Alcatel·Lucent
Certified Application

# Alcatel Lucent Application Partner Program
# Inter-Working Report

## Partner: ATLINKS
## Application type: VoIP SIP Phone
## Application name: Alcatel Temporis IP 800, 600, 200
## Alcatel-Lucent Platform: OmniPCX Office

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

# Certification overview

| | |
|---|---|
| Date of the certification | August 2012 |

| | |
|---|---|
| Alcatel-Lucent's representative | Alain Botti |
| AAPP member representative | Daniel Ananikian |

| | |
|---|---|
| Alcatel-Lucent Communication Platform | OmniPCX Enterprise |
| Alcatel-Lucent Communication Platform Release | R820 / 034.001 |
| AAPP member application version | TEMPORIS Firmware v15.60.0.80, v 14.60.0.78, v13.60.0.77 |
| Application Category | Terminals |

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Reviewer(s): Denis Lienhart, Rachid Himmi

**Revision History**

Edition 1: creation of the document – *August 2012*
Edition 2: extension to OXO R9.0 – *December 2012*

## Test results

☐ Passed ☐ Refused ☐ Postponed

☑ Passed with restrictions

Refer to the section 6 for a summary of the test results.

## IWR validity extension

The validity of this IWR has been extended to the following software releases/products:

- OmniPCX Office Release 9.0 – *December 2012*

# AAPP Member Contact Information


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# 1 Introduction

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

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

Information contained in this document is believed to be accurate and reliable at the time of printing. However, due to ongoing product improvements and revisions, Alcatel-Lucent cannot guarantee accuracy of printed material after the date of certification nor can it accept responsibility for errors or omissions. Updates to this document can be viewed by Business Partners on the Technical Support page of the Enterprise Business Portal (https://businessportal.alcatel-lucent.com) in the Application Partner Interworking Reports corner.

Note: This interworking report does not cover configuration/management and/or mass provisioning of the TEMPORIS SIP Set. For any questions related to these topics, please contact ATLINKS.

# 2 Validity of the InterWorking Report

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

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

A new release is identified as following:

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

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

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

# 3 Limits of the Technical support

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

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

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

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

## 3.1 Case of additional Third party applications

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

# 4 Application information

**Application type:** VoIP SIP Phone

**Application commercial name:** SIP Phone Model TEMPORIS

**Application version:** TEMPORIS Firmware version SIP 800: v15.60.0.80, SIP 600: v14.60.0.78 and SIP 200: v13.60.0.77

**Interface type** : SIP/Ethernet

**Interface version (if relevant):**

**Brief application description:**

Alcatel Phones designs, develops, markets and sells corded and cordless fixed-line telephones to telecom operators and to professional and consumer retail sales channels around the world.
The excellent reputation of Alcatel Phones in the minds of both consumers and businesspeople is the result of our long history of providing high-quality communication terminals with the features people need and the designs they want.

**Type of application/product**:

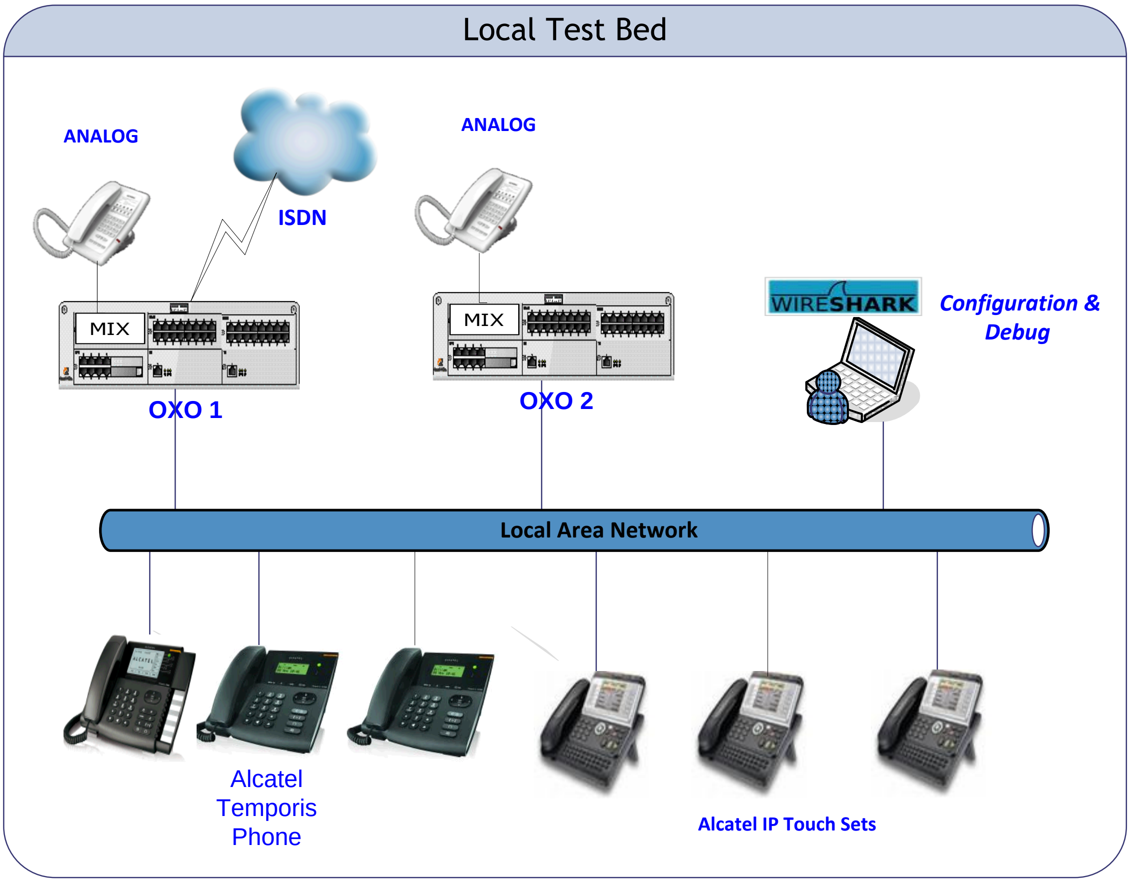
**IP800**

**IP600**

**IP 200**

# 5 Test Environment

Figure 1 **Test Environment**



## 5.1 Hardware configuration

**Alcatel-Lucent Communication Platform:**

- OmniPCX Office Rack
- PowerCPU
- Release: R820 / 034.001
- OMC: R820 / 17.1a

**Setup Details:**

| Setup Information OXO 1 | |
|---|---|
| OXO 1 IP address | 10.130.158.45 |
| Domain name | Oxoone.testandvalidate.com |
| Voicemail No | 500 |
| Attendant No | 0 |
| OXO Extension Details used for test | |
| IP Touch numbers | 322, 323 & 324 |
| TEMPORIS Dir numbers | 397, 398 and 399 |
| UA Set No | 301 |

| Setup Information OXO 2 | |
|---|---|
| Network OXO address | 10.130.158.112 |
| Network OXO Domain name | Oxotwo.testandvalidate.com |
| Network OXO Extension Details used for test | |
| TEMPORIS Dir numbers | 122 & 123 |
| IP Touch numbers | 102,103 & 104 |
| UA Set No | 101 |

**Note:**

1) The Two OXO systems are connected via private SIP Trunk.
2) For some tests we will change the set type from IP Touch to UA set or Analog set.

## 5.2 Software configuration

- **Alcatel-Lucent Communication Platform**: OmniPCX Office R820 / 034.001
- **Partner Application**: TEMPORIS V 15.60.0.80

**Note**: TEMPORIS Phones are registered in the OmniPCX Office as "Open SIP phone".

# 6 Summary of test results

## 6.1 Summary of main functions supported

| Features | Status | Comments |
|---|---|---|
| Initialization including network configuration | OK | |
| SIP registration | OK | DHCP registration is not supported for SIP phones |
| SIP authentication | OK | |
| Voice over IP and RTP codec support | OK | If G723 is enabled in Temporis, framing must be set to 30ms in the device |
| Outgoing Call | OK | RFC 4916 is not supported on OmniPCX Office |
| Incoming Call | OK | |
| Features During Conversation | OK | Only available from device (local) |
| Call Transfer | OK | Semi – Attended / Unattended (Blind) Transfer is not supported for SIP sets on OXO |
| Attendant | OK | |
| Voice mail interaction and indication | OK | |

## 6.2 Summary of problems

- None.

## 6.3 Summary of limitations

- DHCP mode is restricted to ALU IP Phones.
- In Conference we are unable to see the other user information other than the one who initiated the conference.
- No count of new messages (voice mails) available on the display of TEMPORIS.
- Call feature activation in the call server (eg CFU/CFB) is not displayed on the SIP device.
- Semi - Attended and blind Transfer are not supported in OXO.
- SIP Phones are getting registered even if wrong MAC ID provided in OXO

## 6.4 Notes, remarks

- Temporis phones are registered in the OmniPCX Office as "Open SIP phone".

# 7 Test Result 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**<br>• Action<br>• Expected result | ☐ | ☒ | ☐ | |
| 2 | **Test case 2**<br>• Action<br>• Expected result | ☐ | ☒ | ☐ | The application waits for PBX timer or phone set hangs up |
| 3 | **Test case 3**<br>• Action<br>• Expected result | ☒ | ☐ | ☐ | Relevant only if the CTI interface is a direct CSTA link |
| 4 | **Test case 4**<br>• Action<br>• 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 Alcatel-Lucent side or on Application 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.

# 8 Test Results

## 8.1 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).

| Test Case Id | Test Case | N/A | OK | NOK | Comment |
|---|---|---|---|---|---|
| **1** | **SIP sets**<br><br>Configure your SIP sets MCDU number on the OXO as 397, 398 & 399 to register with the OXO IP address<br><br>Check the registration on your sets and the display<br><br>Note that authentication is disabled for these users, the password doesn’t matter. | ☐ | ☒ | ☐ | |
| **2** | **SIP set registration to OXO in static IP addressing**<br><br>For this test we will try to register the SIP phone with authentication enabled.<br><br>SIP phones 397, 398 & 399 are configured with a static IP address of OXO.<br>Check the phone registration and display.<br><br>Redo the same test on one IP phone with a wrong password and check that the phone is rejected. | ☐ | ☒ | ☐ | |
| **3** | **DHCP registration (with OXO internal DHCP server)** | ☒ | ☐ | ☐ | |
| **4** | **NTP registration**<br><br>The SIP phone 399 is configured to retrieve the date and time from the OXO IP address.<br>Check the phone retrieves the right date and time information and displays it. | ☐ | ☒ | ☐ | |
| **5** | **Support of “423 Interval Too Brief” (1)**<br><br>The SIP phone 398 is configured with a value lower than 120 seconds.<br>Check the phone registration and display | ☐ | ☒ | ☐ | |
| **6** | **Signaling TCP-UDP**<br><br>If applicable configure your SIP set 398 to use the protocol SIP over UDP and other TCP<br><br>In the two cases, check the registration and basic calls. | ☐ | ☒ | ☐ | |

## 8.2 Audio codec negotiations/ VAD / Framing

These tests check that the phones are using the configured audio parameters (codec, VAD, framing).

Phone configuration: configure TEMPORIS to use G.722, 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).

| Test Case Id | Test Case | N/A | OK | NOK | Comment |
|---|---|---|---|---|---|
| **1** | Select G711 A-law as 1st codec in Temporis<br>Call from SIP 398 to IP Touch 323<br>Check that the call is established in G711 A-law.<br>Check audio quality<br><br>Call from IP Touch 323 to SIP 398<br>Check that the call is established in G711 A-law.<br>Check audio quality | ☐ | ☒ | ☐ | |
| **2** | Select G729 as 1st codec in Temporis<br>Call from SIP 398 to IP Touch 323<br>Check that the call is established in G729<br>Check audio quality<br><br>Call from IP Touch 323 to SIP 398<br>Check that the call is established in G729<br>Check audio quality | ☐ | ☒ | ☐ | |
| **3** | Select G723 as 1st codec in TEMPORIS<br>Check that the call is established in G723<br>Check audio quality<br><br>Call from IP Touch 323 to SIP 398<br>Check that the call is established in G723<br>Check audio quality | ☐ | ☒ | ☐ | Framing in Temporis must be configured to 30ms (default value : 20ms - G723_20 is not allowed in call server) |
| **4** | **Configure** 398 **to use VAD**<br>**Configure IP Touch** 323 **NOT to use VAD**<br><br>Call from SIP 398 to IP Touch 323<br>Check that the call is established in G711 A-law.<br>Check audio quality<br><br>**Configure SIP** 398 **to use VAD**<br>**Configure IP Touch** 323 **to use VAD**<br>Redo the same tests.<br><br>**Configure SIP** 398 **NOT to use VAD**<br>**Configure IP Touch** 323 **to use VAD**<br>Redo the same tests | ☐ | ☒ | ☐ | |
| **5** | **In OXO enable codec pass through for SIP phones.**<br>**Call from SIP** 397 **to SIP** 398<br>Check that the call is established using G.729<br>Check audio quality. | ☐ | ☒ | ☐ | |

| | | | | | |
|---|---|---|---|---|---|
| **6** | **In OXO 1 and OXO 2 enable codec pass through for SIP phone ; direct RTP and codec pass through for SIP trunk. G722 is preferred codec in TEMPORIS**<br><br>**Call from SIP** 397 **to Network SIP** 122<br>Check that the call is established using direct RTP in G722.<br>Check audio quality. | ☐ | ☒ | ☐ | |
| **7** | **In OXO 1 and OXO 2 disable direct RTP and codec pass through for SIP trunk ARS table is configured with "default" codec. G722 is preferred codec in TEMPORIS**<br><br>**Call from SIP** 397 **to Network SIP** 122<br>Check that the call is established in G711.<br>Check audio quality | ☐ | ☒ | ☐ | |
| **8** | **In OXO 1 and OXO 2 disable direct RTP and codec pass through for SIP trunk**<br>**ARS table is configured with codec G729_30**<br><br>**Call from SIP** 397 **to Network SIP** 122<br>Check that the call is established in G729.<br>Check audio quality | ☐ | ☒ | ☐ | |

## 8.3 Outgoing Calls

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 SEPLOS prefixes are mandatory for several tests of this section. For more information refer to the appendix C.

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

| Test Case Id | Test Case | N/A | OK | NOK | Comment |
|---|---|---|---|---|---|
| 1 | **Call to a local user**<br><br>With SIP Phone 398 call the IP Touch 322. Check that 322 is ringing.<br>Take the call and check ring back tone audio and display. | ☐ | ☒ | ☐ | |
| 2 | **Call to local user with no answer**<br><br>With SIP Phone 399 call the IP Touch 322.<br>And never take the call.<br>Check time out (if any) and display.<br>Note that 322 don't have a Voice Mail | ☐ | ☒ | ☐ | 322 is ringing until 399 releases the call |
| 3 | **Call to another SIP set**<br><br>With the SIP phone 398 call the other SIP Phone 399<br><br>Check the display and audio during all steps (dialing, ring back tone, conversation, and release). | ☐ | ☒ | ☐ | |
| 4 | **Call to wrong number**<br><br>(SIP: "404 Not Found")<br>With the SIP phone 398 call a wrong number<br>Check the ring back tone and display | ☐ | ☒ | ☐ | |
| 5 | **Call to busy user**<br>**(SIP: "486 Busy Here")**<br><br>With the SIP phone 398 call IP Touch 322, take the call and don't hang up.<br>With other SIP phone 399 call 322 which is busy<br>Check the ring back tone and display | ☐ | ☒ | ☐ | |
| 6 | **Call to user in "Out of Service" state**<br>**(SIP: "480 Temporarily Unavailable")**<br><br>With the SIP phone 399 call the IP Touch 322 which is in "Out of Service State"<br>Check the display and ring back tone | ☐ | ☒ | ☐ | |

| Test Case Id | Test Case | N/A | OK | NOK | Comment |
|---|---|---|---|---|---|
| 7 | **Call to user in "Do not Disturb" (DND) state (SIP: "480 Temporarily not available")**<br><br>Dial "*63" on the IP Touch 322 in order to enable the DND. Wait for acknowledgement ring back tone from OXO.<br>With the SIP phone 398 call 322.<br>Check ring back tone and display.<br>Redial *63 on 322 to cancel the DND | ☐ | ☒ | ☐ | 486 busy here message is displayed in Wireshark |
| 8 | **Call to local user, immediate forward (CFU). (SIP: "181 Forwarded")(1)**<br><br>On IP Touch 322 dial the *61323 (*61 + 323) to activate the CFU. Wait for acknowledgement ring back tone from OXO.<br>With the SIP phone 398 call the 322.<br>Check that 323 is ringing and the display. Take the call check audio and hung up.<br>Dial *60 on 322 for forward cancellation. | ☐ | ☒ | ☐ | |
| 9 | **Call to local user, forward on no reply (CFNR). (1)**<br><br>On IP Touch 322 configure with OMC the CFNR using dynamic routing to 323.<br>With 398 call the 322. Check that 322 is ringing but don't take the call and wait the time out (about 30 sec). Time out is defined in 322 dynamic routing of Timer 1.<br>After time out check that 323 is ringing and take the call.<br>Check the audio and display. | ☐ | ☒ | ☐ | In Temporis, Callee ID Header must be set to PAI-RPID<br><br>*Note: RFC4916 is not supported on OmniPCX Office* |
| 10 | **Call to local user, forward on busy (CFB). (1)**<br><br>On IP Touch 322 dial the *62323 (*62+<target MCDU number>) to activate the CFB. Wait for acknowledgement ring back tone from OXO.<br>With SIP phone 398 call 322 and take the call to make it busy.<br>With other SIP phone 399 call 322.<br>Check that 323 is ringing and take the call.<br>Check the audio and display.<br>Dial *60 on 322 for forward cancellation. | ☐ | ☒ | ☐ | In Temporis, Callee ID Header must be set to PAI-RPID<br><br>*Note: RFC4916 is not supported on OmniPCX Office* |
| 11 | **Call to external number (Check ring back tone, called party display)**<br><br>With SIP set 398 dial 9 (9 prefix +external number )<br>Take the call and check audio, display and call release. | ☐ | ☒ | ☐ | OK but Conversation Timer starts during ringing state |
| 12 | **SIP session timer expiration: Check if call is maintained or released after the session timer has expired**<br>With SIP set 398 call IP Touch 322.<br>Take the call on 322 and never hang up, wait for time out expiration.<br>Check that call is maintained or release. | ☐ | ☒ | ☐ | Note :<br>SIP session timer for SIP phones is available since OmniPCX Office R8.2 |

**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.

## 8.4 Incoming Calls

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 SEPLOS prefixes are mandatory for several tests of this section. For more information refer to the appendix C.

| Test Case Id | Test Case | N/A | OK | NOK | Comment |
|---|---|---|---|---|---|
| **1** | **Local /network call to free SIP terminal**<br>Local: with IP Touch 322 call SIP set 398. Check that 398 is ringing and take the call<br><br>Check ring back tone and called party display.<br><br>Network: with IP Touch 322 call SIP set 123 on another Node. Check that 123 is ringing and take the call.<br><br>Check ring back tone and called party display. | ☐ | ☒ | ☐ | |
| **2** | **Local/network call to busy SIP terminal**<br>Local: With SIP set 399 call other SIP set 398 and take the call to make it busy, don't hang up.<br>With IP Touch 323 call 398 which is busy<br><br>Check the ring back tone and display.<br><br>Network: With SIP set 398 call SIP set 123 and take the call to make it busy, don't hang up.<br>With 322 call 123 which is busy<br><br>Check ring back tone and called party display. | ☐ | ☒ | ☐ | |
| **3** | **Local/network call to unplugged SIP terminal**<br>Local: Unplug the 398 SIP set and call it with IP Touch 322.<br><br>Check the ring back tone and display<br><br>Network: Unplug the SIP set 123 and call it with 322<br><br>Check the ring back tone and display | ☐ | ☒ | ☐ | Call to SIP phone is not routed to its VMU until OXO detects SIP phone is unregistered (after register time out)... |
| **4A** | **Local/network call to SIP terminal in Do Not Disturb (DND) mode**<br>**By local feature if applicable:**<br><br>Local: Enable DND on SIP set 398 and call it with IP Touch 322<br>Check the ring back tone and display<br>Cancel the DND on 398.<br><br>Network: Enable DND on SIP set 123 and call it with IP Touch 322<br>Check the ring back tone and display<br>Cancel the DND on 398. | ☐ | ☒ | ☐ | Local:<br>IP Touch 322 display: "released".<br><br>Network:<br>IP Touch 322 display: "Network Error". |

| Test Case Id | Test Case | N/A | OK | NOK | Comment |
|---|---|---|---|---|---|
| **4B** | **By system feature (SEPLOS)**<br><br><u>Local</u>: Enable DND on SIP set 398 using the *63 prefix..<br>Wait for acknowledgement ring back tone from OXO.<br><br>With IP Touch 322 call 398<br>Check the ring back tone and display<br>Cancel the DND on 398 using *63 prefix.<br><br><u>Network</u>: Enable DND on SIP set 123 using the *63 prefix.<br>Wait for acknowledgement ring back tone from OXO.<br><br>With IP Touch 322 call 123<br><br>Check the ring back tone and display<br>Cancel the DND on 123 using * 60 prefix. | ☐ | ☒ | ☐ | DND is activated and working in the call server but OXO returns a wrong SIP msg “500 Internal Server Error” : end user feedback is wrong. |
| **5A** | **Local/network/SIP call to SIP terminal in immediate forward (CFU) to local user:**<br><br>**By local feature if applicable:**<br><u>Local</u>: On SIP set 398 enable CFU to IP Touch 322 With SIP set 399 call 398. Check that 322 is ringing.<br>Take the call and check audio and display.<br><br>Disable CFU on 398.<br><br><u>Network</u>: On SIP set 123 enable CFU to IP Touch 102.<br>With SIP set 398 call 123. Check that 102 is ringing.<br>Take the call and check audio and display.<br><br>Disable CFU on 123. | ☐ | ☒ | ☐ | |
| **5B** | **By system feature:**<br><br><u>Local</u>: On SIP set 398 enable CFU to IP Touch 322 using *61322 prefix (*61 + <target MCDU number>).<br>Wait for acknowledgement ring back tone from OXO.<br>With SIP set 399 call 398. Check that 322 is ringing.<br>Take the call and check audio and display.<br><br>Disable CFU on 398 using *60 prefix.<br><br><u>Network</u>: On SIP Set 123 enable CFU to IP Touch 102 using *61122 prefix (*61 + <target MCDU number>).<br>Wait for acknowledgement ring back tone from OXO.<br>With SIP Set 399 call 123. Check that 102 is ringing.<br>Take the call and check audio and display.<br><br>Disable CFU on 123 using *60 prefix. | ☐ | ☒ | ☐ | CFU is activated and working in the call server but OXO returns a wrong SIP msg “500 Internal Server Error” : end user feedback is wrong.<br><br>Phone display is not updated. |
| **6A** | **Local/network/SIP call to SIP terminal in immediate forward (CFU) to network number:**<br>**By local feature if applicable:**<br><br><u>Local</u>: On SIP Set 399 enable CFU to SIP Set122.With SIP set 398 call 399. Check that 122 is ringing.<br>Take the call and check audio and display.<br><br>Disable CFU on 399.<br><br><u>Network:</u> On SIP Set 398 enable CFU to IP Touch 102.<br>With SIP Set 123 call 398. Check that 102 is ringing.<br>Take the call and check audio and display.<br><br>Disable CFU on 398. | ☐ | ☒ | ☐ | Phone display is not updated. |

| Test Case Id | Test Case | N/A | OK | NOK | Comment |
|---|---|---|---|---|---|
| **6B** | **By system feature:**<br><br><u>Local</u>: On SIP Set 398 enable CFU to SIP Set 122 using *61122 prefix (*61 + <target MCDU number>).<br>Wait for acknowledgement ring back tone from OXO.<br>With SIP set 399 call 398. Check that 122 is ringing.<br>Take the call and check audio and display.<br><br>Disable CFU on 398 using *60 prefix.<br><br><u>Network</u>: On SIP Set 398 enable CFU to IP Touch 102 using *61102 prefix (*61 + <target MCDU number>).<br>Wait for acknowledgement ring back tone from OXO.<br>With SIP Set 123 call 398. Check that 102 is ringing.<br>Take the call and check audio and display.<br><br>Disable CFU on 398 using *60 prefix. | ☐ | ☒ | ☐ | CFU is activated and working in the call server but OXO returns a wrong SIP msg “500 Internal Server Error” : end user feedback is wrong.<br><br>Phone display is not updated. |
| **7A** | **Local/network/SIP call to SIP terminal in immediate forward (CFU) to another SIP user**<br><br>**By local feature if applicable:**<br><u>Local</u>: On SIP set 398 enable CFU to SIP set 122 With 399 call 398. Check that 122 is ringing.<br>Take the call and check audio and display.<br><br>Disable CFU on 398.<br><br><u>Network</u>: On SIP set 398 enable CFU to IP Touch 103.<br>With SIP Set 122 call 398. Check that 103 is ringing.<br>Take the call and check audio and display.<br><br>Disable CFU on 398. | ☐ | ☒ | ☐ | .<br>Phone display is not updated. |
| **7B** | **By system feature:**<br><br><u>Local</u>: On SIP Set 399 enable CFU to SIP Set 122 using *61122 prefix (*61 + <target MCDU number>).<br>Wait for acknowledgement ring back tone from OXO.<br>With SIP Set 398 call 399. Check that 122 is ringing.<br>Take the call and check audio and display.<br><br>Disable CFU on 399 using *60 prefix.<br><br>Network: On SIP Set 399 enable CFU to IP Touch 103 using *61123 prefix (*61 + <target MCDU number>).<br>Wait for acknowledgement ring back tone from OXO.<br>With SIP Set 122 call 399. Check that 103 is ringing.<br>Take the call and check audio and display.<br>Disable CFU on 399 using *60 prefix | ☐ | ☒ | ☐ | CFU is activated and working in the call server but OXO returns a wrong SIP msg “500 Internal Server Error” : end user feedback is wrong.<br><br>Phone display is not updated. |
| **8A** | **Local call to SIP terminal in “forward on busy” (CFB) state:**<br>**By local feature if applicable**<br><br>On SIP Set 398 enable CFB to IP Touch 322<br>With 398 call the voice mail at 500 to make it busy.<br>With SIP Set 399 call 398 which is busy.<br>Check that 322 is ringing<br>Take the call and check audio and display.<br><br>Disable CFU on 398. | ☐ | ☒ | ☐ | *Note: on the device CallWaiting must be set to OFF* |

| Test Case Id | Test Case | N/A | OK | NOK | Comment |
|---|---|---|---|---|---|
| 8B | **By system feature:**<br><br>On SIP Set 398 enable CFB to IP Touch 322 using *62322 prefix (*62 + <target MCDU number>).<br>Wait for acknowledgement ring back tone from OXO.<br>With 398 call the voice mail at 500 to make it busy.<br>With SIP Set 399 call 398 which is busy.<br>Check that 322 is ringing<br>Take the call and check audio and display.<br><br>Disable CFB on 398 using *60 prefix. | ☐ | ☒ | ☐ | |
| 9A | **Local call to SIP terminal in "forward on no reply" (CFNR)**<br>**By local feature if applicable**<br>On SIP Set 399 enable CFNR to IP Touch 322<br>With SIP Set 398 call 399.<br>Check that 399 is ringing and don't take the call, wait for time out (about 30 seconds).<br><br>After time out expiration the 322 is ringing, take the call and check audio and display. | ☐ | ☒ | ☐ | Phone display is not updated. |
| 9B | **By system feature:** | ☐ | ☒ | ☐ | CNFR via prefix not available on OXO (dynamic routing has to be used) |
| 10 | **Call to busy user, Call waiting.**<br>(Camp-on), local feature if applicable:<br>With SIP Set 398 call other SIP Set 399 (multiline set) to make it busy, take the call and don't hang up.<br><br>With IP Touch 323 call 399 (on 399 camp-on feature is enabled).<br>Check the Call waiting or ring back tones and display | ☐ | ☒ | ☐ | |
| 11 | **External call to SIP terminal.**<br>Check that external call back number is shown correctly:<br>With SIP Set 399 dial 9 + target MCDU number.<br><br>Check that external is ringing and the external call number is shown correctly<br>Take the call and check audio, display and call release. | ☐ | ☒ | ☐ | |
| 12 | **Calling Line Identity Restriction (CLIR): Local call to SIP terminal.**<br>On IP Touch 323 enable mask Identity and call SIP Set 399 in order to hide 323 identity.<br>Check that 399 is ringing, take the call and check that 323 identity is hidden. | ☐ | ☒ | ☐ | Display on Phone shows "Anonymous" |
| 13 | Display: Call to free SIP terminal from IP Touch user with a name containing non-ASCII characters (eg éëêèè).<br>Check caller display.<br><br>Check that SIP set is ringing and check on its display that the characters are correctly printed. | ☐ | ☒ | ☐ | |
| 14 | Display: Call from IP Touch to SIP which has the name containing non-ASCII characters, eg &@(#?+)=.<br>Check caller display.<br><br>Check that SIP set is ringing and check that the characters are correctly printed. | ☐ | ☒ | ☐ | |

| Test Case Id | Test Case | N/A | OK | NOK | Comment |
|---|---|---|---|---|---|
| **15** | SIP set is part of a sequential hunt group **(1)**. Call to hunt group. Check call/release.<br>With IP Touch 322 call the sequential hunt group MCDU number 328<br>Check that 398 is ringing<br>Take the call and don't hang up.<br><br>And with IP Touch 323 call the sequential hunt group MCDU number 328 Check that 323 is ringing<br>Take the call and don't hang up.<br><br>And with SIP Set 397 call the sequential hunt group MCDU number 328<br>Check that 399 is ringing<br>Take the call and don't hang up. | ☐ | ☒ | ☐ | |
| **16** | SIP set is part of a cyclic hunt group **(2)**. Call to hunt group. Check call/release.<br>With IP Touch 322 call the cyclic hunt group MCDU number 323<br>Check that 301 is ringing<br>Take the call and hang up.<br><br>And with 322 call the cyclic hunt group MCDU number 323<br>Check that 399 is ringing<br>Take the call and hang up.<br><br>And with SIP Set 397 call the cyclic hunt group MCDU number 323<br>Check that 398 is ringing<br>Take the call and don't hang up. | ☐ | ☒ | ☐ | |
| **17** | SIP set is declared as a MultiSet. Call to main set and see if twin set rings. Take call with twin set.<br><br>With IP Touch 323 call IP Touch 322 which is in MultiSet with SIP Set 399.<br>Check that 399 and 322 both ringing.<br><br>Take the call from 399 and check that 322 stop ringing.<br>Check audio and display. | ☐ | ☒ | ☐ | |

**Notes:**
(1) Sequential Hunt Group behavior: the endpoint n+1 is ringing **only** if the endpoint n is now in call (busy).

(2) 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.

## 8.5 Features during Conversation

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

| Test Case Id | Test Case | N/A | OK | NOK | Comment |
|---|---|---|---|---|---|
| 1A | **Hold and resume with local feature** (if applicable)<br>With 399 call 322 take the call, check audio and display.<br><br>With 399 put 322 on hold check tones and display on both and resume the call.<br><br>With 322 put 399 on hold check tones and display on both and resume the call.<br><br>Keep this call for the next test. | ☐ | ☒ | ☐ | |
| 1B | **Enquiry call to another local user** (if applicable)<br>Distant user is put on hold with local feature<br><br>With 399 (multi-lines) call 323 and take the call.<br>322 will be put on hold when making second call to 323<br><br>Put 323 on hold and check tones and display on both.<br><br>Keep these two calls for the next test. | ☐ | ☒ | ☐ | |
| 1C | **Broker request, toggle back and forth between both lines with local feature** (if applicable)<br><br>With 399 switch between 322 and 323 lines.<br><br>Check the tones and display on sets on hold state.<br><br>Keep these two calls for the next test. | ☐ | ☒ | ☐ | |
| 1D | Release first call. Keep second call.<br>Hang up 322 and only 399 and 323 are in call<br>Check that 399 & 323 are still in a call, check display. | ☐ | ☒ | ☐ | |
| 2 | Repeat the test 1C to 1D but using the call server feature | ☒ | ☐ | ☐ | Hold, enquiry, broker call functionality are not supported within call server for SIP device |
| 3 | **Three party conferences initiated from OXO set**<br>With 322 call 398, take the call and don't release it.<br><br>With 322 call 324, take the call and don't release it too.<br><br>With 322 start a conference.<br><br>Check that 322, 323 and 398 are in conference.<br>Check audio and display. | ☐ | ☒ | ☐ | SIP device display is not updated when OXO users initiate the conference |
| 4A | **Three party conferences initiated from SIP set with local feature** (if applicable)<br><br>With 398 call 322 take the call and don't release it.<br><br>With 398 call 323, take the call and don't release it too. | ☐ | ☒ | ☐ | Display on IP Touch sets doesn't show the second user in the conference initiated on the SIP device |

| Test Case Id | Test Case | N/A | OK | NOK | Comment |
|---|---|---|---|---|---|
| | With 398 start a conference by the local feature<br><br>Check that 322, 323 and 398 are in conference.<br>Check audio and display. | | | | |
| **4B** | **Three party conferences initiated from SIP set with local feature** | ☒ | ☐ | ☐ | Conference feature is not supported within call server for SIP device |
| **5** | **Meet Me conference**<br><br>With 399 call the Meet me Conference bridge dialing prefix 68 and follow instruction to open the bride.<br><br>With 398 join the conference bridge by dialing prefix 69 and enter access code.<br><br>With 322 join the conference bridge by dialing prefix 69 and enter access code.<br><br>Check that 322, 398 and 399 are in conference. | ☐ | ☒ | ☐ | |

## 8.6 Call Transfer

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 and Semi Attended transfer are not supported for SIP phones on OmniPCX Office.

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.

| 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 | OK | |
| 4 | SIP | SIP | SIP | OK | |
| 5 | SIP | OXO | OXO | OK | |
| 6 | Ext Call | OXO | SIP | OK | |
| 7 | SIP | OXO | SIP | OK | |

## 8.7 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 set call to attendant**<br><br>From SIP set 398 dial “9” (attendant call prefix)<br><br>Check audio and display | ☐ | ☒ | ☐ | Display on 398 shows internal number/name of Attendant extension |
| 2 | **2nd incoming call while in conversation with attendant**<br><br>While SIP set 398 is in conversation with the attendant, from IP Touch 323 call 398<br>Answer the call and check audio and display | ☐ | ☒ | ☐ | |
| 3 | **SIP set call to attendant, attendant transfers to OXO set, semi-attended**<br><br>From SIP set 398 dial “9” (attendant call prefix) and answer.<br><br>Attendant transfer semi-attended to IP Touch 323<br>Answer the call and check audio and display | ☐ | ☒ | ☐ | |
| 4 | **SIP set call to attendant, attendant transfers to OXO set, attended**<br><br>From SIP set 398 dial “9” (attendant call prefix) and answer<br><br>Attendant transfer attended to IP Touch 323<br>Check audio and display | ☐ | ☒ | ☐ | |
| 5 | **OXO set calls to attendant, attendant transfers to SIP set, attended**<br><br>From IP Touch 323 dial “9” (attendant call prefix) and answer<br><br>Attendant transfer attended to SIP set 398<br>Check audio and display | ☐ | ☒ | ☐ | |
| 6 | **External ISDN Call to attendant, attendant transfers to SIP set, attended**<br><br>ISDN incoming call to the attendant.<br><br>From the attendant call SIP set 398 and transfer attended<br>Check audio and display | ☐ | ☒ | ☐ | |
| 7 | **SIP set call to attendant, attendant transfers to External**<br><br>From SIP set 398, dial “9” (attendant call prefix) and answer<br><br>From the attendant, call an external ISDN destination and transfer semi-attended<br>Answer and check audio and display. | ☐ | ☒ | ☐ | |

## 8.8 Voice Mail

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

The default Voice Mail number is 500, and this service is enabled on SIP sets 398, 399 and OXO 322.

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<br>With SIP set 399 call the Voice Mail at 500 and follow the Voice guide in order to modify the default password.<br><br>When modification is accepted hang-up.<br><br>Recall the voice mail and try to log with a wrong password. Check the rejection.<br><br>Recall the voice mail and try to log with the right password. Check the service access. | ☐ | ☒ | ☐ | |
| 2 | Message display activation, MWI (1):<br>With SIP set 398 call the Voice Mail at 500.<br>Follow the instructions in order to send a voice message in SIP set 399 boxes.<br><br>Check that the MWI on 399 is activated. | ☐ | ☒ | ☐ | . |
| 3 | Message consultation<br>With SIP set 399 call the Voice Mail at 500.<br>Follow the instructions in order to listen your voice message leaved during the previous test. Check that your can listen it and delete.<br><br>Check that MWI display is disabled on 399 after message cancellation. | ☐ | ☒ | ☐ | |
| 4 | SIP call to a OXO user forwarded to Voice Mail<br>Forward the IP Touch 322 to Voice Mail by dialing *61500 (*61 prefix + <Voice Mail number>).<br><br>With SIP set 399 call 322 and check that you are immediately forwarded to Voice Mail.<br>Check that you can leave a message<br><br>On 322 disable Voice Mail forwarding with *60 prefix. | ☐ | ☒ | ☐ | |
| 5 | OXO set call to a SIP user forwarded to Voice Mail<br>Forward the SIP set 399 to Voice Mail by dialing *61500 (*61 prefix + <Voice Mail number>).<br><br>With IP Touch 322 call 399 and check that you are immediately forwarded to Voice Mail.<br>Check that you can leave a message<br><br>On 399 disable Voice Mail forwarding with *60 prefix. | ☐ | ☒ | ☐ | |

**Notes:**
(1) On SIP sets, in order to enable the MWI feature, you have to configure the Voice Mail number.

Show the SIP set will react in case of a OXO reboot, Ethernet link failure.

| Test Case Id | Test Case | N/A | OK | NOK | Comment |
|---|---|---|---|---|---|
| 1 | **OXO Reboot**<br><br>Establish an incoming ISDN call with SIP set-1.<br><br>Reboot the OXO.<br><br>When the OXO is up again, re-establish an incoming ISDN call with SIPset-1 and check the audio. | ☐ | ☒ | ☐ | |
| 2 | **Ethernet link failure**<br><br>Establish an incoming ISDN call with SIP set-1.<br><br>Disconnect the Ethernet link of SIP set-1.<br><br>Check that the incoming call is presented to the attendant.<br><br>Reconnect the Ethernet link of SIP set-1.<br><br>Re-establish an incoming ISDN call with SIP set-1 and check the audio. | ☐ | ☒ | ☐ | ISDN call is rerouted to attendant after register time-out |

# 9 Appendix A: AAPP member's Application Description

**TEMPORIS:**


IP800


IP600


IP200

**Configure phone using web interface**

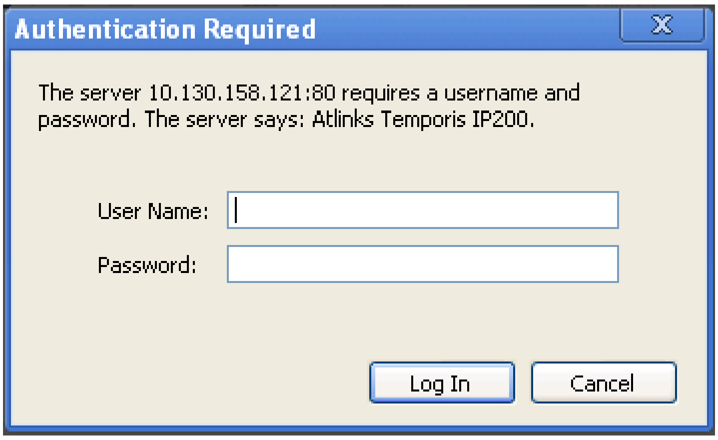
Find the phone IP address

1. Press OK key
2. The IP address is displayed on the screen

# 10 Appendix B: AAPP member's: Application Configuration Requirements

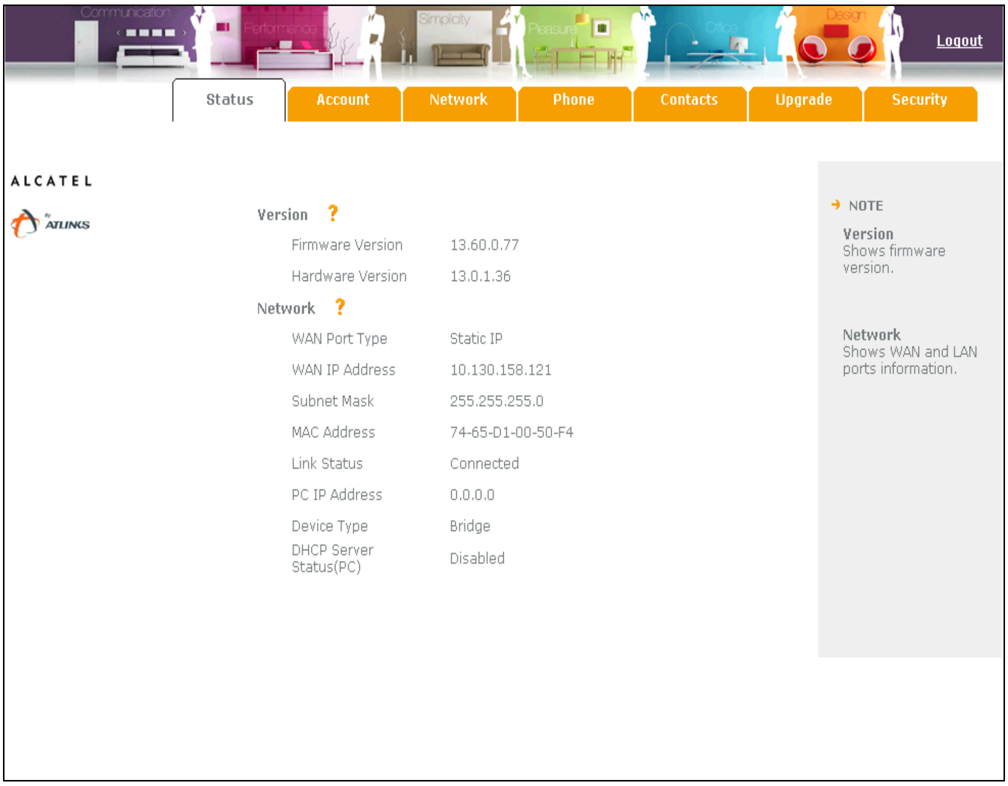
**Access to Admin Homepage (web interface)**

1. Open a web browser (Firefox, Internet Explorer, Safari…)
2. Enter the TEMPORIS IP address in the address bar of it. Ex: http://10.130.158.122



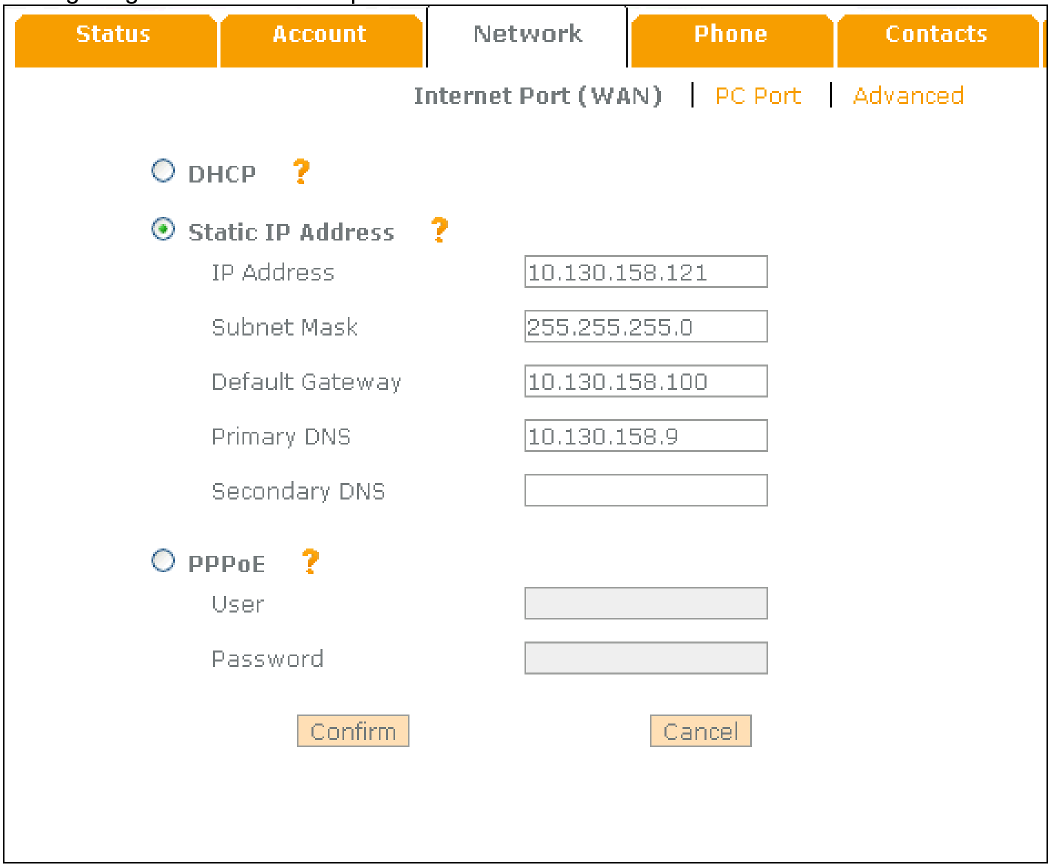

3. Enter the administrator login "admin"
4. Enter the administrator password (default value is admin)
5. You will access the Homepage of the TEMPORIS

**Main Page Details**

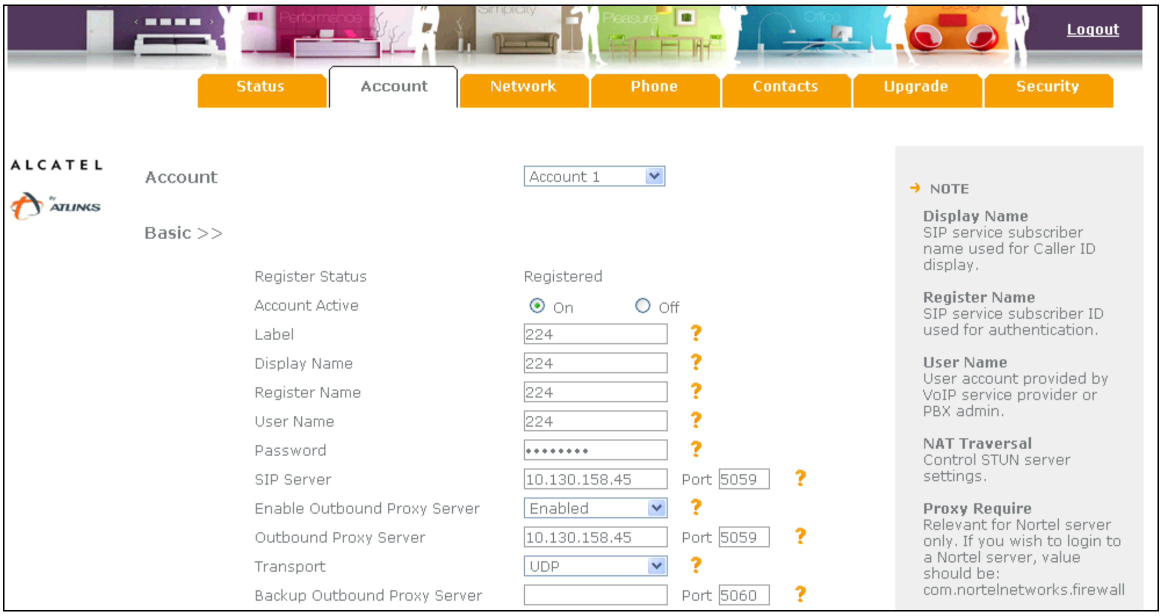


## Network Configuration Details

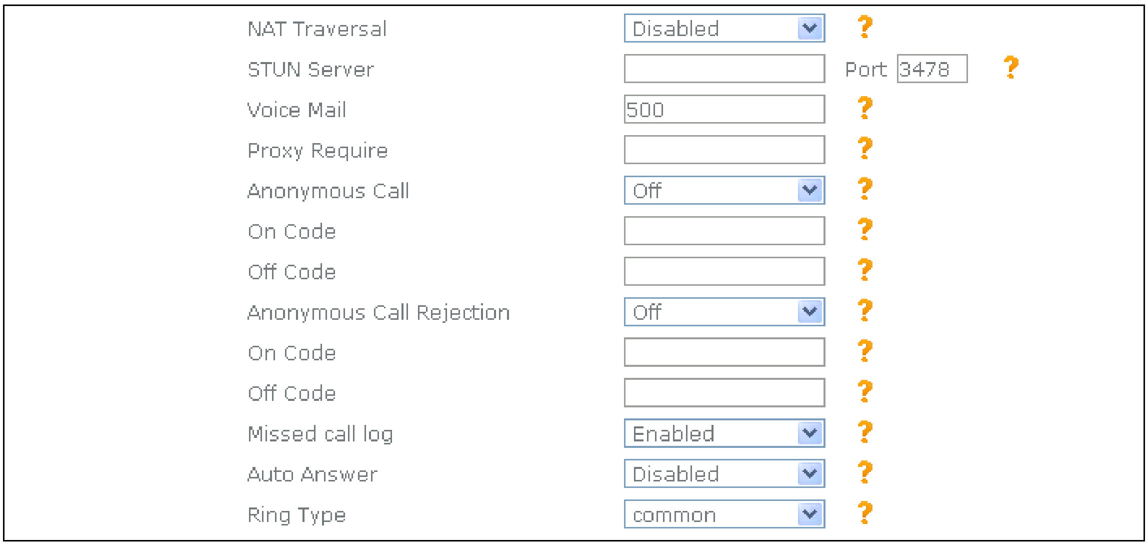
Configuring Various Network parameters



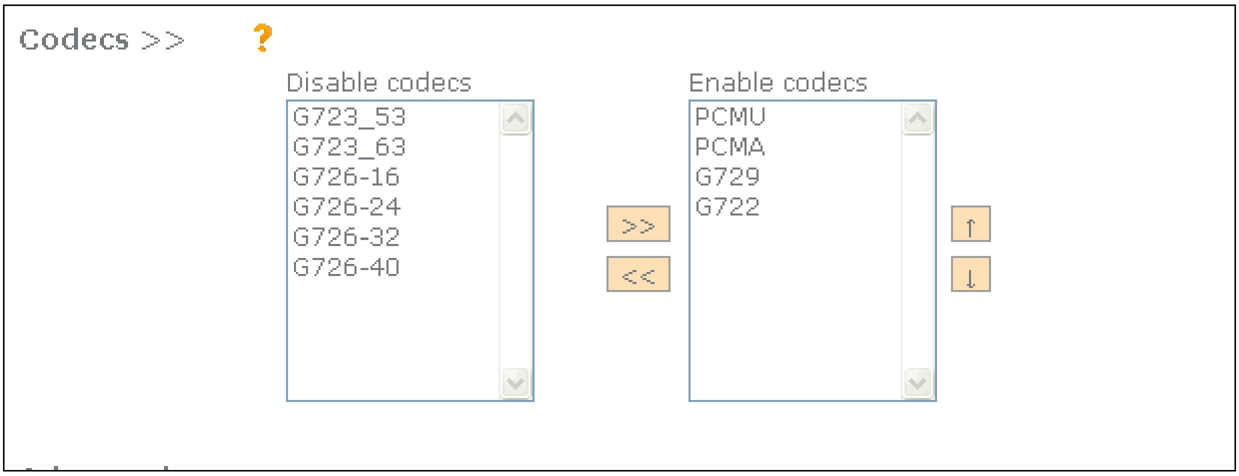

## SIP parameters configuration




Enable SIP profile and configure various parameters like registrar server, Proxy server, SIP transport protocol, Phone number, authentication Id and password.
The proxy and registrar server information can be name based or IP address based.




## CODEC configuration

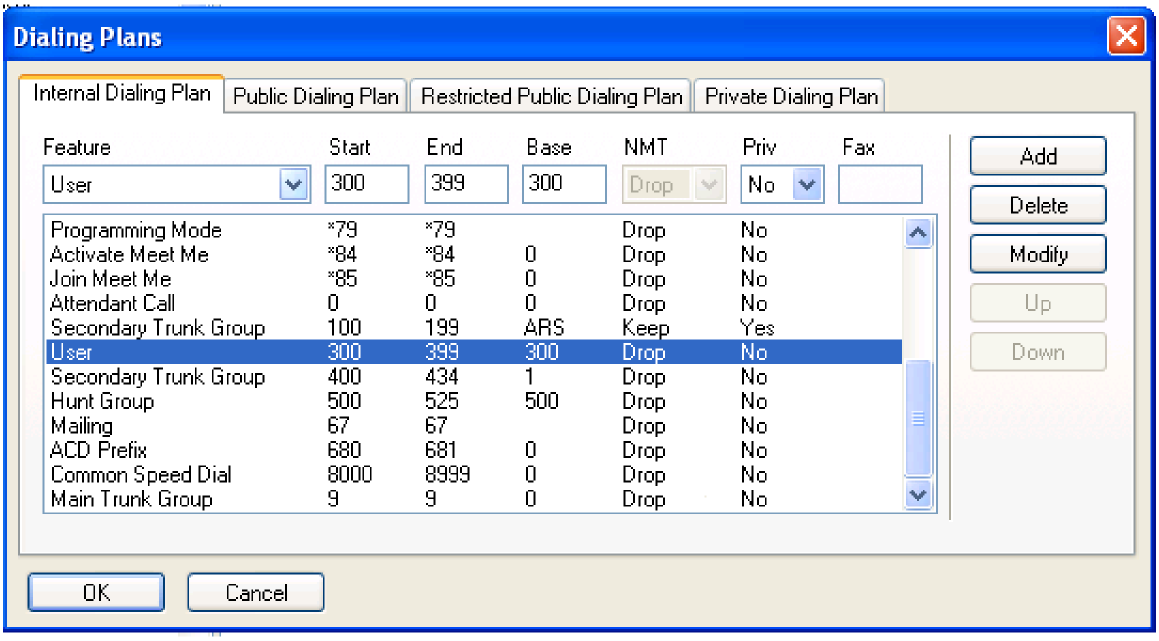



Enable the required CODECs enabled and keep the priority of the CODEC as required.

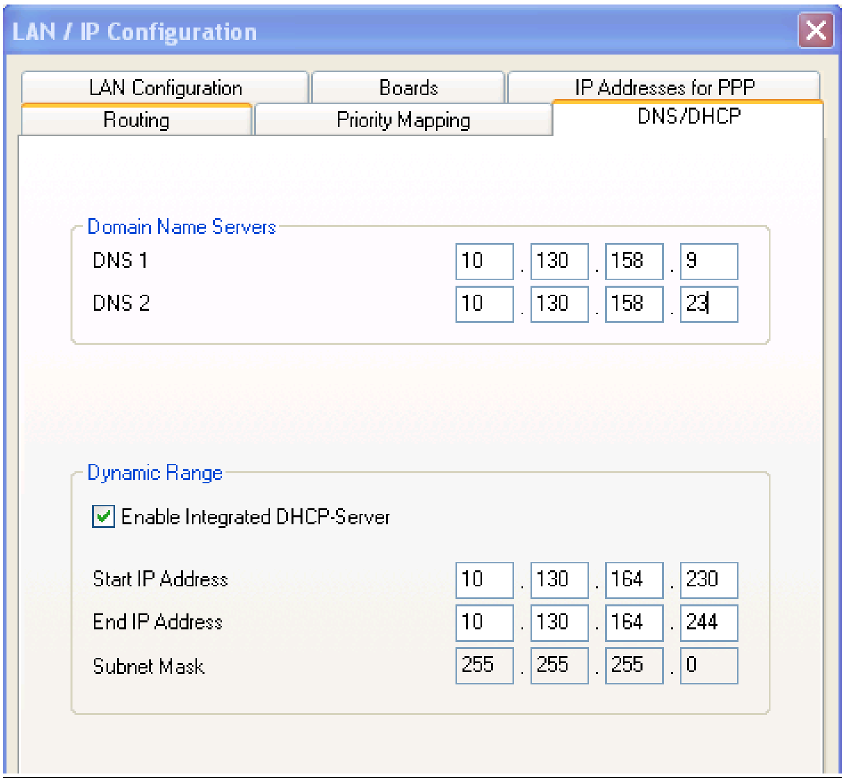
# 11 Appendix C: Alcatel-Lucent Platform: Configuration Requirements

## OXO Configuration

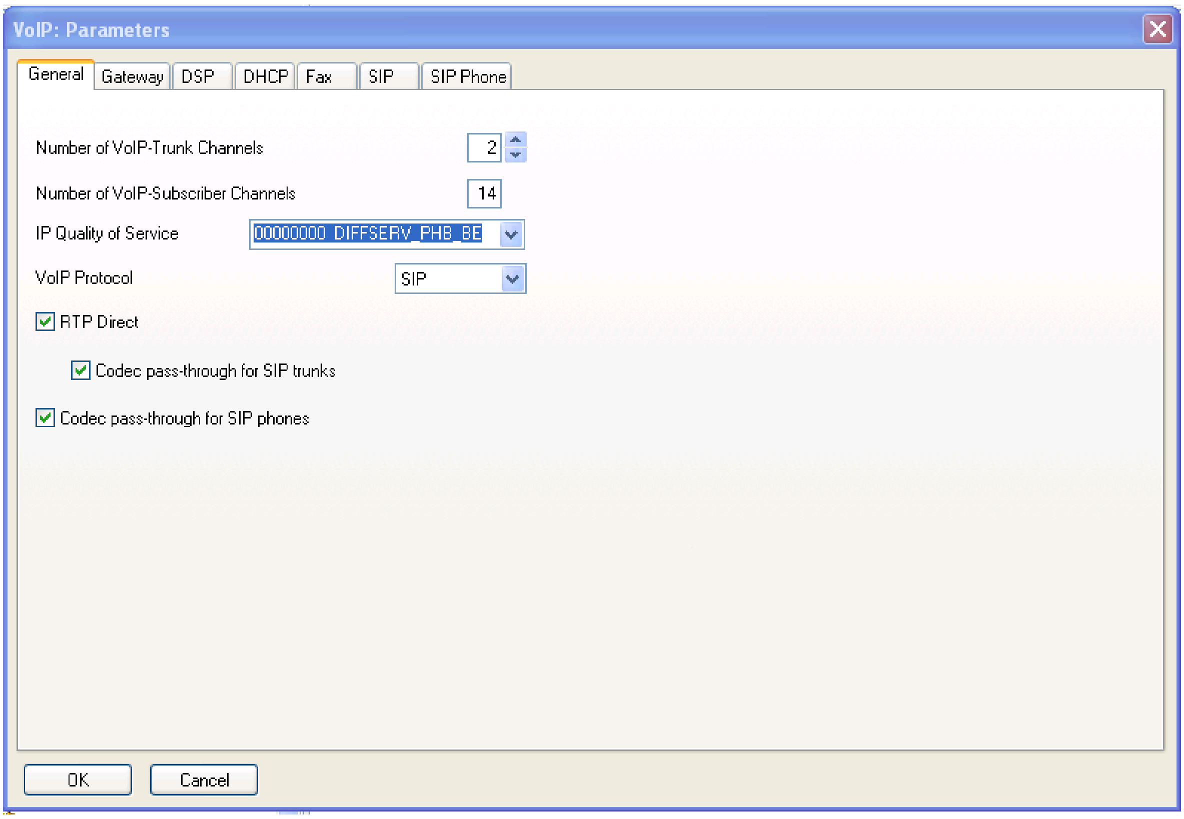
### 1. Dialing Plan:

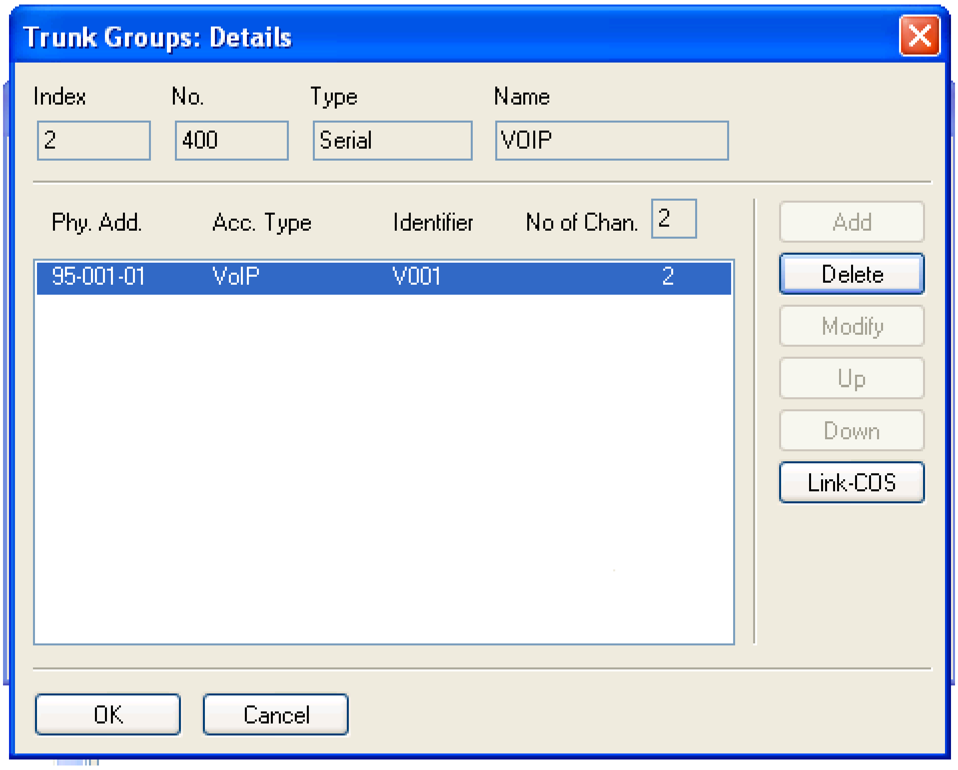



### 2. DNS/DHCP Configuration:

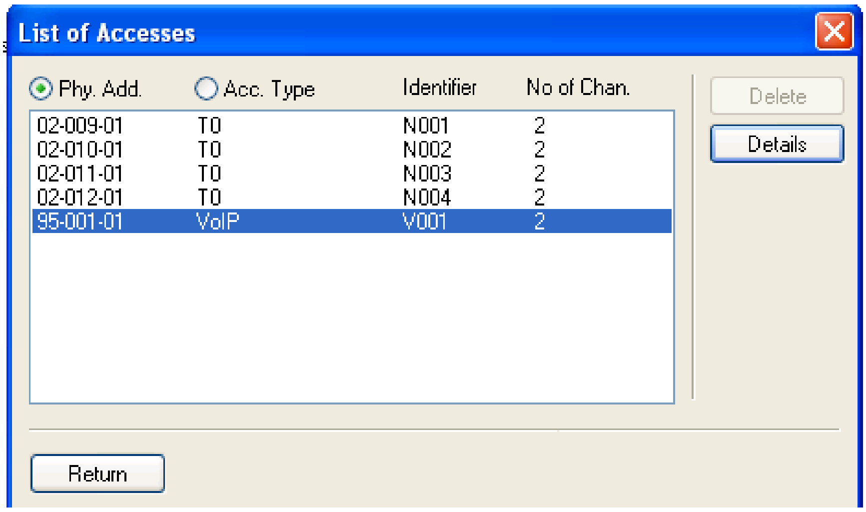


## 3. Trunk Configuration:

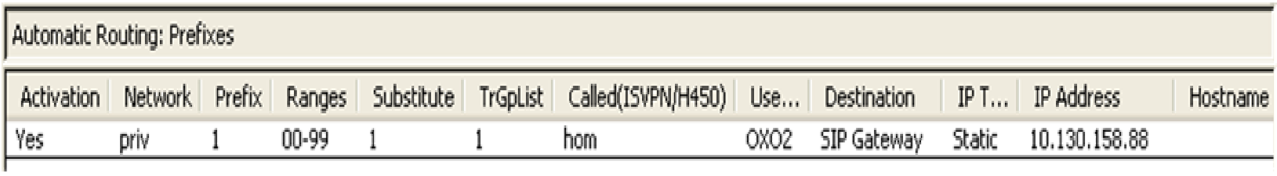


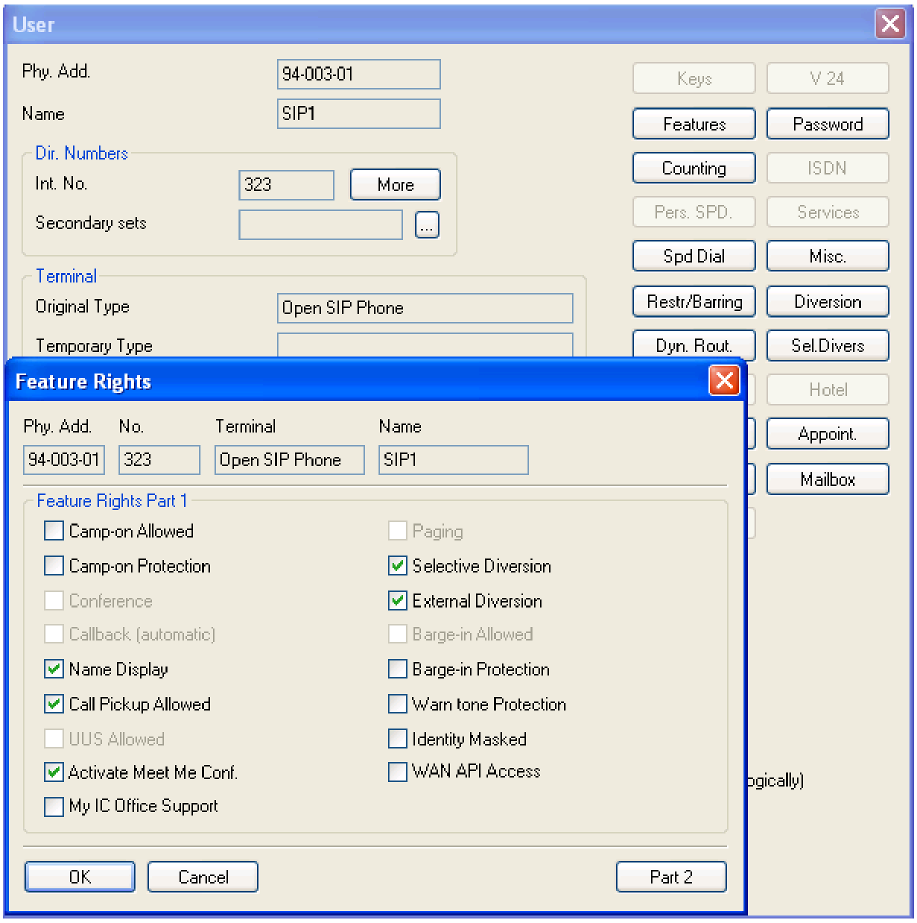
## 4. Trunk Access:



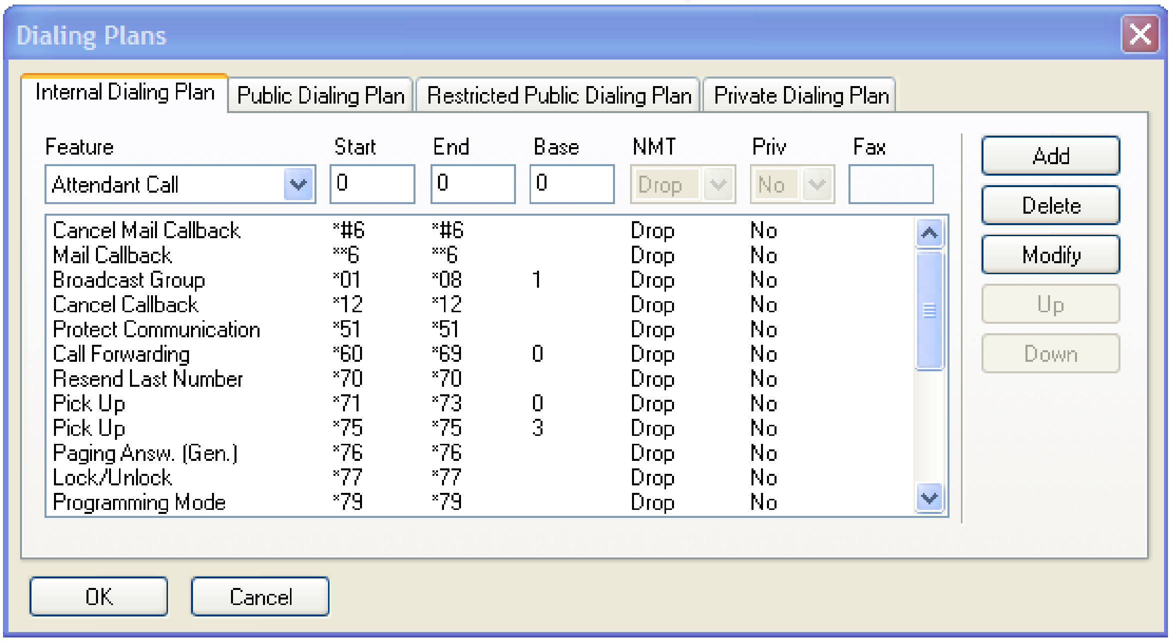

## 5. Network Call Configuration:

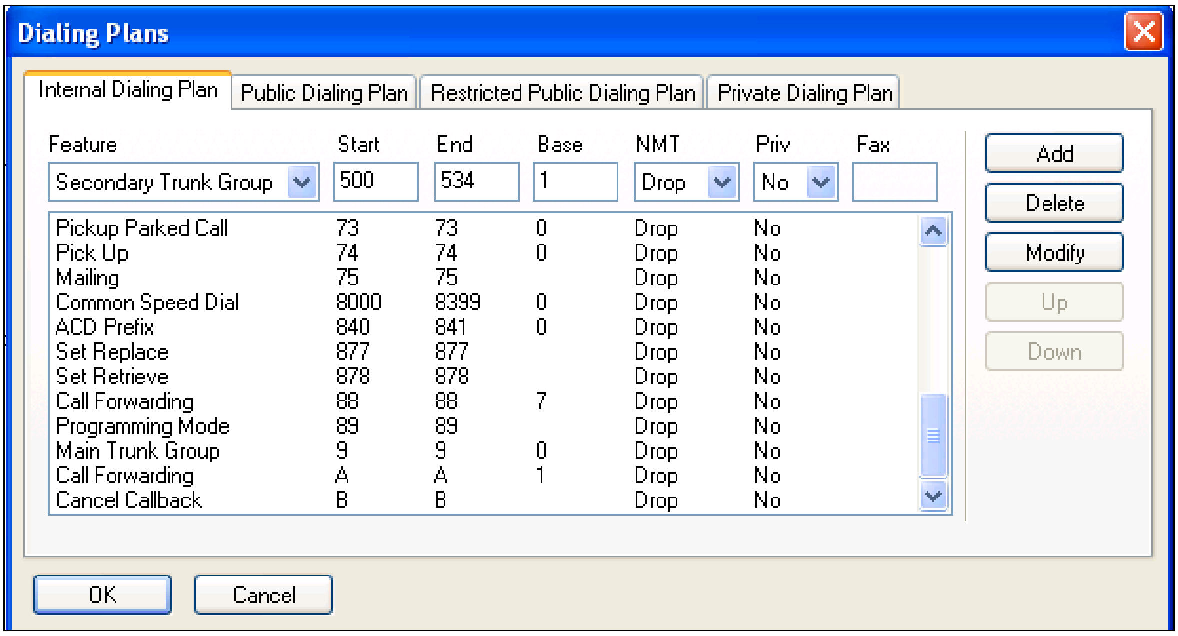



## 6. SIP Set Configuration:




## 7. Lists of OXO prefixes used in tests


Dialing Plans
Internal Dialing Plan
Public Dialing Plan
Restricted Public Dialing Plan
Private Dialing Plan
Feature
Start
End
Base
NMT
Priv
Fax
Attendant Call
0
0
0
Drop
No
Add
Delete
Modify
Up
Down
Cancel Mail Callback *#6 *#6 Drop No
Mail Callback **6 **6 Drop No
Broadcast Group *01 *08 1 Drop No
Cancel Callback *12 *12 Drop No
Protect Communication *51 *51 Drop No
Call Forwarding *60 *69 0 Drop No
Resend Last Number *70 *70 Drop No
Pick Up *71 *73 0 Drop No
Pick Up *75 *75 3 Drop No
Paging Answ. (Gen.) *76 *76 Drop No
Lock/Unlock *77 *77 Drop No
Programming Mode *79 *79 Drop No
OK
Cancel


Dialing Plans
Internal Dialing Plan
Public Dialing Plan
Restricted Public Dialing Plan
Private Dialing Plan
Feature
Start
End
Base
NMT
Priv
Fax
Secondary Trunk Group
500
534
1
Drop
No
Add
Delete
Modify
Up
Down
Pickup Parked Call 73 73 0 Drop No
Pick Up 74 74 0 Drop No
Mailing 75 75 Drop No
Common Speed Dial 8000 8399 0 Drop No
ACD Prefix 840 841 0 Drop No
Set Replace 877 877 Drop No
Set Retrieve 878 878 Drop No
Call Forwarding 88 88 7 Drop No
Programming Mode 89 89 Drop No
Main Trunk Group 9 9 0 Drop No
Call Forwarding A A 1 Drop No
Cancel Callback B B Drop No
OK
Cancel

# 12 Appendix D: AAPP member's escalation process

In Case of problem please contact

For more update information on phone & contact:
http://www.atlinks.com/en/alcatel-temporis-IP-range

# 13 Appendix E: AAPP program

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

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

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

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

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

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

### Web site

The Application Partner Portal is a website dedicated to the AAPP members and potential candidates. It can be accessed at this URL: http://applicationpartner.alcatel-lucent.com

## 13.2 Alcatel-Lucent.com

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

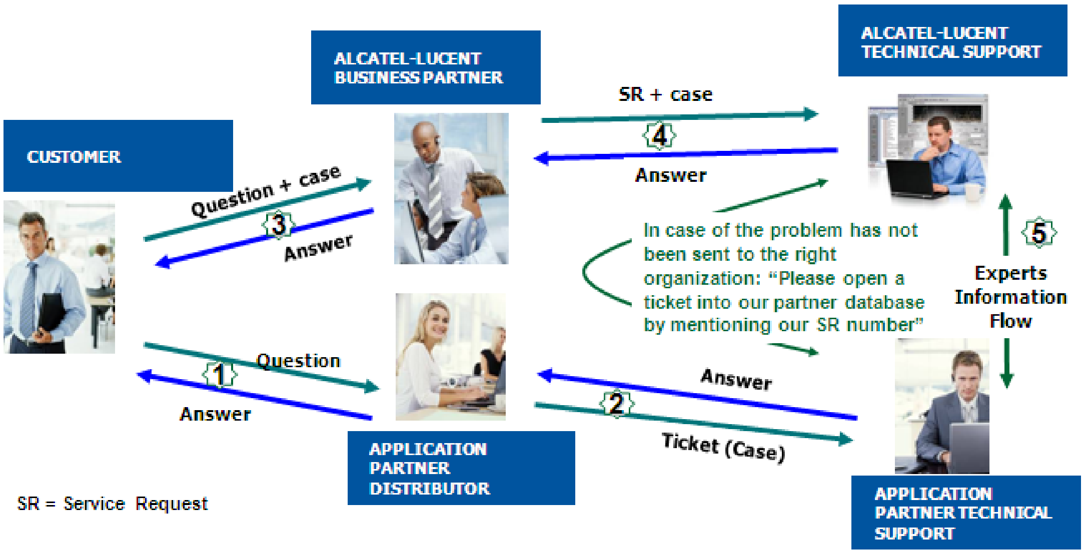
# 14 Appendix F: AAPP Escalation process

## 14.1 Introduction

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

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

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




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

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

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

This defines the scope of what has been certified.

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

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

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

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

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

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

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

Alcatel-Lucent reserves the right to close the case opened on his side if the investigations made on the Application Partner side are insufficient or do no exist.

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

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

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

Please check the availability of the Inter-Working Report on the AAPP (URL: https://private.applicationpartner.alcatel-lucent.com) or Enterprise Business Portal (Url: Enterprise Business Portal) web sites.

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

## 14.3 Escalation in all other cases

These cases can cover following situations:

1. An InterWorking Report exist but is not valid (see Chap 2 “Validity of an Interworking Report”)

2. The 3rd party company is referenced as AAPP participant but there is no official InterWorking Report (no IWR published on the Enterprise Business Portal for Business Partners or on the Alcatel-Lucent Application Partner web site) ,

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

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

## 14.4 Technical support access

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

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

Alcatel-Lucent Business Partners Support Center for countries:

<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>
<tr><td>Australia</td></tr>
<tr><td>Denmark</td></tr>
<tr><td>Ireland</td></tr>
<tr><td>Netherlands</td></tr>
<tr><td>South Africa</td></tr>
<tr><td>Norway</td></tr>
<tr><td>Poland</td></tr>
<tr><td>Sweden</td></tr>
<tr><td>Czech Republic</td></tr>
<tr><td>Estonia</td></tr>
<tr><td>Finland</td></tr>
<tr><td>Greece</td></tr>
<tr><td>Slovakia</td></tr>
<tr><td>Portugal</td></tr>
<tr><td>Spain</td><td>Spanish</td></tr>
</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 |

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