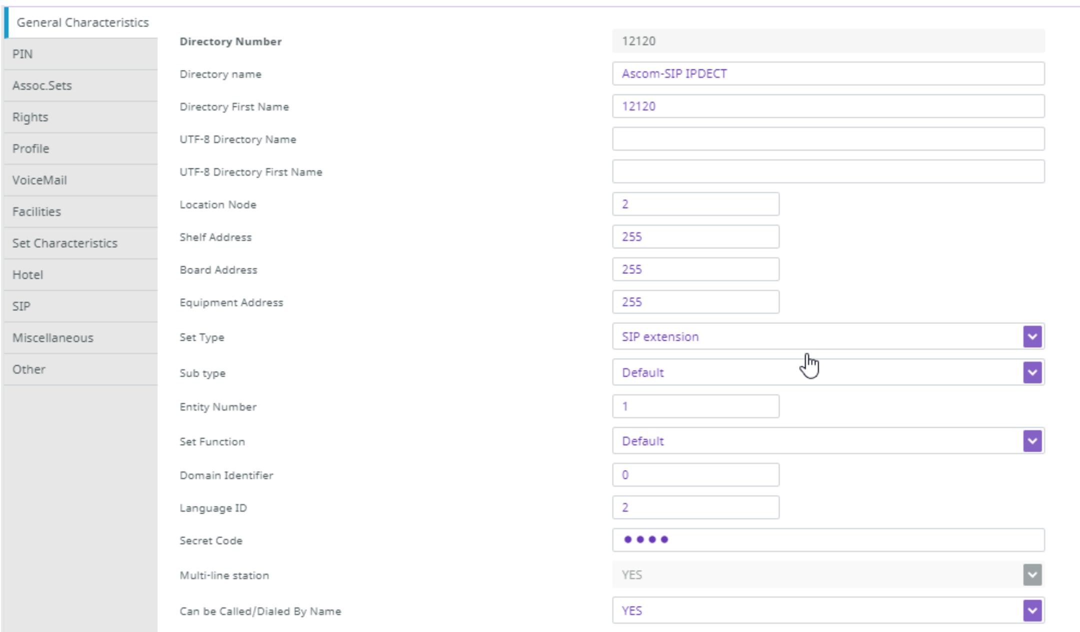
Remark: Configuration of a “Time Server” is **strongly** recommended.

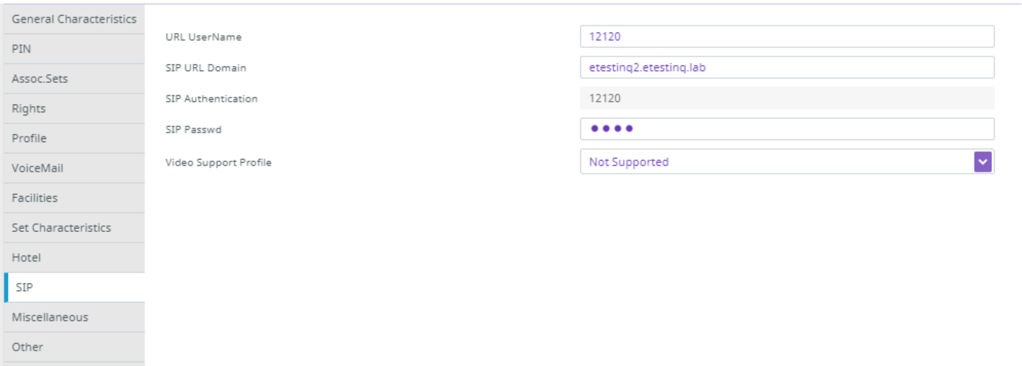
Chapter

7 Appendix C: ALE side CONFIGURATION

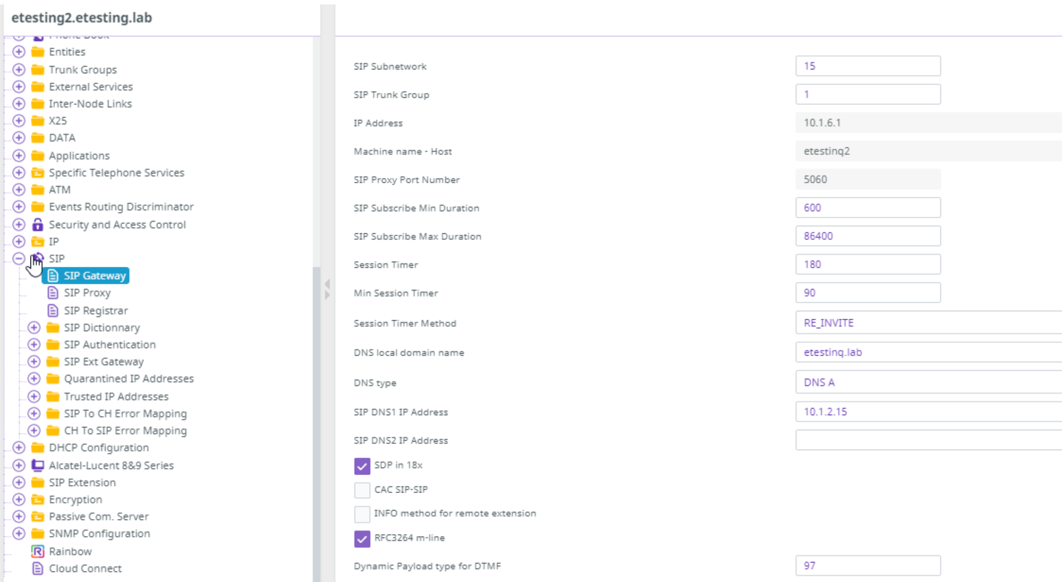
7.1 SIP Users management for DECT handsets





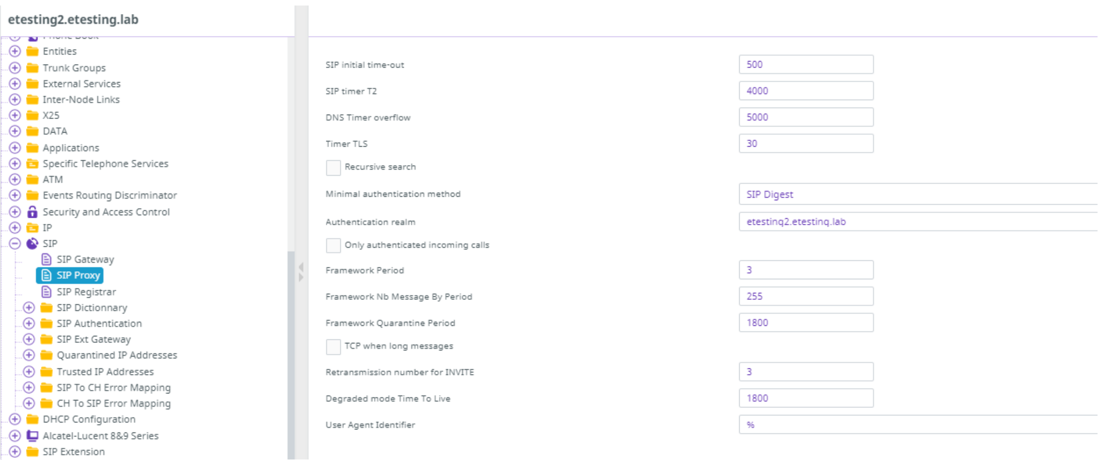


7.2 SIP Gateway management

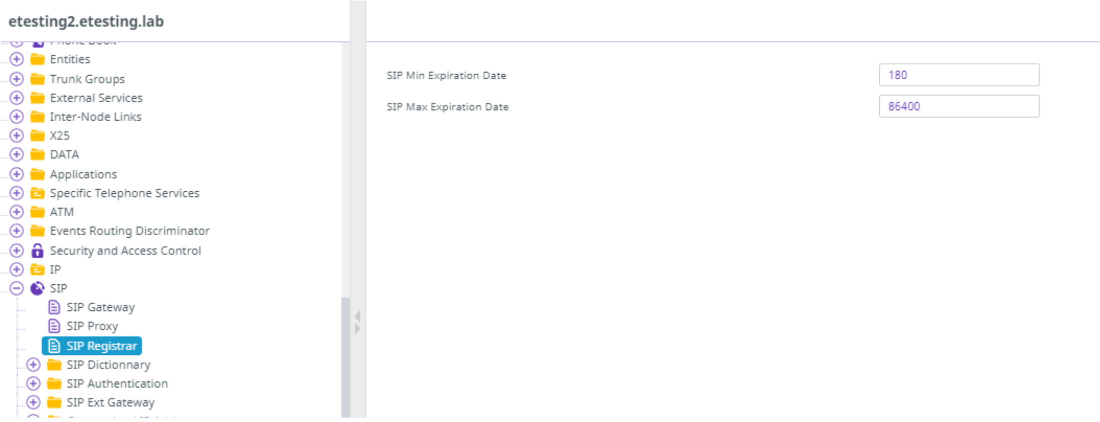


7.3 SIP Proxy management

Authentication method can be “none” or “SIP Digest”.

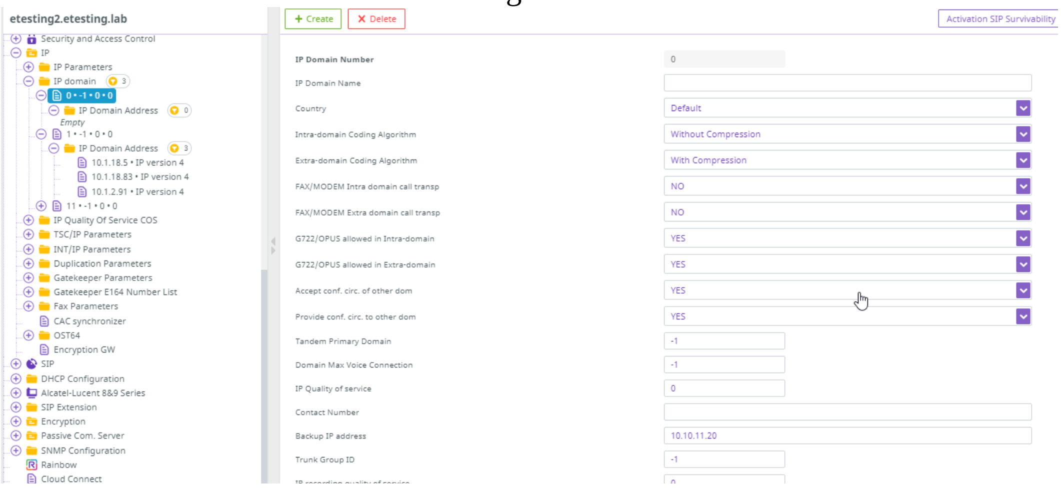


7.4 SIP Registrar timers

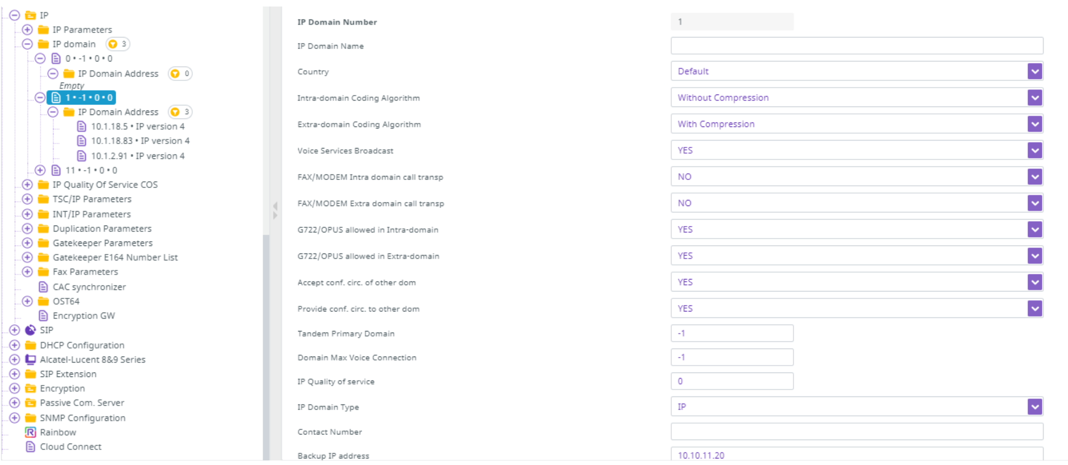


7.5 IP Domains management

Domain 0 is default where the SIP sets goes.



Domain 1 is configured with IP range of the belonging devices.



7.6 Software locks

Use spadmin command to check locks of your system.

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

Chapter

8

Appendix D: PARTNER SUPPORT 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 Australia	Simon Squire	Simon.Squire@ascom.com
Ascom Austria	Torsten Wolf	Torsten.Wolf@ascom.com
Ascom Belgium	Peter Moens	Peter.Moens@ascom.com
Ascom Denmark	Behdad Bakhshaie	Behdad.Bakhshaie@ascom.com
Ascom Finland	Mikko Hagström	Mikko.Hagstrom@ascom.com
Ascom France	Thierry MORAEL	Thierry.MORAEL@ascom.com
Ascom Germany	Torsten Wolf	Torsten.Wolf@ascom.com
Ascom Italy	Francesco Naldini	Francesco.Naldini@ascom.com
Ascom Malaysia	Hans Tysk	Hans.Tysk@ascom.com
Ascom Netherlands	Gerty Everts	Gerty.Everts@ascom.com
Ascom Norway	Håkon Storm	Hakon.Storm@ascom.com
Ascom Romania	Marko Savinainen	Marko.savinainen@ascom.com
Ascom Singapore	Hans Tysk	Hans.Tysk@ascom.com
Ascom Sweden	Niclas Holmblad	Niclas.holmblad@ascom.com
Ascom Switzerland	Torsten Wolf	Torsten.Wolf@ascom.com
Ascom United Arab Emirates	Marko Savinainen	Marko.savinainen@ascom.com
Ascom United Kingdom	Luke Blackmoore	Luke.Blackmoore@ascom.com
Ascom United States	Steven Zachary	Steven.Zachary@ascom.com
International	Marko Savinainen	Marko.savinainen@ascom.com

Chapter

9

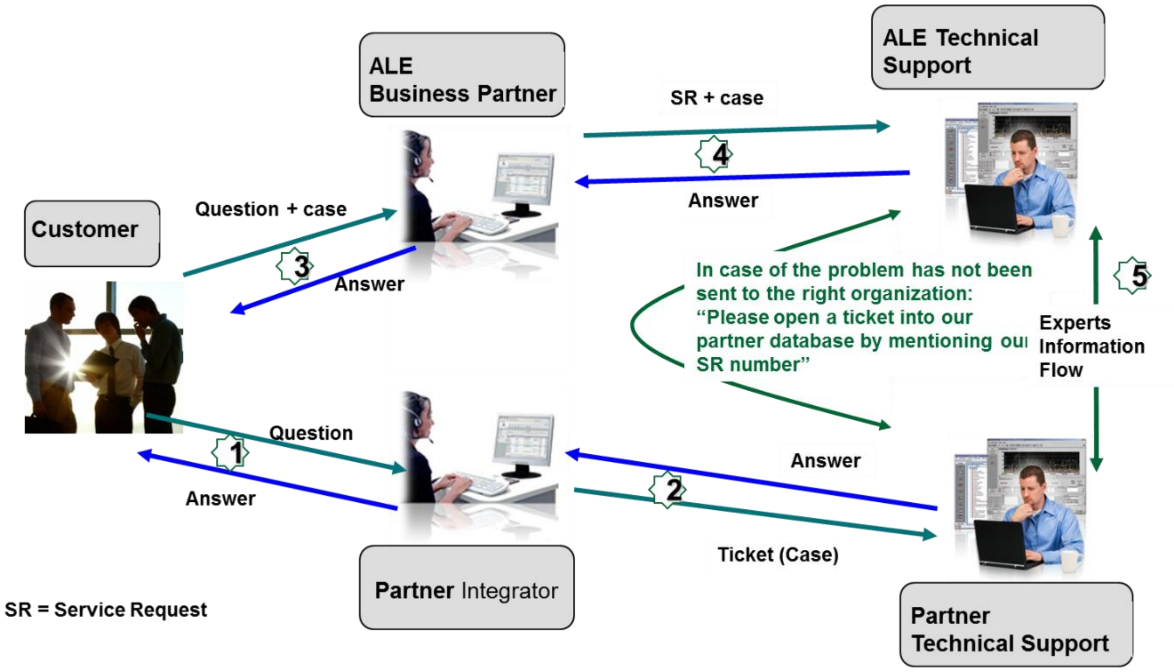
Appendix E: ALE SUPPORT PROCESS

9.1 Introduction

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

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

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



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

9.2 Escalation in case of a valid Inter-Working Report

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

This defines the scope of what has been certified.

If the issue is in the scope of the IWR, both parties, ALE and the Solution or Developer Partner, are engaged:

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

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

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

- The Solution or Developer Partner shall be contacted first by the ALE Business Partner (responsible for the application, see figure in previous page) for an analysis of the problem.

- The ALE Business Partner will escalate the problem to the ALE Support Center only if the Solution or Developer Partner has demonstrated with traces a problem on the ALE side or if the Solution or Developer Partner (not the Business Partner) needs the involvement of ALE

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

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

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

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

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

Please check the availability of the Inter-Working Report on DSPP (URL: https://www.al-enterprise.com/en/partners/dspp) or Enterprise Business Portal (Url: Enterprise Business Portal) web sites.

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

9.3 Escalation in all other cases

For non-certified solutions, no valid InterWorking Report is available and the integrator is expected to troubleshoot the issue. If the ALE Business Partner finds out the reported issue is maybe due to one of the Alcatel-Lucent Enterprise solutions, the ALE Business Partner opens a ticket with ALE Support and shares all trouble shooting information and conclusions that shows a need for ALE to analyse.

Access to technical support requires a valid ALE maintenance contract and the most recent maintenance software revision deployed on site. The resolution of those non-DSPP solutions cases is based on best effort and there is no commitment to fix or enhance the licensed Alcatel-Lucent Enterprise software.

For information, for non-certified solution and if the ALE Business Partner is not able to find out the issues, ALE offers an “On Demand Diagnostic” service where assistance will be provided for a fee.

9.4 Technical support access

The ALE **Support Center** is open 24 hours a day; 7 days a week:

- e-Support from the DSPP Web site (if registered as Solution or Developer Partner): https://www.al-enterprise.com/en/partners/dspp
- e-Support from the ALE Business Partners Web site (if registered Alcatel-Lucent Enterprise Business Partners): https://businessportal2.alcatel-lucent.com click under "Contact us" the *eService Request* link
- e-mail: Ebg_Global_Supportcenter@al-enterprise.com
- Fax number: +33(0)3 69 20 85 85
- Telephone numbers:

ALE 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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