Alcatel·Lucent
Certified Application

# Alcatel Lucent Application Partner Program
# Inter-Working Report

## Partner: Zenitel
## Application type: VoIP DoorPhone
## Application name: TCIS-6, TCIS-2 and TCIS-1
## Alcatel-Lucent Platform: OmniPCX Enterprise

zenitel

The product and version listed have been tested with the Alcatel-Lucent Communication Server and the version specified hereinafter. The tests concern only the inter-working between the Application Partner product and the Alcatel-Lucent Communication platforms. The inter-working report is valid until the Application Partner issues a new version of such product (incorporating new features or functionality), or until Alcatel-Lucent issues a new version of such Alcatel-Lucent product (incorporating new features or functionality), whichever first occurs.

# Tests Identification

| | |
|---|---|
| Date of the tests | September 2013 |

| | |
|---|---|
| Alcatel-Lucent's representative | Residori Florian |
| Partner's representative | Adil Ashraf |

| | |
|---|---|
| Alcatel-Lucent Communication Platform | OmniPCX Enterprise |
| Alcatel-Lucent compatibility release | R10.1.1 - J2.603.20.i<br>R11.0 - K1.400.12.d |
| Partner's application version | V3.0.3.4 |
| Application Category | Terminals |

Author(s): Residori Florian
Reviewer(s): Denis Lienhart

**History:**

Edition 1: 01/09/2013: Creation of document

## Test results

☐ Passed ☐ Refused ☐ Postponed

☑ Passed with restrictions

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

## IWR validity extension

None

# 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 on:

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

# 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 0 "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**: Door Phone

**Application commercial name**: TCIS-6 ,TCIS-2 and TCIS-1

**Application version**: V3.0.3.4

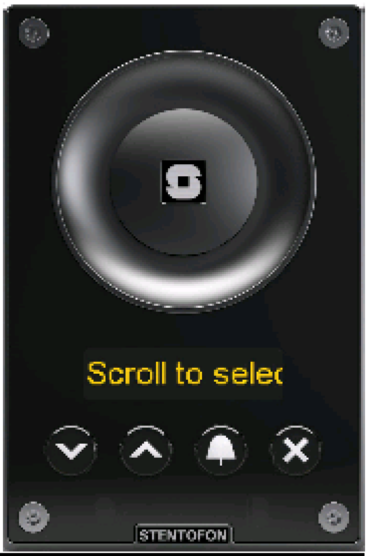
**Interface type**: SIP / Ethernet

**Interface version (if relevant)**:

**Brief application description**:

All IP stations in the STENTOFON Turbine series utilize the latest technology to create unparalleled audio quality. Some of the many features include: HD voice quality, Open Duplex, Active Noise Cancellation, MEMS microphone, a 10W Class D amplifier and our unique speaker grille design. These features, in conjunction with STENTOFON's 65+ years of experience with acoustic technology are only a few of the many factors that contribute to our superior audio quality.

**TCIS – 6**

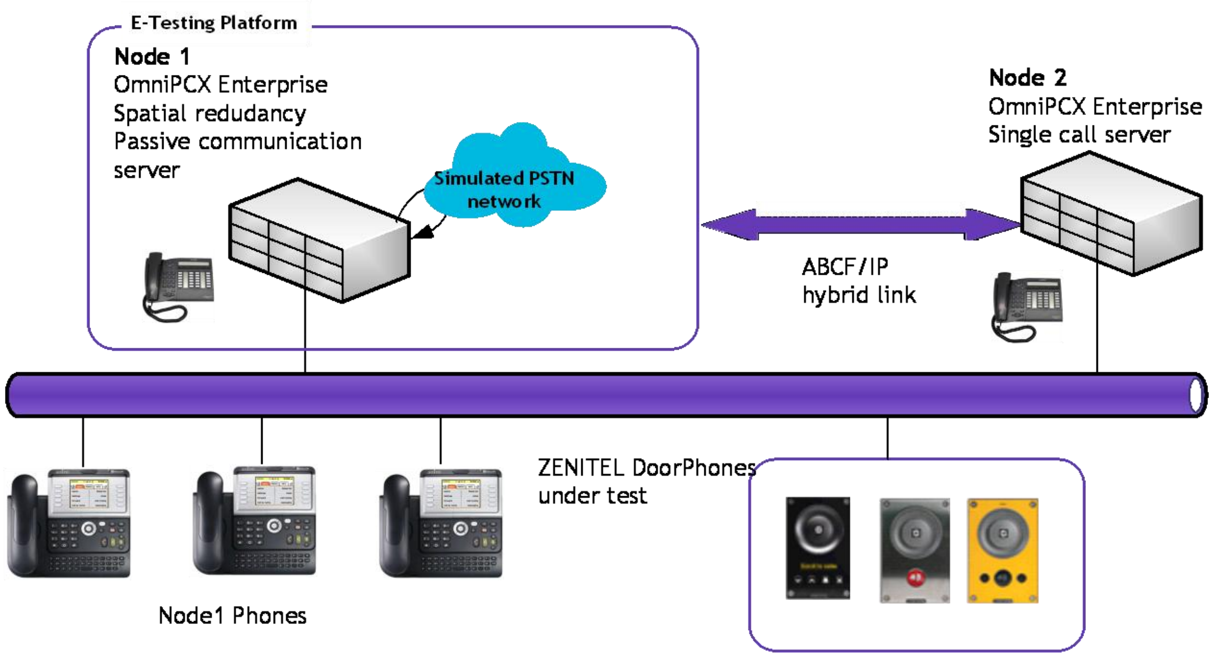


**TCIS – 2**


**TCIS – 1**

# 5 Tests environment

Figure 1 **Tests Environnent**



## 5.1 Hardware configuration

- **OmniPCX Entreprise**:
    - CS (Call Server Processing Unit)
    - GD (Gateway driver processing Unit
    - PRA T2 (ISDN Access)
    - MIX 2/4/4 (ISDN T0, digital & analog interfaces)
    - IPTouch, UA and Analog sets

## 5.2 Software configuration

**Alcatel-Lucent Communication Platform**: OmniPCX Enterprise R10.1.1 (J2.603.20.i), R11.0 (K1.400.12.d)

- **Partner Application**: V3.0.3.4

## 5.3 Video Testing

These DoorPhones doesn't have camera and hence video calls are not applicable.

# 6 Summary of test results

## 6.1 Summary of main functions supported

| Features | Status | Comments |
|---|---|---|
| Initialization including network configuration (static) | OK | |
| SIP registration | OK | |
| SIP authentication | OK | |
| Outgoing Call | OK | |
| Incoming Call | OK | |
| Trigger the relay during Outgoing call (DTMF code = 01) | OK | |
| Trigger the relay during Incoming call (DTMF code = 02) | OK | |
| Features During Conversation | OK | |
| Disconnect call after phone hang up | OK | |
| Transparent Video with SIP device | NA | |

## 6.2 Summary of problems

NIL

## 6.3 Summary of limitations

- Spatial redundancy with alternate proxy is not fully supported. Zenitel tries to use the first proxy even after the switchover. No calls are possible until a new registration. The doorphone should be deployed with spatial redundancy with DNS method.
- There is no led for the door status in TCIS-2 model.
- There is no timeout for disconnection in all the models. If we call the DoorPhone and it is not responding, Zenitel doorphone stays ringing. The calling user needs to end the call.

## 6.4 Notes, remarks

- These DoorPhones doesn't have camera and hence video calls are not applicable.
- RFC2833 is used for DTMF signaling.

**Environment setup for Generic Test:**

- The phone is configured as SIP Device
- It is configured as Administrator + Normal

# 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 | Door Phone IP configuration in DHCP mode | ☐ | ☒ | ☐ | |
| 2 | Door Phone IP configuration in Static mode | ☐ | ☒ | ☐ | |
| 3 | **SIP registration**<br><br>Configure DoorPhone with following parameters :<br>- Local IP address and mask<br>- OXE IP address and port 5060<br>- Extension Number and SIP password<br><br>Deactivate SIP authentication on OXE<br><br>Check the registration on the DoorPhone and on the wireshark Traces | ☐ | ☒ | ☐ | |
| 4 | SIP re-registration after timer expiry | ☐ | ☒ | ☐ | |
| 5 | **SIP registration**<br><br>Configure DoorPhone with following parameters :<br>- Local IP address and mask<br>- OXE IP address and port 5060<br>- Extension number and SIP password<br><br>Activate SIP authentication on OXE<br><br>Check the registration on the DoorPhone and on the wireshark traces. | ☐ | ☒ | ☐ | |
| 6 | **SIP set registration to OXE using a DNS or alternate proxy**<br><br>The phone is configured to use a domain name as registrar / proxy server address. The DNS IP addresses are the OXE CPU address.<br>In case of alternate proxy possibilities, the main and alternate proxy addresses are the OXE CPU address.<br>Tests are performed when first Call Server is active and then when second Call Server is active | ☐ | ☒ | ☐ | |

## 8.2 Duplication and Robustness

### 8.2.1 Test results

Check how the system will react in case of a CPU reboot, switchover or link failure etc.
The test system is configured with spatial redundancy (duplicate call servers on two different IP subnetworks).

Spatial redundancy can be configured in two ways:

- "Alternate Proxy method": Specify both CS MAIN addresses as primary and alternative proxy respectively. Requires that on non availability of primary proxy, secondary proxy is used. Requires ability to accept incoming calls from secondary proxy.
- "DNS method": Do not specify a proxy address, only SIP domain. Specify the CS MAIN address as first and second DNS server, respectively. Requires that (at least on non availability of current proxy) a new DNS request is issued for every message. Only MAIN CS will respond. Requires ability to accept incoming calls from secondary CS when it becomes new MAIN.

For each configuration, check:
Can new outgoing calls be made immediately after switchover?
Are existing calls maintained after switchover?
Are incoming calls (from new MAIN CS) accepted immediately after switchover?
Can existing call be modified (transfer, hang-up, etc.) after switchover?
Check if a session that has been started before switchover is maintained after switchover, i.e. does the new MAIN CS send session updates and is this accepted by the client?

| Test Case Id | Test Case | N/A | OK | NOK | Comments |
|---|---|---|---|---|---|
| **1** | Spatial redundancy with alternate proxy method | | | | |
| **A** | **Spatial redundancy, using "Alternate Proxy method", two SIP sets in conversation**<br>Configure Zenitel doorphone to use two SIP proxies (OXE call server a and OXE call server b IP addresses).<br><br>With Zenitel call another OXE user (IPTouch for example). Answer the call and check audio and display.<br><br>Switchover to standby call server using OXE "bascul" command (check first the database replication using OXE "twin" command). Check that the existing call is maintained.<br><br>Wait for a "session timer" expiration (this timer is negotiated between the INVITE and the OK message). Check that the call is maintained after this timer expiration. | ☐ | ☒ | ☐ | |

| Test Case Id | Test Case | N/A | OK | NOK | Comments |
|---|---|---|---|---|---|
| **B** | **Spatial redundancy, using “Alternate Proxy method”, one SIP set in conversation with a external party**<br>Configure Zenitel doorphone to use two SIP proxies (OXE call server a and OXE call server b IP addresses).<br><br>Zenitel doorphone calls a PSTN user.<br>Answer the call and check audio and display.<br>Switchover to standby call server using OXE “bascul” command (check first the database replication using OXE “twin” command). Check that the call is maintained.<br><br>Wait for a “session timer” expiration (this timer is negotiated between the INVITE and the OK message). Check that the call is maintained after this timer expiration. | ☐ | ☒ | ☐ | |
| **C** | **Spatial redundancy, using “Alternate Proxy method”, call after the switchover**<br>Configure Zenitel doorphone to use two SIP proxies (OXE call server a and OXE call server b IP addresses).<br><br>Switchover to standby call server using OXE “bascul” command (check first the database replication using OXE “twin” command).<br><br>Just after the switchover, with Zenitel doorphone call another OXE user (IPTouch for example). Answer the call and check audio and display. | ☐ | ☐ | ☒ | Zenitel tries to use the first proxy. No calls are possible until a new registration. |
| **D** | **Spatial redundancy, using “Alternate Proxy method”, call after the switchover and a registration timeout**<br>Configure Zenitel doorphone to use two SIP proxies (OXE call server a and OXE call server b IP addresses).<br><br>Switchover to standby call server using OXE “bascul” command (check first the database replication using OXE “twin” command).<br><br>Wait for a Registration period timeout. After this period, with Zenitel doorphone call another OXE user (IPTouch for example). Answer the call and check audio and display. | ☐ | ☒ | ☐ | |
| **2** | Spatial redundancy with alternate DNS method | | | | |

| Test Case Id | Test Case | N/A | OK | NOK | Comments |
|---|---|---|---|---|---|
| **A** | **Spatial redundancy, using "DNS method" and OXE used as DNS server, two SIP sets in conversation**<br>Configure Zenitel doorphone to use to use one SIP proxies (OXE call server node name). Configure Zenitel doorphone to use two DNS servers (OXE call server a and OXE call server b IP addresses).<br><br>With Zenitel call another OXE user (IPTouch for example). Answer the call and check audio and display.<br><br>Switchover to standby call server using OXE "bascul" command (check first the database replication using OXE "twin" command). Check that the existing call is maintained.<br><br>Wait for a "session timer" expiration (this timer is negotiated between the INVITE and the OK message). Check that the call is maintained after this timer expiration. | ☐ | ☒ | ☐ | NTP server should be configured with an IP address. |
| **B** | **Spatial redundancy, using "DNS method" and OXE used as DNS server, one SIP set in conversation with a external party**<br>Configure Zenitel doorphone to use to use one SIP proxies (OXE call server node name). Configure Zenitel doorphone to use two DNS servers (OXE call server a and OXE call server b IP addresses).<br><br>Zenitel doorphone calls a PSTN user.<br>Answer the call and check audio and display.<br>Switchover to standby call server using OXE "bascul" command (check first the database replication using OXE "twin" command). Check that the call is maintained.<br><br>Wait for a "session timer" expiration (this timer is negotiated between the INVITE and the OK message). Check that the call is maintained after this timer expiration. | ☐ | ☒ | ☐ | NTP server should be configured with an IP address. |
| **C** | **Spatial redundancy, using "DNS method" and OXE used as DNS server, call after the switchover**<br>Configure Zenitel doorphone to use to use one SIP proxies (OXE call server node name). Configure Zenitel doorphone to use two DNS servers (OXE call server a and OXE call server b IP addresses).<br><br>Switchover to standby call server using OXE "bascul" command (check first the database replication using OXE "twin" command).<br><br>Just after the switchover, with Zenitel doorphone call another OXE user (IPTouch for example). Answer the call and check audio and display. | ☐ | ☒ | ☐ | NTP server should be configured with an IP address. |

| Test Case Id | Test Case | N/A | OK | NOK | Comments |
|---|---|---|---|---|---|
| D | **Spatial redundancy, using “DNS method” and OXE used as DNS server, call after the switchover and a registration timeout**<br>Configure Zenitel doorphone to use to use one SIP proxies (OXE call server node name). Configure Zenitel doorphone to use two DNS servers (OXE call server a and OXE call server b IP addresses).<br><br>Switchover to standby call server using OXE “bascul” command (check first the database replication using OXE “twin” command).<br><br>Wait for a Registration period timeout. After this period, with Zenitel doorphone call another OXE user (IPTouch for example). Answer the call and check audio and display. | ☐ | ☒ | ☐ | NTP server should be configured with an IP address. |
| E | **Spatial redundancy, using “DNS method” with a delegation DNS server, two SIP sets in conversation**<br>Configure Zenitel doorphone to use to use one SIP proxies (OXE call server node name). Configure Zenitel doorphone to use to use one DNS server (An external DNS server delegates the DNS request to OXE call server a and OXE call server b).<br><br>With Zenitel call another OXE user (IPTouch for example). Answer the call and check audio and display.<br><br>Switchover to standby call server using OXE “bascul” command (check first the database replication using OXE “twin” command). Check that the existing call is maintained.<br><br>Wait for a “session timer” expiration (this timer is negotiated between the INVITE and the OK message). Check that the call is maintained after this timer expiration. | ☐ | ☒ | ☐ | |
| F | **Spatial redundancy, using “DNS method” with a delegation DNS server, one SIP set in conversation with a external party**<br>Configure Zenitel doorphone to use to use one SIP proxies (OXE call server node name). Configure Zenitel doorphone to use to use one DNS server (An external DNS server delegates the DNS request to OXE call server a and OXE call server b).<br><br>Zenitel doorphone calls a PSTN user.<br>Answer the call and check audio and display.<br>Switchover to standby call server using OXE “bascul” command (check first the database replication using OXE “twin” command). Check that the call is maintained.<br><br>Wait for a “session timer” expiration (this timer is negotiated between the INVITE and the OK message). Check that the call is maintained after this timer expiration. | ☐ | ☒ | ☐ | |

| Test Case Id | Test Case | N/A | OK | NOK | Comments |
|---|---|---|---|---|---|
| G | **Spatial redundancy, using "DNS method" with a delegation DNS server, call after the switchover**<br>Configure Zenitel doorphone to use to use one SIP proxies (OXE call server node name). Configure Zenitel doorphone to use to use one DNS server (An external DNS server delegates the DNS request to OXE call server a and OXE call server b).<br><br>Switchover to standby call server using OXE "bascul" command (check first the database replication using OXE "twin" command).<br><br>Just after the switchover, with Zenitel doorphone call another OXE user (IPTouch for example). Answer the call and check audio and display. | ☐ | ☒ | ☐ | |
| H | **Spatial redundancy, using "DNS method" with a delegation DNS server, call after the switchover and a registration timeout**<br>Configure Zenitel doorphone to use to use one SIP proxies (OXE call server node name). Configure Zenitel doorphone to use to use one DNS server (An external DNS server delegates the DNS request to OXE call server a and OXE call server b).<br><br>Switchover to standby call server using OXE "bascul" command (check first the database replication using OXE "twin" command).<br><br>Wait for a Registration period timeout. After this period, with Zenitel doorphone call another OXE user (IPTouch for example). Answer the call and check audio and display. | ☐ | ☒ | ☐ | |
| 3 | Switchover to PCS | | | | |
| A | **Passive call server backup, using "Alternate Proxy method", two SIP sets in conversation**<br>Configure Zenitel doorphone to use two SIP proxies (OXE call server a and OXE call server PCS IP addresses). Zenitel doorphone is part of IP domain with a PCS backup.<br><br>With Zenitel doorphone call another OXE user (IPTouch for example). Answer the call and check audio and display.<br><br>Stop OXE call server a and call server b. Check that the existing call is maintained.<br><br>Wait for a "session timer" expiration (this timer is negotiated between the INVITE and the OK message). Check that the call is maintained after this timer expiration. | ☐ | ☒ | ☐ | |

| Test Case Id | Test Case | N/A | OK | NOK | Comments |
|---|---|---|---|---|---|
| B | **Passive call server backup, using "Alternate Proxy method", one SIP set in conversation with a external party**<br>Configure Zenitel doorphone to use two SIP proxies (OXE call server a and OXE call server PCS IP addresses). Zenitel doorphone is part of IP domain with a PCS backup.<br><br>Zenitel doorphone call a PSTN user.<br>Answer the call and check audio and display.<br>Stop OXE call server a and call server b. Check that the call is maintained.<br><br>Wait for a "session timer" expiration (this timer is negotiated between the INVITE and the OK message).<br>Check that the call is maintained after this timer expiration. | ☐ | ☒ | ☐ | |
| C | **Passive call server backup, using "Alternate Proxy method", call after the switchover**<br>Configure Zenitel doorphone to use two SIP proxies (OXE call server a and OXE call server PCS IP addresses). Zenitel doorphone is part of IP domain with a PCS backup.<br><br>Stop OXE call server a and call server b.<br><br>Just after the switchover, with Zenitel doorphone call another OXE user (IPTouch for example). Answer the call and check audio and display. | ☐ | ☐ | ☒ | Zenitel tries to use the first proxy. No calls are possible until a new registration. |
| **4** | Partner SIP endpoint reboot | | | | |
| A | **Partner SIP set reboot**<br>Reboot Zenitel doorphone. When Zenitel doorphone comes back in service, call another OXE user (IPTouch for example).<br><br>Check that Zenitel doorphone is registered and the call establishment. | ☐ | ☒ | ☐ | |
| **5** | Network failure | | | | |
| A | **Temporary Link between OXE and the partner SIP set**<br>Disconnect the link between Zenitel doorphone and OXE.<br>Check that SIPset-1 becomes out of service after a keep-alive or a registration period.<br>When Zenitel doorphone comes back in service, call Zenitel doorphone.<br>Reconnect the link between Zenitel doorphone and OXE.<br>Check that Zenitel doorphone becomes in service registration period. | ☐ | ☒ | ☐ | |

### 8.2.2 Recommendation

Zenitel doorphone supports spatial redundancy via "DNS method" and delegation dns server and supports PCS. The next tests will be done with the OXE domain name configured as SIP proxy/registrar.

## 8.3 Calls from DoorPhone

These tests check that the phones can interact with the DoorPhone.

| Test Case Id | Test Case | N/A | OK | NOK | Comment |
|---|---|---|---|---|---|
| **1** | Call from DoorPhone to IP Touch<br><br>Configure the system law to A-law<br>Check that the call is established in G711 A-law<br>Check audio quality and hold option<br><br>Release the call from IP Touch | ☐ | ☒ | ☐ | |
| **2** | Call from DoorPhone to UA Phone<br><br>Configure the system law to A-law<br>Check that the call is established in G711 A-law<br>Check audio quality and hold option<br><br>Release the call from UA Phone | ☐ | ☒ | ☐ | |
| **3** | Call from DoorPhone to SIP device<br><br>Configure the system law to A-law<br>Check that the call is established in G711 A-law<br>Check audio quality and hold option<br><br>Release the call from SIP device | ☐ | ☒ | ☐ | |
| **4** | Call from DoorPhone to IP Touch<br><br>Configure the system law to µ-law<br>Check that the call is established in G711 µ-law<br>Check audio quality and hold option<br><br>Release the call from IP Touch | ☐ | ☒ | ☐ | |
| **5** | Call from DoorPhone to UA Phone<br><br>Configure the system law to µ-law<br>Check that the call is established in G711 µ-law<br>Check audio quality and hold option<br><br>Release the call from UA Phone | ☐ | ☒ | ☐ | |
| **6** | Call from DoorPhone to SIP device<br><br>Configure the system law to µ-law<br>Check that the call is established in G711 µ-law<br>Check audio quality and hold option<br><br>Release the call from SIP device | ☐ | ☒ | ☐ | |
| **7** | Call from DoorPhone to IP Touch<br><br>Wait for the DoorPhone timer to release the call | ☐ | ☒ | ☐ | Timer can be configured in the GUI of the Doorphone and the call is released properly after the specified time. |
| **8** | Call from DoorPhone to UA Phone<br><br>Wait for the DoorPhone timer to release the call | ☐ | ☒ | ☐ | |
| **9** | Call from DoorPhone to SIP device<br><br>Wait for the DoorPhone timer to release the call | ☐ | ☒ | ☐ | |

| | | | | | |
|---|---|---|---|---|---|
| **10** | Call from DoorPhone to IP Touch<br><br>Open the latch by DTMF (Call is released) | ☐ | ☒ | ☐ | We checked the circuit continuity with the help of Multimeter.<br><br>The door latch LED is present only in two models TCIS-6 &TCIS-1 and not in TCIS-2.<br><br>When we press the dtmf for opening the door the LED glows and if we press the dtmf for closing then the LED stops. |
| **11** | Call from DoorPhone to UA<br><br>Open the latch by DTMF(Call is released) | ☐ | ☒ | ☐ | |
| **12** | Call from DoorPhone to SIP device<br><br>Open the latch by DTMF(Call is released) | ☐ | ☒ | ☐ | |

## 8.4 Calls to DoorPhone

These tests check that the phones can interact with the DoorPhone.

| Test Case Id | Test Case | N/A | OK | NOK | Comment |
|---|---|---|---|---|---|
| **1** | Call to DoorPhone from IP Touch<br><br>Check audio quality<br>Release the call from IP Touch | ☐ | ☒ | ☐ | |
| **2** | Call to DoorPhone from UA Phone<br><br>Check audio quality<br>Release the call from UA Phone | ☐ | ☒ | ☐ | |
| **3** | Call to DoorPhone from SIP device<br><br>Check audio quality<br>Release the call from SIP device | ☐ | ☒ | ☐ | |
| **4** | Call to DoorPhone from IP Touch<br><br>Wait for the DoorPhone timer to release the call | ☐ | ☒ | ☐ | |
| **5** | Call to DoorPhone from UA Phone<br><br>Wait for the DoorPhone timer to release the call | ☐ | ☒ | ☐ | |
| **6** | Call to DoorPhone from SIP device<br><br>Wait for the DoorPhone timer to release the call | ☐ | ☒ | ☐ | |
| **7** | Call to DoorPhone from IP Touch<br><br>Open the latch by DTMF (Call is released) | ☐ | ☒ | ☐ | When we press the dtmf for opening the door the LED glows and if we press the dtmf for closing then the LED stops. |
| **8** | Call to DoorPhone from UA Phone<br><br>Open the latch by DTMF(Call is released) | ☐ | ☒ | ☐ | |
| **9** | Call to DoorPhone from SIP device<br><br>Open the latch by DTMF(Call is released) | ☐ | ☒ | ☐ | |
| **10** | Call to DoorPhone from IP Touch<br><br>Check that the call can be continued after pressing digit from IPTouch | ☐ | ☒ | ☐ | |
| **11** | Call to DoorPhone from UA Phone<br><br>Check that the call can be continued after pressing digit from UA Phone | ☐ | ☒ | ☐ | |
| **12** | Call to DoorPhone from SIP device<br><br>Check that the call can be continued after pressing digit from SIP device | ☐ | ☒ | ☐ | |
| **13** | Call to DoorPhone from IP Touch<br><br>Check that the mode (day/night mode) of the DoorPhone can be changed from IP Touch. | ☒ | ☐ | ☐ | |
| **14** | Call to DoorPhone from UA Phone<br><br>Check that the mode (day/night mode) of the DoorPhone can be changed from UA Phone | ☒ | ☐ | ☐ | |

| | | | | | |
|---|---|---|---|---|---|
| **15** | Call to DoorPhone from SIP device<br><br>Check that the mode (day/night mode) of the DoorPhone can be changed from SIP device | ☒ | ☐ | ☐ | |
| **16** | Call to DoorPhone from SIP device<br><br>Configure the DoorPhone to answer the incoming INVITE with a 180 RINGING.<br><br>Check ring back tone on the SIP device. | ☐ | ☒ | ☐ | |
| **17** | Call to DoorPhone from SIP device<br><br>Configure the DoorPhone to answer the incoming INVITE with a 183 SESSION PROGRESS.<br><br>Check ring back tone on the SIP device. | ☒ | ☐ | ☐ | |
| **18** | Call from external number(T0/T2) to DoorPhone | ☐ | ☒ | ☐ | |
| **19** | Call from attendant to DoorPhone | ☐ | ☒ | ☐ | |
| **20** | Incoming external call (T0/T2 for example) to an attendant phone set which transfers the call to the Door Phone.<br><br>Call is properly established. | ☐ | ☒ | ☐ | |

## 8.5 In conversation scenarios

| Test Case Id | Test Case | N/A | OK | NOK | Comment |
|---|---|---|---|---|---|
| **1** | Call from DoorPhone to UA Phone after the call is attended press the DTMF prefix and On the DoorPhone press the same call button (it releases the first call, and the second one gets busy) | ☐ | ☒ | ☐ | |
| **2** | Call from DoorPhone to busy UA Phone<br><br>Check that the call is released (gets busy) | ☐ | ☒ | ☐ | |
| **3** | Call from DoorPhone to IP Touch<br>Put on hold<br>Take back the call and check the audio<br>Open the Latch<br>Release the call | ☐ | ☒ | ☐ | |
| **4** | Call from DoorPhone to UA Phone<br>Put on hold<br>Take back the call and check the audio<br>Open the Latch<br>Release the call | ☐ | ☒ | ☐ | |
| **5** | Call from DoorPhone to SIP device<br>Put on hold<br>Take back the call and check the audio<br>Open the Latch<br>Release the call | ☐ | ☒ | ☐ | |
| **6** | Call from DoorPhone to IP Touch<br>From IPTouch call UA Phone (answer) and press transfer button in IP Touch<br>Check the audio<br>Open the Latch<br>Release the call | ☐ | ☒ | ☐ | |
| **7** | Make call forwarding from SIP device to other extension.<br>Call from DoorPhone to SIP device.<br>Check the audio<br>Open the Latch<br>Release the call | ☐ | ☒ | ☐ | |
| **8** | OXE Call Server CPU switches over while SIP phone in idle.<br><br>Check the SIP phone behavior after a switch from the OXE main to standby CPU.<br>The phone must be able to make and receive a call after the switch over. | ☐ | ☒ | ☐ | |
| **9** | OXE Call Server CPU switch over while SIP phone in conversation with an IPTouch.<br><br>Check the SIP phone behavior after a switch from the OXE main to standby CPU.<br>The call is still active. The phone can make and receive a second call and switch from one to another.<br>After on hook, the phone must be able to make and receive a call after the switch over. | ☐ | ☒ | ☐ | DoorPhone is getting registered to the main OXE with the managed registration timer and calls can be made. |

| | | | | | |
|---|---|---|---|---|---|
| **10** | OXE Call Server reboot while SIP phone in idle.<br><br>Check the phone behavior when the OXE Call Server reboots (without standby CPU).<br>As soon as the Call Server is running again, the phone is able to make and receive a call. | ☐ | ☒ | ☐ | |
| **11** | OXE Call Server reboot while SIP phone in conversation with an IPTouch.<br><br>Check the phone behavior when the OXE Call Server reboots (without standby CPU).<br>The call is released.<br>As soon as the Call Server is running again, the phone is able to make and receive a call | ☐ | ☒ | ☐ | |

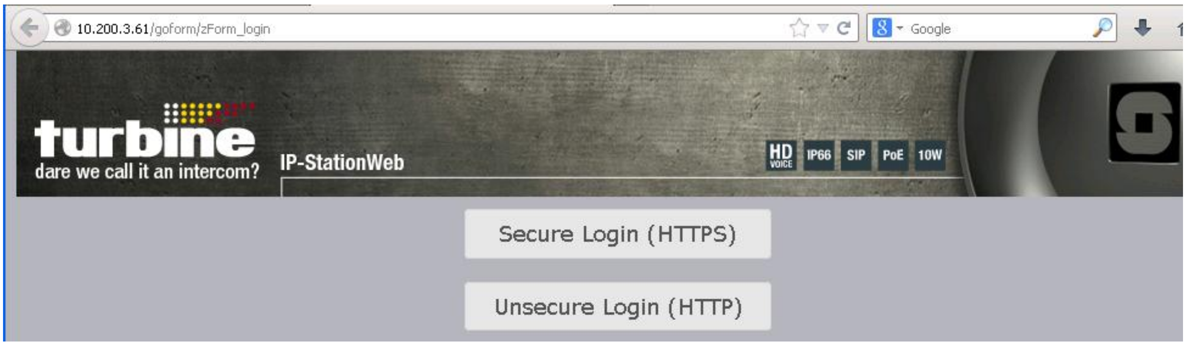
## 8.6 Video calls

| Test Case Id | Test Case | N/A | OK | NOK | Comment |
|---|---|---|---|---|---|
| **1** | Call from DoorPhone to SIP device (Counterpath Bria)<br>Check that the call is established in audio and video<br>Open the Latch<br>Release the call | ☒ | ☐ | ☐ | |
| **2** | Call to DoorPhone from SIP device (Counterpath Bria)<br>Check that the call is established in audio and video<br>Open the Latch<br>Release the call | ☒ | ☐ | ☐ | |
| **3** | Call from DoorPhone to SIP device (My IC 8082)<br>Check that the call is established in audio and video<br>Open the Latch<br>Release the call | ☒ | ☐ | ☐ | |
| **3** | Call from DoorPhone to SIP device (My IC 8082)<br>Check that the call is established in audio and video<br>Open the Latch<br>Release the call | ☒ | ☐ | ☐ | |

# 9 Appendix A : Partner Application Description

**Configure the DoorPhone using the web interface**

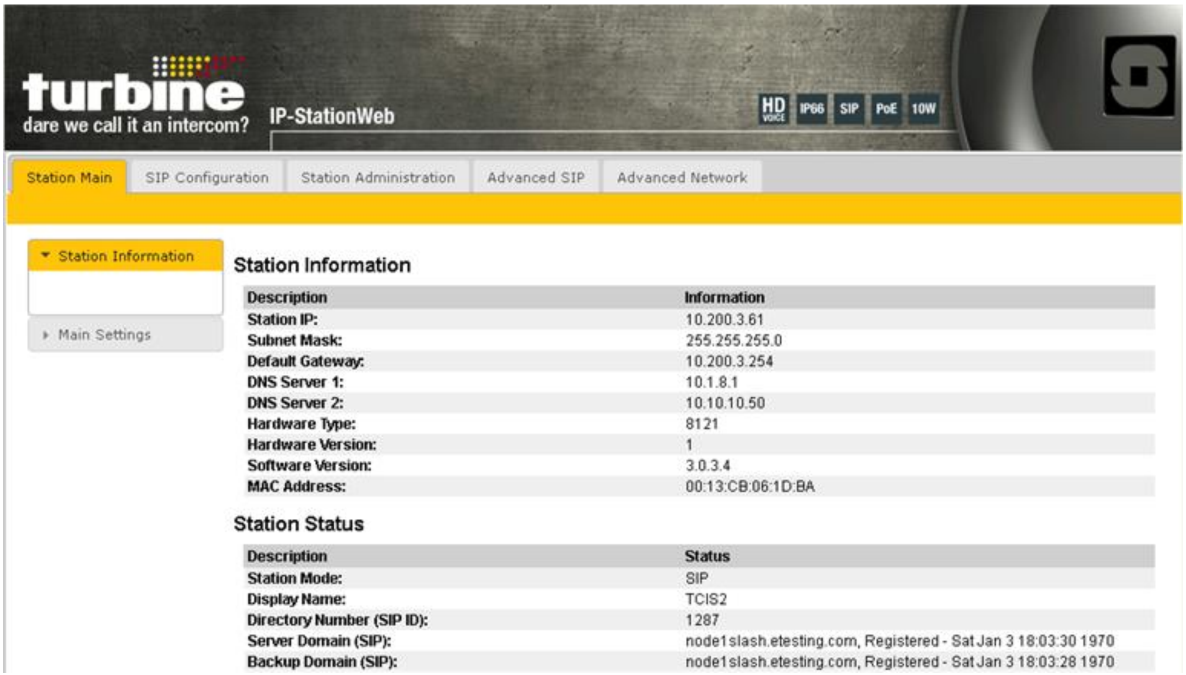
1. Open a web browser
2. Enter the TCIS DoorPhone IP address http://10.200.3.61 in the address bar. You will get the below screen




3. Click on any one above to start the configuration
4. Enter the username "admin" and the password "alphaadmin"

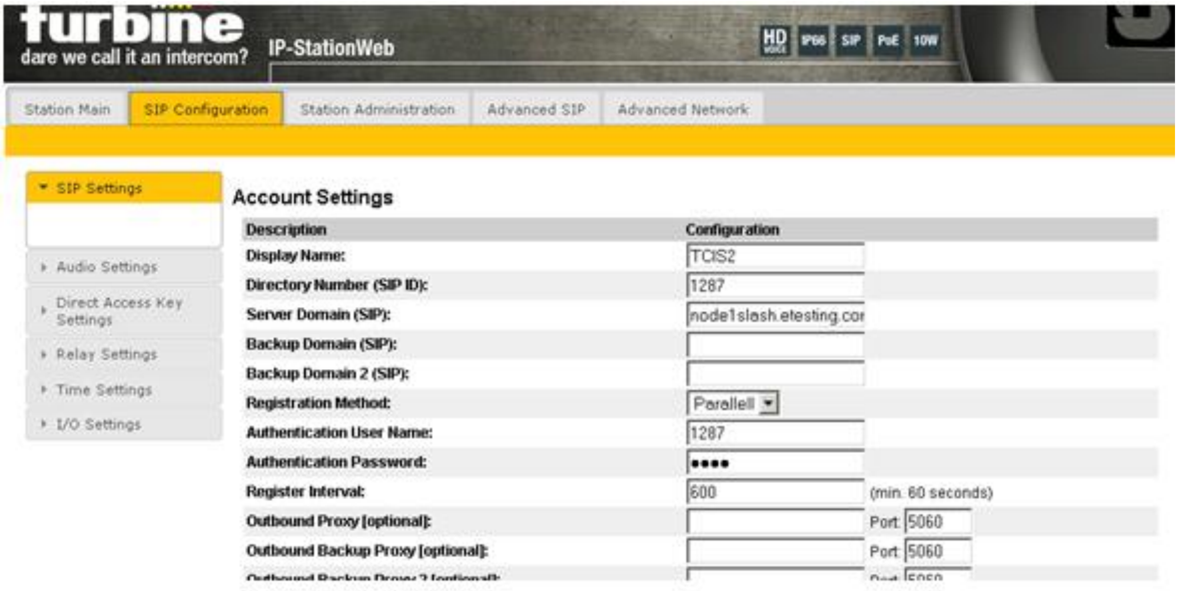
# 10 Appendix B: Partner Application: Configuration Requirements

**Main Page Details**

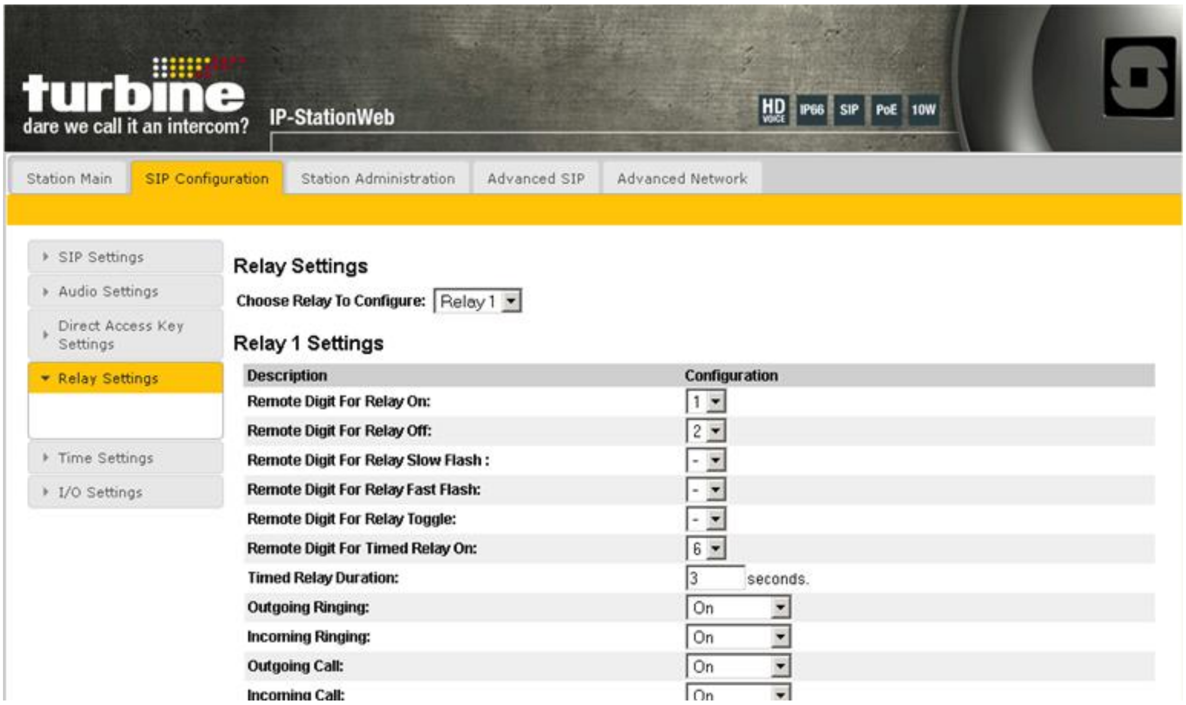


## SIP Configuration

Click on “SIP Configuration” and configure the OXE parameters:




Configure the code for relay opening and closing in Relay settings.

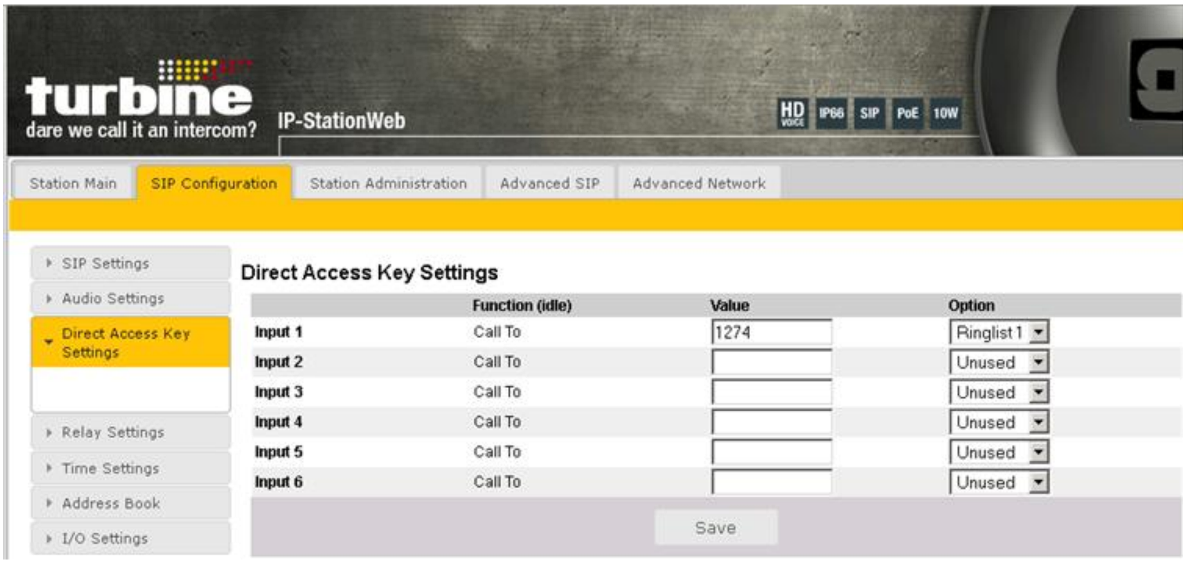


## Codec Configuration:

### Call Settings

| Description | Configuration |
|---|---|
| Enable Auto Answer: | ☐ |
| Auto Answer Delay: | 2 seconds. Max 30 seconds. |
| Delay Call Setup: | 1 seconds. Max 60 seconds. Delays call setup using DAK/Input buttons. |
| Overlap dialing: | ☐ |
| DTMF method: | RFC 2833 |
| RTP Timeout value: | 15 seconds. 0 = RTP Timeout Disabled. |
| Codec g722: | Low Priority |
| Codec g711a: | Medium Priority |
| Codec g711u: | High Priority |

## User configuration to dial and call:



# 11 Appendix C: Alcatel-Lucent Communication Platform: configuration requirements

**Configuration of OmniPCX Enterprise**

Creation of a SIP Device user:

```
Review/Modify: Users
              Node Number (reserved) : 101
                    Directory Number : 1269

                      Directory name : 1269_
                Directory First Name : --------------------
                UTF-8 Directory Name : ---------------------------------------
          UTF-8 Directory First Name : ---------------------------------------
                       Location Node : 1
                       Shelf Address : 255
                       Board Address : 255
                   Equipment Address : 255
                            Set Type + SIP device
                       Entity Number : 1
                        Set Function + Default
                        Profile Name : ---------------
                        Key Profiles + None
                   Domain Identifier : 0
                         Language ID : 1

                         Secret Code : ****
                             Confirm : ****
```

Creation of the trunk group: add a trunk group under mgr / Trunk groups.

```
             Node Number (reserved) : 101
                     Trunk Group ID : 100
                Instance (reserved) : 1

                   Trunk Group Type + T2
                   T2 Specification + SIP
                Public Network Ref. : ------
          VG for non-existent No. + YES
                      Entity Number : 0
             Supervised by Routing + NO
     VPN Cost Limit for Incom.Calls : 0
Immediate Trk Listening if VPNCall + YES
                           VPN TS % : 50
                     CSTA-Monitored + NO
```

Configuration of the SIP gateway: under mgr / SIP

```
        Node Number (reserved) : 101
           Instance (reserved) : 1
           Instance (reserved) : 1

                SIP Subnetwork : 10
               SIP Trunk Group : 100
                    IP Address : 10.10.10.50
         Machine name - Host : node1slash
         SIP Proxy Port Number : 5060
    SIP Subscribe Min Duration : 15
    SIP Subscribe Max Duration : 5060
                 Session Timer : 300
             Min Session Timer : 100
          Session Timer Method + UPDATE
         DNS local domain name : etesting.com
                      DNS type + DNS SRV
           SIP DNS1 IP Address : 10.1.8.1
           SIP DNS2 IP Address : 10.10.10.50
                    SDP in 18x + True
                  Cac SIP-SIP + False
```

Configuration of the SIP proxy: under mgr / SIP

```
            Node Number (reserved) : 101
               Instance (reserved) : 1
               Instance (reserved) : 1

              SIP initial time-out : 500
                      SIP timer T2 : 4000
                Dns Timer overflow : 5000
                  Recursive search + True
      Minimal authentication method + SIP None
              Authentication realm : ----------------
Only authenticated incoming calls + False
                  Framework Period : 3
     Framework Nb Message By Period : 25
        Framework Quarantine Period : 1800
            TCP when  long messages + True
```

# 12 Appendix D: Partner escalation process

Please contact the Customer service for any queries :

Customer service: +47 4000 2700

Opening hours 08.00 - 16.00 (CET / +1 UTC)

# 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:

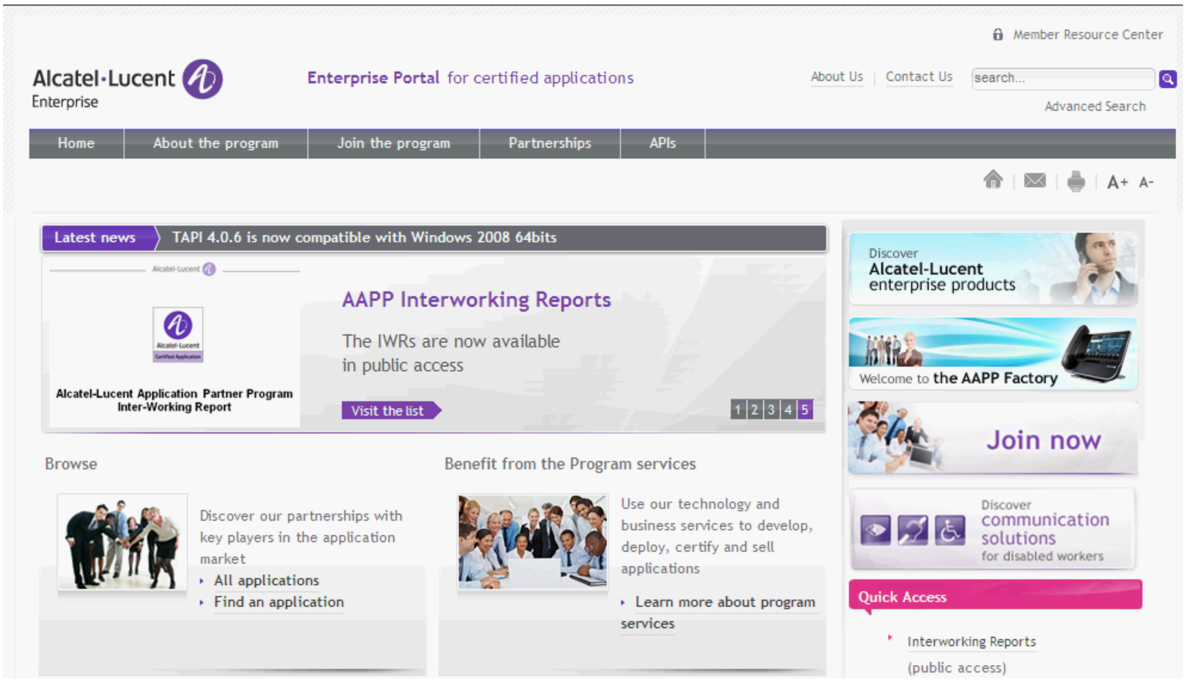
- **Provide easy interfacing for Alcatel-Lucent communication products**: Alcatel-Lucent's communication products for the enterprise market include infrastructure elements, platforms and software suites. To ensure easy integration, the AAPP provides a full array of standards-based application programming 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 program and where the InterWorking Reports can be consulted. Its access is free at
http://applicationpartner.alcatel-lucent.com




## 13.2 Alcatel-Lucent.com

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

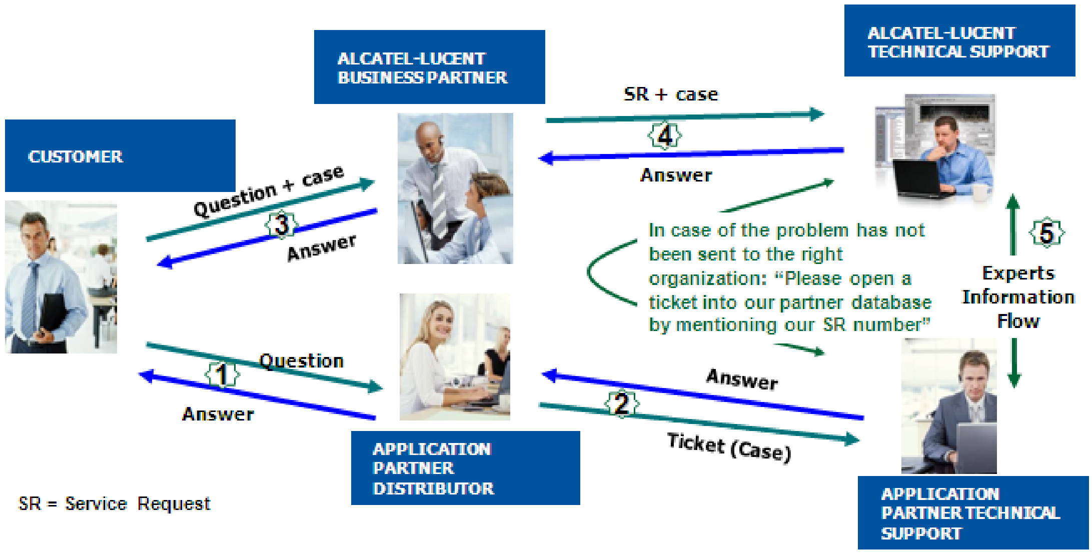
# 14 Appendix F: AAPP Escalation process

## 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 0 “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:

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

For other countries:

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

**END OF DOCUMENT**