Developer and Solution Partner Program
Inter-Working Report

Partner: **Ascom**

Solution name: **IP-DECT Solution for hospital and senior care segment**

Alcatel-Lucent Enterprise Platform: **OXO Connect**

August 2019

Legal notice

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	20 Sept 19	Thierry Chevert	Creation

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Tests Overview

Date	September, 23rd to 25th 2019
ALE representative	Thierry Chevert
Partner representative	Matthew Williams
ALE platform	OXO Connect
ALE release	R3.1
Partner solution	IP-DECT
Partner release	R10.3.5
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

Partner contact information

Name	Matthew Williams
Title	Technology Expert Interoperability
Address	Grimbodalen 2 Box 8783
Zip code	40276
City	Göteborg
Country	Sweden
Phone	+46 31 55 93 58
Fax	+46 31 55 83 13
Mobile phone	
Email address	http://www.ascom.com
Web site	matthew.williams@ascom.com

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

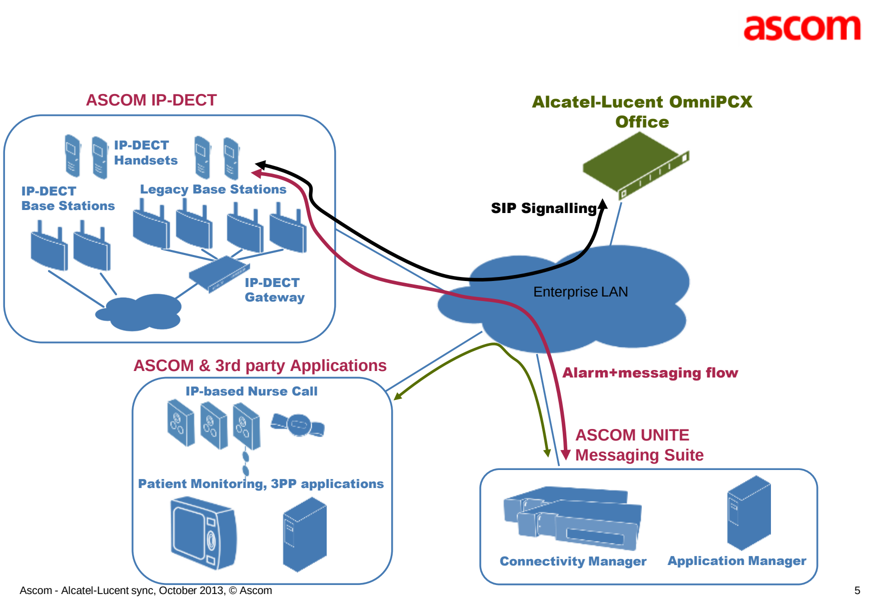
SOLUTION INFORMATION

Solution name	IP-DECT d81, d62, d63, d41, d43 hand-sets for hospital and senior care segment
Solution version	R10.3.5
Interface/API	SIP with OXO Connect / DECT with Ascom handsets
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 OXO via SIP protocol. All telephony features as provided by SIP Endpoint Level of Service are available to the handsets. The Ascom & 3rd party applications and the Ascom Unite Messaging Suite complete the solution.



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

IPBS2

IPBS1 (DECT radio only)

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

Ascom d81

Ascom d81ex

Ascom d62

Ascom d41

Ascom d43

Ascom d63

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

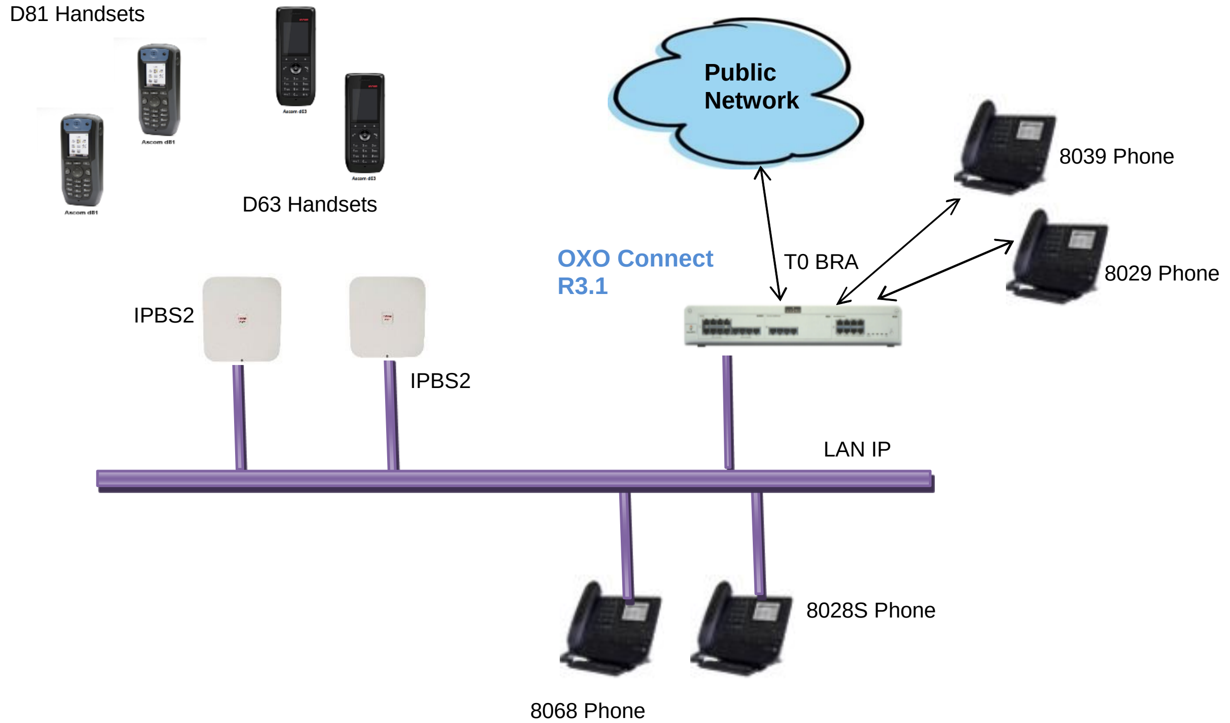


Figure 1 **Test environment**

3.1 Hardware configuration

Ascom handsets are registered in the OmniPCX Office as "Open SIP Handset".
There were2 Base Stations used for the tests of the IP-DECT. The BS are mirrored and the SIPand RTP are on one Base Station (active one).
The DECT handsets used were 2x D81 and 2x D63.

3.2 Software configuration

- **Alcatel-Lucent Communication Platform**: OXO Connect R3.1
- **Partner Application**:
  - IP-DECT IPBS2 version 10.3.5
  - Ascom handsets:
    - D63 SW version 2.4.0 - 21/01/2019
    - D81 SW version 4.7.2 15/05/2018

Chapter

4 SUMMARY OF TESTS

4.1 Summary of main functions supported

Features	Status	Comments
Initialization including network configuration	OK	
SIP registration	OK	
SIP authentication	OK	
Voice over IP and RTP codec support	OK	
Outgoing Call	OK	
Incoming Call	OK	
Features During Conversation	OK	
Call Transfer	OK	
Attendant	OK	
Voice mail interaction and indication	OKBut	Issue found with Message Waiting Indication see 4.2 .

4.2 Summary of problems

- Issue found with MWI (Message Waiting Indication) on OXO connect, while deleting the voice messages accumulated in the voice mail box the OXO system sends "SIP Notify" with updated information:

```
Message Body
      Messages-Waiting: no\r\n
      voice-message: 1/0\r\n
```

This issue is still under analysis by ALE R&D as case # 00393372.

4.3 Summary of limitations

- None

4.4 Notes, remarks

- None

Chapter

5 TESTS RESULT

5.1 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 and the expected result
N/A: when checked, means the test case is not applicable in the scope of the application
OK: when checked, means the test case performs as expected
NOK: when checked, means the test case has failed. In that case, describe in the field "Comment" the reason for the failure and the reference number of the issue either on ALE side or on partner side
Comment: to be filled in with any relevant comment. Mandatory in case a test has failed especially the reference number of the issue.

5.2 Test results

5.2.1 Connectivity and Setup

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

Management of the IP-DECT and DECT Handset is done via the Master Base Station by connecting to http://IP@ofBS.

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Ascom Base Station Connectivity** Configure the Base Stations (mirrored as Active/Master, Non-Active/Standby Master) with Static IP Address, Subnet mask, gateway and DNS.	☐	☒	☐	Web based console to configure the Base Station and the attached SIP Handsets.
2	**Ascom Base Stations interaction with OXO IP address** Configure the Base Stations Master with IP PBX Proxy set to IP address of OXO.	☐	☒	☐	Use of OXO IP address for registrar on port 5059.
3	**Ascom Base Stations interaction with OXO Connect Name** Configure the Base Stations Master with IP PBX Proxy set to FQDN of OXO. Check DNS entry for this FQDN.	☐	☒	☐	Use of OXO FQDN and DNS server for registration on port 5059.
4	**Ascom handset SIP Registration to OXO Connect with correct passwords** Configure your SIP sets MCDU number on the OXO as 220, 221 and 222 and register with the OXO IP address Check the registration on your sets and the display	☐	☒	☐	SIP Password have to be entered into base station for all handsets. All SIP Handsets register OK. All SIP Handsets registered with IP address of Master Base Station when active.
5	**Ascom handset SIP registration to OXO with wrong SIP Password** For this test, we will try to register the SIP Handset with authentication enabled. And a wrong SIP password. Check the phone registration and display. Redo the same test on one SIP Handset with a wrong password and check that the phone is rejected.	☐	☒	☐	SIP Handset with wrong password rejected ("Failed" in GUI on active IP-DECT base station)

Test Case Id	Test Case	N/A	OK	NOK	Comment
6	**DHCP registration with OXO Connect internal DHCP server.**	☒	☐	☐	It is **strongly recommended by Ascom** that the IP-DECT master Base Station is configured with a Static IP address into the LAN subnet.
7	**NTP registration and synchronization** The Ascom Base Stations are configure to register for NTP to OXO Connect IP address. Ascom handsets are getting the date & time from base station synchronized with OXO connect. configured to retrieve the date and time from the OXO IP address.	☐	☒	☐	Correct time displayed on SIP Handset.
8	**Support of "423 Interval Too Brief" (1)** The SIP Handsets are configured with a value lower than 120 seconds, 60 seconds for instance. Check the phones registration and phones operation	☐	☒	☐	Register negotiates 120s, while subscribe will use 60s.
9	**SIP authentication method** **OXO connect has its IP address configured as Realm.** OxO connect will send 401 unauthorized for messages without authentication.	☐	☒	☐	
10	**SIP Sets Signaling UDP** If applicable configure your SIP set xxx to use the protocol SIP over UDP Check the registration and basic calls (RTP are still with UDP).	☐	☒	☐	Change done on Master Base Station and require a restart of this Base Station. (SIP over UDP used during testing)
11	**SIP Sets Signaling TCP** If applicable configure your SIP set xxx to use the protocol SIP over TCP. Check the registration and basic calls (RTP are still with UDP).	☐	☒	☐	Change done on Master Base Station and require a restart of this Base Station. (SIP over UDP used during testing)

5.2.2 Audio codec negotiations and Voice Activity Detection

Voice Activity Detection VAD is managed and Silence Suppression into the SIP SDP.
These tests check that the phones are using the configured audio parameters (codec, VAD).

Phone configuration: configure IP-DECT to use G.722.2 (G722 not support by this Base Station), G.711 A-law, G.711 mu-law, G.729, G.723 in this order (unless otherwise stated). Configure IP Touch with codec G.711 and to NOT use VAD (unless otherwise stated).

In the tests, we'll use "**Open SIP Handset**" for the DECT Handset connected to OXO via the Master IP-DECT Base Station.

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Select G711 A-law as 1st codec in IP-DECT** Call from Dect handset SIP Handset to Premium deskphone Check that the call is established in G711 A-law. Check audio quality Call from IP Touch to Dect Handset SIP Handset Check that the call is established in G711 A-law. Check audio quality	☐	☒	☐	
2	**Select G729 as 1st codec in IP-DECT** Call from SIP Handset to IP Touch Check that the call is established in G729 Check audio quality Call from IP Touch to SIPPhone Check that the call is established in G729 Check audio quality	☐	☒	☐	
3	**Select G723 as 1st codec in IP-DECT** Check that the call is established in G723 Check audio quality Call from IP Touch xxx to SIP xxx Check that the call is established in G723 Check audio quality	☒	☐	☐	Not available for Open SIP Handsets in OXO Connect.
4	**Select G722.2 as 1st codec – Call with OXO IP Premium Phone with G722 enabled.** Check that the call is established in G722 Check audio quality Call from IP Touch xxx to SIP xxx Check that the call is established in G723 Check audio quality	☒	☐	☐	G722.2 is not available in OXO Connect for users but can be pass-through.

5	**Select G722.2 as 1st codec – Call between 2 Ascom handsets – CODEC pass-through for SIP Handsets inhibited** Check that the call could not be established in G722.2 Check audio quality	☐	☒	☒	OXO will not be transparent for SDP and remove G722.2, Media Comfort Noise and Silence Suppression attribute. Call won't be established in G722.2 between 2 SIP Handsets, instead G.711A used.
6	**Select G722.2 as 1st codec – Call between 2 Ascom handsets – CODEC pass-through for SIP Handsets validated** Check that the call is established in G722.2 Check audio quality	☐	☒	☐	The SDP information are not changed by OXO and remain the same. G722.2 is seen as AMR-WB into BS configuration.
7	**Configure Ascom handset to use VAD** **Configure Premium Deskphone to use VAD** Call from SIP Handset to IP Touch Check that the call is established in G711 A-law. Check audio quality	☒	☐	☐	See Nota 1
8	**Configure Ascom handset to use VAD** **Configure Premium Deskphone NOT to use VAD** Call from SIP Handset to IP Touch Check that the call is established in G711 A-law. Check audio quality	☒	☐	☐	See Nota 1
9	**Configure Ascom handset NOT to use VAD** **Configure Premium Deskphone to use VAD** Call from SIP Handset to IP Touch Check that the call is established in G711 A-law. Check audio quality	☒	☐	☐	See Nota 1

Nota 1:

- OXO Phones 80x8 do not handle the VAD for G711 but only for G722 and G729.
- OXO phones 40x8 do handle the G711 with VAD into configuration OMC but then into the SIP-SDP there is no information related to silence suppression (either while called or calling).
- The Ascom handsets d81 and d63 can be configured for “Silence Compression” and they handle the SIP-SDP information (Media attribute: SilenceSupp: on) for this feature but only with Codec G711 / PCM.
- OXO let calls with unsupported codec while not in Pass-through mode because it check the presence of G711 codec (to allow Voice prompt

5.2.3 Outgoing Calls – Ascom Handsets → OXO Connect Phones

The calls are generated to several users belonging to the same network.
Called party can be in different states: free, busy, out of service, do not disturb, etc.
Calls to data devices are refused.
Points to be checked: tones, voice during the conversation, display (on caller and called party), hang-up phase.
OXO prefixes are mandatory for several tests of this section. For more information refer to the expert doc.

Note: dialing will be based on direct dialing number.

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Call to local user with answer** With Ascom Dect Handset call the Premium Deskphone. Check that is ringing and handset get ring-back tone. Take the call and check audio and display.	☐	☒	☐	
2	**Call to local user without answer** With Ascom handset call the premium Deskphone. And never take the call. Check time out (if any) and display. Note that 211 don't have a Voice Mail	☐	☒	☐	Diverted to Voice Mail after 15s (Expected: Local user is ringing until SIP Handset releases the call – No Time out.)
3	**Call to another Ascom DECT handset** With the SIP Handset call the other SIP Handset Check the display and audio during all steps (dialing, ring back tone, conversation, and release).	☐	☒	☐	Name display is correct between 2 Ascom handsets.
4	**Call to wrong number** With the SIP Handset call a wrong number Check the ring back tone and display	☐	☒	☐	OXO Connect sends 404 not found. Ascom set shows "Vacant" on display, got local tone and get released after 30 secondes..
5a	**Ascom handset Call to busy OXO user** **OXO connect configuration with no protection for camp on and camp on allowed** With the Ascom handset call Premium Deskphone, take the call and don't hang up. With other Ascom handset call the same oxo user which is busy. Check the ring back tone and display (free)	☐	☒	☐	Called party is busy in a call and second caller is waiting. Called party handset can take the waiting call (R + 1) and switch over (back and forth) between first and second

Test Case Id	Test Case	N/A	OK	NOK	Comment
5b	**Ascom handset Call to busy OXO user** **OXO connect configuration with protection for camp on** The call waiting feature is up to SIP Handset behavior. In this case the SIP Handset is configured with “call waiting”	☐	☒	☐	OXO connect will always send 486 Busy Here to second call.
6	**Ascom handsetCall to Out of Service OXO user** Put an OXO connect premium Deskphone “Out of Service” from OMC (Logically out of service) With the SIP Handset call the premium Deskphone which is in “Out of Service State” Check the display and tones.	☐	☒	☐	If OXO user is an IP Phone 8038 then OXO send 486-Busy Here. Handset displays “Busy” If OXO user is digital phone 8039 then OXO send 480 Not available. Handset displays “Not Reachable”.
7	**Call to OXO user in “Do not Disturb” (DND) state** Dial “793” on the premium Deskphone in order to enable the DND. Wait for acknowledgement ring back tone from OXO. With the SIP Handset call this premium Deskphone Check ring back tone and display. Redial 790 on OXO Phone to cancel the DND	☐	☒	☐	OXO sends 480- Temporarily unavailable Display on SIP Handset is: “Not Reachable”.
8	**Call to local user with immediate forward (CFU).** On premium Deskphone activate the CFU. Wait for acknowledgement ring back tone from OXO. With the Dect handset call the premium Deskphone with forwarding. Check that destination is ringing and the display. Take the call check audio and hung up.	☐	☒	☐	
9	**Call to local user, forward on no reply (CFNR).** On premium Deskphone configure with OMC the CFNR using dynamic routing to other station for local calls. Ascom handset call the oxo user. Check that destination is ringing but don’t take the call and wait the time out. Time out is defined in xxx dynamic routing of Timer 1. After time out check that destination is ringing and take the call. Check the audio and display.	☐	☒	☐	Timer is 12 sec Tested also with forward to other SIP Handset

Test Case Id	Test Case	N/A	OK	NOK	Comment
10	**Call to local user, forward on busy (CFB).** On premium Deskphone dial the prefix or use dynamic keys to activate the CFB. Wait for acknowledgement ring back tone from OXO. With SIP Handset call premium Deskphone and take the call to make it busy. With other SIP Handset call the busy premium Deskphone. Check that destination is ringing and take the call. Check the audio and display.	☐	☒	☐	
11	**Call to external number** **(Check ring back tone, called party display)** With Ascom handset dial 0 (0 prefix +external number) Take the call and check audio, display and call release.	☐	☒	☐	

5.2.4 SIP Session Timer

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**SIP session timer update** Check if call is maintained or released after the session timer has expired Make call between oxo user and SIP Handset. Wait for time-out expiration. Check that call is maintained.	☐	☐	☐	*OXO send "timer" and session expires : 1800 into the Invite then SIP Handset sends session-expires into 200-OK.* *Update sent by OXO to SIP Handset every 15min (900 sec, half timer 1800 sec)*

5.2.5 Incoming Calls and forwarding

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.

Network calls are made using SIP private trunk established between two OXO's.
OXO prefixes are mandatory for several tests of this section. For more information refer to the appendix C.

SIP Handset stands for DECT Handset behind the IP-DECT Base Station.

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**OXO user call to free Ascom handset** with premium Deskphone call Ascom handset Check that Ascom handset is ringing and take the call Check ring back tone and called party display.	☐	☒	☐	
2	**OXO user call to busy Ascom handset** With SIP set 220 call other SIP set 222 and take the call to make it busy, don't hang up. With IP Touch 211 call 220 which is busy Check the ring back tone and display. Check ring back tone and called party display.	☐	☒	☐	
3a	**Local call to unplugged Battery Ascom handset** Unplug the battery of an Handset and call it from OXO user. Check the tone and display	☐	☒	☐	If handset still registered then caller hear ring-back tone and Base Station sends: "503- service unavailable" after time. Display on OXO phone: "Release" If SIP Handset unregistered then OXO release the call and do not send Invite to BS.

Test Case Id	Test Case	N/A	OK	NOK	Comment
3b	**Local call to Ascom handset that is Powered-Off** Power-Off the Ascom handset and call it with IP Touch. Check the tone and display	☐	☒	☐	As handset is unregistered is SIP then OXO give direct answer to the OXO user “Unobtainable”. If calls come from another handset then OXO send 486 – busy here to Base Station and SIP handset.
4A	**Call to Ascom handset in Do Not Disturb (DND) mode - By local feature of SIP Handset if applicable.** Enable DND (*42#) on SIP Handset and call it with Premium deskphone. Check the tone and display Cancel the DND (#42#) on.	☐	☒	☐	”Released” is displayed on OXO phone
4B	**Call to Ascom handset in Do Not Disturb (DND) mode - By system feature.** Enable DND on SIP Handset using the 793 prefix. Wait for acknowledgement ring back tone from OXO. With IP Touch call The SIP Handset in DND. Check the tone and display Cancel the DND on SIP Handset using 790 prefix.	☒	☐	☐	Not possible in handset
5A	**Local SIP call to Ascom handset in immediate forward (CFU) to local user** By local feature if applicable: On **Ascom handset** enable CFU to OXO user Place a call to handset forwarded. Check that call follow the forwarding. Take the call and check audio and display.	☐	☒	☐	Local CFU feature was done with combination of keys on Ascom handset yyy with *21*xxx# (xxx is destination). Display is: “yyy > xxx”

Test Case Id	Test Case	N/A	OK	NOK	Comment
5B	**SIP call to Ascom handset in immediate forward (CFU) to local user - By system feature** On **Ascom handset** enable CFU toOXO user using 791 + <target MCDU number>. Wait for acknowledgement ring back tone from OXO. With another handset call this forwarded Ascom handset. Check that Destination is ringing. Take the call and check audio and display. Disable CFU on SIP Handset using 790 prefix..	☐	☒	☐	No display of Forwarding on the Ascom handset. On oxo user called party we'll see name of fwd extension ← name of caller during call.
6A	Local call to **Ascom handset** in forward on busy (CFB) state - By local feature if applicable On **Ascom handset** enable CFB to 0X0 user. With this handset call another user to make it busy. With oxo user call the handset which is busy and cfb. Check that destination of forward is ringing Take the call and check audio and display. Disable CFU using #67# prefix.	☐	☒	☐	*67*xxx# Test case passes, but when SIP set is in call with OXO invite not sent to BS (feature fails). The handset shows information about forwarding.
6B	**Local call to SIP Handset in "forward on busy" (CFB) state - By system feature** On Ascom handset enable CFB to one oxo user using 792 + <target MCDU number>. Wait for acknowledgement tone from OXO. With oxo user call This CFB SIP Handset. Check that destination is ringing. Take the call and check audio and display. Disable CFU using 790 prefix.	☐	☒	☐	It works when call is initiated by OXO user or if done from another Ascom handset. There is no indication of forwarding on the handset.

Test Case Id	Test Case	N/A	OK	NOK	Comment
7A	**Call to Ascom handset in “forward on no reply” (CFNR) - By local feature if applicable** On enable CFNR on handset to oxo user. With oxo user call this handset. Check that handset is ringing and don’t take the call, wait for time out (about 30 seconds). After time-out expiration the oxo user is ringing, take the call and check audio and display. To cancel uses #61#	☐	☒	☐	*61*xxx# Base Station will send to OXO a 302 – Moved Temporarily.
7B	**Call to Ascom handset in “forward on no reply” (CFNR)** - By system feature CNFR via prefix not available on OXO dynamic routing has to be used.	☐	☒	☐	It rings on the requested handset then after 12 sec (timer) it rings also on the routing number (can be oxo user or another handset).
8a	Ascom handset Call to busy Ascom handset OXO connect configuration with no protection for camp on and camp on allowed Ascom BS set with call waiting on With the Ascom handset call IP Touch, take the call and don’t hang up. With other Ascom handset call the first Ascom handset which is busy. Check the ring back tone and display (free)	☐	☒	☐	Called party is busy in a call and second caller is waiting. Called party handset can take the waiting call (R + 1) and switch over (back and forth) between first and second
8b	Ascom handset Call to busy Ascom handset OXO connect configuration with no protection for camp on and camp on allowed Ascom BS set with call waiting off With the SIP Handset call IP Touch, take the call and don’t hang up. With other SIP Handset call this SIP Handset which is busy Check the ring back tone and display.	☐	☒	☐	Ascom Base Station will send the 486 Busy Here

Test Case Id	Test Case	N/A	OK	NOK	Comment
8c	Ascom handset Call to busy Ascom handset OXO connect configuration with protection for camp on and camp on disabled Ascom BS set with call waiting off The call waiting feature is up to SIP Handset behavior. In this case the SIP Handset is configured with “call waiting”	☐	☒	☐	OXO connect will always send 486 Busy Here to second call.
8d	Ascom handset Call to busy Ascom handset OXO connect configuration with no protection for camp on and camp on allowed Ascom BS set with call waiting on Handset is busy in a call then another call is waiting on. Try a third call. Check the behavior	☐	☒	☐	The third call is waiting and called party can switch between second and third call. The first call remain on-hold. When third call release then called party can switch between first and second.
9	**Calling Line Identity Restriction (CLIR): Local call to Ascom handset.** On premium deskphone enable mask Identity (from OMC subscriber features part1: Identity Secrecy) and call handset in order to hide identity. Check that handset is ringing, take the call and check that caller identity is hidden. In OMC do Ascom handset Identity masking and call premium deskphone to test identity masking. Check that called party is ringing, take the call and check that caller handset identity is hidden.	☐	☒	☐	Manage Identity secrecy into user features. SIP Handset will display “************” and call seen as “external call” and ring with external ring melody.
10	**Display: Call to free Ascom handset from oxo phone with a name containing non-ASCII characters (eg éëêèè).** Check caller display. Check that Handset Ascom is ringing and check on its display that the characters are correctly printed.	☐	☒	☐	User name managed has a name: **àéöäß$IP8028** Display is correct on Ascom handset. Issue on handset display when calling-back from call log.

Test Case Id	Test Case	N/A	OK	NOK	Comment
11	**Display: Call from OXO Phone to Ascom handset which has the name containing non-ASCII characters, eg &@(#+)=.** Check caller display. Check that SIP set is ringing and check that the characters are correctly printed.	☐	☒	☐	Name displayed, is the one configured by OXO. Ascom sends an empty display info.

Nota: OXO does not provide music on-hold internally and sets do generate a cadenced tone. The music on-hold is used only for external calls.

5.2.6 Public external calls

Office System is equipped with BRA interfaces and an ISDN T0 Basic access is connected to the system.
Trunk group is managed and prefix for seizure is “0”.

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Public Incoming External call to Ascom handset** Manage a DDI number routing to SIP terminal. Place the call from external user to this DDI. Check that SIP terminal is ringing and the external number is shown correctly Take the call and check audio, display and call release.	☐	☒	☐	
2	**Public External Call-Back from Ascom handset** Check the previous call into call log and call-back from SIP terminal. Check that external called party is ringing. Take the call and check audio, display and call release.	☐	☒	☐	Public Prefix is 0 and into the call log in front of Public number.
3	**Public Outgoing External call from Ascom SIP handset** From SIP Handset, dial 0 + public number. Check tones and voice quality.	☐	☒	☐	

5.2.7 Hunting group tests

There are 2 types of Hunting Groups: Cyclic and Sequential.

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Ascom handset is part of a sequential hunting group** Call to hunt group. Check call/release. With Premium Deskphone call the **sequential hunting group** number. Check that handset is ringing Take the call and hang-up And with same phone call the sequential hunt group MCDU number. Check that same first group handset is ringing.	☐	☒	☐	Sequential hunting group is number 501 and has 221 and 222
2	**Ascom handset is part of a cyclic hunting group** Call to hunt group. Check call/release. With Premium Deskphone call the **cyclic hunting group** number. Check that first handset is ringing Take the call and hang-up And with same phone call the sequential hunt group MCDU number. Check that second handset of group is ringing.	☐	☒	☐	Cyclic hunting group is number 502 and has 221and 222

Notes:

- Sequential Hunt Group behavior: the endpoint n+1 is ringing **only** if the endpoint n is now in call (busy).
- Cyclic Hunt Group behavior: the endpoint n+1 is ringing if previously the endpoint n has been reached (ringing only or in call). The actual state of the n endpoint doesn’t matter.

5.2.8 MultiSet configuration

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Ascom handset is declared as a MultiSet.** Manage the secondary sets into Main Set. Call to main set and see if twin set rings. Take call with twin set. With one other oxo user call main extension number which is in MultiSet with handset. Check that both are ringing. Take the call from handset and check that main phone stop ringing. Check audio and display.	☐	☒	☐	While Secondary call, it sends num-ber of Primary set and not its own number.

5.2.9 Features during Conversation

Features during conversation between local user and SIP user must be checked.
Check that right tones are generated on the SIP Handset. A multiline SIP set is mandatory for tests 2, 3, 4 and 8.
OXO prefixes are mandatory for several tests of this section. For more information refer to the expert doc.

Test Case Id	Test Case	N/A	OK	NOK	Comment
1A	**Hold and resume with local feature** With Ascom handset call oxo phone and answer call Check audio and display. Now handset put on hold the phone check tones and display on both and resume the call. Keep this call for the next test.	☐	☒	☐	OXO does not provide Music on Hold. SIP Handset uses Rxxx to make enquiry call to xxx.
1B	**Enquiry call to another handset** (if applicable) **– Use of Local Feature** With SIP Handset1 call other SIP Handset2 that will answer. OXO Phone1 will be put on hold when SIP Handset1 make second call to SIP Handset2 Put SIP Handset2 on hold and check tones and display on both. Keep these two calls for the next test.	☐	☒	☐	SIP Handset1 dial R+2 to make enquiry call. Display “on hold” on SIP Hand-set2, “Please wait” on OXO
1C	**Broker request, toggle back and forth between both lines with local feature** (if applicable) With handset 1 switch between phone and handset 2. Check the tones and display on sets on hold state. Keep these two calls for the next test.	☐	☒	☐	SIP Handset1 uses R+2 to make Broker call (back & forth).
1D	There will be two active calls from the previous test case execution. In that release the first call. Keep second call. Hang up 140 and only 138 and 139 are in call Check that 138 & 139 are still in a call, check display.	☐	☒	☐	
2	**Same tests with all 3 Ascom SIP handsets.** Repeat the tests 1A to 1D still using handset call features (suffix)	☒	☒	☐	Hold, Enquiry call, and Broker call features are not supported within call server for SIP Hand-sets.

Test Case Id	Test Case	N/A	OK	NOK	Comment
3	**Three party conferences initiated by OXO phone** With OXO Phone1 call SIP Handset1, take the call and don't release it. With OXO Phone1 call SIP Handset2, take the call and don't release it too. With OXO Phone1 start a conference. Check that all sets are in the conference. Check audio and display.	☐	☒	☐	SIP device display is updated (previously not updated) when OXO users initiate the conference
4A	**Three party conferences initiated by Ascom handset with local Ascom features** With ascom handset 1 call oxo phone take the call and don't release it. With ascom handset 1 call ascom handset 2, take the call and don't release it too. With ascom handset 1 start a conference by the local feature Check that all phones are in conference. Check audio and display.	☒	☐	☐	No Conference local feature into Ascom DECT handsets.
4B	**Three party conferences initiated from Ascom handset with OXO features.** With xxx call xxx take the call and don't release it. With xxx call xxx, take the call and don't release it too. With xxx start a conference by the OXO suffix code.	☒	☐	☐	Conference feature is not supported within call server for SIP devices with OXO suffix
5	**Meet Me conference** With oxo phone call the Meet me Conference bridge dialing prefix 77 and follow instruction to open the bride (enter your number then your password). Ascom handset 1 join the conference bridge by dialing prefix 76 and enter access code. With Ext join the conference bridge by dialing prefix and enter access code. Check that 138, 139 and Ext are in conference.	☒	☐	☐	Not tested.

5.2.10 Call Transfers

During the consultation call step, the transfer service can be requested and must be tested. Several transfer services exist: Unattended Transfer, Semi-Attended Transfer and Attended Transfer.
Audio, tones and display must be checked.
We use the following scenario, terminology and notation:
There are three actors in a given transfer event:

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

There are three sorts of transfers in the SIP world:

- ***Unattended Transfer*** or *Blind transfer* : The Transferor provides the Transfer Target's contact to the Transferee. The Transferee attempts to establish a session using that contact and reports the results of that attempt to the Transferor.

- ***Semi-Attended Transfer*** or *Transfer on ringing*:

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

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

Note: Unattended transfer (Blind transfer) is not supported by OmniPCX Office system.

In the below table, SIP means a partner SIP set, OXO means a proprietary OXO (Z/UA/IP) set, Ext. Call means an External Call, ISDN for example.

Summary of Attended transfers results:

Test	Action			Result	Comment
	A	B	C		
	Transferee	Transferor	Transfer Target		
1	OXO	SIP Handset	OXO	OK	
2	Ext Call	SIP Handset	OXO	OK	
3	Ext Call	SIP Handset	Ext Call	OK	
4	SIP	SIP	SIP	OK	
5	SIP	OXO	OXO	OK	
6	Ext Call	OXO	SIP	OK	
7	SIP	OXO	SIP	OK	
8	OXO	SIP	SIP	OK	

Summary of Semi-Attended transfers results:

The REFER is sent by Base Station only after it received the 200-OK (answer of last call party). The caller never receive the ring-back tone after transfer initiated.

Test	Action			Result	Comment
	A	B	C		
	Transferee	Transferor	Transfer Target		
1	OXO	SIP	OXO	**OK**	
2	Ext Call	SIP	OXO	**OK**	
3	Ext Call	SIP	Ext Call	**NA**	Not tested
4	SIP	SIP	SIP	**OK**	
5	SIP	OXO	OXO	**OK**	
6	Ext Call	OXO	SIP	**OK**	
7	SIP	OXO	SIP	**OK**	
8	OXO	SIP	SIP	**OK**	

Test cases:

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**SIP Handset call to OXO Phone1 and be transferred by OXO Phone1 to OXO Phone2 after answer – Attended transfer.**	☐	☒	☐	
2	**SIP Handset call to OXO Phone1and be transferred by OXO Phone1 to OXO Phone2 on ringing – Semi-Attended transfer.**	☐	☒	☐	
3	**SIP Handset call to OXO Phone1and be transferred by OXO Phone1 to OXO Phone2 directly – Unattended transfer.**	☒	☐	☐	Not possible to Blind transfer on Oxo Users.
4	**OXO Phone1 call to SIP Handset1 and be transferred by SIP Handset1 to OXO Phone2 after answer – Attended transfer.**	☐	☒	☐	SIP Handset in conversation with second user, uses R+4 to transfer.
5	**OXO Phone1 call to SIP Handset1 and be transferred by SIP Handset1 to OXO Phone2 – Semi-Attended transfer.**	☐	☒	☐	Dial R+4 while it is ringing. The REFER is sent by Base Station only after it received the 200-OK (answer of last call party). The caller never receive

Test Case Id	Test Case	N/A	OK	NOK	Comment
					the ring-back tone after transfer initiated.
6	**OXO Phone1 call to SIP Handset1 and be transferred by SIP Handset1 to OXO Phone2 – Unattended transfer.**	☒	☐	☐	Oxo answered with 403-Forbidden to Refer message from SIP Handset. Initial call is maintained.
7	**Public External call to SIP Handset and be transferred to OXO Phone1 – Attended transfer**	☐	☒	☐	
8	**Public External call to SIP Handset and be transferred to OXO Phone1 – Semi-Attended transfer**	☐	☒	☐	The REFER is sent by Base Station only after it received the 200-OK (answer of last call party). The caller never receive the ring-back tone after transfer initiated.
9	**Public External call to SIP Handset and be transferred to OXO Phone1 – Unattended transfer**	☒	☐	☐	Oxo answered with 403-Forbidden to Refer message from SIP Handset. Initial call is maintained.
10	**SIP Handset1 call to SIP Handset2 and be transferred to other OXO Phone1 – Attended transfer**	☐	☒	☐	
11	**SIP Handset1 call to SIP Handset2 and be transferred to other OXO Phone1 – Semi-Attended transfer**	☐	☒	☐	OXO updated of A after pickup (same behavior as DECT). The REFER is sent by Base Station only after it received the 200-OK (answer of last call party). The caller never receive ring-back tone after transfer initiated.
12	**SIP Handset1 call to SIP Handset2 and be transferred to other OXO Phone1 – Unattended transfer**	☒	☐	☐	Oxo answered with 403-Forbidden to Refer message from SIP Handset. Initial call is maintained.
13	**SIP Handset1 call to SIP Handset2 and be transferred to SIP Handset3 – Attended transfer**	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
14	**SIP Handset1 call to SIP Handset2 and be transferred to SIP Handset3 – Semi-Attended transfer**	☐	☒	☐	SIP set C updated of A after pickup (Previously NOK: After transfer, the target hang-up and transferee is on-hold.)
15	**SIP Handset1 call to SIP Handset2 and be transferred to SIP Handset3 – Unattended transfer**	☒	☐	☐	Oxo answered with 403-Forbidden to Refer message from SIP Handset. Initial call is maintained.
16	**Public External call to SIP Handset and be transferred to other Public External – Attended transfer** SIP Handset user have to be configured with “join incoming – outgoing” into features.	☐	☒	☐	
17	**Public External call to SIP Handset1 and be transferred to SIP Handset2 – Attended transfer**	☐	☒	☐	
18	**Public External call to SIP Handset1 and be transferred to SIP Handset2 – Semi-Attended transfer**	☐	☒	☐	

5.2.11 Call transfer by Attendant

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

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**SIP Handset call to attendant** From SIP Handset1 dial “9” (attendant call prefix) Check audio and display	☐	☒	☐	Display shows internal Number / name of Attendant extension
2	**2nd incoming call while in conversation with attendant** While SIP Handset1 is in conversation with the attendant, from oxo phone call SIP Handset1 Answer the call on SIP handset1 and check audio and display	☐	☒	☐	
3	**SIP handset call to attendant, attendant transfers to OXO set, semi-attended**	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
	From SIP Handset1 dial “9” (attendant call prefix) and answer. Attendant transfer semi-attended to IP Touch2 Answer the call and check audio and display				
4	**SIP handset call to attendant, attendant transfers to OXO set, attended transfer** From SIP Handset1 dial “9” (attendant call prefix) and answer Attendant transfer attended to IP Touch2 Check audio and display	☐	☒	☐	
5	**OXO set calls to attendant, attendant transfers to SIP handset, attended transfer** From oxo phone, dial “9” (attendant call prefix) and answer Attendant transfer attended to SIP Handset1 Check audio and display	☐	☒	☐	
6	**External ISDN Call to attendant, attendant transfers to SIP set, attended** ISDN incoming call to the attendant. From the attendant call SIP Handset1 and transfer attended Check audio and display	☐	☒	☐	
7	**SIP set call to attendant, attendant transfers to External** From SIP Handset2, dial “9” (attendant call prefix) and answer From the attendant, call an external ISDN destination and transfer semi-attended Answer and check audio and display.	☐	☒	☐	

Nota: In order to have correct display of Public according to your country, you have to change the language from Ascom Device manager or from the set configuration itself.

5.2.12 Voice Mail

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

The default Voice Mail direct call number is 500 and prefix to consult voicemail is 67.

For these tests, DTMF sending (RFC 2833) has to be validated in order to use Voice Mail menu.

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Password modification in Voice Mail** With SIP Handset 1 call the Voice Mail at 500 and follow the Voice guide in order to modify the default password. When modification is accepted hang-up. Recall the voice mail and try to log with a wrong password. Check the rejection. Recall the voice mail and try to log with the right password. Check the service access.	☐	☒	☐	Default password was changed in Voice Mail. 6 digits password.
2	**Message display activation, MWI** With SIP Handset 1 call the Voice Mail at 500. Follow the instructions in order to send a voice message in SIP Handset2 voicemail box. Check that the MWI on SIP Handset2 is activated.	☐	☒	☐	SIP Handset / Base Station will SIP-Subscribe (Message-Summary) to get MWI. OXO sends SIP-Notify with Message-Waiting: Yes and Voice messages: 1/0.
3a	**Message consultation** With SIP Handset1 (extn. 220) call the Voice Mail at 500 (or 67). Follow the instructions in order to listen your voice message left during the previous test. Check that you can listen it and delete. Check that MWI display is disabled on 220 after message cancellation.	☐	☒	☐	Note: Test case also OK when message left by OXO phone
3b	**Message consultation after Base Station restart** With SIP Handset1 (extn. 220) call the Voice Mail at 500 (or 67). Restart base station and check MWI on SIP Handset 1 display Follow the instructions in order to listen your voice message left during the previous test. Check that you can listen it and delete.	☐	☒	☐	Note: Voice Mail left by OXO phone

Test Case Id	Test Case	N/A	OK	NOK	Comment
	Check that MWI display is disabled on 220 after message cancellation.				
4	**SIP Handset calling to forwarded OXO user** Forward the oxo phone to Voice Mail . With SIP Handset1 call OXO Phone and check that you are immediately forwarded to Voice Mail. Check that you can leave a message	☐	☒	☐	The NOTIFY contains: ***Message Waiting: yes\r\n*** At least one new message. ***Voice-message: 2/7r\n*** 2 new messages and 7 old messages.
5	**OXO user calling SIP Handset forwarded to Voice Mail** Forward the SIP Handset1 to Voice Mail by dialing 791+ 500. With oxo phone call SIP Handset1 and check that you are immediately forwarded to Voice Mail. Check that you can leave a message Listen to Voice Mail from SIP Handset 1 and then delete. On SIP Handset1 disable Voice Mail forwarding with 790 prefix.	☐	☒	☐	No counter on Ascom SIP Handset for multiple messages

5.2.13 Defence tests

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

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**OXO Reboot while SIP Handsets are in a call** Establish an incoming ISDN call to SIP Handset1. Reboot the OXO. When the OXO is up again, re-establish an incoming ISDN call to SIP Handset1 and check the audio.	☐	☒	☐	Trunk was released after time but no audio path to SIP Handset. SIP handset did try to register every 120 sec.
2	**IP-DECT Base Station - Ethernet link failure or Power failure** Disconnect the Ethernet link of Base Station or Power it down.	☐	☒	☐	IPBS2 is POE and we just disconnected the uplink of switch to disconnect BS from OXO.

	Check that the incoming call is presented to the attendant or Voice Mail. Reconnect the Ethernet link of BS or power it up. Test that you can call to or from SIP Handsets and check the audio.				Display on SIP Handset was “PBX Out of Service”
3	**IP-DECT Base Station redundancy – Fail-over between 2 Base Stations** Disconnect the Ethernet link of Active Base Station or Power it down. Check that Stand-by Base station took the service. Check that the incoming call is presented to the attendant or Voice Mail. Reconnect the Ethernet link of BS or power it up. Test that you can call to or from SIP Handsets and check the audio.	☐	☒	☐	During the Fail-Over, the DECT service is temporarily not available. The SIP Handset will register back to OXO connect with IP address of the new base station.

Chapter

6

Appendix A: SOLUTION DESCRIPTION

Ascom IP-DECT Overview

Ascom IP-DECT combines Voice over IP with Digitally Enhanced Cordless Telephony (DECT) technology. Ascom IP-DECT a reliable wireless communication solution that offers enterprise-grade telephony, professional messaging, personal alarm, and positioning over secure dedicated frequency bands. It is developed based on open standards, such as SIP, which maximizes interoperability with leading vendors.

The best of both worlds

- **IP** (Internet Protocol) – Universal standard for inter-networking that maximizes scalability and interoperability.
- **DECT** (Digital Enhanced Cordless Telecommunications) - Secure radio communication standard that delivers superior voice quality over reserved radio frequency bands.*

*Frequency Range	Max. Output Power	Region
1880-1895 MHz	10mW	Taiwan
1880-1900 MHz	10mW	Europe, Middle East & Africa, Australia, New Zeeland, and parts of Asia
1900-1906 MHz	10mW	Thailand
1910-1930 MHz	10mW	South America
1920-1930 MHz	4mW	North America

Ascom IP-DECT Systems Architecture

The basic building blocks of an Ascom IP-DECT system include:

Ascom IP-DECT infrastructure

- IP-DECT Access Points
- TDM-DECT Base Stations
- IP-DECT Gateways

kmzslykxcntsgvlh

Ascom DECT handsets

- Ascom d81
- Ascom d63
- Ascom d43

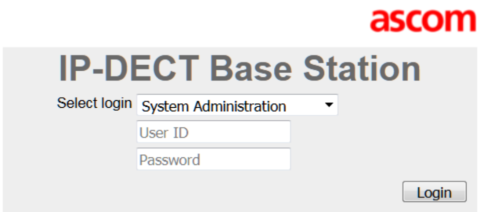
Ascom Unite Middleware

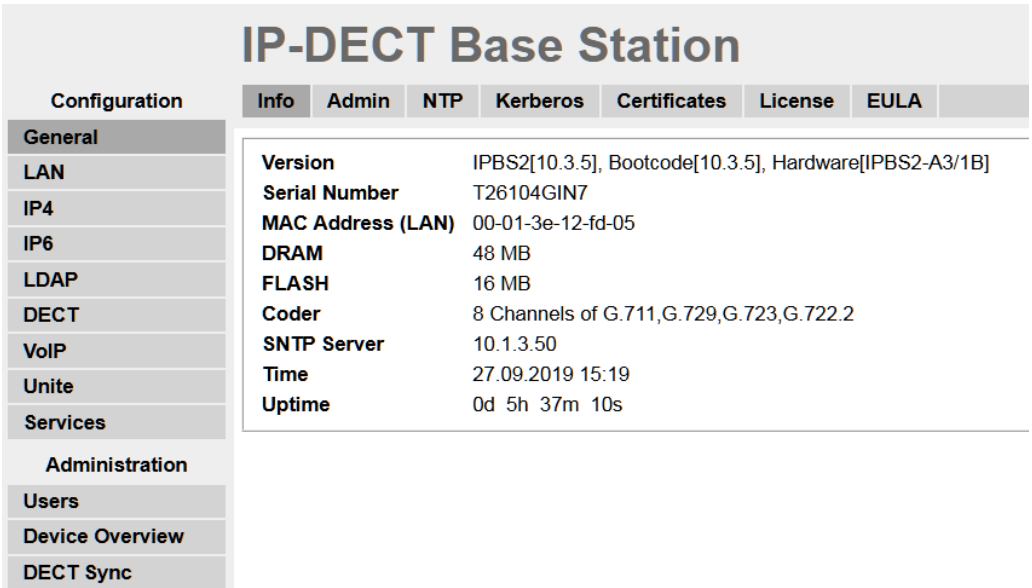
For more information, please refer to the following link: https://www.ascom.com/products/technology/ip-dect.html

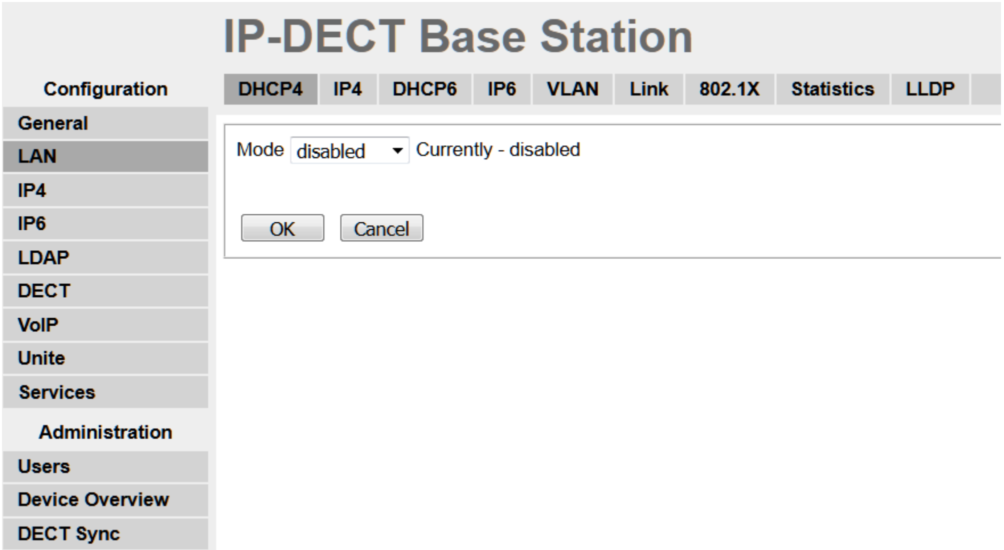
Chapter

7

Appendix B: PARTNER side CONFIGURATION



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DHCP6
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Link
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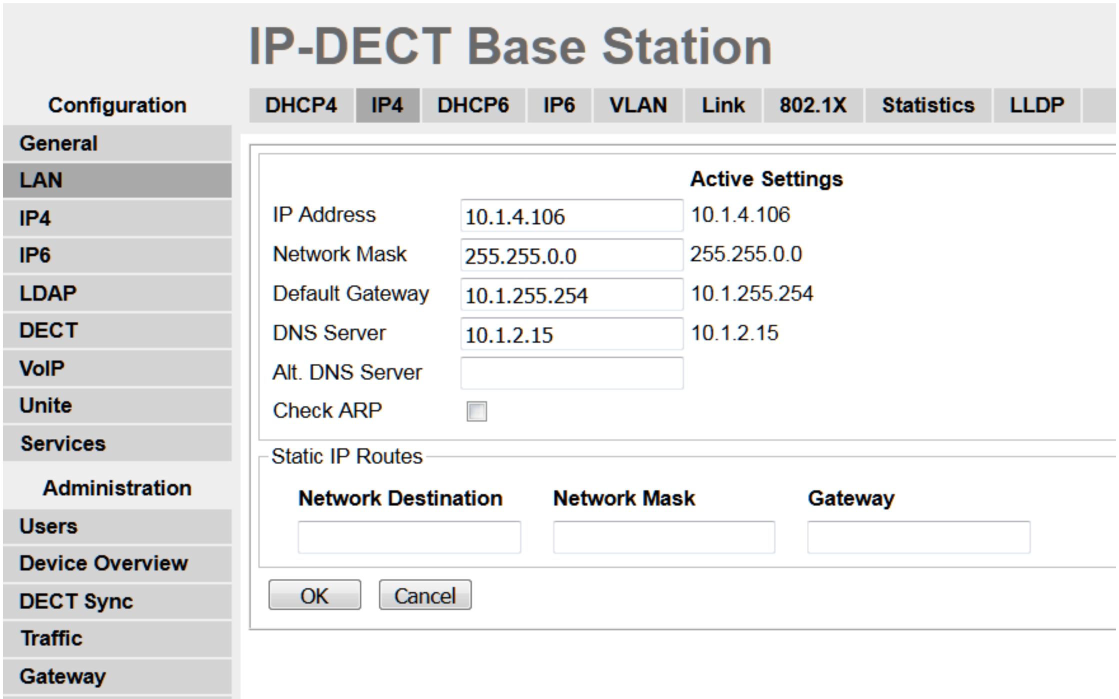
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LDAP
DECT
VoIP
Unite
Services
Administration
Users
Device Overview
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Traffic
Gateway
Active Settings
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Network Mask 255.255.0.0 255.255.0.0
Default Gateway 10.1.255.254 10.1.255.254
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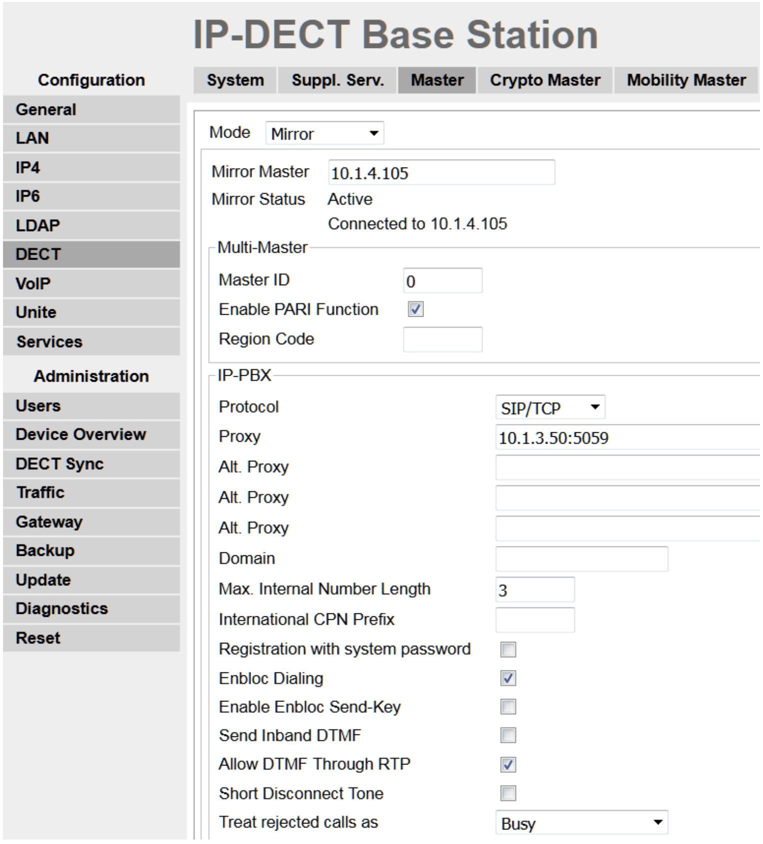
Network Mask
Gateway
OK
Cancel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Update
Diagnostics
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System
Suppl. Serv.
Master
Crypto Master
Mobility Master
Mode
Mirror
Mirror Master
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Max. Internal Number Length
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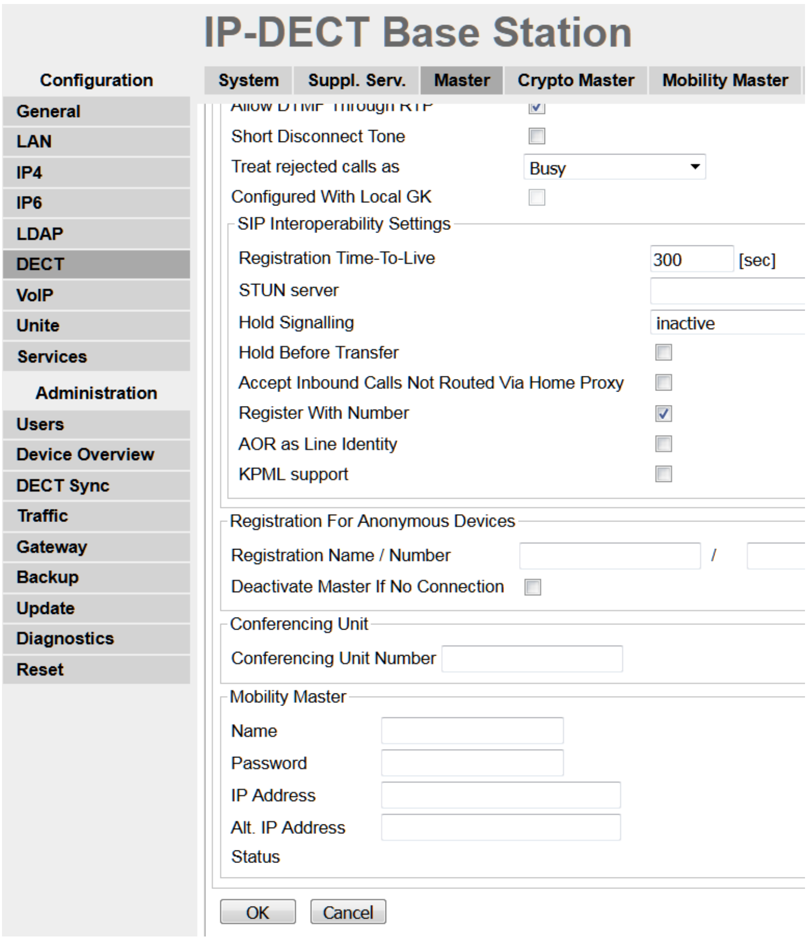
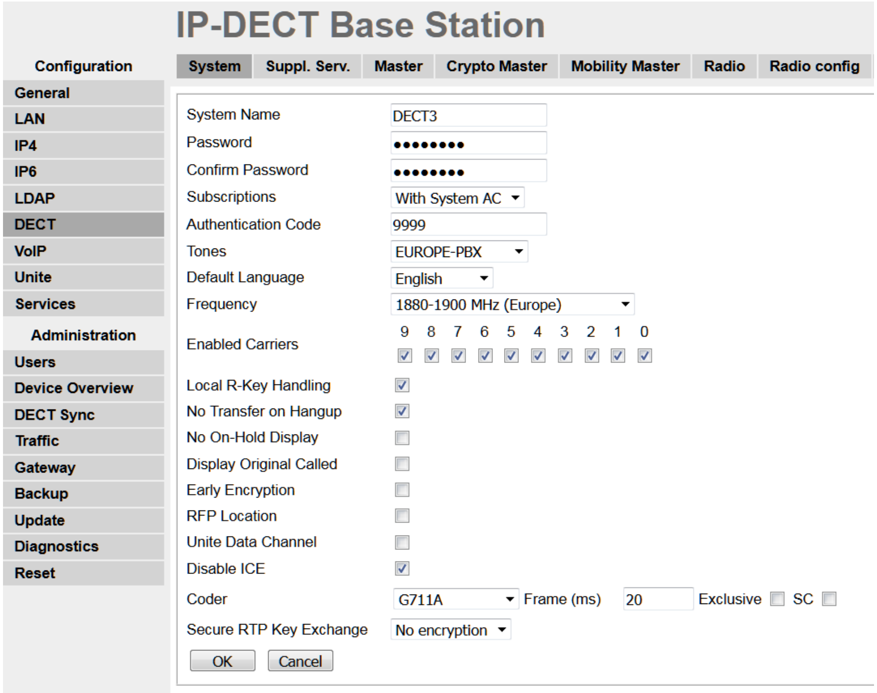
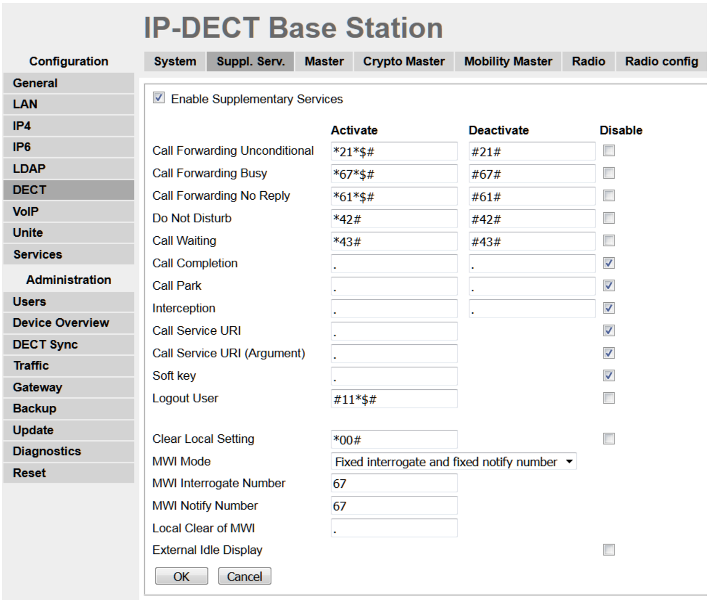
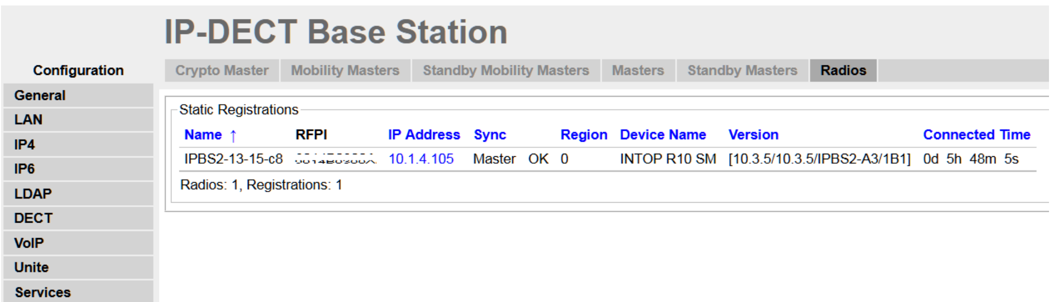
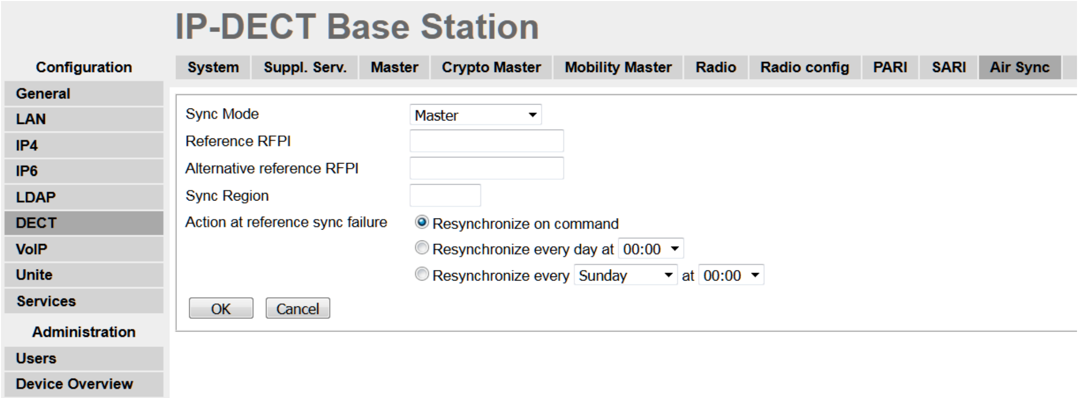
International CPN Prefix
Registration with system password
Enbloc Dialing
Enable Enbloc Send-Key
Send Inband DTMF
Allow DTMF Through RTP
Short Disconnect Tone
Treat rejected calls as
Busy

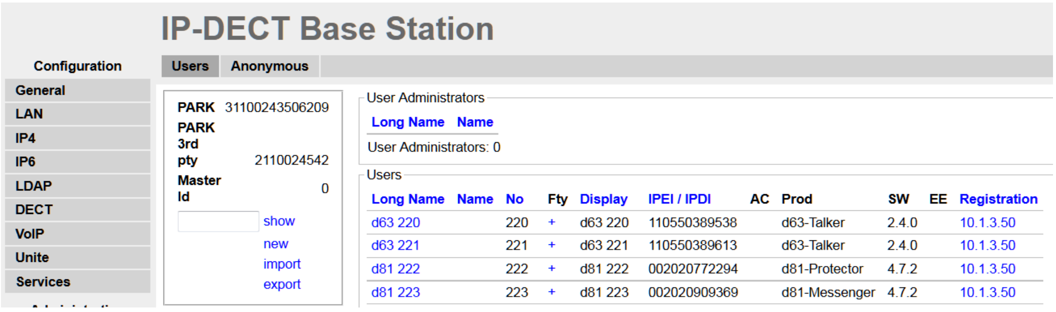

IP-DECT Base Station
Configuration
General
LAN
IP4
IP6
LDAP
DECT
VoIP
Unite
Services
Administration
Users
Device Overview
DECT Sync
Traffic
Gateway
Backup
Update
Diagnostics
Reset
System
Suppl. Serv.
Master
Crypto Master
Mobility Master
Short Disconnect Tone
Treat rejected calls as
Busy
Configured With Local GK
SIP Interoperability Settings
Registration Time-To-Live
300
[sec]
STUN server
Hold Signalling
inactive
Hold Before Transfer
Accept Inbound Calls Not Routed Via Home Proxy
Register With Number
AOR as Line Identity
KPML support
Registration For Anonymous Devices
Registration Name / Number
/
Deactivate Master If No Connection
Conferencing Unit
Conferencing Unit Number
Mobility Master
Name
Password
IP Address
Alt. IP Address
Status
OK
Cancel


IP-DECT Base Station
Configuration
System
Suppl. Serv.
Master
Crypto Master
Mobility Master
Radio
Radio config
General
LAN
IP4
IP6
LDAP
DECT
VoIP
Unite
Services
Administration
Users
Device Overview
DECT Sync
Traffic
Gateway
Backup
Update
Diagnostics
Reset
System Name
DECT3
Password
Confirm Password
Subscriptions
With System AC
Authentication Code
9999
Tones
EUROPE-PBX
Default Language
English
Frequency
1880-1900 MHz (Europe)
Enabled Carriers
9 8 7 6 5 4 3 2 1 0
Local R-Key Handling
No Transfer on Hangup
No On-Hold Display
Display Original Called
Early Encryption
RFP Location
Unite Data Channel
Disable ICE
Coder
G711A
Frame (ms)
20
Exclusive
SC
Secure RTP Key Exchange
No encryption
OK
Cancel


IP-DECT Base Station
Configuration
System
Suppl. Serv.
Master
Crypto Master
Mobility Master
Radio
Radio config
General
LAN
IP4
IP6
LDAP
DECT
VoIP
Unite
Services
Administration
Users
Device Overview
DECT Sync
Traffic
Gateway
Backup
Update
Diagnostics
Reset
Enable Supplementary Services
Activate
Deactivate
Disable
Call Forwarding Unconditional
21$#
#21#
Call Forwarding Busy
67$#
#67#
Call Forwarding No Reply
61$#
#61#
Do Not Disturb
*42#
#42#
Call Waiting
*43#
#43#
Call Completion
Call Park
Interception
Call Service URI
Call Service URI (Argument)
Soft key
Logout User
#11*$#
Clear Local Setting
*00#
MWI Mode
Fixed interrogate and fixed notify number
MWI Interrogate Number
67
MWI Notify Number
67
Local Clear of MWI
External Idle Display
OK
Cancel


IP-DECT Base Station
Configuration
Crypto Master
Mobility Masters
Standby Mobility Masters
Masters
Standby Masters
Radios
General
LAN
IP4
IP6
LDAP
DECT
VoIP
Unite
Services
Static Registrations
Name ↑
RFPI
IP Address
Sync
Region
Device Name
Version
Connected Time
IPBS2-13-15-c8
10.1.4.105
Master
OK
0
INTOP R10 SM
[10.3.5/10.3.5/IPBS2-A3/1B1]
0d 5h 48m 5s
Radios: 1, Registrations: 1



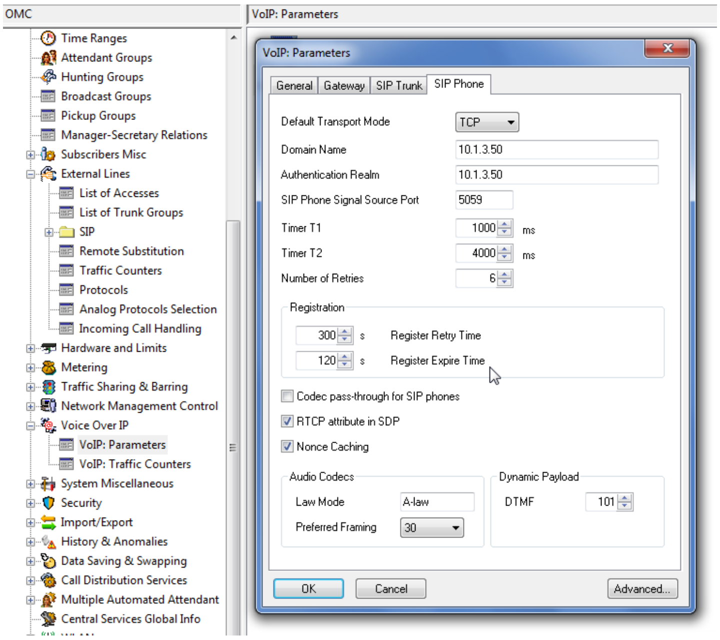


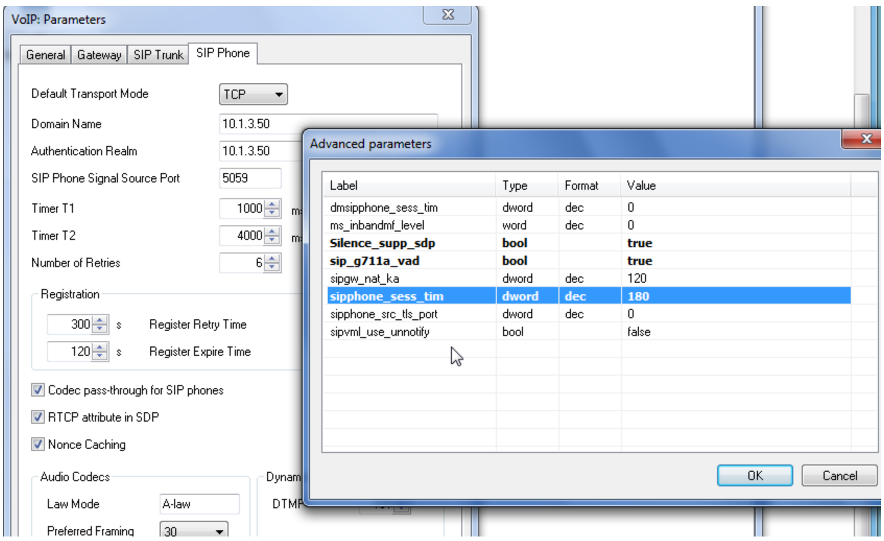
Chapter 8 Appendix C: ALE side CONFIGURATION

8.1 Configuration of OXO Connect

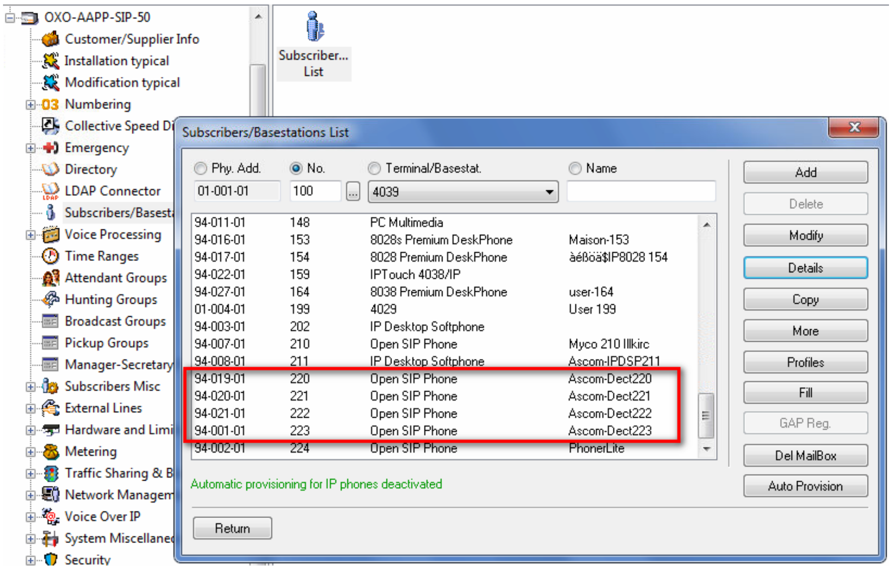
The configuration and management of parameters for the OXO Connect is done using the OMC program (Windows application). OMC stands for OmniPCX Management Console.

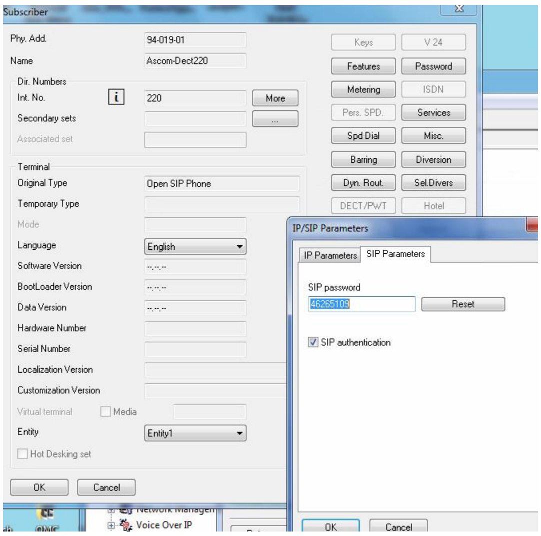
8.2 Configuration of VOIP parameters

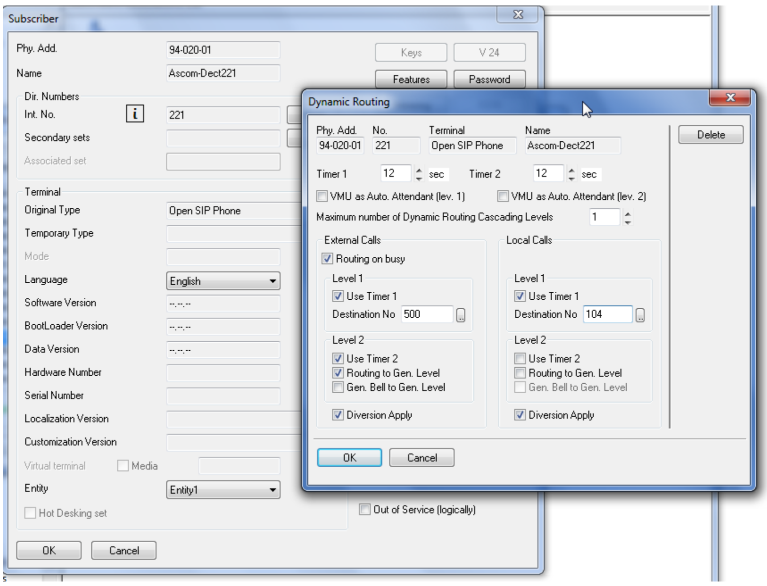


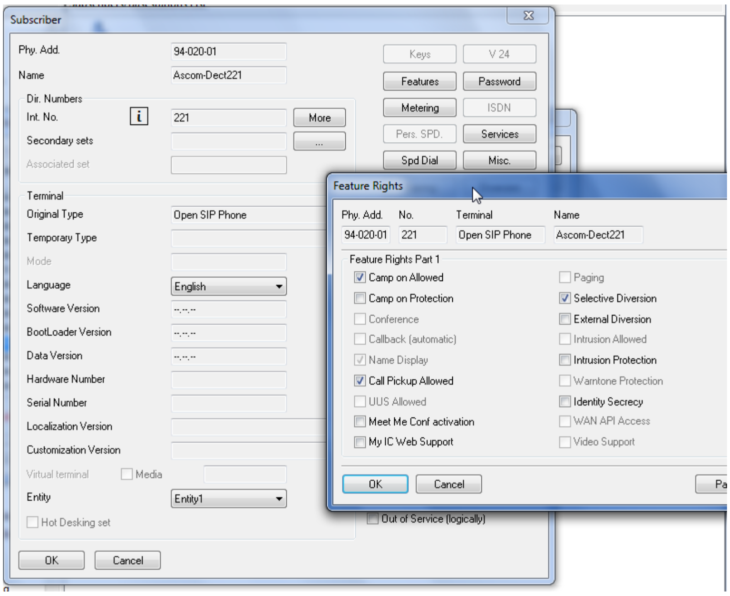


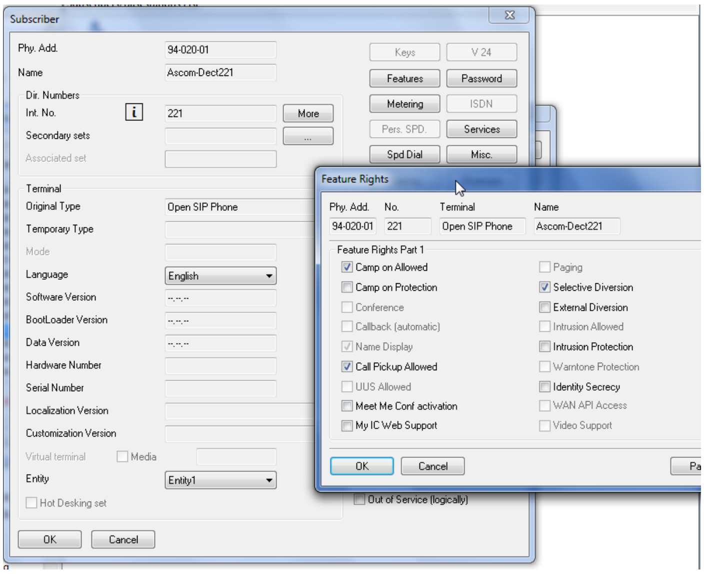
8.3 Configuration of the Open SIP Phones for DECT handsets

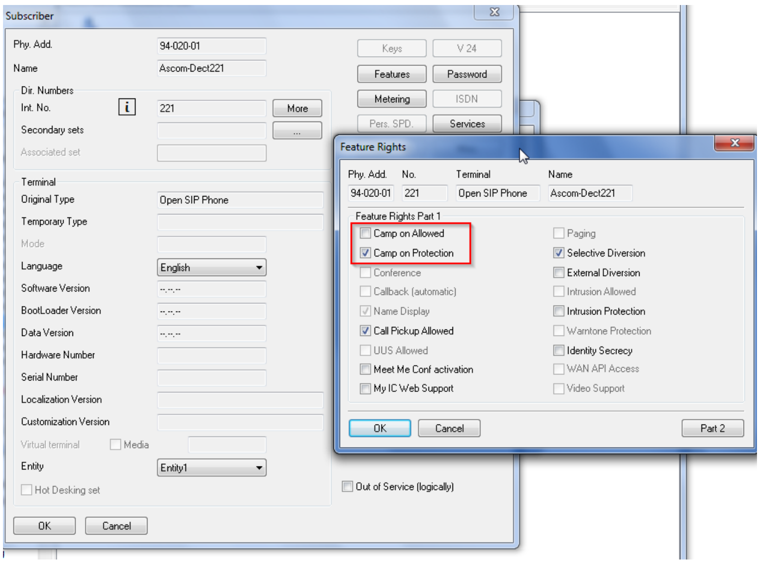






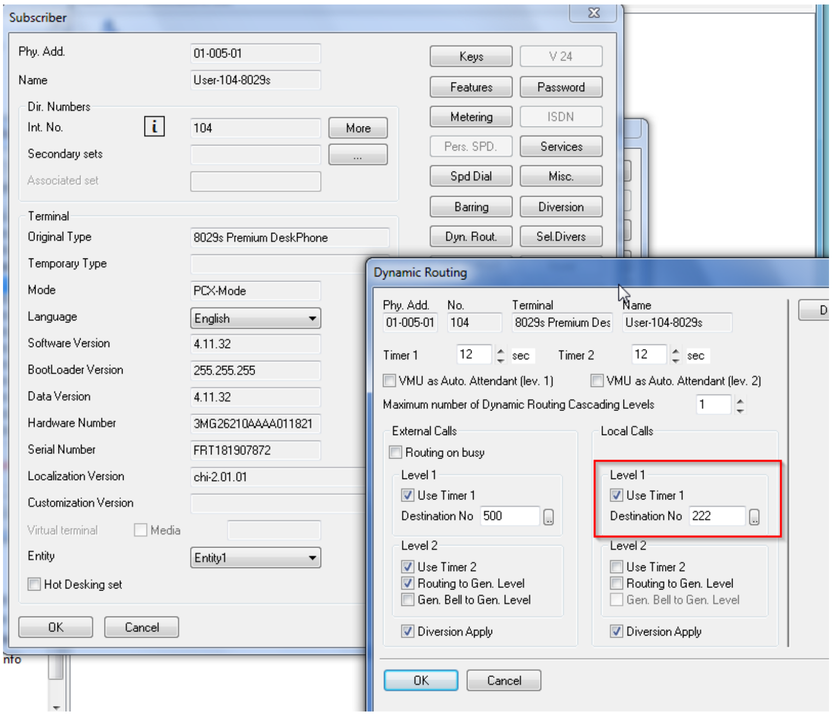


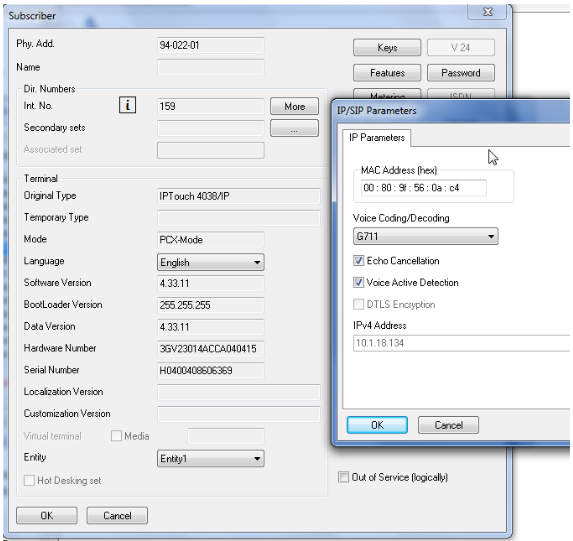


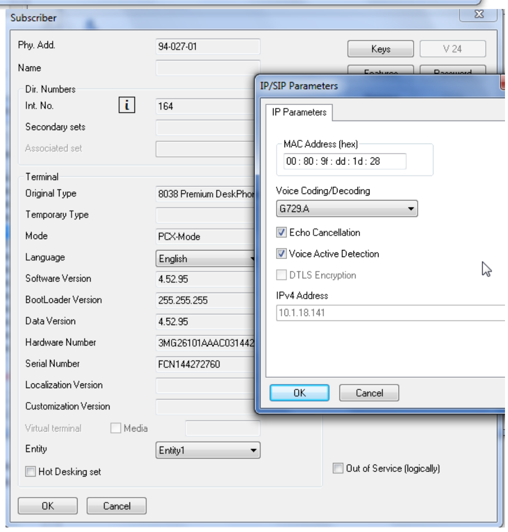


This has to be done for all 4 handsets D81 and D83.

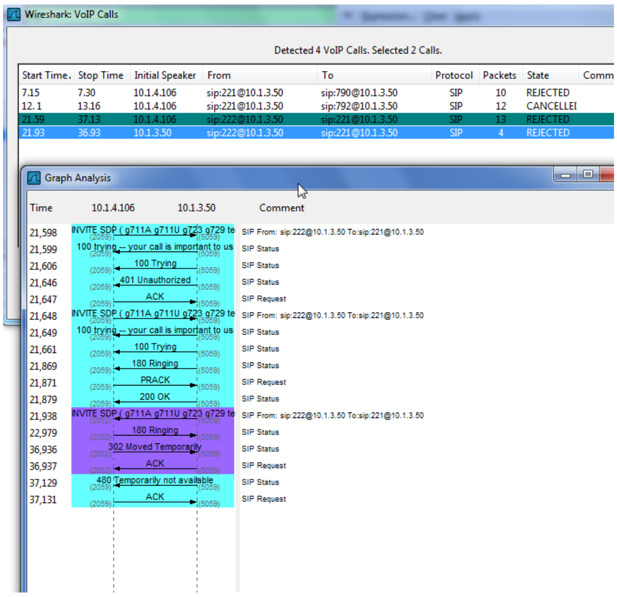
8.4 Configuration of premium Deskphones used for testing



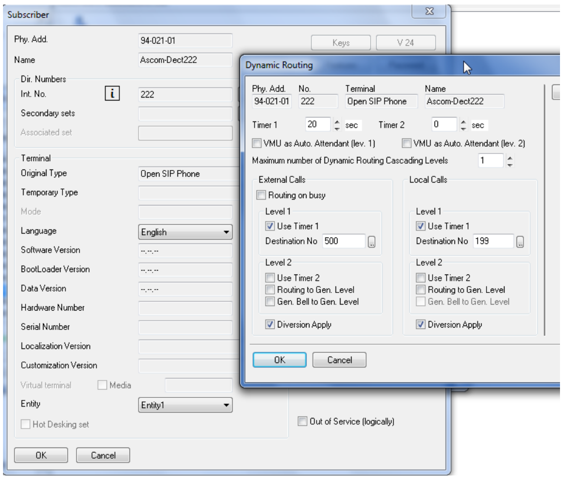


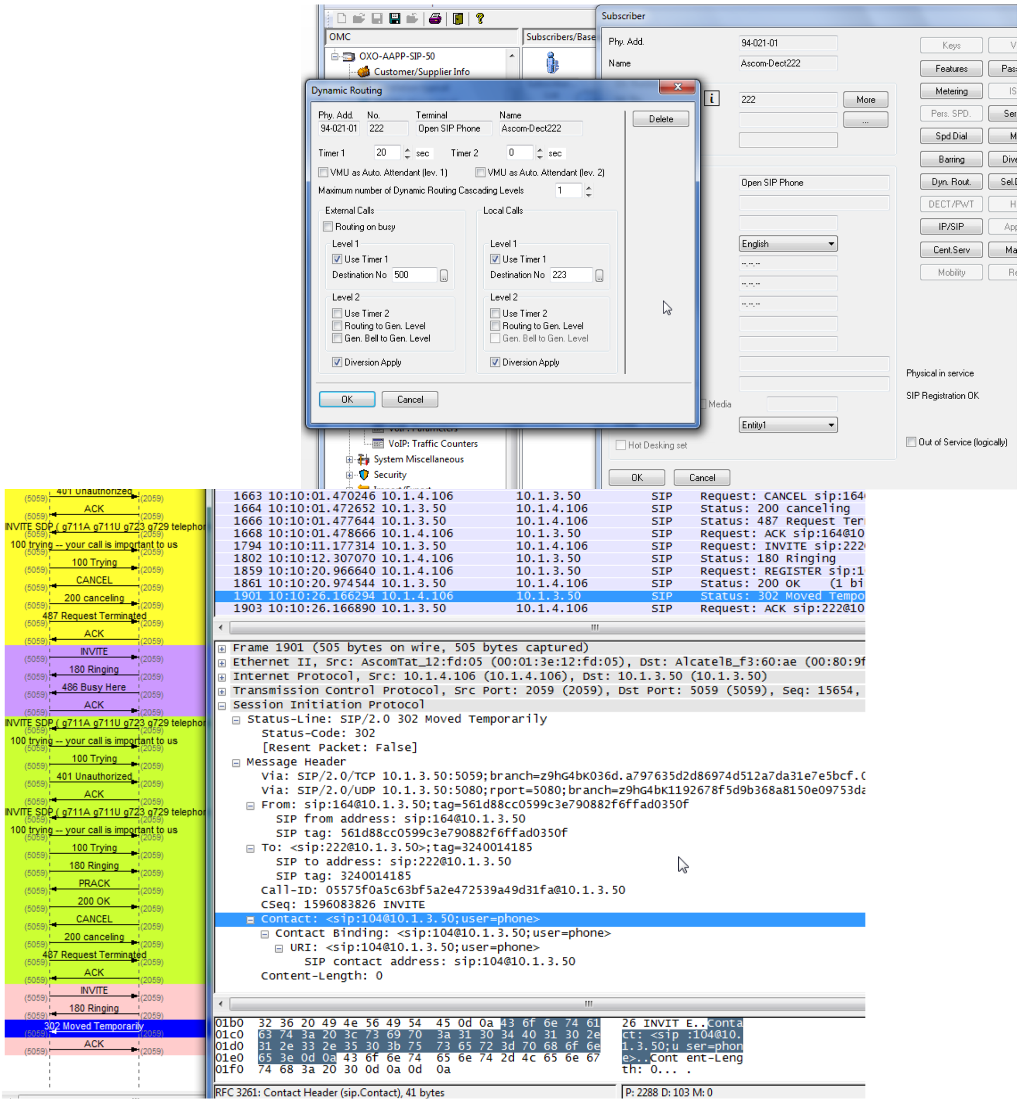


8.5 Test of forwarded call

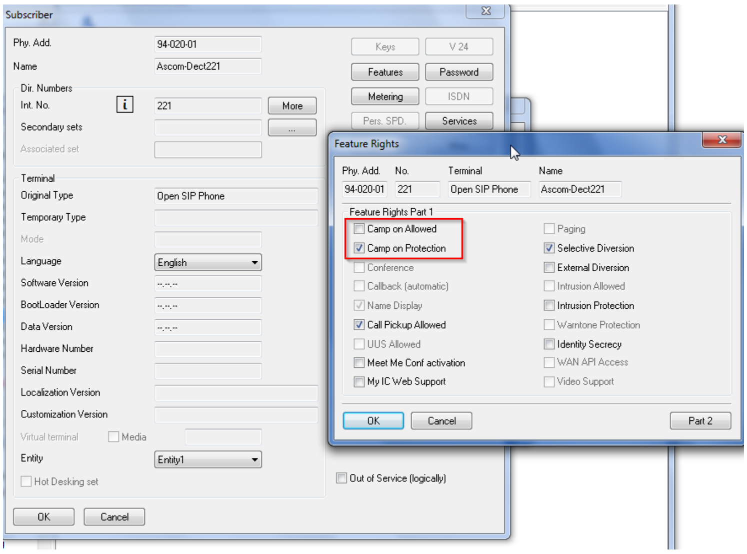


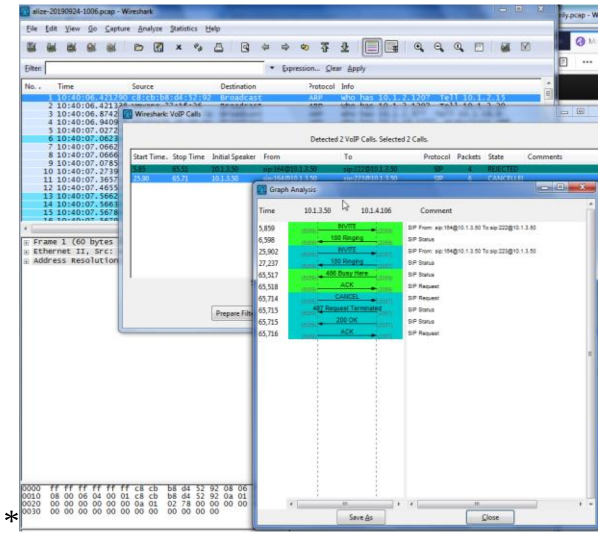
8.6 Test of diversion call



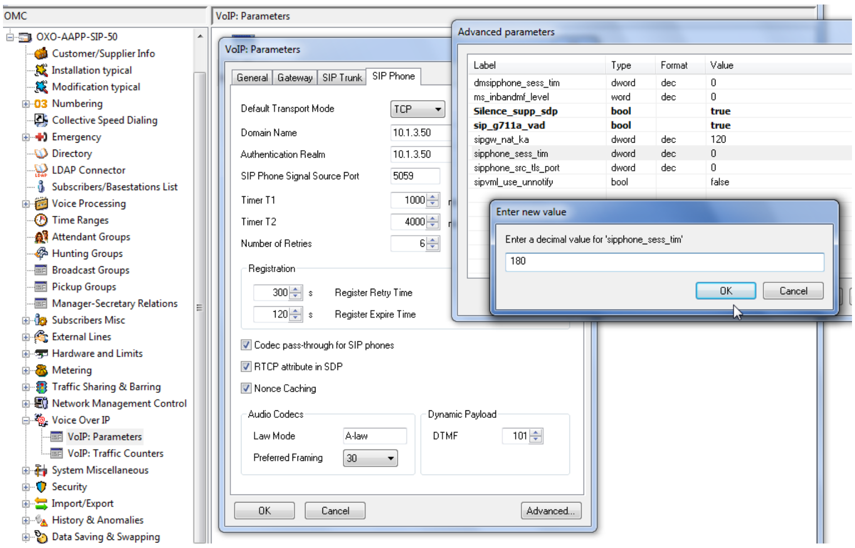


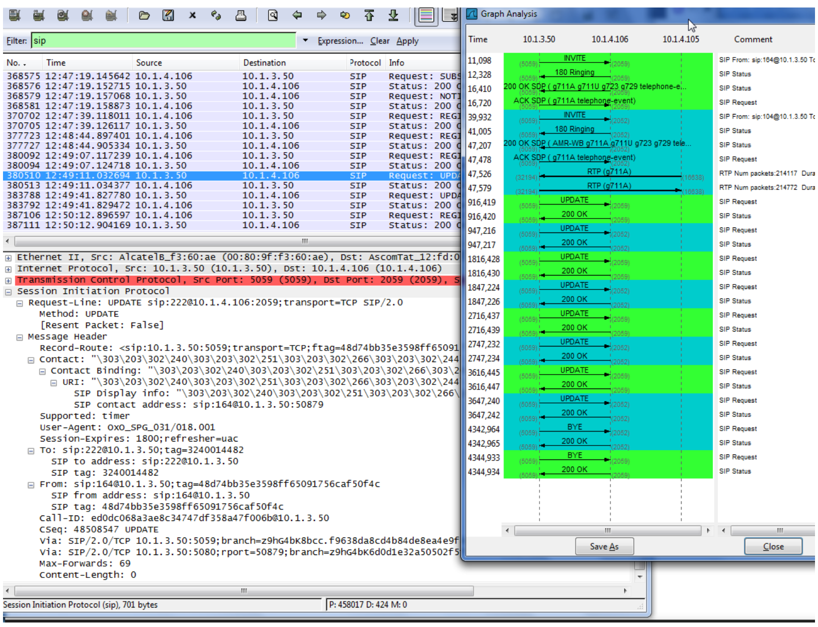
8.7 Test of call to Busy user





8.8 SIP Session timer for long time call :





Chapter

9

Appendix D: PARTNER SUPPORT PROCESS

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

Company/Country	Technical Manager/ Service Manager	e-mail
Ascom Australia	Simon Squire	Simon.Squire@ascom.com
Ascom Austria	Torsten Wolf	Torsten.Wolf@ascom.com
Ascom Belgium	Gerty Everts	Gerty.Everts@ascom.com
Ascom Denmark	Tarja Brusila	Tarja.Brusila@ascom.com
Ascom Finland	Tarja Brusila	Tarja.Brusila@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	Tarja Brusila	Tarja.Brusila@ascom.com
Ascom Romania	Johan Andrén	Johan.Andren@ascom.com
Ascom Singapore	Hans Tysk	Hans.Tysk@ascom.com
Ascom Sweden	Tarja Brusila	Tarja.Brusila@ascom.com
Ascom Switzerland	Torsten Wolf	Torsten.Wolf@ascom.com
Ascom United Arab Emirates	Johan Andrén	Johan.Andren@ascom.com
Ascom United Kingdom	Luke Blackmoore	Luke.Blackmoore@ascom.com
Ascom United States	Steven Zachary	Steven-Zachary@ascom.com
International	Johan Andrén	Johan.Andren@ascom.com

Chapter

10

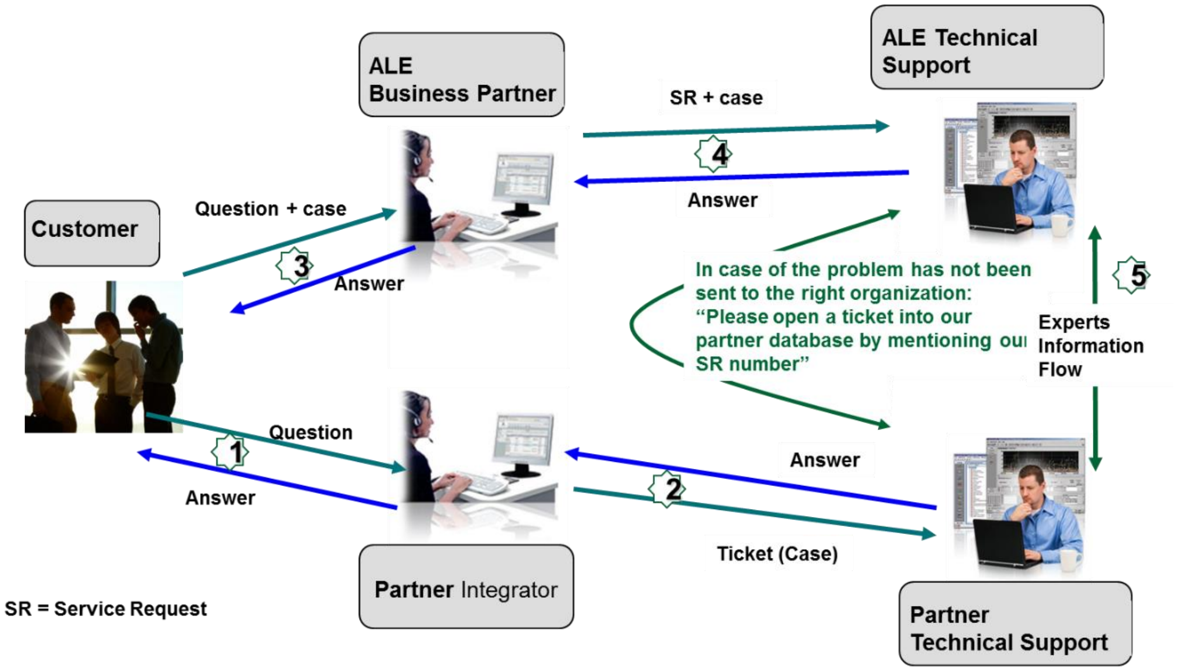
Appendix E: ALE SUPPORT PROCESS

10.1Introduction

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

10.2Escalation 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.

10.3 Escalation in all other cases

For non-certified solutions, no valid InterWorking Report is available and the integrator is expected to trouble-shoot 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.

10.4 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:

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

For other countries:

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

END OF DOCUMENT