ALE Application Partner Program Inter-Working Report

Partner: **Amphitech**
Application type: **SIP Door Phone**
Application name: **IPAC 101, IPAC 500**
Alcatel-Lucent Enterprise Platform:
OmniPCX Enterprise™

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

Certification overview

Date of the certification	July 2019

ALE International representative	HIMMI RACHID
AAPP member representative	Marc Labouille

Alcatel-Lucent Enterprise Communication Platform	OmniPCX Enterprise
Alcatel-Lucent Enterprise Communication Platform release	R12.0 - M1.403.12a R12.2 – M3.402.25a
AAPP member application release	IPAC101-2v - 0.27 IPAC500-21 - 1.66 / 1.71 Softphone ASIP 3.2
Application Category	Terminals

Author(s): Karthik Padmarajan, Mudassir Ahmed, Sebastien Ehrhard
Reviewer(s): Thierry Chevert, Rachid Himmi, Krassimira Atanassov

Revision History

Edition 1: creation of the document – May 2017
Edition 2 : extension for IP-GAP-02V – using the same SIP stack – July 2017
Edition 3 : tests with IPAC 500 and 8088 NOE Android softphone ASIP – July 2019

Test results

☑ Passed ☐ Refused ☐ Postponed

☐ Passed with restrictions

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

IWR validity extension

This report is also valid for **IP-GAP-02V (v1.27)** which share the same SIP stack than other devices

AAPP Member Contact Information

Contact name: Marc Labouille

Title: IP Project manager

Address: Amphitech SAV, 1 rue Robert et Sonia Delaunay,
75011 Paris, France

Zip Code: 75011
City: Paris

Country: France

Phone: +33 (0)1 43 67 98 09
Fax: +33 (0)1 43 67 13 97
Mobile Phone:

Web site: www.amphitech.fr
Email address: mlabouille@amphitech.fr , jgalle@amphitech.fr

TABLE OF CONTENTS

1 Introduction

This document is the result of the certification tests performed between the AAPP member's application and Alcatel-Lucent Enterprise'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, ALE International 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 ALE International issues a new major release of such Alcatel-Lucent Enterprise product (incorporating new features or functionalities), whichever first occurs.

A new release is identified as following:

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

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

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

3 Limits of the Technical support

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

The certification does not verify the functional achievement of the 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 analysed by the AAPP member before being escalated to ALE International. Access to technical support by the Business Partner requires a valid ALE maintenance contract

For details on all cases (3rd party application certified or not, request outside the scope of this IWR, etc.), please refer to Appendix 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 ALE International is included in the solution between the certified Alcatel-Lucent Enterprise and AAPP member products such as a Session Border Controller or a firewall for example, ALE International will consider that situation as to that where no IWR exists. ALE International will handle this situation accordingly (for more details, please refer to Appendix F “AAPP Escalation Process”).

4 Application information

Application commercial name: IPAC 101, IPAC 500

Application version: IPAC101-2V, IPAC500_21

Interface type: SIP

Brief application description:

Amphitech has been specialized in the design and manufacture of communications equipment such as telephone gateways, emergency call stations, elevator telegrams. Amphitech is now a leader in its field of activity.

Specialized in communication systems, AMPHITECH is aimed at professionals with weak currents. Its expertise, innovation, the reliability of its equipment have made AMPHITECH. The reference in the fields of the telephone, the emergency call and the elevator telealarm.

IPAC 101

- 1 call button
- Simplified configuration; Advanced configuration on dedicated WEB interface.
- Peer-to-peer communication
- Communication via SIP server (multiple calls, conferences, Queue management, mail ...)
- Time slot management
- HD audio

IPAC 500

- Configuration in 4 easy steps
- Peer-to-peer network scan
- Day/night operation mode
- Realtime display of the door phone screen on the web pages
- LDAP-udpate of the phonebook
- HD audio
- Video codec H264 or streaming

IP-GAP-02V

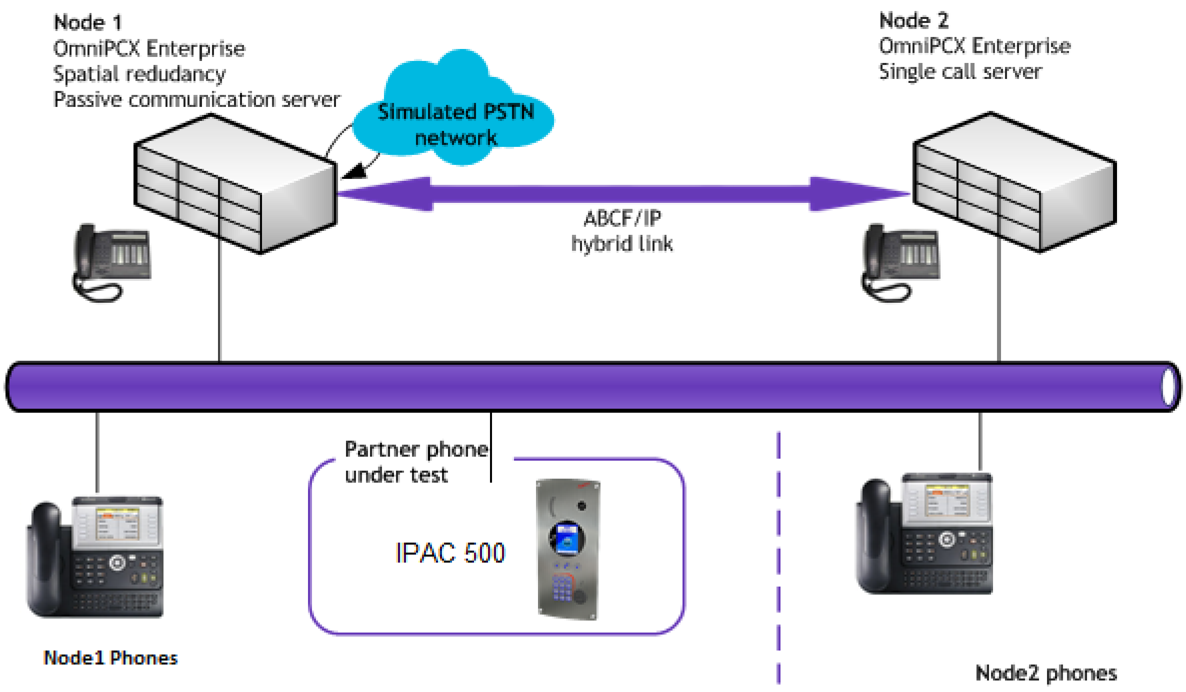
- 1 call button
- Simplified configuration; Advanced configuration on dedicated WEB interface.
- Peer-to-peer communication
- Communication via SIP server (multiple calls, conferences, Queue management, mail ...)
- Time slot management
- HD audio
- Inductive loop
- 2 relays

ADD ASIP SOFTPHONE DESCRIPTION

5 Test environment

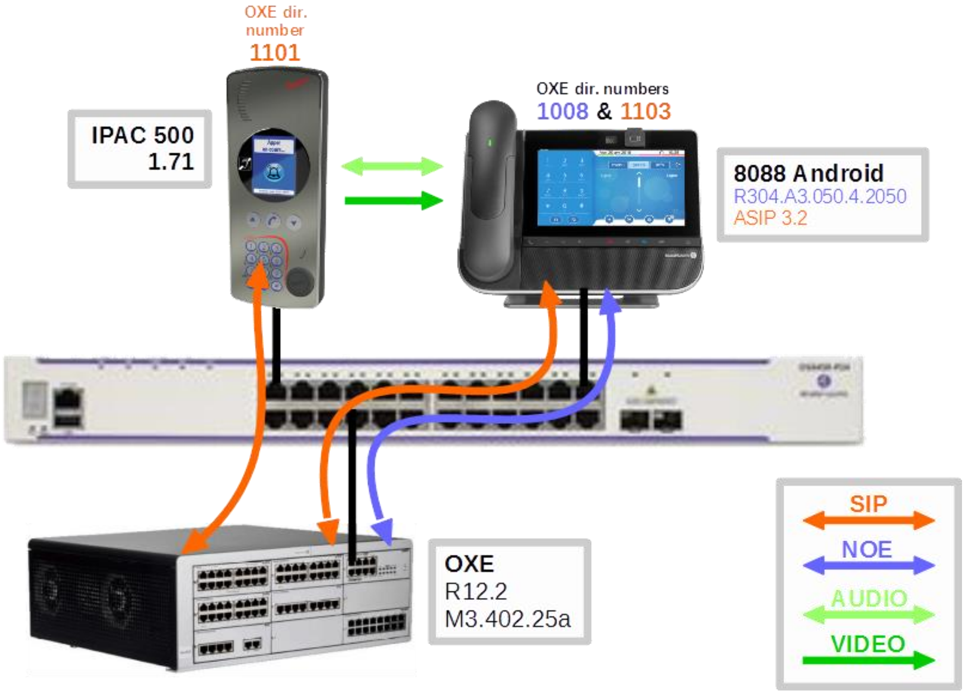
5.1 Test environment

Figure 1 **Test environment**



5.2 Test environment for IPAC 500 and 8088 NOE Android ASIP softphone

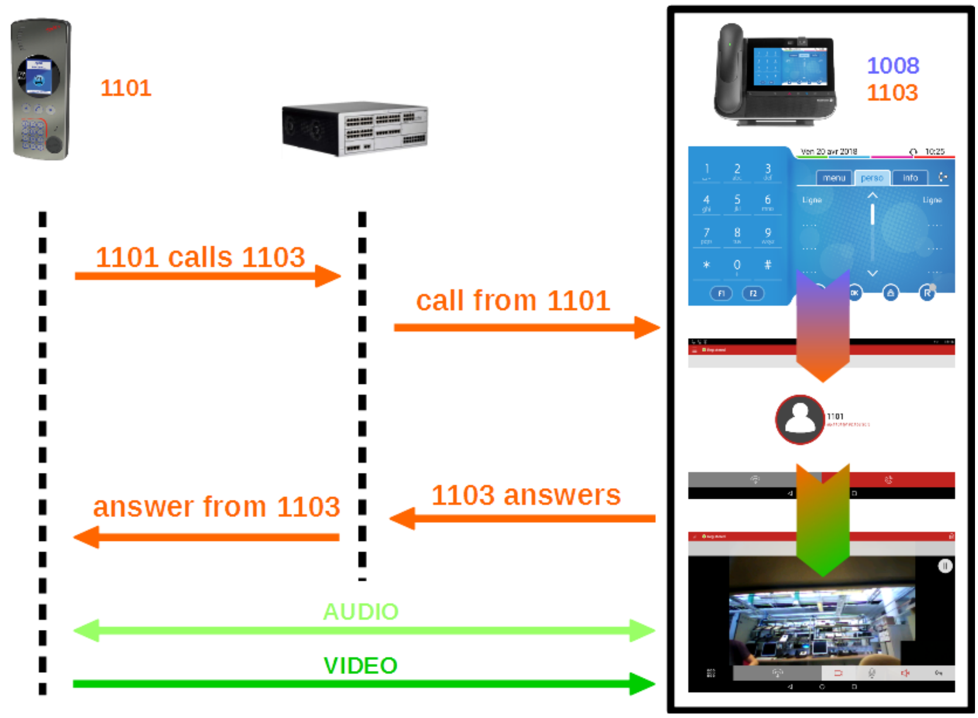
Figure 2 **Test environment with 8088 NOE Android**



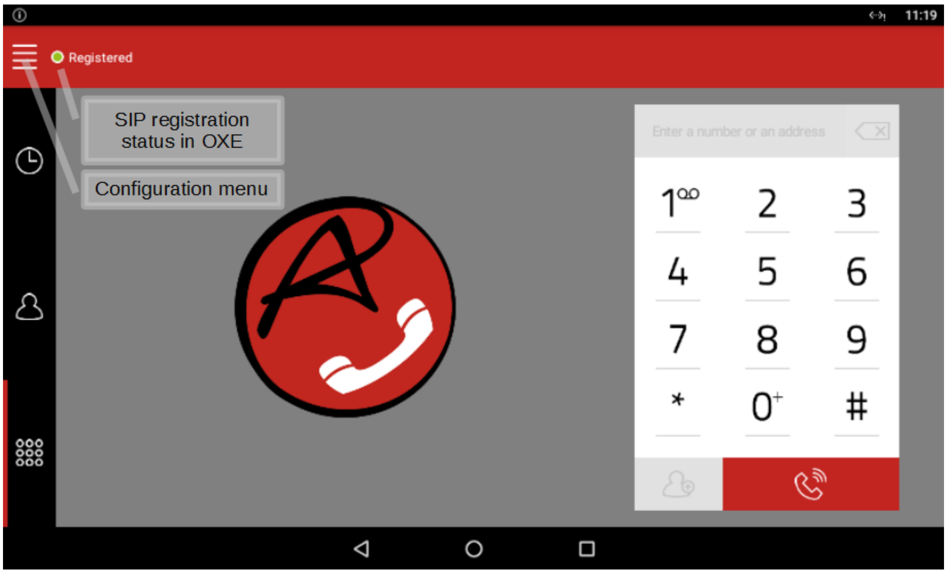
The 8088 is registered twice in the OXE (see section **Erreur ! Source du renvoi introuvable.**):

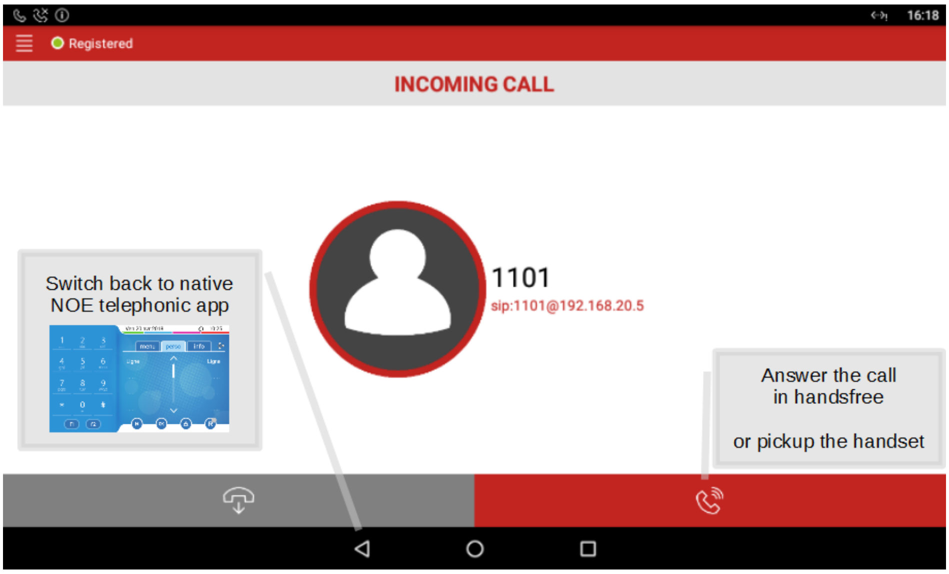
- In NOE mode for the native telephonic application (1008 directory number)
- In SIP mode for the ASIP application (1103 directory number)

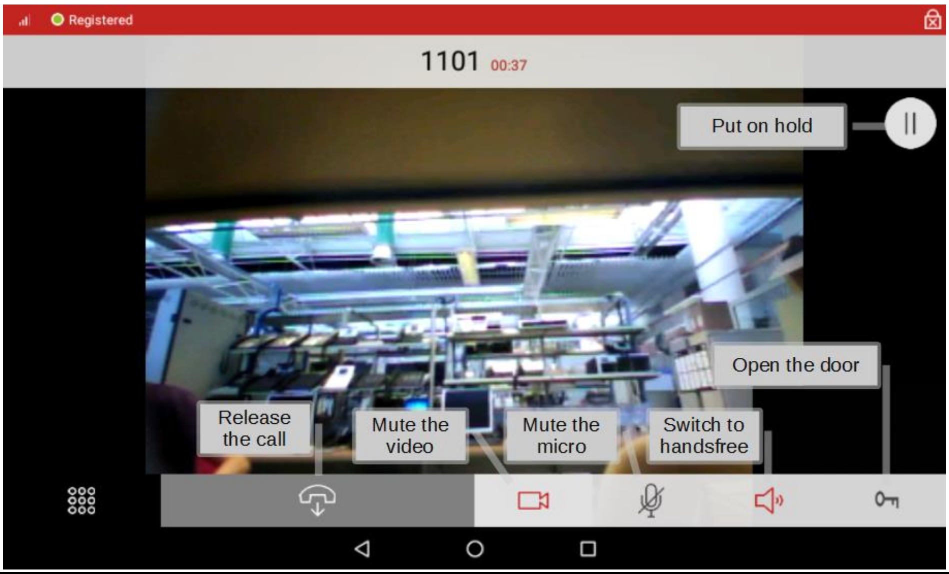
5.2.1 Call flow



5.2.2 User interface of the ASIP application







5.3 Hardware configuration

List main hardware equipments used for testing

- **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)
  - UA digital and analog sets

OXE setup	
OXE 1 IP address	10.9.224.238 **/** 10.9.223.238
Domain name	r12.proservtesting.com
Attendant No	6666
OXE Extension Details used for test	
IP touch and UA extensions	1001 to 1009
SIP users	1010 to 1020

5.4 Software configuration

List main softwares used for testing

- **Alcatel-Lucent Enterprise Communication Platform**: OmniPCX Enterprise R12 M1.403.12a

- **Partner Application** : IPAC101-2v - 0.27
  IPAC500_21 - 1.66

5.4.1 Software configuration for tests with IPAC 500 and 8088 NOE Android ASIP softphone

- **Alcatel-Lucent Enterprise Communication Platform**: OmniPCX Enterprise R12.2 M3.402.25a M1.403.12a
- **8088 NOE Android** : A3.050.4.2050

- **Partner Application** : ASIP 3.2
  IPAC500 - 1.71

6 Summary of test results

6.1 Summary of main functions supported

This section is a summary of the main features tested. This is not a complete description of all the tests performed. If the status is “OK but” or “NOK”, please refer to the below paragraphs or to the detailed test results.

Features	Statu	Comments
Initialization including network configuration	OK	
SIP registration	OK	
SIP authentication	OK	
Outgoing Call	OK	
Incoming Call	OK	
Trigger the relay during Outgoing call	OK	
Trigger the relay during Incoming call	OK	
Features During Conversation	OK	
Disconnect call after phone hang up	OK	
Defence	OK	New calls cannot be initiated from the door phone after an OXE call server switchover. New calls can be initiated only after the next SIP registration.
Video	OK	Tests performed with 8088 on OmniPCX standalone and OpenTouch

6.2 Summary of main functions supported for IPAC with 8088 NOE Android ASIP softphone

Features	Status	Comments
Application deployment, installation and configuration	OK	
Defenses	OK	
SIP registration	OK	
Basic calls	OK	
Handset and handsfree	OK but	See restriction in 8.6.3.58.6.3.5 section
Mute and volume management	OK	
Interaction with the native telephonic application	OK but	See restrictions in 8.6.3.78.6.3.5 section

6.3 Summary of problems

- None

6.4 Summary of limitations

- No Hold tone is heard in the device.
- Full attendant transfer is not working.
-
- The 8088 NOE Android ASIP has an established audio and video call with the IAPC 500 in handset mode. The user switches to handsfree then puts the handset back on the phone: the call is released.
- The 8088 NOE Android (NOE native telephonic application) is already in an established call with another OXE phone when there is a new incoming call from the IPAC 500: the IPAC 500 call fails with “call failed” error message displayed on its screen.
- The 8088 NOE Android (NOE native telephonic application) user answers a new OXE incoming call while already in an established audio and video call with the IPAC 500 door cam. The door cam call is put on hold and the OXE audio call is established: but the user is not able switch the audio modes on the 8088 phone using the “audio” sensitive key (just below the screen and above the loudspeaker).

6.5 Notes, remarks

- None

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

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

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

8.2 Calls from DoorPhone

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	Call from DoorPhone to IP Touch Configure the system law to A-law Check that the call is established in G711 A-law Check audio quality and hold option Release the call from IP Touch	☐	☒	☐	
2	Call from DoorPhone to UA Phone Configure the system law to A-law Check that the call is established in G711 A-law Check audio quality and hold option Release the call from UA Phone	☐	☒	☐	
3	Call from DoorPhone to SIP device Configure the system law to A-law Check that the call is established in G711 A-law Check audio quality and hold option Release the call from SIP device	☐	☒	☐	
4	Call from DoorPhone to IP Touch Configure the system law to µ-law Check that the call is established in G711 µ-law Check audio quality and hold option Release the call from IP Touch	☐	☒	☐	
5	Call from DoorPhone to UA Phone Configure the system law to µ-law Check that the call is established in G711 µ-law Check audio quality and hold option Release the call from UA Phone	☐	☒	☐	
6	Call from DoorPhone to SIP device Configure the system law to µ-law Check that the call is established in G711 µ-law Check audio quality and hold option Release the call from SIP device	☐	☒	☐	
7	Communication timeout Call from DoorPhone to IP Touch Wait for the DoorPhone timer to release the call	☐	☒	☐	Release timer can be configured (Max communication time)
8	Communication timeout Call from DoorPhone to UA Phone Wait for the DoorPhone timer to release the call	☐	☒	☐	Release timer can be configured (Communication timeout timer)

9	Communication timeout Call from DoorPhone to SIP device Wait for the DoorPhone timer to release the call	☐	☒	☐	Release timer can be configured (Communication timeout timer)
10	Call from DoorPhone to IP Touch Open the latch by DTMF	☐	☒	☐	
11	Call from DoorPhone to UA Open the latch by DTMF	☐	☒	☐	
12	Call from DoorPhone to SIP device Open the latch by DTMF	☐	☒	☐	

8.3 Calls to DoorPhone

These tests check that the phones can interact with the DoorPhone (tested with audio only – no video).

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	Call to DoorPhone from IP Touch Check audio quality Release the call from IP Touch	☐	☒	☐	
2	Call to DoorPhone from UA Phone Check audio quality Release the call from UA Phone	☐	☒	☐	
3	Call to DoorPhone from SIP device Check audio quality Release the call from SIP device	☐	☒	☐	
4	Call to DoorPhone from IP Touch Wait for the DoorPhone timer to release the call	☐	☒	☐	
5	Call to DoorPhone from UA Phone Wait for the DoorPhone timer to release the call	☐	☒	☐	
6	Call to DoorPhone from SIP device Wait for the DoorPhone timer to release the call	☐	☒	☐	
7	Call to DoorPhone from IP Touch Open the latch by DTMF (Call is released)	☐	☒	☐	
8	Call to DoorPhone from UA Phone Open the latch by DTMF(Call is released)	☐	☒	☐	
9	Call to DoorPhone from SIP device Open the latch by DTMF(Call is released)	☐	☒	☐	
10	Mode of the DoorPhone. Check Day and night mode using time zone	☐	☒	☐	**IPAC101** Tested with the call button. During day mode call goes to one extension and during night mode call goes to another extension using time zone feature. **IPAC500** Tested with the Lighting. During day/night mode, key & screen brightness updated.

13	Call to DoorPhone from SIP device Configure the DoorPhone to answer the incoming INVITE with a 180 RINGING. Check ring back tone on the SIP device.	☐	☒	☐	
14	Call from external number(T0/T2) to DoorPhone	☐	☒	☐	
15	Call from attendant to DoorPhone	☐	☒	☐	
16	Incoming external call (T0/T2 for example) to an attendant phone set which transfers the call to the Door Phone. Check the Call is properly established.	☐	☒	☐	Only semi attendant transfer is working not full attendant trasnfer.

8.4 In conversation scenarios

Tested with audio only – no video.

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 to disconnect the call and On the DoorPhone press the same call button (it releases the first call, and the second call is made)	☐	☒	☐	
2	Call from DoorPhone to busy UA Phone Check that call is released (gets busy)	☐	☒	☐	(SIP: " 183 Session progress" reason=is busy):
3	Call from DoorPhone to IP Touch Put on hold Take back the call and check the audio Open the Latch Release the call	☐	☒	☐	
4	Call from DoorPhone to UA Phone Put on hold Take back the call and check the audio Open the Latch Release the call	☐	☒	☐	
5	Call from DoorPhone to SIP device Put on hold Take back the call and check the audio Open the Latch Release the call	☐	☒	☐	
6	Call from DoorPhone to IP Touch and once the conversation is established make a call from the same IPTouch to a UA Phone (answer the call in UA phone) and press transfer button in IP Touch Check the audio Open the Latch Release the call	☐	☒	☐	
7	**Call from a Door phone to other sip extension** (In which forwarding is Enabled) Call from Door Phone to SIP extension (In which call forward is enabled) and check the conversation is established. Check the audio Open the Latch Release the call	☐	☒	☐	We tested with SIP phone local feature and it is working fine.

8.5 Duplicated call servers and passive call server

Below test cases were checked only with audio call.

1	OXE Call Server CPU switches over while door phone in idle. Check the door phone behavior after a switch over from the OXE main to standby CPU. The phone must be able to make and receive a call after the switch over.	☐	☒	☐	BYE is not sending properly after switchover. After re-register (session expires timeout) everything works perfectly.
2	OXE Call Server CPU switch over while door phone in conversation with an IPTouch. Check the SIP phone behavior after a switch from the OXE main to standby CPU. The call is still active. The phone can make and receive a second call and switch from one to another. After on hook, the phone must be able to make and receive a call after the switch over.	☐	☒	☐	
3	OXE Passive Communication Server activation while DoorPhone in idle. Check the DoorPhone behavior secured by a Passive Communication Server after its activation. The phone must be able to make and receive a call after Passive Communication Server activation.	☐	☒	☐	
4	OXE Passive Communication Server activation while DoorPhone in conversation with an IPTouch. Check the SIP phone behavior secured by a Passive Communication Server after its activation. The call is still active. The phone can make and receive a second call and switch from one to another. After on hook, the phone must be able to make and receive a call after the Passive Communication Server activation.	☐	☒	☐	
5	OXE Call Server reboot while Door phone in idle. Check the phone behavior when the OXE Call Server reboots (without standby CPU). As soon as the Call Server is running again, the phone is able to make and receive a call.	☐	☒	☐	
6	OXE Call Server reboot while Door phone in conversation with an IPTouch. Check the phone behavior when the OXE Call Server reboots (without standby CPU). The call is released. As soon as the Call Server is running again, the phone is able to make and receive a call	☐	☒	☐	

8.6 Video

Only IPAC500 supports video calls. It has an embedded camera to send out video stream and is able to display incoming video stream as well.
Tests are performed with a 8082 configured as an hotel phone on the OmniPcx Entreprise (see **Erreur ! Source du renvoi introuvable. Erreur ! Source du renvoi introuvable.** for the configuration details). Depending on the tests, the 8082 does or does not have its video camera connected to send out its video stream.

You cannot have at the same time the "Open door" button on the 8082 and video from the 8082 to the Amphitech station (see **Erreur ! Source du renvoi introuvable. Erreur ! Source du renvoi introuvable.** and **Erreur ! Source du renvoi introuvable. Erreur ! Source du renvoi introuvable.** for the details).

8.6.1 Video calls with 8082 and 8088 deskphone on OmniPCX Enterprise

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	Call from the DoorPhone to a 8088 without video camera enable. The DoorPhone calls the 8088. The 8088 picks the call up. Check that an audio call is established in both ways.	☐	☒	☐	
2	Repeat test 1 but release the call from the DoorPhone.	☐	☒	☐	
3	Call from the DoorPhone to a 8088 with video camera enable. The DoorPhone calls the 8088. The 8088 picks the call up. Check that an audio call is established in both ways. Check the video call. Release the call from the 8088	☐	☒	☐	
4	Repeat test 3 but release the call from the DoorPhone.	☐	☒	☐	
5	Open the latch by DTMF from the 8088 Check that the DoorPhone triggers the "open the door".	☐	☒	☐	
6	Repeat tests 1 to 5 but this time, with My IC 8088 extension configured as hotel extension.	☐	☒	☐	

8.6.2 Video calls with 8088 deskphone on Open Touch

Repeat tests of section 8.6.1 (Video calls with 8082 and 8088 deskphone on OmniPCX), but this time use a 8088 desk phone connected to an Open Touch.
The Amphitech DoorPhone is configured on the OmniPCX Enterprise associated to the Open Touch. Configuration is the same as for the OmniPCX Enterprise standalone tests.

Test Case Id	Test Case	N/A	OK	NOK	Comment
1 to 6	Repeat tests 1 to 5 but this time, with My 8088 extension connected to Open touch	☐	☒	☐	

8.6.3 Tests with IPAC 500 and 8088 NOE Android ASIP softphone application

8.6.3.1 ASIP softphone application deployment, installation and configuration

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Application deployment and installation** Check that the application is installed after deployment through the “private store”		☒		
2	**Application deployment and installation** Check that the application can be configured (SIP Account) to register to the OXE		☒		

8.6.3.2 Defenses

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**8088 phone reboot** Reboot the 8088 and check that the ASIP application is started automatically after the 8088 initialization		☒		

8.6.3.3 SIP registration

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**ASIP registration in the OXE** Check that the ASIP application is registrating in SIP		☒		

	in the OXE (OXE “sipregister” command can be used to check).				
2	**ASIP registration after expiration** Check that the ASIP application is registrating again in SIP in the OXE after the current registration has expired (OXE “sipregister” command can be used to check).		☒		

8.6.3.4 Basic calls

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**IPAC 500 door phone 1101 calls the 8088 1103** Check that the 8088 switches from the native telephonic application to the ASIP application, rings and displays the incoming call. Check that the 8088 user can answer the call by picking the 8088 handset from the phone. Check that the ringing stops and that the ASIP application switches to the ongoing call screen. Check that there is audio from and to the IPAC 500. Check that there is video from the IPAC 500 to the 8088 ASIP application: the ASIP application displays the video from the IPAC 500		☒		

8.6.3.5 Handset and handsfree support

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Handset** Repeat test case 1 from 8.6.3.4 section.		☒		
2	**Handsfree** Repeat test case 1. But this time, 8088 user answers the call with the ASIP application “answer call” button (see section 5.2.2). Check that the audio is established on the 8088 handsfree (in both direction). And check that the ASIP application displays the video from the IPAC 500.		☒		
3	**Switch from handsfree to handset** Repeat test case 2. Once in communication, the 8088 user picks the handset from the phone. Check that the audio is now in the 8088 handset and no more in the 8088 handsfree.		☒ but		When in handsfree, putting the handset back on the 8088 releases the call.

	Switch back to handsfree thanks to the ASIP button. Once in handsfree, put the 8088 handset back on the phone.				

8.6.3.6 Mute and volume management

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Mute** Establish a call (see section 8.6.3.48.6.3.4). Press ASIP mute button (see section 5.2.2). Check that there is no more audio from the ASIP to the IPAC 500. Press ASIP mute button. Check that there is again audio from the ASIP to the IPAC 500. Switch to handsfree by pressing the ASIP audio button (see section 5.2.25.2.2). Check that the audio is established on the 8088 handsfree (in both direction). Press ASIP mute button. Check that there is no more audio from the ASIP to the IPAC 500. Press ASIP mute button. Check that there is again audio from the ASIP to the IPAC 500.		☒		
2	**Volume** Establish a call (see section 8.6.3.4). Press 8088 + and – touch keys (just below the screen and above the phone loudspeaker). Check that the volume in the handset is changed. Switch to handsfree. And repeat the + and – key presses to check that the volume in the handsfree is changed.		☒		

8.6.3.7 Interaction with the native telephonic application

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Ongoing "native" OXE call when incoming door cam call** The 8088 (1008) is already in an established call with another OXE phone (in NOE mode). IPAC 500 door phone 1101 calls the 8088 1103.		☒ but		The IAPC 500 call is not processed. "Call failed" is displayed on the device.
2	**Ongoing door cam call when incoming native OXE call**		☒		It is not possible to switch the audio

	The 8088 (1103) is already in an established audio call with the IPAC 500 door cam. Another OXE phones calls the 8088 1008. The 8088 switches back to the native OXE telephonic application to display the new incoming call. The ongoing audio and video call with the IPAC 500 is put on hold. The 8088 user touches the “Answer” softkey to answer the new incoming call.		but		modes on the 8088 phone using the “audio” sensitive key (just below the screen and above the loudspeaker).

9 Appendix A: AAPP member's Application description

DATA SHEET

Amphitech

N°F000 0830B - 1/2

IPAC 500 21 S

VoIP Door Entry Phone with video camera, « hands-free », vandal-resistant:

- **Meeting the requirements for accessibility for people with disabilities**
- with name search by phone book scrolling.

IPAC 500 21 S for:

- a point-to-point communication (Peer to Peer) **or**
- the connection via a SIP server.

Functions

- Door entry phone
- Full duplex voice communication

Technical data

- Acoustic inductive loop amplifier as communication aid for hearing aid users
- 2 navigation buttons (arrows) for name search in the phone book (1000 contacts)
- 1 keypad with the functions: access code, dialling (abridged or free according to configuration) and alphabetical name search in the phone book
- Video camera (active during communication or streaming) – viewing angle 90° - CMOS Sensor IR Cut Filter
- Management of call parameters: communication time, button activation time, ring time for outgoing calls, volume...
- Management of time lock zones
- Operation mode « Porter »
- Pictograms display associated with product functions
- Automatic speech announcements (dialling, communication..., door opening)
- Audio-quality HD
- Data encryption (audio and video): SRTP / ZRTP / SIP-TLS
- 2 relays for door open command **or** remote control of external elements (lighting, etc...)
- 2 inputs for external contacts **or** voltage **with the possibility to define time lock zones**
- Day/night operation (adjustment of volume and brightness)
- Updates:
  - LDAP-update of the phonebook
  - system update by downloadable file
- Real-time monitoring of device status:
  - → On access code keying, outgoing calls, door opening, loss of SIP server
  - → In case of power failure
- Real-Time display of the device screen on the web page

Power supply

- Network: POE+

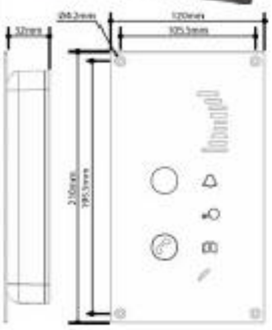
  or
- External power supply unit: 24 VDC - **AMPHITECH BAS 2415**

Mechanical design

- Surface mount
- Dimensions 300 x 120 x 30 mm - weight 2,2 kg
- Degree of protection: IP 55 - IK 08
- Temperature range: -20°C to +50°C
- Stainless steel face plate 2,5 mm, ZAMAK housing

DATA SHEET

N°F000 0993B -1/2



IPAC 101-2VE

VoIP Door Entry Phone **for audio-video transmission,** « hands-free », vandal-resistant:

- **Pictograms display and automatic speech announcements to meet the requirements for accessibility for people with disabilities**
- 1 direct call button

The IPAC 101-2VE allows for:

- a point-to-point communication (Peer to Peer) **or**
- the connection via a SIP server.

Functions

- Telephone
- Full duplex voice communication

Technical data

- 1 call button
- Caméra vidéo (mode en communication, mode streaming) - Angle de vision 90°- Capteur CMOS - IR Cut Filter
- Redial if busy or if no answer (1 - 4 call numbers)
- Management of call parameters: communication time, button activation time, ring time for outgoing calls, volume...
- Management of time lock zones
- Pictograms display associated with product functions
- Automatic speech announcements (dialling, communication ..., door opening)
- HD audio quality
- Media incryption (audio and video): SRTP / ZRTP / SIP-TLS
- 1 relay for door open command **or** remote control of external elements (line seizure information)
- 1 input for external contact or voltage with the possibility to define time lock zones
- LDAP update of the IPAC 100 contacts
- Monitoring of the device status:
  → On access code keying, outgoing calls, door opening, loss of SIP server....
  → In case of power failure
- Real-time display of the device screen on the web page

Power supply

- Network: POE

  or
- External power supply unit: 24 VDC - **BAS 2415 AMPHITECH**

Mechanical design

- Flush mount
- Dimensions 210 x 120 x 32 mm
- Degree of protection: IP 55 - IK 08
- Temperature range: -20°C to +50°C
- Stainless steel faceplate 2.5 mm, ZAMAK housing
- Flush mount housing BM 100 included in delivery

FICHE PRODUIT

Conception et fabrication françaises depuis 1988

N°F000 1042x

IP-GAP-02V

Portier **audio-vidéo** full-IP "mains-libres", anti-vandales pour **ENVIRONNEMENTS BRUYANTS :**

- **Pictogrammes lumineux et synthèse vocale pour accessibilité des personnes avec handicap**
- avec 1 bouton d'appel.

L'IP-GAP-02V permet :

- la communication Point à Point (Peer to Peer) **ou**
- la communication via un serveur SIP.

Fonctions

- Téléphone
- Communication vocale full duplex

Caractéristiques techniques

- 1 bouton d'appel
- Caméra vidéo (mode en communication, mode streaming) - Angle de vision 90° - Capteur CMOS - IR Cut Filter
- **Boucle inductive pour aide à la communication des personnes malentendantes appareillées**
- Appel cyclique en cas d'occupation ou de non réponse (1 à 4 numéros)
- **Amplificateur audio 80 dB à 1 m au niveau max.**
- Gestion des paramètres d'appels : temps de communication, temps d'appui bouton, délais appel sortant, volume...
- Gestion de plages horaires
- Affichage de pictogrammes en fonction de l'état de l'appel
- Synthèse vocale (appel en cours, communication en cours, ouverture de la porte)
- Audio HD
- Cryptage média (audio et vidéo) : SRTP / ZRTP / SIP-TLS
- 2 relais pour la commande d'ouverture de porte **ou** le pilotage d'éléments externes (Information **Prise De Ligne**)
- 1 entrée contact **ou** tension **avec définition possible de plages horaires**
- Mises à jour :
  - LDAP des contacts IP-GAP
  - système par fichier téléchargeable
- Surveillance de l'état du portier :
  → Sur saisie des codes d'accès, appel sortant, ouverture porte, perte serveur SIP....
  → En cas de défaut alimentation secteur
- Visualisation en temps réel de l'état des pictogrammes

Alimentations

- Réseau : POE+ **ou** Alimentation externe : 24 VDC - **BAS 2415 AMPHITECH**

Caractéristiques mécaniques

- Montage en saillie **ou** Montage encastré
- Dimensions 210 x 120 x 43 mm
- Indices de protection **IP 65 - IK 09**
- Températures de fonctionnement -20°C à +50°C
- Façade inox 2.5 mm

Options et périphériques

- Montage en saillie avec boîtier BSV100 - 213 x 123 x 71 mm (à prévoir)
- Montage encastré en maçonnerie avec boîtier BM100 (fourni)

Garantie

- 2 ans, retour usine

CE

Conception & Fabrication Françaises

FICHE PRODUIT

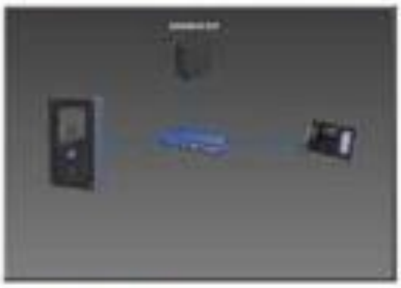
Conception et fabrication françaises depuis 1988

N°F000 1042x

IP-GAP-02V

Communication Point à Point (Peer to Peer)

Communication via un serveur SIP



Configuration

- Accès par serveur WEB
- Configuration simplifiée ; Configuration avancée sur interface WEB dédiée
- Paramétrage réseau IP, compte SIP, codecs audio et vidéo
- Communication point à point (Peer-to-Peer)
- Communication via serveur SIP (appels multiples, conférences, gestion file d'attente, messagerie...)
- Gestion de 3 comptes SIP sur différents IP-PBX
- Protocoles de communication : SIP (RFC 3261), IPV4, TCP/UDP, HTTP, HTTPS, RTP, DHCP/STATIC/NAT, RFC 6086 INFO Method, DTMF RFC 2833, RFC 2976 SIP INFO
- Codecs audio : G.722, G.711u, G.711a, GSM, Speex 8k, Speex 16k, Speex 32k, G.726-16, G.726.32, G.726-24, G.726-40, AAL2-G.726-16, AAL2-G.726-32, AAL2-G.726-24, AAL2-G.726-40, opus, AMR, -32
- Codecs vidéo : H264, H263, H263p, VP8 - Vidéo en communication : Qcif ou Cif - Vidéo en streaming : résolution 320 x 240 ou 640 x 480 (accès sécurisé)
- Rapport des évènements système : fichiers téléchargeables, SYSLOG, notifications par e-mails (client smtp)
- Choix de la langue (configuration, exploitation) : Français, Allemand, Anglais, Espagnol, Portugais
- Configuration de l'heure mode manuel ou serveur NTP
- Passage automatique heure d'hiver / heure d'été

Conception & Fabrication Françaises

ADD ASIP DESCRIPTION

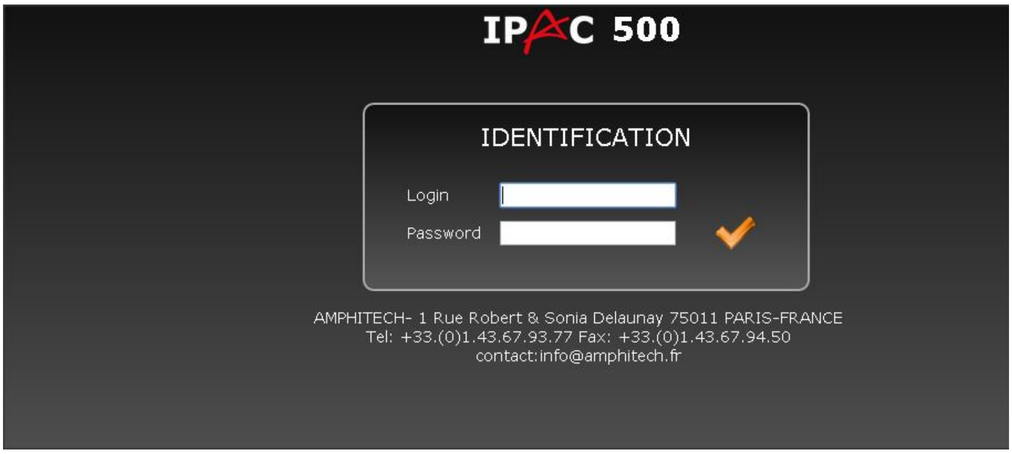
10 Appendix B: Configuration requirements of the AAPP member's application

10.1 IPAC 500

Access to the Admin Home page (Web interface)

1. Access your web browser. Enter the Static IP address on your browser. Example: http://10.9.224.198 (Phone IP Address).

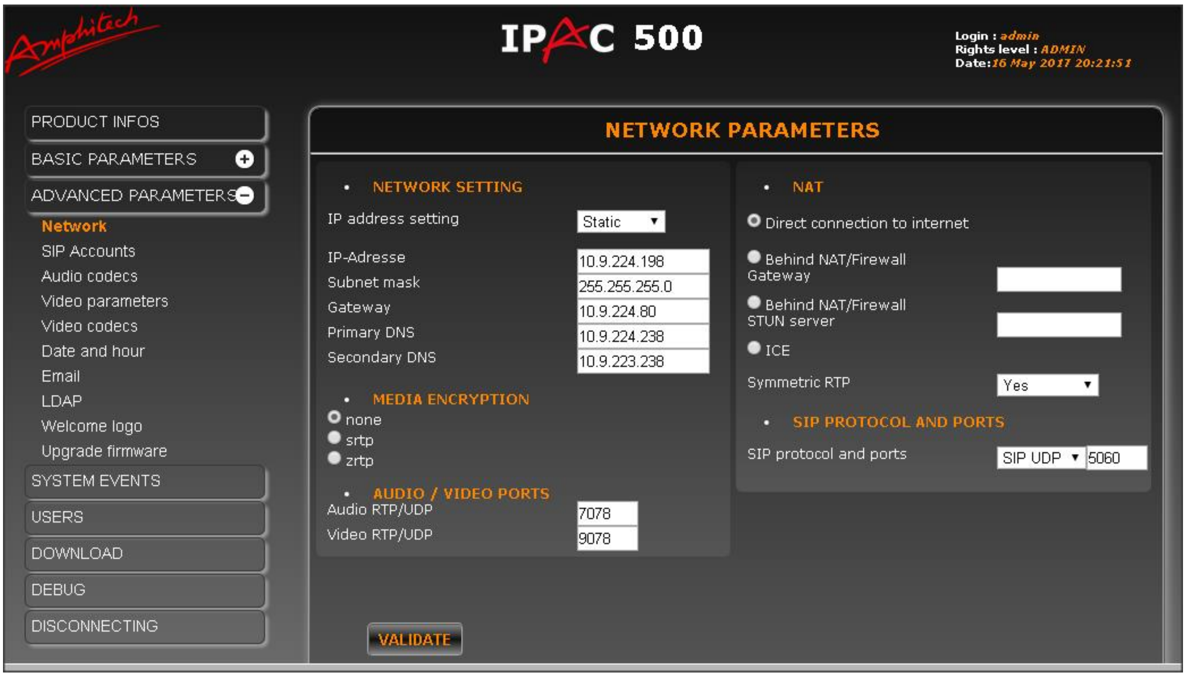
2. The Web language page will be displayed. Select the language.
3. The Web login page will be displayed. Enter the user name and the password and click **Login**. The administrator's default user name and password are "**admin**" and "**admin**" respectively.



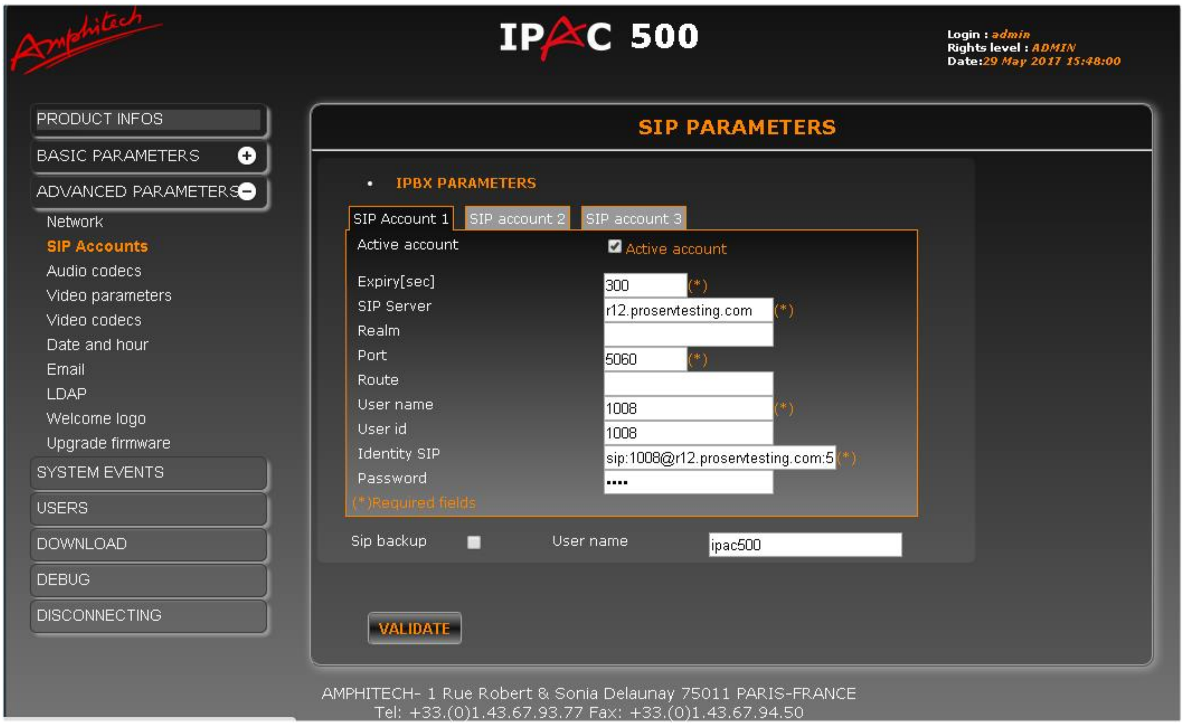
Enter the IP address of the device in your browser, then log with admin account.

Advance setup

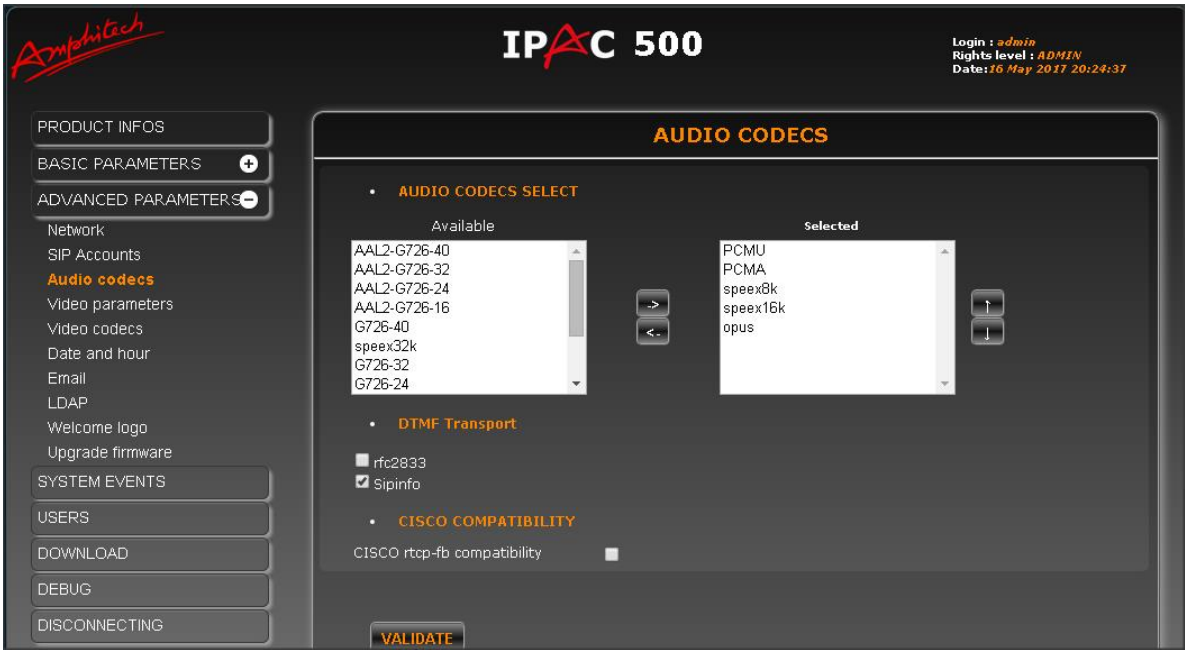
Network page information (Advance parameters→Network)



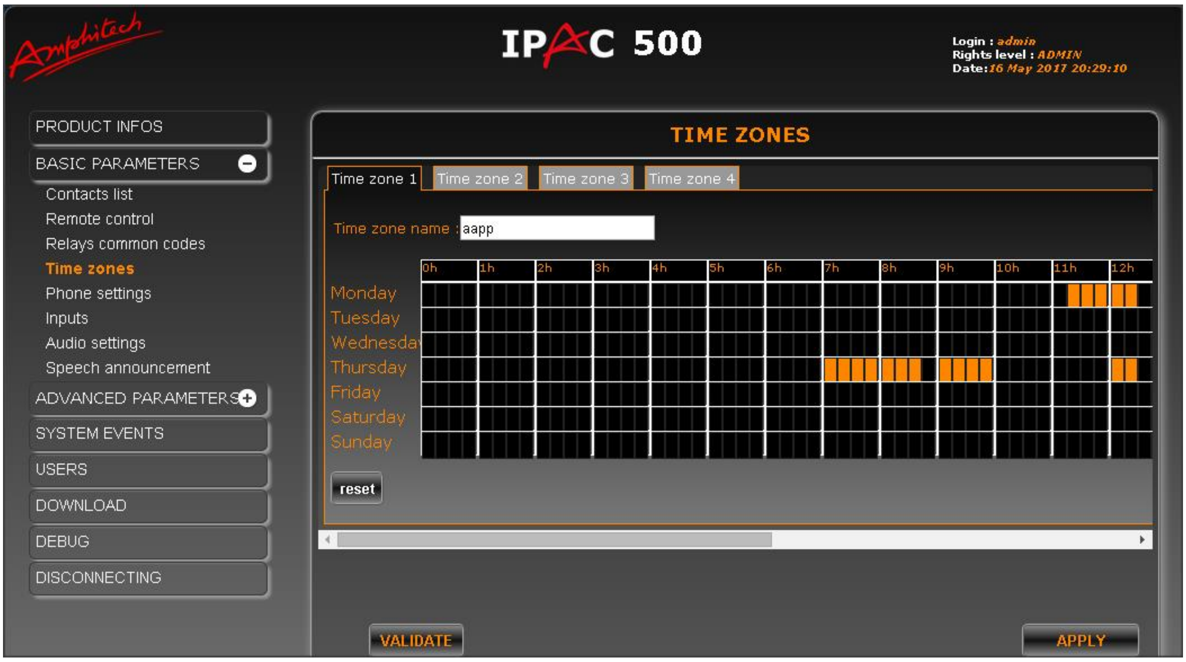
SIP Settings for phone (Advanced Parameters→SIP Accounts)



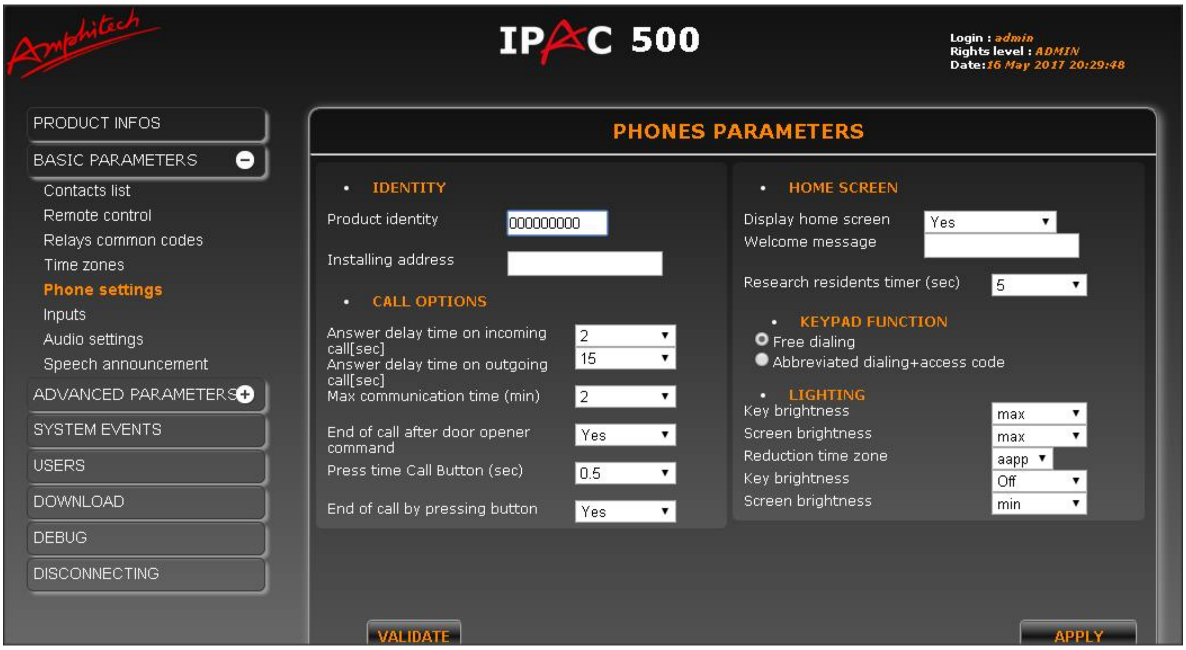
Audio Codecs(Advanced Parameters→Audio codecs)



Time Zone Setting (Basic Parameters→Time Zones)

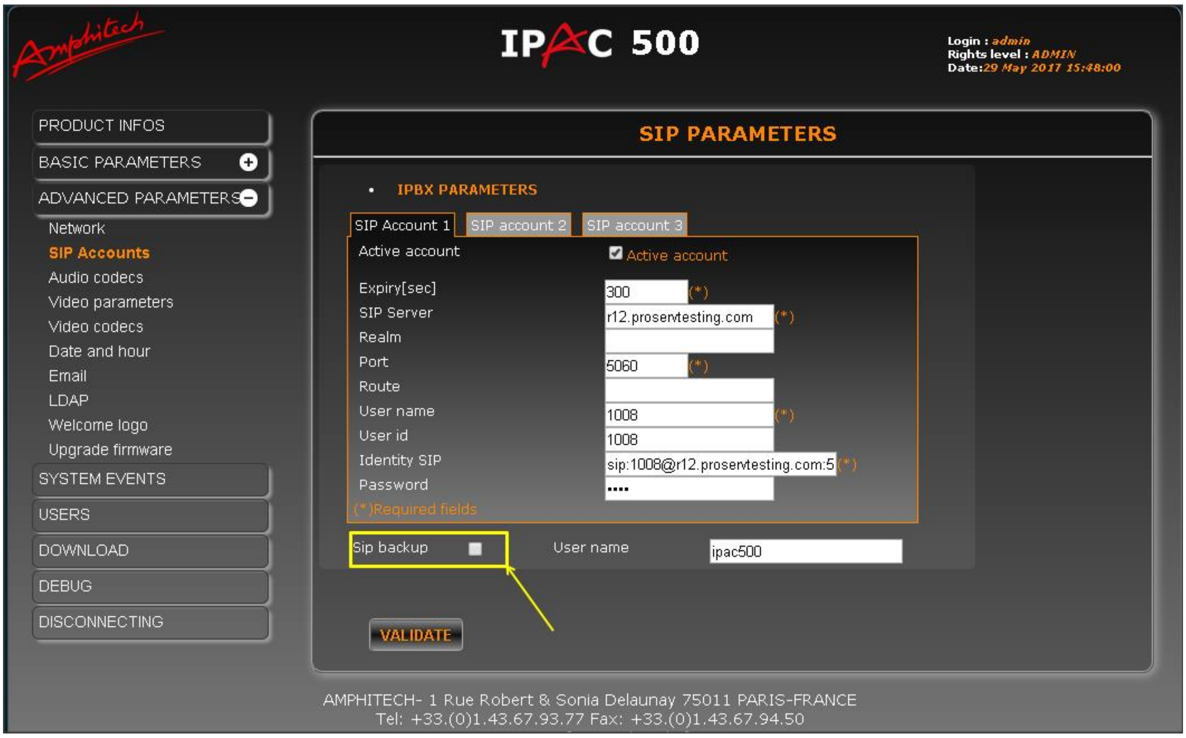


Phone Parameters (Basic Parameters→Phone Settings)



SIP Backup server configuration.

Select the SIP backup option in the configuration page.



10.2 ASIP application deployment and installation on the 8088 phone

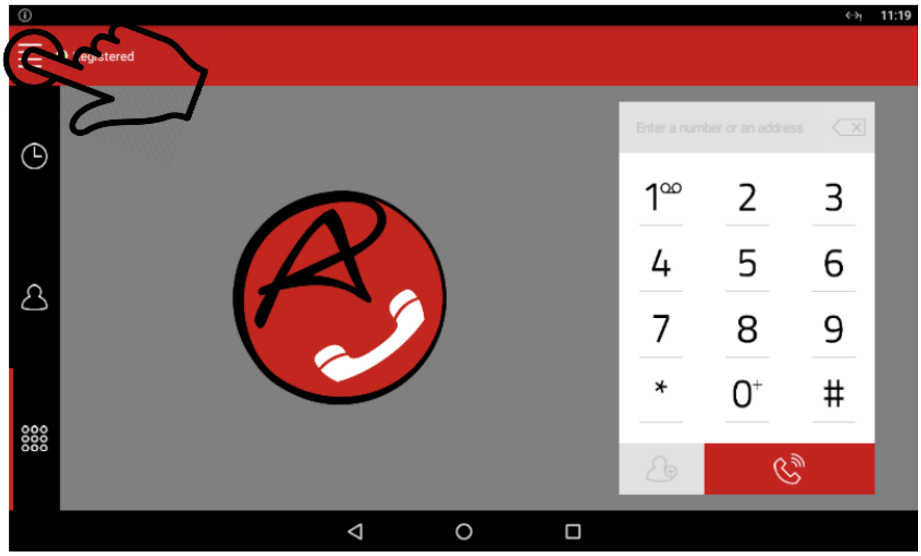
To deploy the application, the phone “private store” has to be used.
See technical communication TC2461 on the Enterprise Portal which explains how to deploy a private store.
The private must embed the Amphitech ASIP apk file.

Then, on the 8088 phone, just switch to the private store screen and install the ASIP application by pressing its icon.

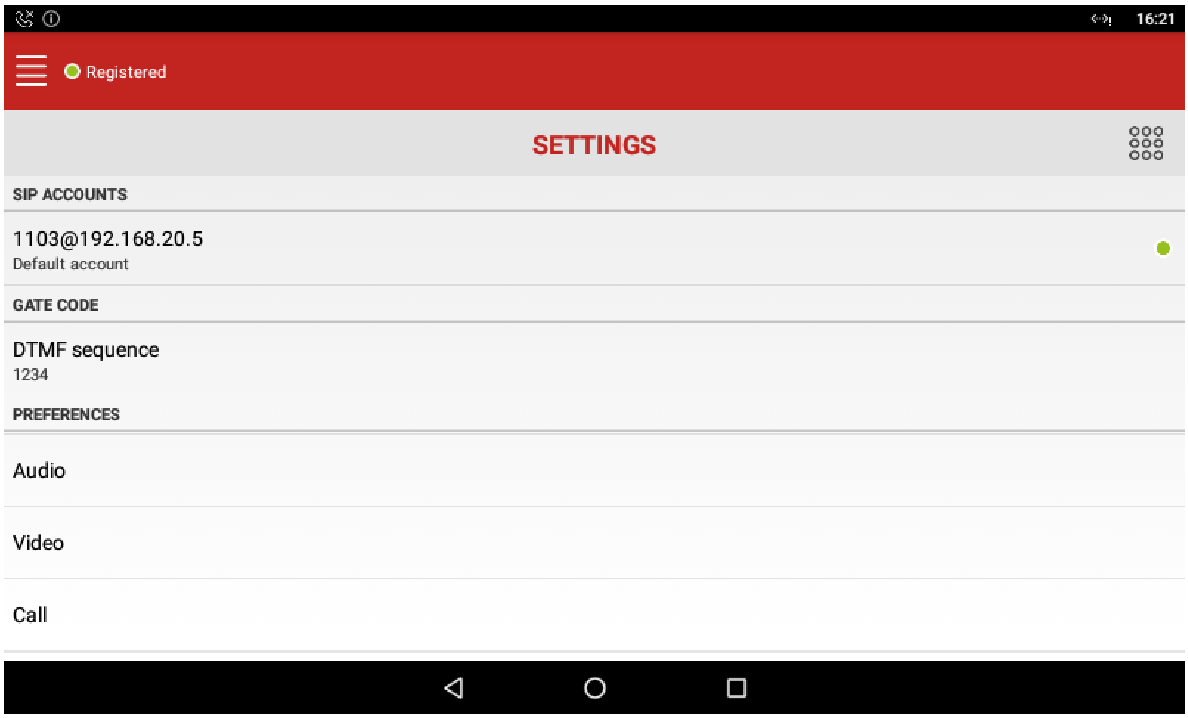
The phone will then download the application and install it.

10.3 ASIP application configuration

Go to the configuration menu:

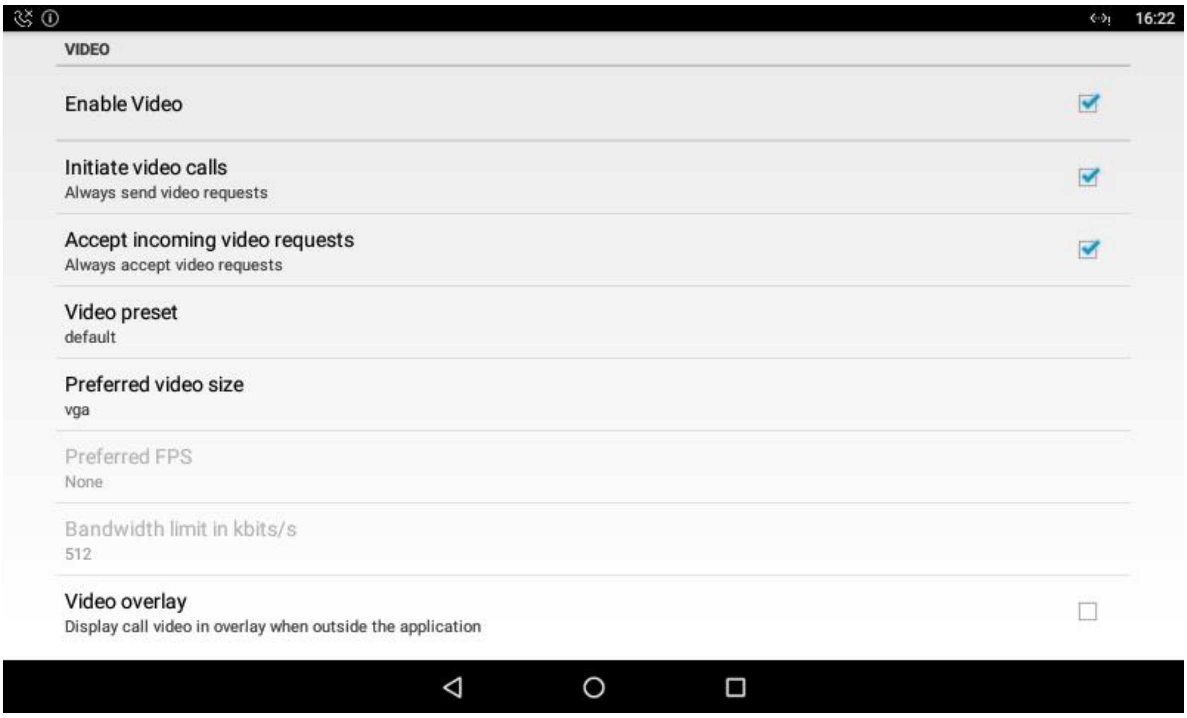


Configure the SIP account (for example, 1103 is the OXE SIP device user directory number and 192.168.20.5 the OXE IP address) and the DTMF to send to “open the door” (the same as the one defined in the IPAC 500 door phone):

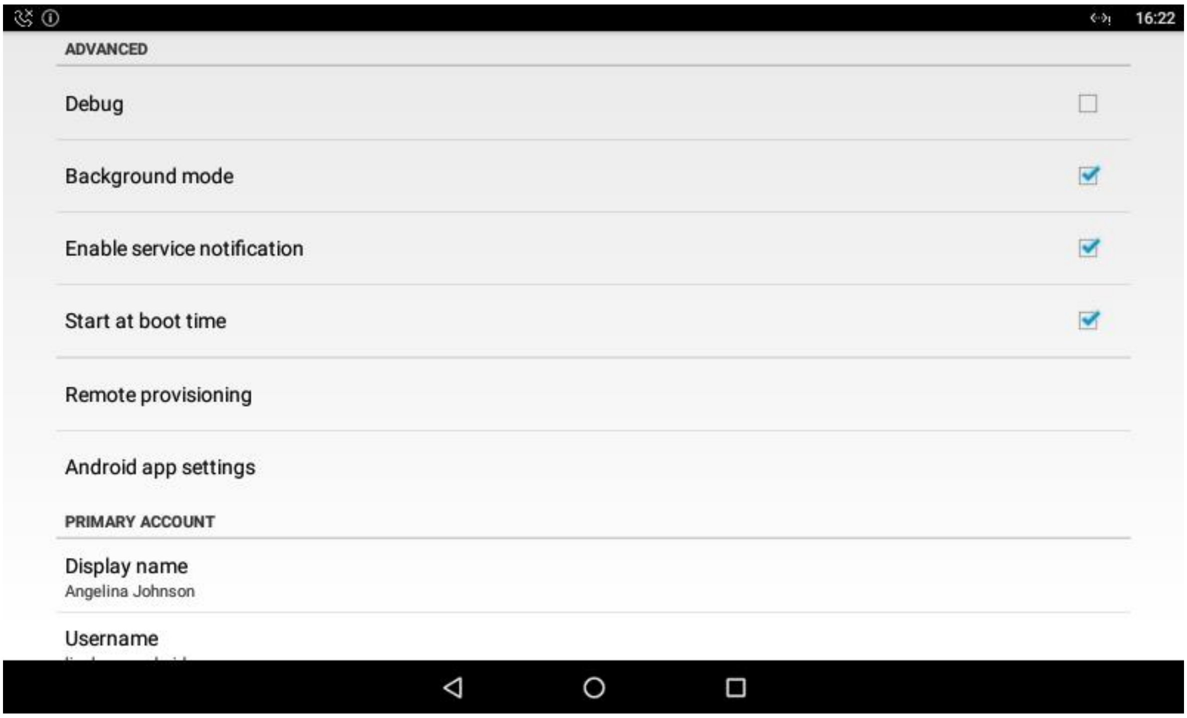


For IPAC 500 configuration, see [IWR-0254 Amphitech IPAC500 & IPAC101 / OmniPCX Enterprise R12] on the Enterprise Portal.

Enable the *video*, the *Initiate video calls* and *Accept incoming video requests*:



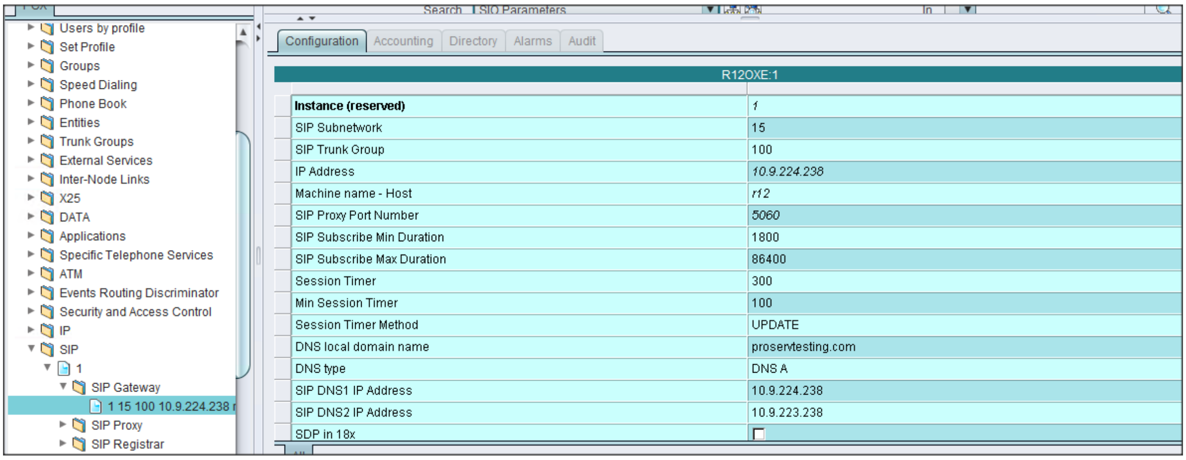
Enable the *Start at boot time* to automatic start the application after a phone reboot:



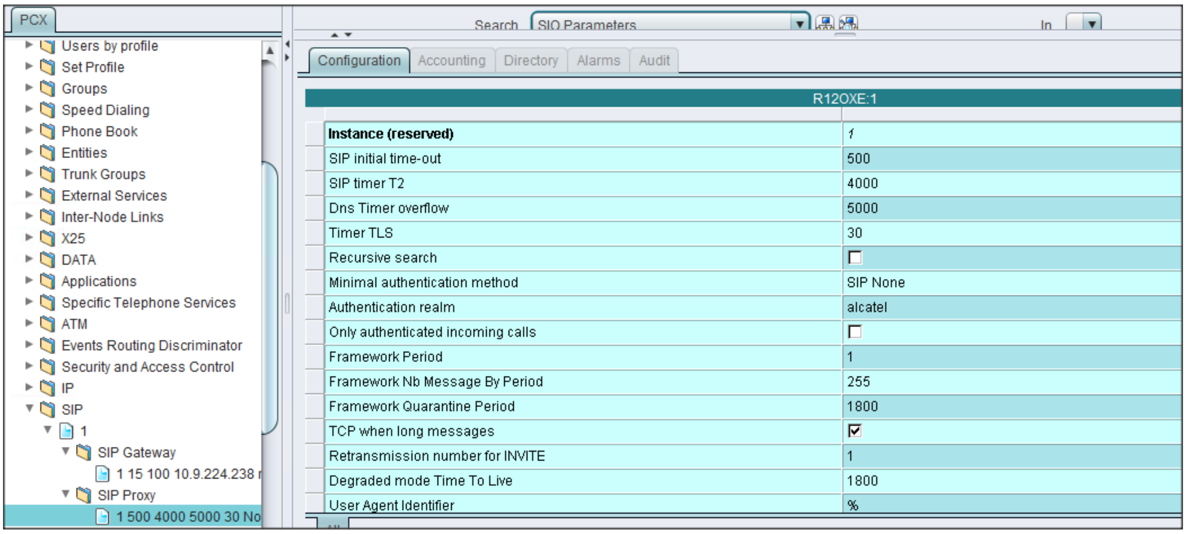
11 Appendix C: Alcatel-Lucent Enterprise Communication Platform: configuration requirements

Launch OXE configuration application.

11.1 SIP gateway

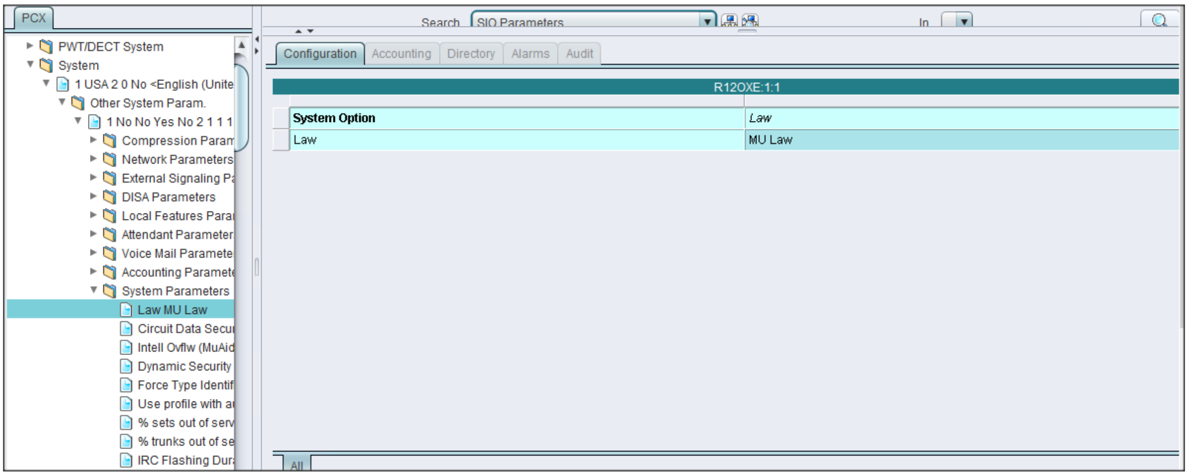


11.2 SIP Proxy

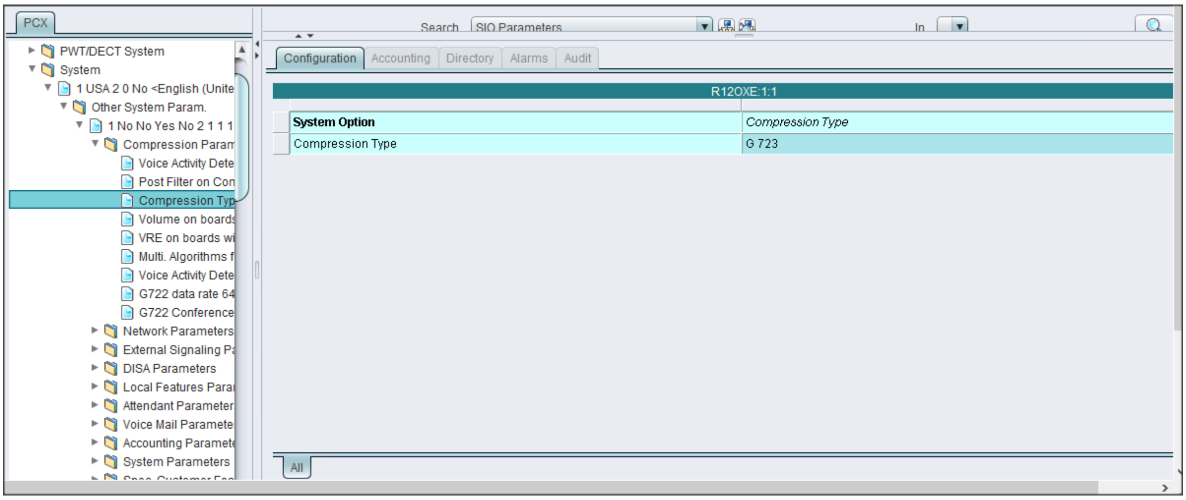


11.3 Codec:

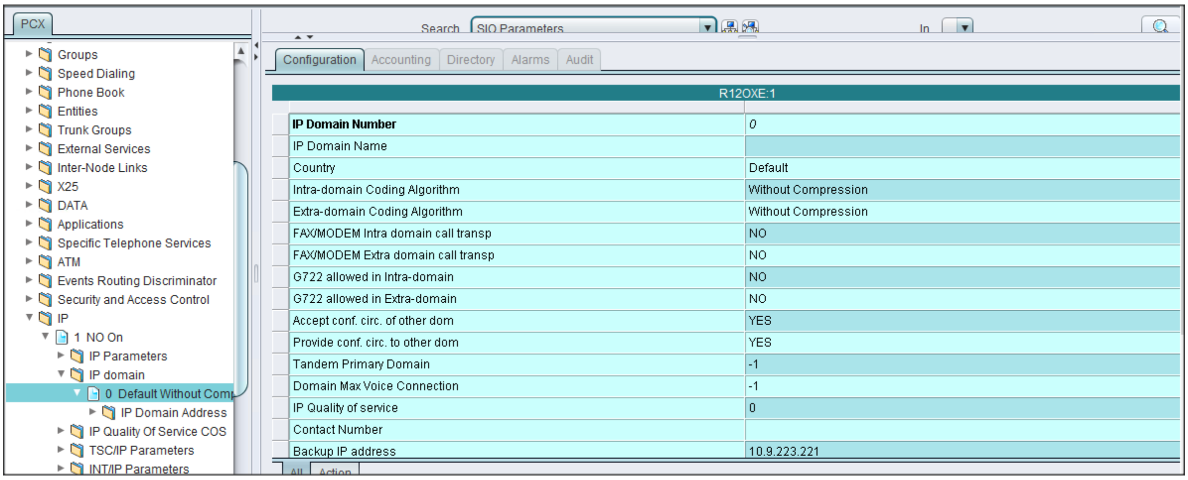
A Law/ Mu Law



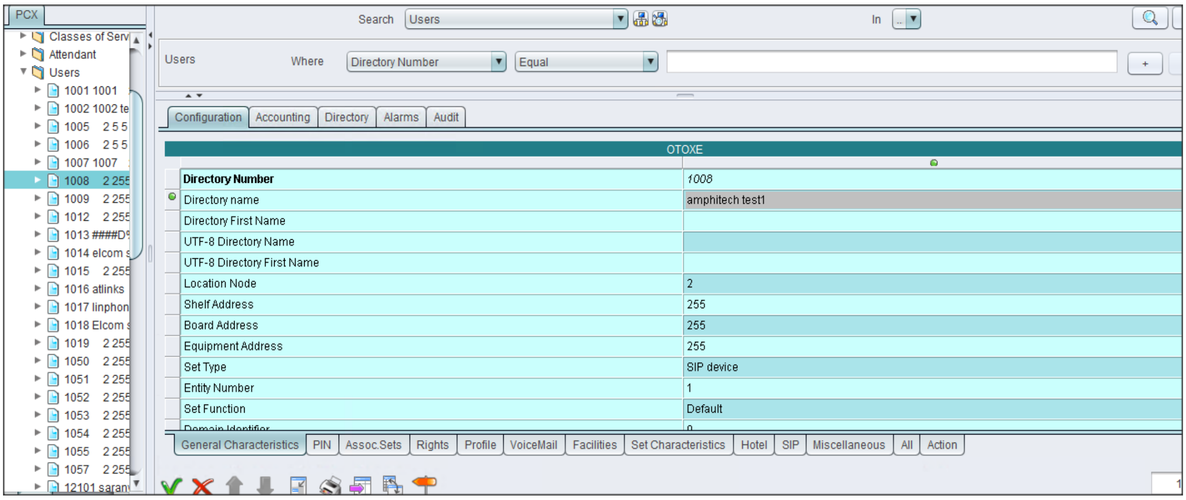
Select: **System > Other System Param. > Compression Parameters**
Compression Type Select: **G723**



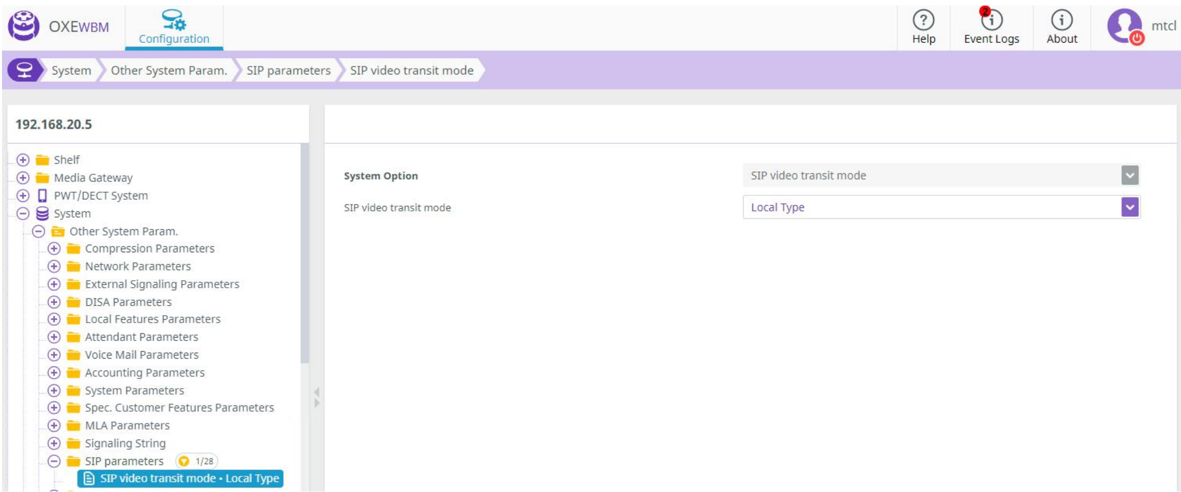
11.4 OXE domain:



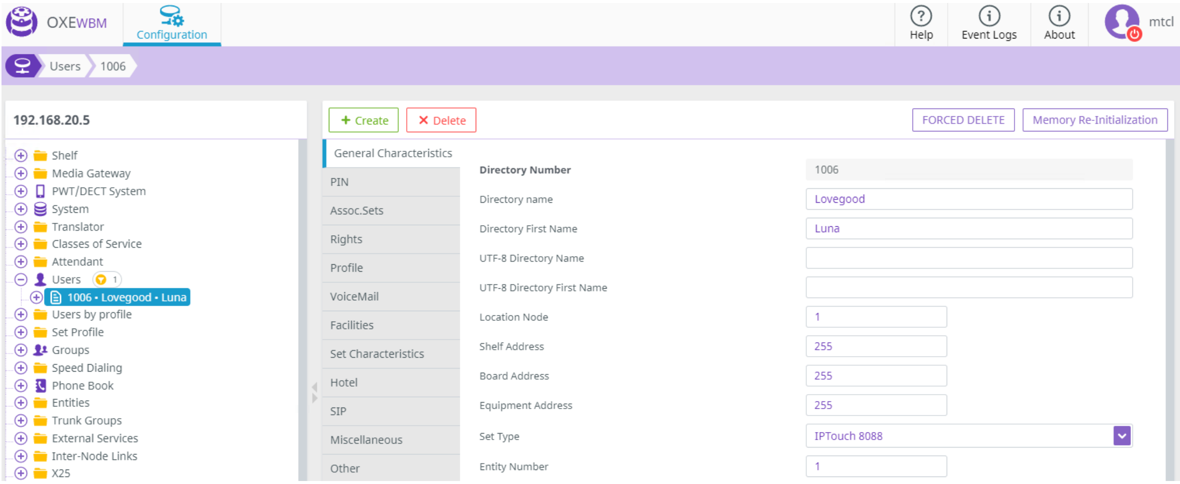
11.5 SIP user configuration:



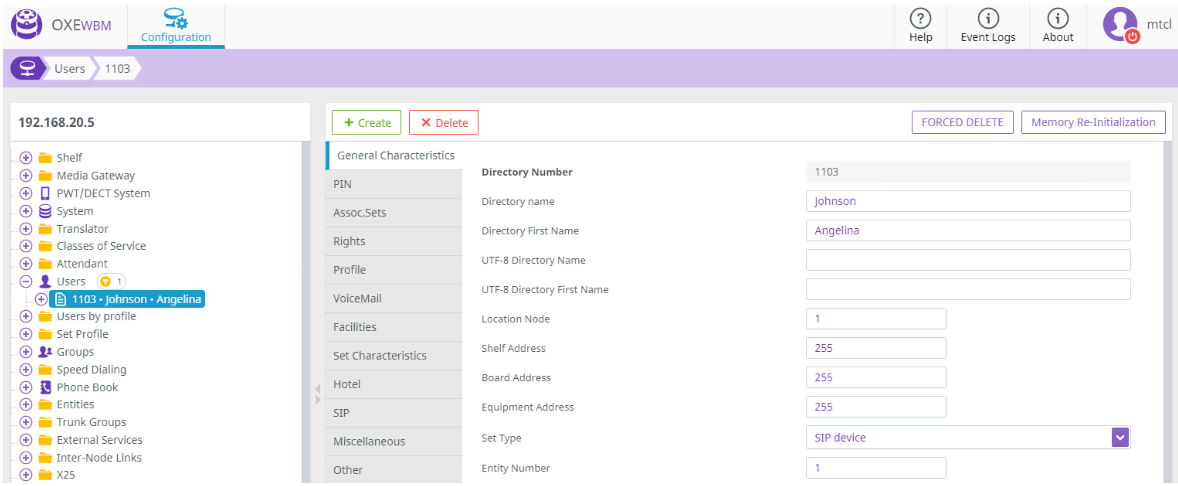
11.6 OXE SIP VIDEO SUPPORT

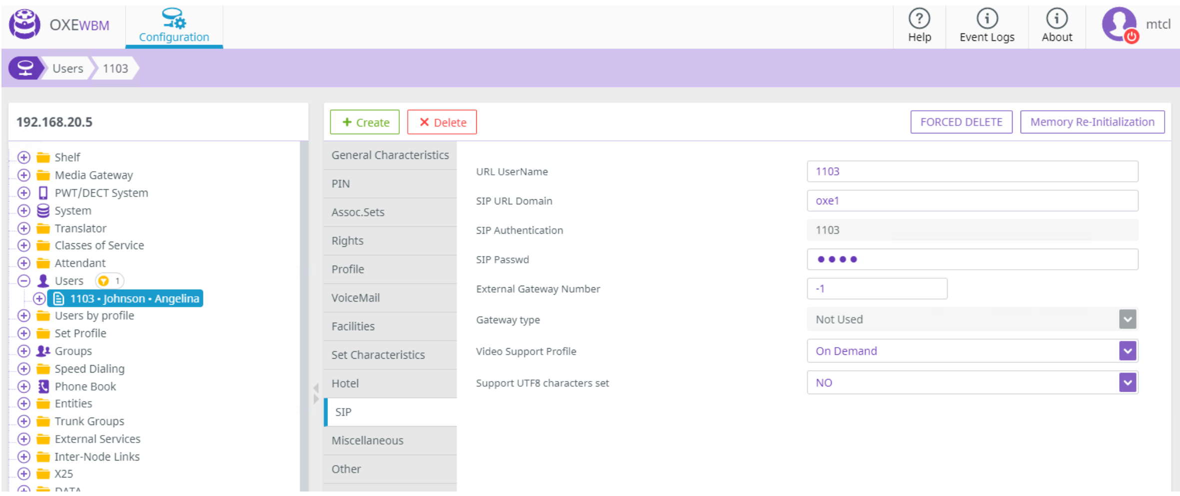


11.7 8088 phone “NOE”

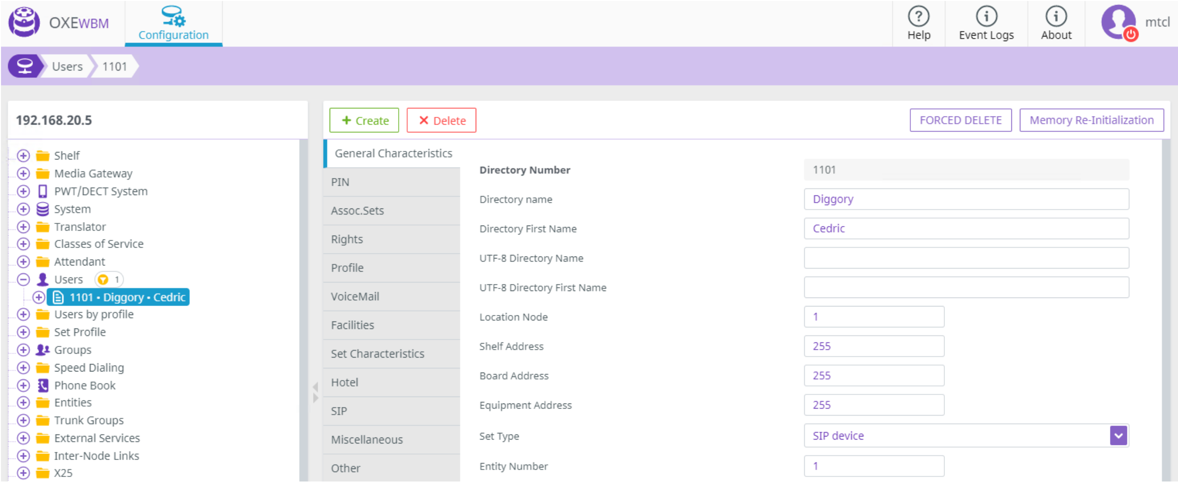


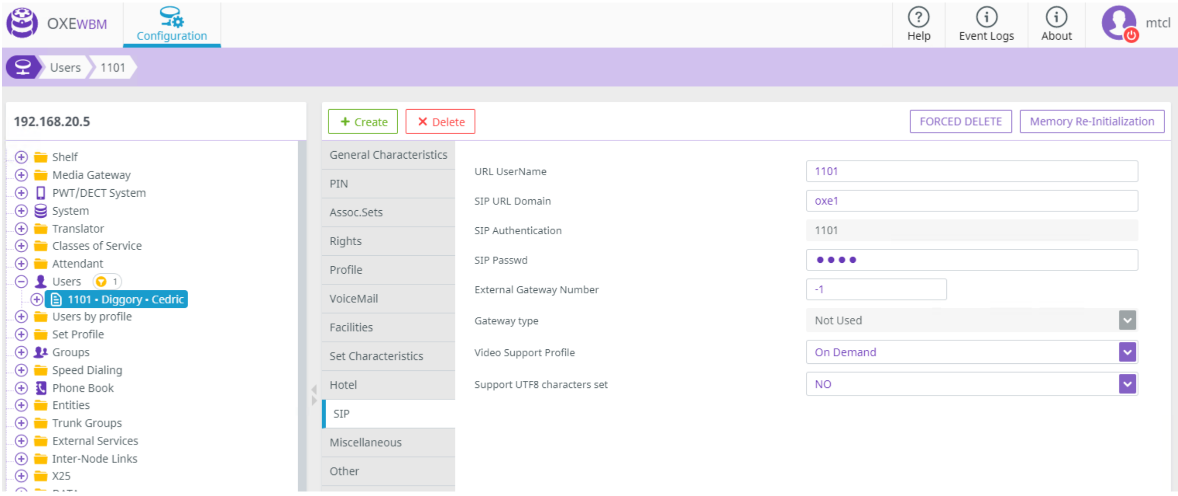
11.8 ASIP SIP device





11.9 IPAC 500 SIP device





12 Appendix D: AAPP member's escalation process

Person to contact for any questions :

- Marc Labouille : IP Project manager : mlabouille@amphitech.fr
- Jérôme Galle : Production manager : jgalle@amphitech.fr

Web site : www.amphitech.fr   and information on : wiki.amphitech.fr

AMPHITECH FRANCE
SAV/Support : Phone : +33 (0)1 43 67 96 74
1, rue Robert et Sonia Delaunay
F - 75011 Paris - FRANCE
Phone : +33 (0)1 43 67 98 09
Fax : +33 (0)1 43 67 13 97

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 Enterprise's product family. The program provides tools and support for developing, verifying and promoting compliant third-party applications that complement Alcatel-Lucent Enterprise's product family. ALE International facilitates market access for compliant applications.

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

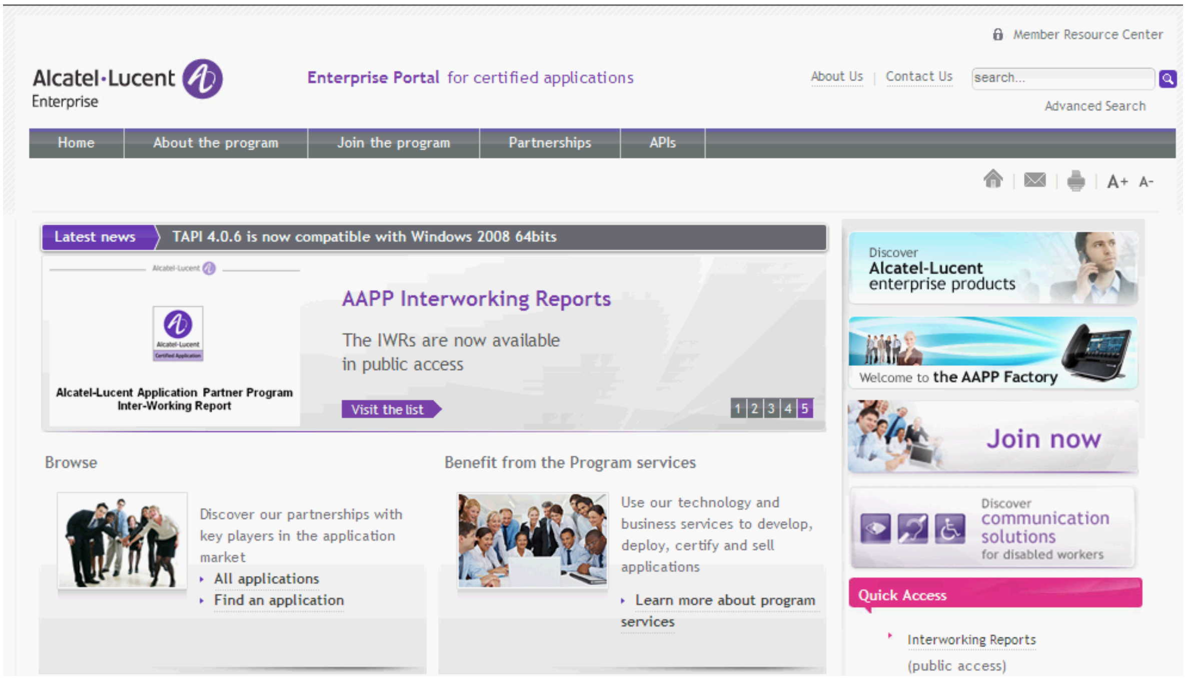
- **Provide easy interfacing for Alcatel-Lucent Enterprise communication products**: Alcatel-Lucent Enterprise's communication products for the enterprise market include infrastructure elements, platforms and software suites. To ensure easy integration, the AAPP provides a full array of standards-based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

- **Test and verify a comprehensive range of third-party applications**: to ensure proper inter-working, ALE International tests and verifies selected third-party applications that complement its portfolio. Successful candidates, which are labelled Alcatel-Lucent Enterprise 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 Enterprise.Alcatel-Lucent.com

You can access the Alcatel-Lucent Enterprise website at this URL: http://enterprise.alcatel-lucent.com/

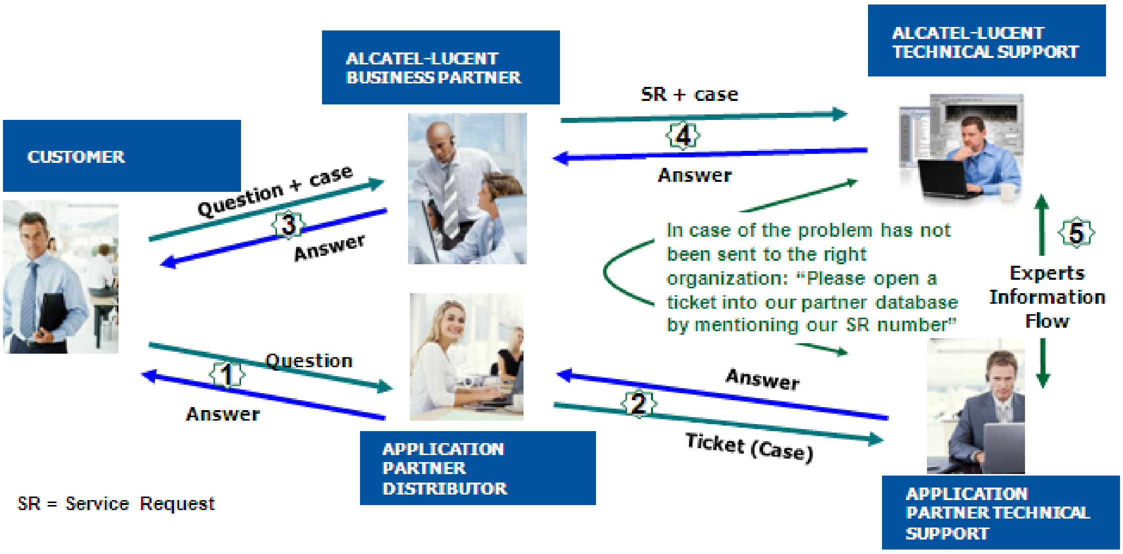
14 Appendix F: AAPP Escalation process

14.1 Introduction

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

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

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



(*) The Application Partner Business Partner can be a Third-Party company or the ALE International 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, ALE International and the Application Partner, are engaged:

Case 1: the responsibility can be established 100% on ALE International side.
In that case, the problem must be escalated by the ALE Business Partner to the ALE International 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 ALE International Business Partner will escalate the problem to the ALE International Support Center only if the Application Partner has demonstrated with traces a problem on the ALE International side or if the Application Partner (not the Business Partner) needs the involvement of ALE International

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

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

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

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

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

Please check the availability of the Inter-Working Report on 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 ALE International Business Partner is mandatory, the access to the Alcatel-Lucent Enterprise platform (remote access, login/password) being the Business Partner responsibility.

14.3 Escalation in all other cases

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

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

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

14.4 Technical support access

The ALE International **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 ALE International Business Partners Web site (if registered Alcatel-Lucent Enterprise Business Partners): https://businessportal2.alcatel-lucent.com click under "Contact us" the *eService Request* link
- e-mail: Ebg_Global_Supportcenter@al-enterprise.com
- Fax number: +33(0)3 69 20 85 85
- Telephone numbers:

ALE International 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

END OF DOCUMENT