Alcatel-Lucent Application Partner Program
Inter-Working Report

Partner: Mediatrix
Application type: Analog equipments SIP Gateway
Application name: Mediatrix 4102 & 4116
Alcatel-Lucent Platform: OmniPCX Office

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

Certification overview

Date of the certification	March 2013

Alcatel-Lucent's representative	Alain Botti
AAPP member representative	Roberto Roman

Alcatel-Lucent Communication Platform	OmniPCX Office
Alcatel-Lucent Communication Platform Release	900/042.001
AAPP member application version	Dgw 2.0.24.402
Application Category	Gateway

Author(s): Florian Residori, Alain Botti
Reviewer(s): Denis Lienhart, Roberto Roman, Bertrand Goldfarb, Eric Rodrique

Revision History

Edition 1: creation of the document – *March 2013*
Edition 2: minor updates of the document *and firmware update* Dgw 2.0.24.402– *April 2013*

Test results

☐ Passed ☐ Refused ☐ Postponed

☑ Passed with restrictions

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

IWR validity extension

The tests have been performed with Mediatrix 4102 and 4116 gateways. The certification is extended to the following devices powered by the save firmware version:

- Mediatrix 4104 (based on the 4102 results)
- Mediatrix 4108 (based on the 4116 results)
- Mediatrix 4124 (based on the 4116 results)

AAPP Member Contact Information

Contact name:	Roberto Roman
Title:	VoIP System Specialist
Address:	4229, Garlock Street Quebec
Zip Code:	J1L 2C8
City:	Sherbrooke
Country:	Canada
Country code:	1
Phone:	1-819-829-8749
Fax:	1-819-829-5100
Web site:	http://www.media5corp.com
E-mail address:	tac@media5corp.com

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1 Introduction

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

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

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

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 Limits of the Technical support*

3 Limits of the Technical support

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

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

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

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

3.1 Case of additional Third party applications

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

4 Application information

Application type: Telephone Adapter / VoIP Gateway for Analog equipments

Application commercial name: Mediatrix 4102 and 4116

Application version: Dgw 2.0.24.402

Interface type : SIP/Ethernet

Interface version (if relevant): -

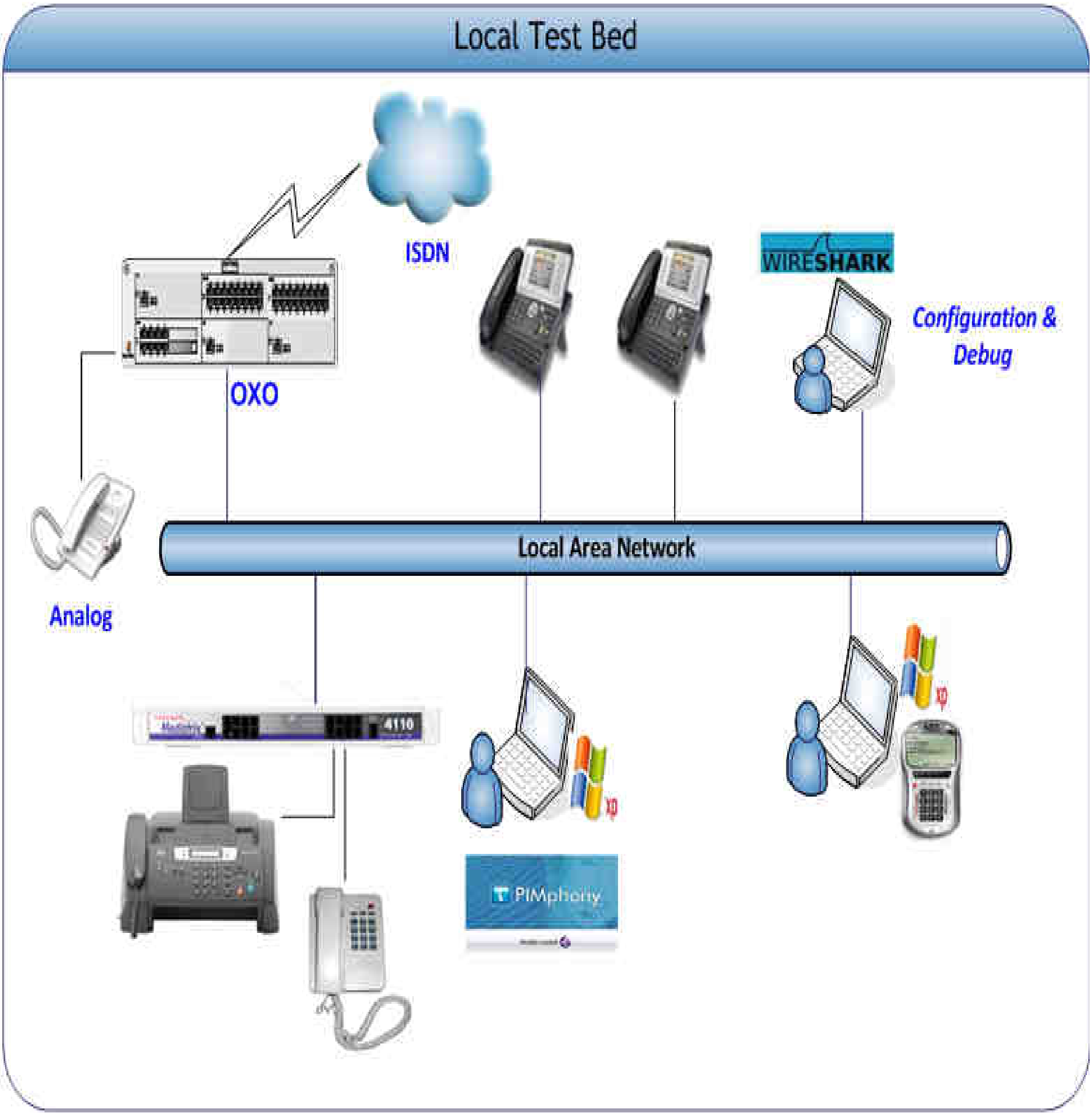
Brief application description:

Mediatrix:

The Mediatrix® 4100 Series provides a solution for the transparent integration of faxes, modems, point of sales and analog phones to IP Communication Systems. The 4100 Series are available in 2, 4, 8, 16 and 24 FXS ports designed to meet the varied application requirements of Service Providers.

The Mediatrix 4100 Series VoIP adaptors are high-quality, cost efficient VoIP gateways connecting small to large branch offices and multi-tenant buildings to an IP network, while preserving existing investments in analog telephones and faxes.

5 Test Environment



Alcatel-Lucent Communication Platform:

- OmniPCX Office Rack
- PowerCPU
- Release: 900/042.001
- OMC: 900/11.1a

Setup Details:

Setup Information OXO 1	
OXO 1 IP address	10.130.158.45
Domain name	Oxo1testing.proservtesting.com
Voicemail numbers	214 -221
Attendant number	200
OXO Extension Details used for test	
IP Touch extension numbers	IPset-1 : 230 IPset-2 : 229 IPset-3 : 231
Analog phone extension numbers	Zset-1 :208 Zset-1 : 209
OXO fax extension numbers	FAXset-1: 208
Analog gateway fax extension numbers	GWFAXset-1 :228

Setup Information OXO 2	
OXO 1 IP address	10.130.158.48
Domain name	Oxo2testing.proservtesting.com
Voicemail number	314
Attendant number	0
OXO Extension Details used for test	
IP Touch extension numbers	IPset-1 :310 IPset-2 : 311 IPset-3 : 312
Analog phone extension numbers	Zset-1 : 325 Zset-1 : 326
OXO fax extension number	FAXset-1: 327
Analog gateway fax extension numbers	GWFAXset-1 :328

5.1 Software configuration

- **Alcatel-Lucent Communication Platform**: OmniPCX Office 900/042.001
- **Partner Application**:

Note: Analog phones are registered in the OmniPCX Office as "Open SIP phone".
Fax phones are registered in the OmniPCX Office as "Basic SIP phone".

6 Summary of test results

6.1 Summary of main functions supported

Features	Status	Comments
Initialization including network configuration	OK	
SIP registration	OK	
SIP authentication	OK	
Voice over IP and RTP codec support	OK	
Outgoing Call	OK	
Incoming Call	OK	
Features During Conversation	OK	
Call Transfer	OK	
Attendant	OK	
Voice mail interaction and indication	OK	
Fax test	OK	

6.2 Summary of problems

-

6.3 Summary of limitations

- DND Option is not available in Mediatrix GUI Interface.
- G723 is not supported by the Gateway
- The Mediatrix 4102 supports the FSK signal to enable the visual indicator and does not provide voltage for the indication. **So message waiting indication will not work in phones which require voltage for the indication with 4102**. Mediatrix 4116 provides both voltage and FSK signal and so MWI will work in this model alone.

6.4 Notes, remarks

- Analog phone are registered in the OmniPCX Office as "Open SIP phone".
- Fax phone are registered in the OmniPCX Office as "Basic SIP phone".

7 Test Result Template

The results are presented as indicated in the example below:

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Test case 1** • 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 Alcatel-Lucent side or on Application Partner side</u>
Comment: to be filled in with any relevant comment. Mandatory in case a test has failed especially the reference number of the issue.

8 Test Results

8.1 Analog phones tests

In this section analog phones are connected as Open SIP device on OXO though the analog gateway. These phones acts as OXO sets, so system features are available (prefix, suffix for example)

8.1.1 Connectivity and Setup

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

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

8.1.2 Audio codec negotiations/ VAD / Framing

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

Phone configuration: configure the analog gateway to use G.722, G.711 A-law, G.711 mu-law, G.729, G.723 in this order (unless otherwise stated).
Configure IP Touch with codec G.711 and to NOT use VAD (unless otherwise stated).

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	Select G711 A-law as 1st codec in the analog gateway Call from SIP Zset-2 to IP Touch IPset-2 Check that the call is established in G711 A-law. Check audio quality Call from IP Touch IPset-2 to SIP Zset-2 Check that the call is established in G711 A-law. Check audio quality	☐	☒	☐	
2	Select G729 as 1st codec in the analog gateway Call from SIP Zset-2 to IP Touch IPset-2 Check that the call is established in G729 Check audio quality Call from IP Touch IPset-2 to SIP Zset-2 Check that the call is established in G729 Check audio quality	☐	☒	☐	
3	Select G723 as 1st codec in the analog gateway Call from SIP Zset-2 to IP Touch IPset-2 Check that the call is established in G723 Check audio quality Call from IP Touch IPset-2 to SIP Zset-2 Check that the call is established in G723 Check audio quality	☒	☐	☐	G723 is not available on Mediatrix gateway
4	**Configure** Zset-2 **to use VAD** **Configure IP Touch** IPset-2 **NOT to use VAD** Call from SIP Zset-2 to IP Touch IPset-2 Check that the call is established in G711 A-law. Check audio quality **Configure SIP** Zset-2 **to use VAD** **Configure IP Touch** IPset-2 **to use VAD** Redo the same tests **Configure SIP** Zset-2 **NOT to use VAD** **Configure IP Touch** IPset-2 **to use VAD** Redo the same tests	☐	☒	☐	
5	**In OXO enable codec pass through for SIP phones** **Call from SIP** Zset-1 **to SIP** Zset-2 Check that the call is established using G.722 Check audio quality	☐	☒	☐	G722 is not available on Mediatrix gateway Tested with G711

6	**In OXO 1 and OXO 2 enable codec pass through for SIP phone ; direct RTP and codec pass through for SIP trunk. G723 is preferred codec in the analog gateway** **Call from SIP** Zset-1 **to Network SIP** NwkZset-1 Check that the call is established using direct RTP in G723. Check audio quality	☐	☒	☐	G723 is not available on Mediatrix gateway Tested with G729
7	**In OXO 1 and OXO 2 disable direct RTP and codec pass through for SIP trunk ARS table is configured with “default” codec. G723 is preferred codec in the analog gateway** **Call from SIP** Zset-1 **to Network SIP** NwkZset-1 Check that the call is established in G711. Check audio quality	☐	☒	☐	
8	**In OXO 1 and OXO 2 disable direct RTP and codec pass through for SIP trunk** **ARS table is configured with codec G729_30** **Call from SIP** Zset-1 **to Network SIP** NwkZset-1 Check that the call is established in G729. Check audio quality	☐	☒	☐	

8.1.3 Outgoing Calls

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

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

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

Test Case Id	Test Case	N/A	OK	NOK	Comment
7	**Call to user in “Do not Disturb” (DND) state** **(SIP: “480 Temporarily not available”)** Dial “*63” on the IP Touch IPset-1 in order to enable the DND. Wait for acknowledgement ring back tone from OXO. With the SIP phone Zset-2 call IPset-1. Check ring back tone and display. Redial *60 on IPset-1 to cancel the DND	☐	☒	☐	
8	**Call to local user, immediate forward (CFU).** **(SIP: “181 Forwarded”)(1)** On IP Touch IPset-1 dial the *61IPset-2 to activate the CFU. Wait for acknowledgement ring back tone from OXO. With the SIP phone Zset-2 call the IPset-1. Check that IPset-2 is ringing and the display. Take the call check audio and hung up. Dial *60 on IPset-1 for forward cancellation.	☐	☒	☐	
9	**Call to local user, forward on no reply (CFNR). (1)** On IP Touch IPset-1 configure with OMC the CFNR using dynamic routing to IPset-2. With Zset-2 call the IPset-1. Check that IPset-1 is ringing but don’t take the call and wait the time out (about 30 sec). Time out is defined in IPset-1 dynamic routing of Timer 1. After time out check that IPset-2 is ringing and take the call. Check the audio and display.	☐	☒	☐	
10	**Call to local user, forward on busy (CFB). (1)** On IP Touch IPset-1 dial the *62IPset-2 (*62+<target MCDU number>) to activate the CFB. Wait for acknowledgement ring back tone from OXO. With SIP phone Zset-2 call IPset-1 and take the call to make it busy. With other SIP phone Zset-3 call IPset-1. Check that IPset-2 is ringing and take the call. Check the audio and display. Dial *60 on IPset-1 for forward cancellation.	☐	☒	☐	
11	**Call to external number** **(Check ring back tone, called party display)** With SIP set Zset-2 dial 9 (9 prefix +external number) Take the call and check audio, display and call release.	☐	☒	☐	
12	**SIP session timer expiration: Check if call is maintained or released after the session timer has expired** With SIP set Zset-2 call IP Touch IPset-1. Take the call on IPset-1 and never hang up, wait for time out expiration. Check that call is maintained or release.	☐	☒	☐	

Notes:

(1) For test cases with call to forwarded user: User is forwarded to another local user. Special case of forward to Voice Mail is tested in another section.

8.1.4 Incoming Calls

Calls will be generated using the numbers or the name of the SIP user.
SIP terminal will be called in different states: free, busy, out of service, forward.
The states are to be set by the appropriate system prefixes unless otherwise noted.
Points to be checked: tones, voice during the conversation, display (on caller and called party), hang-up phase.

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

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Local /network call to free SIP terminal** Local: with IP Touch IPset-1 call SIP set Zset-2. Check that Zset-2 is ringing and take the call Check ring back tone and called party display. Network: with IP Touch IPset-1 call SIP set NwkZset-2 on another Node. Check that NwkZset-2 is ringing and take the call. Check ring back tone and called party display.	☐	☒	☐	
2	**Local/network call to busy SIP terminal** Local: With SIP set Zset-3 call other SIP set Zset-2 and take the call to make it busy, don't hang up. With IP Touch IPset-2 call Zset-2 which is busy Check the ring back tone and display. Network: With SIP set Zset-2 call SIP set NwkZset-2 and take the call to make it busy, don't hang up. With IPset-1 call NwkZset-2 which is busy Check ring back tone and called party display.	☐	☒	☐	
3	**Local/network call to unplugged SIP terminal** Local: Unplug the Zset-2 SIP set and call it with IP Touch IPset-1. Check the ring back tone and display Network: Unplug the SIP set NwkZset-2 and call it with IPset-1 Check the ring back tone and display	☐	☒	☐	
4A	**Local/network call to SIP terminal in Do Not Disturb (DND) mode** **By local feature if applicable:** Local: Enable DND on SIP set Zset-2 and call it with IP Touch IPset-1 Check the ring back tone and display Cancel the DND on Zset-2. Network: Enable DND on SIP set NwkZset-2 and call it with IP Touch IPset-1 Check the ring back tone and display Cancel the DND on Zset-2.	☒	☐	☐	DND Feature is not available in Mediatrix

Test Case Id	Test Case	N/A	OK	NOK	Comment
4B	**By system feature** <u>Local</u>: Enable DND on SIP set Zset-2 using the *63 prefix.. Wait for acknowledgement ring back tone from OXO. With IP Touch IPset-1 call Zset-2 Check the ring back tone and display Cancel the DND on Zset-2 using *63 prefix. <u>Network</u>: Enable DND on SIP set NwkZset-2 using the *63 prefix. Wait for acknowledgement ring back tone from OXO. With IP Touch IPset-1 call NwkZset-2 Check the ring back tone and display Cancel the DND on NwkZset-2 using * 60 prefix.	☐	☒	☐	
5A	**Local/network/SIP call to SIP terminal in immediate forward (CFU) to local user:** **By local feature if applicable:** <u>Local</u>: On SIP set Zset-2 enable CFU to IP Touch IPset-1 With SIP set Zset-3 call Zset-2. Check that IPset-1 is ringing. Take the call and check audio and display. Disable CFU on Zset-2. <u>Network</u>: On SIP set NwkZset-2 enable CFU to IP Touch NwkIPset-1. With SIP set Zset-2 call NwkZset-2. Check that NwkIPset-1 is ringing. Take the call and check audio and display. Disable CFU on NwkZset-2.	☐	☒	☐	
5B	**By system feature:** <u>Local</u>: On SIP set Zset-2 enable CFU to IP Touch IPset-1 using *61IPset-1 prefix (*61 + <target MCDU number>). Wait for acknowledgement ring back tone from OXO. With SIP set Zset-3 call Zset-2. Check that IPset-1 is ringing. Take the call and check audio and display. Disable CFU on Zset-2 using *60 prefix. <u>Network</u>: On SIP Set NwkZset-2 enable CFU to IP Touch NwkIPset-1 using *61 + <target MCDU number>. Wait for acknowledgement ring back tone from OXO. With SIP Set Zset-3 call NwkZset-2. Check that NwkIPset-1 is ringing. Take the call and check audio and display. Disable CFU on NwkZset-2 using *60 prefix.	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
B	**Local/network/SIP call to SIP terminal in immediate forward (CFU) to network number:** **By local feature if applicable:** Local: On SIP Set Zset-3 enable CFU to SIP Set NwkZset-1.With SIP set Zset-2 call Zset-3. Check that NwkZset-1 is ringing. Take the call and check audio and display. Disable CFU on Zset-3. Network: On SIP Set Zset-2 enable CFU to IP Touch NwkIPset-1. With SIP Set NwkZset-2 call Zset-2. Check that NwkIPset-1 is ringing. Take the call and check audio and display. Disable CFU on Zset-2.	☐	☒	☐	
6B	**By system feature:** Local: On SIP Set Zset-2 enable CFU to SIP Set NwkZset-1 using *61NwkZset-1 prefix (*61 + <target MCDU number>). Wait for acknowledgement ring back tone from OXO. With SIP set Zset-3 call Zset-2. Check that NwkZset-1 is ringing. Take the call and check audio and display. Disable CFU on Zset-2 using *60 prefix. Network: On SIP Set Zset-2 enable CFU to IP Touch NwkIPset-1 using *61 + <target MCDU number>. Wait for acknowledgement ring back tone from OXO. With SIP Set NwkZset-2 call Zset-2. Check that NwkIPset-1 is ringing. Take the call and check audio and display. Disable CFU on Zset-2 using *60 prefix.	☐	☒	☐	
7A	**Local/network/SIP call to SIP terminal in immediate forward (CFU) to another SIP user** **By local feature if applicable:** Local: On SIP set Zset-2 enable CFU to SIP set NwkZset-1. With Zset-3 call Zset-2. Check that NwkZset-1 is ringing. Take the call and check audio and display. Disable CFU on Zset-2. Network: On SIP set Zset-2 enable CFU to IP Touch NwkIPset-3. With SIP Set NwkZset-1 call Zset-2. Check that NwkIPset-3 is ringing. Take the call and check audio and display. Disable CFU on Zset-2.	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
7B	**By system feature:** Local: On SIP Set Zset-3 enable CFU to SIP Set NwkZset-1 using *61 + <target MCDU number>. Wait for acknowledgement ring back tone from OXO. With SIP Set Zset-2 call Zset-3. Check that NwkZset-1 is ringing. Take the call and check audio and display. Disable CFU on Zset-3 using *60 prefix. Network: On SIP Set Zset-3 enable CFU to IP Touch NwkIPset-3 using *61 + <target MCDU number>. Wait for acknowledgement ring back tone from OXO. With SIP Set NwkZset-1 call Zset-3. Check that NwkIPset-3 is ringing. Take the call and check audio and display. Disable CFU on Zset-3 using *60 prefix	☐	☒	☐	
8A	**Local call to SIP terminal in "forward on busy" (CFB) state:** **By local feature if applicable** On SIP Set Zset-2 enable CFB to IP Touch IPset-1 With Zset-2 call the voice mail to make it busy. With SIP Set Zset-3 call Zset-2 which is busy. Check that IPset-1 is ringing Take the call and check audio and display. Disable CFU on Zset-2.	☐	☒	☐	
8B	**By system feature:** On SIP Set Zset-2 enable CFB to IP Touch IPset-1 using *62 + <target MCDU number>. Wait for acknowledgement ring back tone from OXO. With Zset-2 call the voice mail to make it busy. With SIP Set Zset-3 call Zset-2 which is busy. Check that IPset-1 is ringing Take the call and check audio and display. Disable CFB on Zset-2 using *60 prefix.	☐	☒	☐	
9A	**Local call to SIP terminal in "forward on no reply" (CFNR)** **By local feature if applicable** On SIP Set Zset-3 enable CFNR to IP Touch IPset-1 With SIP Set Zset-2 call Zset-3. Check that Zset-3 is ringing and don't take the call, wait for time out (about 30 seconds). After time out expiration the IPset-1 is ringing, take the call and check audio and display.	☐	☒	☐	
9B	**By system feature** CNFR via prefix not available on OXO (dynamic routing has to be used)	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
10	**Call to busy user, Call waiting.** (Camp-on), local feature if applicable: With SIP Set Zset-2 call other SIP Set Zset-3 (multiline set) to make it busy, take the call and don't hang up. With IP Touch IPset-2 call Zset-3 (on Zset-3 camp-on feature is enabled). Check the Call waiting or ring back tones and display	☐	☒	☐	Busy tone is heard
11	**External call to SIP terminal.** Check that external call back number is shown correctly: With SIP Set Zset-3 dial 9 + target MCDU number. Check that external is ringing and the external call number is shown correctly Take the call and check audio, display and call release.	☐	☒	☐	External number is shown correctly in the external device
12	**Calling Line Identity Restriction (CLIR): Local call to SIP terminal.** On IP Touch IPset-2 enable mask Identity and call SIP Set Zset-3 in order to hide IPset-2 identity. Check that Zset-3 is ringing, take the call and check that IPset-2 identity is hidden.	☐	☒	☐	
13	Display: Call to free SIP terminal (Analog Phone connected behind the Sip Gateway) from IP Touch user with a name containing non-ASCII characters (eg éêêèè). Check that SIP set is ringing	☐	☒	☐	
14	Display: Call from IP Touch to SIP (Analog Phone connected behind the Sip Gateway) which has the name containing non-ASCII characters, eg &@(#?+)=. Check that SIP set is ringing	☐	☒	☐	
15	SIP set is part of a sequential hunt group **(1)**. Call to hunt group. Check call/release. With IP Touch IPset-1 call the sequential hunt group MCDU number 328 Check that Zset-2 is ringing Take the call and don't hang up. And with IP Touch IPset-2 call the sequential hunt group MCDU number 328 Check that IPset-2 is ringing Take the call and don't hang up. And with SIP Set Zset-1 call the sequential hunt group MCDU number 328 Check that Zset-3 is ringing Take the call and don't hang up.	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
16	SIP set is part of a cyclic hunt group **(2)**. Call to hunt group. Check call/release. With IP Touch IPset-1 call the cyclic hunt group MCDU number IPset-2 Check that 301 is ringing Take the call and hang up. And with IPset-1 call the cyclic hunt group MCDU number IPset-2 Check that Zset-3 is ringing Take the call and hang up. And with SIP Set Zset-1 call the cyclic hunt group MCDU number IPset-2 Check that Zset-2 is ringing Take the call and don't hang up.	☐	☒	☐	
17	SIP set is declared as a MultiSet. Call to main set and see if twin set rings. Take call with twin set. With IP Touch IPset-2 call IP Touch IPset-1 which is in MultiSet with SIP Set Zset-3. Check that Zset-3 and IPset-1 both ringing. Take the call from Zset-3 and check that IPset-1 stop ringing. Check audio and display.	☐	☒	☐	

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

(2) Cyclic Hunt Group behavior: the endpoint n+1 is ringing if previously the endpoint n has been reached (ringing only or in call). The actual state of the n endpoint doesn't matter.

8.1.5 Features during Conversation

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

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

Test Case Id	Test Case	N/A	OK	NOK	Comment
	With Zset-2 call IPset-2, take the call and don't release it too. With Zset-2 start a conference by the local feature Check that IPset-1, IPset-2 and Zset-2 are in conference. Check audio and display.				
4B	**Three party conferences initiated from SIP set with OXO feature** With Zset-2 call IPset-1 takes the call and don't release it. With Zset-2 call IPset-2, take the call and don't release it too. With Zset-2 start a conference by the OXO feature Check that IPset-1, IPset-2 and Zset-2 are in conference. Check audio and display.	☒	☐	☐	Conference feature is not supported within call server for SIP device
5	**Meet Me conference** With Zset-3 call the Meet me Conference bridge dialing prefix 68 and follow instruction to open the bride. With Zset-2 join the conference bridge by dialing prefix 69 and enter access code. With IPset-1 join the conference bridge by dialing prefix 69 and enter access code. Check that IPset-1, Zset-2 and Zset-3 are in conference.	☐	☒	☐	

8.1.6 Call Transfer

During the consultation call step, the transfer service can be requested and must be tested.
Several transfer services exist: Unattended Transfer, Semi-Attended Transfer and Attended Transfer.
Audio, tones and display must be checked.

We use the following scenario, terminology and notation:
There are three actors in a given transfer event:

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

There are three sorts of transfers in the SIP world:

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

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

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

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

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

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

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

8.1.7 Attendant

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

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**SIP set call to attendant** From SIP set Zset-2 dial "9" (attendant call prefix) Check audio and display	☐	☒	☐	
2	**2nd incoming call while in conversation with attendant** While SIP set Zset-2 is in conversation with the attendant, from IP Touch IPset-2 call Zset-2 Check audio and display	☐	☒	☐	
3	**SIP set call to attendant, attendant transfers to OXO set, semi-attended** From SIP set Zset-2 dial "9" (attendant call prefix) and answer. Attendant transfer semi-attended to IP Touch IPset-2 Answer the call and check audio and display	☐	☒	☐	
4	**SIP set call to attendant, attendant transfers to OXO set, attended** From SIP set Zset-2 dial "9" (attendant call prefix) and answer Attendant transfer attended to IP Touch IPset-2 Check audio and display	☐	☒	☐	
5	**OXO set calls to attendant, attendant transfers to SIP set, attended** From IP Touch IPset-2 dial "9" (attendant call prefix) and answer Attendant transfer attended to SIP set Zset-2 Check audio and display	☐	☒	☐	
6	**External ISDN Call to attendant, attendant transfers to SIP set, attended** ISDN incoming call to the attendant. From the attendant call SIP set Zset-2 and transfer attended Check audio and display	☐	☒	☐	
7	**SIP set call to attendant, attendant transfers to External** From SIP set Zset-2, dial "9" (attendant call prefix) and answer From the attendant, call an external ISDN destination and transfer semi-attended Answer and check audio and display.	☐	☒	☐	

8.1.8 Voice Mail

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

Voice mail service is enabled on SIP sets Zset-2, Zset-3 and OXO IPset-1.

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

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	Password modification With SIP set Zset-3 call the Voice Mail and follow the Voice guide in order to modify the default password. When modification is accepted hang-up. Recall the voice mail and try to log with a wrong password. Check the rejection. Recall the voice mail and try to log with the right password. Check the service access.	☐	☒	☐	
2	Message display activation, MWI (1): With SIP set Zset-2 call the Voice Mail. Follow the instructions in order to send a voice message in SIP set Zset-3 boxes. Check that the MWI on Zset-3 is activated.	☐	☒	☐	**4102:** the Mediatrix 4102 supports the FSK signal to enable the visual indicator and does not provide voltage for the indication. So message waiting indication will not work in phones which require voltage for the indication. **4116:** OK
3	Message consultation With SIP set Zset-3 call the Voice Mail. Follow the instructions in order to listen your voice message leaved during the previous test. Check that your can listen it and delete. Check that MWI display is disabled on Zset-3 after message cancellation.	☐	☒	☐	Message can be consulted No message waiting indication
4	SIP call to a OXO user forwarded to Voice Mail Forward the IP Touch IPset-1 to Voice Mail by dialing *61 prefix + <Voice Mail number>. With SIP set Zset-3 call IPset-1 and check that you are immediately forwarded to Voice Mail. Check that you can leave a message On IPset-1 disable Voice Mail forwarding with *60 prefix.	☐	☒	☐	
5	OXO set call to a SIP user forwarded to Voice Mail Forward the SIP set Zset-3 to Voice Mail by dialing *61 prefix + <Voice Mail number>. With IP Touch IPset-1 call Zset-3 and check that you are immediately forwarded to Voice Mail. Check that you can leave a message On Zset-3 disable Voice Mail forwarding with *60 prefix.	☐	☒	☐	

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

8.1.9 Defence

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

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

8.1 Fax tests

In this section fax modules are connected as Basic SIP device on OXO though the analog gateway. These fax modules are limited to G711 pass-through (G711 transparency) sending method. The OmniPCX Office does not support T.38 protocol on SIP subscriber.

All tests performed in this chapter are made exclusively with G711.

8.1.1 Basic Fax Tests

8.1.1.1 Test objectives

These tests shall verify that the basic communication between FAX can be made on different conditions.

8.1.1.2 Authentication between GATEWAY Fax and OmniPCX Office

Description: *Check the behavior of GATEWAY endpoints registration without/with authentication*

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Register with no authentication** On OXO and on the analog gateway, configure GWFAXset-1 and GWFAXset-2 registration with no authentication. Check registration.	☐	☒	☐	
2	**Fax sending with no authentication** Send a fax from an FAXset-1 to GWFAXset-1	☐	☒	☐	
3	**Fax receiving with no authentication** Send a fax from GWFAXset-1 to FAXset-1	☐	☒	☐	
4	**Register with authentication** On OXO and on the analog gateway, configure GWFAXset-1 and GWFAXset-2 registration with digest authentication mode. Check registration.	☐	☒	☐	
5	**Fax receiving with authentication** Send a fax from an FAXset-1 to GWFAXset-1	☐	☒	☐	
6	**Receiving fax with authentication** Send a fax from GWFAXset-1 to FAXset-1	☐	☒	☐	
7	**Register time out** On the analog gateway, configure 120 seconds as registration period. Wait for a registration timeout and check that the gateway registers again	☐	☒	☐	

8.1.1.3 Basic communication between Gateway fax and Exernal fax

Description: *Check the behavior of a basic fax transmission*

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Fax sending to an external fax** Send a fax from an External fax to GWFAXset-1	☐	☒	☐	
2	**Fax receiving to an external fax** Send a fax from GWFAXset-1 to an External fax	☐	☒	☐	

8.1.1.4 Loop-back communication from GATEWAY Fax to GATEWAY Fax through OmniPCX Office.

***Description**: Check the behavior of loop-back fax transmission*

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	Send a fax between two gateway fax devices Send a fax from GWFAXset-1 to GWFAXset-2	☐	☒	☐	
2	Send a fax between two gateway fax devices via PSTN T0 Send a fax from GWFAXset-1 to GWFAXset-2 via PSTN with T0	☐	☒	☐	

8.1.1.5 Multiple pages exchanged between GATEWAY Fax and OmniPCX Office.

***Description**: Check the behavior of multiple page fax transmission*

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Fax receiving with 5 pages** Send a fax (5pages) from FAXset-1 to GWFAXset-1	☐	☒	☐	
2	**Fax sending with 5 pages** Send a fax (5 pages) from GWFAXset-1 to FAXset-1	☐	☒	☐	
3	**Fax receiving with 5 pages from an external fax** Send a fax (5pages) from an external fax device to GWFAXset-1	☐	☒	☐	
4	**Fax sending with 5 pages to an external fax** Send a fax (5pages) from GWFAXset-1 to an external fax device	☐	☒	☐	
5	**Fax sending with 5 pages to an between two gateway fax devices** Send a fax (5 pages) from GWFAXset-1 to GWFAXset-2	☐	☒	☐	
6	**Fax sending with 5 pages to an between two gateway fax devices via PSTN** Send a fax (5 pages) from GWFAXset-1 to GWFAXset-2 via PSTN with T0	☐	☒	☐	

8.1.2 Surveillance/Recovery

8.1.2.1 Test objectives

These tests shall verify that the basic communication between faxes can be made when the network or equipments are stressed.

8.1.2.2 Perturbations

Description*: Check the solution behaviors when network is perturb*

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Fax receiving stop after the first page** Send a fax from FAXset-1 to GWFAXset-1. Stop the transmission after sending the first page. Check the fax receiving is correctly stopped.	☐	☒	☐	
2	**Fax sending stop after the first page** Send a fax from GWFAXset-1 to FAXset-1. Stop the transmission after sending the first page. Check the fax sending is correctly stopped.	☐	☒	☐	
3	**Fax receiving when busy** Send one fax from FAXset-1 to GWFAXset-1 Send one fax from FAXset-2 to GWFAXset-1 Check the FAXset-2 receives a busy tone.	☐	☒	☐	
4	**Fax sending when no answer** Send one fax from GWFAXset-1 to FAXset-1. Verify that the behavior is correct when there is no answer	☐	☒	☐	

8.1.2.3 OmniPCX Office system phones call GATEWAY Fax

Description*: Check the behavior when a phone calls the GATEWAY Fax*

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	**Fax receiving stop after the first page** Make a call from the Ipset-1 to the GWFAXset-1, verify that the call is released after a time out Verify that no issues are generated	☐	☒	☐	

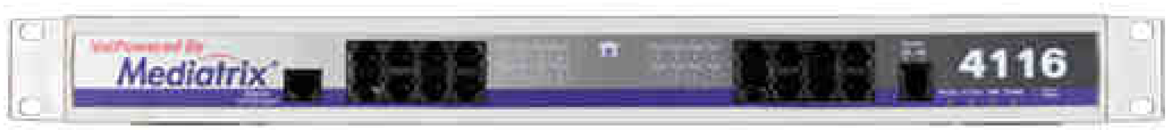
9 Appendix A: AAPP member's Application Description

Mediatrix 41xx

The Mediatrix Series Analog VoIP Gateways are cost-effective, best-of-breed technology products. These stand-alone analog VoIP Gateways provide superior voice technology for connecting legacy telephones, fax machines and PBX systems with IP-based telephony networks, as well as for integration with new IP-based PBX systems. These products are designed and tested to be fully interoperable with leading Soft switches, SIP servers and H.323

Mediatrix Gateway 4102

Mediatrix Gateway 4116



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

Here the configuration of Mediatrix 4102 &4116 is provided in both GUI and INI formats

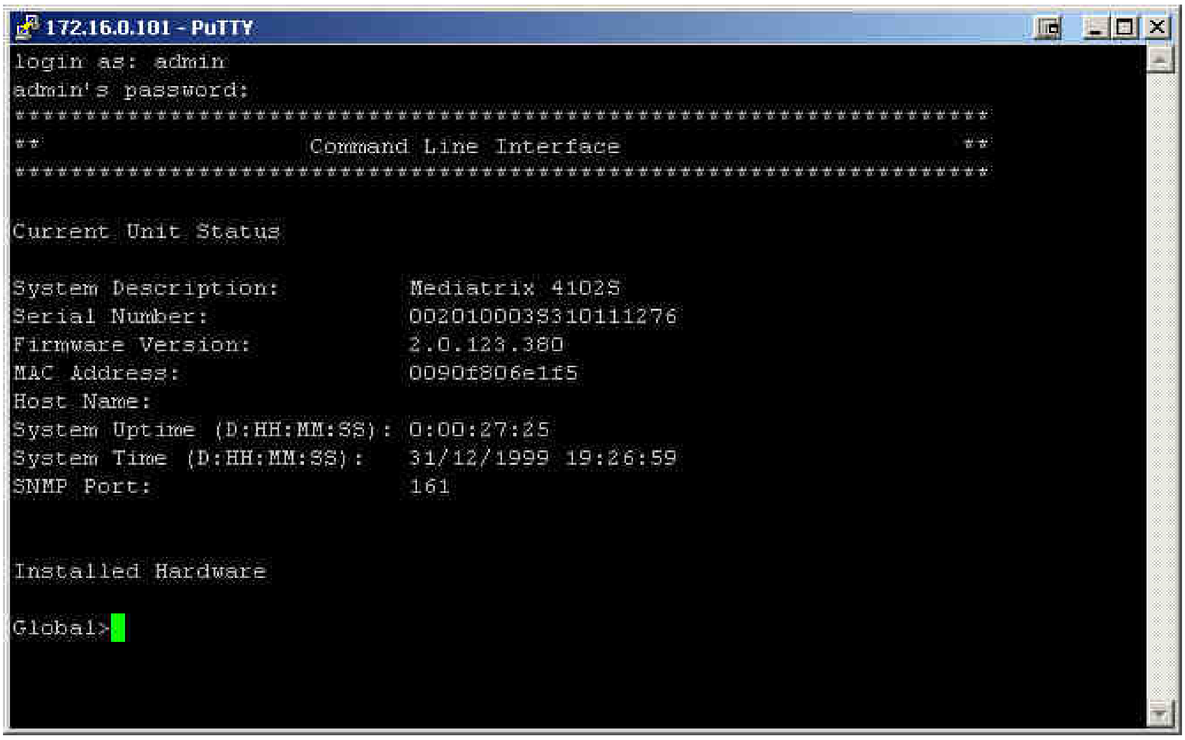
Section 1: Configurations done at Mediatrix 4102 & 4116 GUI based

Notes:

Apart from those parameters that are highlighted/shown here, all other parameters remain at default values.

Both 4102 and 4116 has the same GUI interface apart from the User authentication page as the number of user that can be connected and registered in 4116 is more.

1. Accessing the Mediatrix 4102 CLI Interface (via SSH)



2. Accessing the Mediatrix 4116 CLI Interface (via SSH)

```
172.16.0.102 - PuTTY
login as: admin
admin's password:
*********************************************************************
**                    Command Line Interface                       **
*********************************************************************

Current Unit Status

System Description:          Mediatrix 4116
Serial Number:               002350002S351111185
Firmware Version:            2.0.123.380
MAC Address:                 0090f807b7fb
Host Name:
System Uptime (D:HH:MM:SS):  0:00:25:13
System Time (D:HH:MM:SS):    31/12/1999 19:24:57
SNMP Port:                   161


Installed Hardware

Global>
```

3. Set Variables though the CLI in the Mediatrix 4102

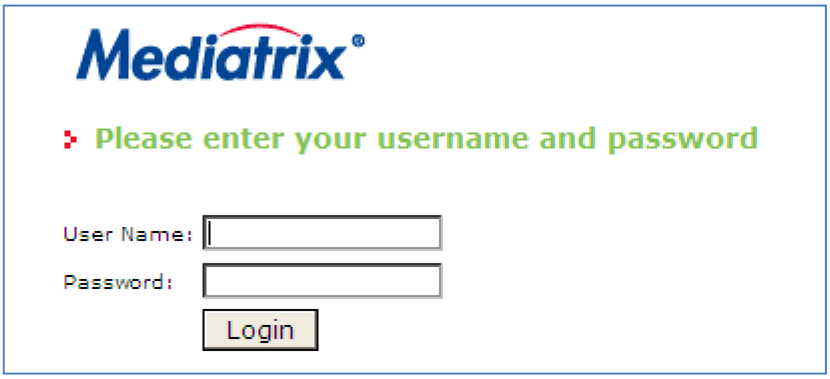
```
Global>
Global>set SipEp.InteropSdpDetectPeerDirectionAttributesupportEnable ="Disable"

Global>set SipEp.InteropOnholdSdpStreamDirection = "Sendonly"

Global>set SipEp.InteropOnholdAnswerSdpStreamDirection = "Inactive"

Global>set SipEp.InteropEscapePoundInSipUriUsername = "Enable"

Global>set SipEp.InteropSipContactDisplayNamePresence = "Disable"
```

4. Set variables through the CLI in the Mediatrix 4116

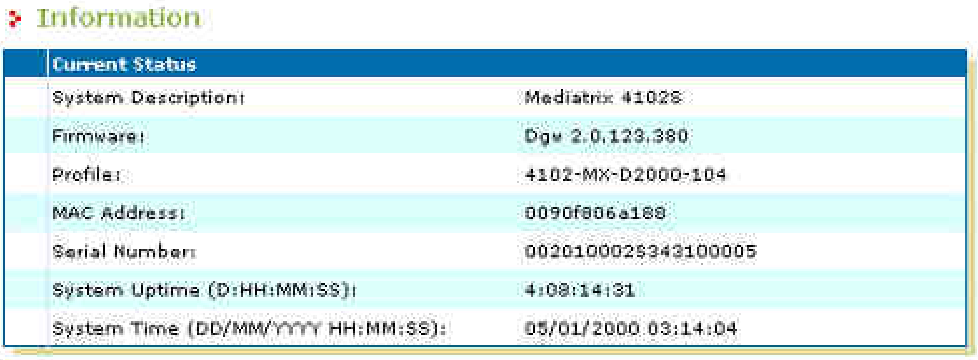
```
Global>set SipEp.InteropSdpDetectPeerDirectionAttributeSupportEnable = "Disable"

Global>set SipEp.InteropOnHoldSdpStreamDirection = "Sendonly"

Global>set SipEp.InteropOnHoldAnswerSdpStreamDirection = "Inactive"

Global>set SipEp.InteropEscapePoundInSipUriUsername = "Enable"

Global>set Pots.FxsDefaultVisualmessageWaitingIndicatorType = "FskAndVoltage"

Global>set SipEp.InteropSipContactDisplayNamePresence = "Disable"
```

The parameter “SIPEp.InteropSipContactDisplayNamePresence” should be disabled in order to get a proper display in the Iptouch phones. If not disabled the extesnion numbers of the mediatrix SIP extension cannot be seen.

5. Accessing web Interface of Mediatrix 4102 & 4116



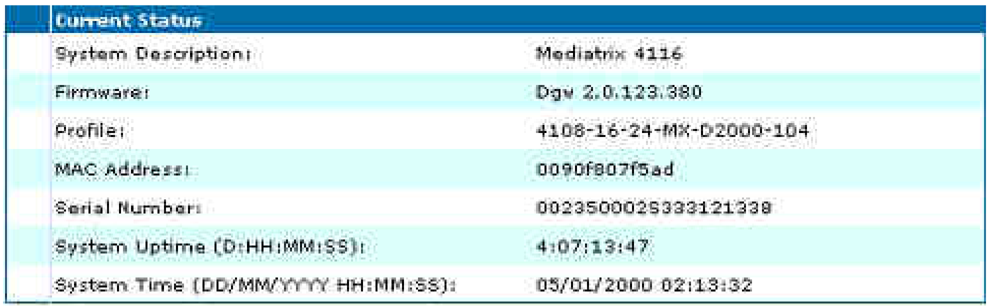
6. Home Page of Mediatrix 4102



Information

Current Status	
System Description:	Mediatrix 4102S
Firmware:	Dgw 2.0.123.380
Profile:	4102-MX-D2000-104
MAC Address:	0090f806a188
Serial Number:	002010002S343100005
System Uptime (D:HH:MM:SS):	4:08:14:31
System Time (DD/MM/YYYY HH:MM:SS):	05/01/2000 03:14:04

Home Page of Mediatrix 4116



Current Status	
System Description:	Mediatrix 4116
Firmware:	Dgw 2.0.123.380
Profile:	4108-16-24-MX-D2000-104
MAC Address:	0090f807f5ad
Serial Number:	002350002S333121338
System Uptime (D:HH:MM:SS):	4:07:13:47
System Time (DD/MM/YYYY HH:MM:SS):	05/01/2000 02:13:32

7. Network page of Mediatrix 4102 & 4116

Status

Interfaces Status					
Name	Link	IP Address	Default Router	VLAN Override	Status
Uplink	lan	10.130.158.57/24	10.130.158.100	Disable	Active

Host Status	
General Configuration:	
IPv4 Automatic Configuration Interface:	Uplink
IPv6 Automatic Configuration Interface:	UplinkV6
Host Name Configuration	
Host Name:	
Domain Name:	(none)
Default Gateway Configuration	
IPv4 Default Gateway:	10.130.158.100
IPv6 Default Gateway:	0.0.0.0
DNS Configuration	
SNTP Configuration	
SNTP Host:	

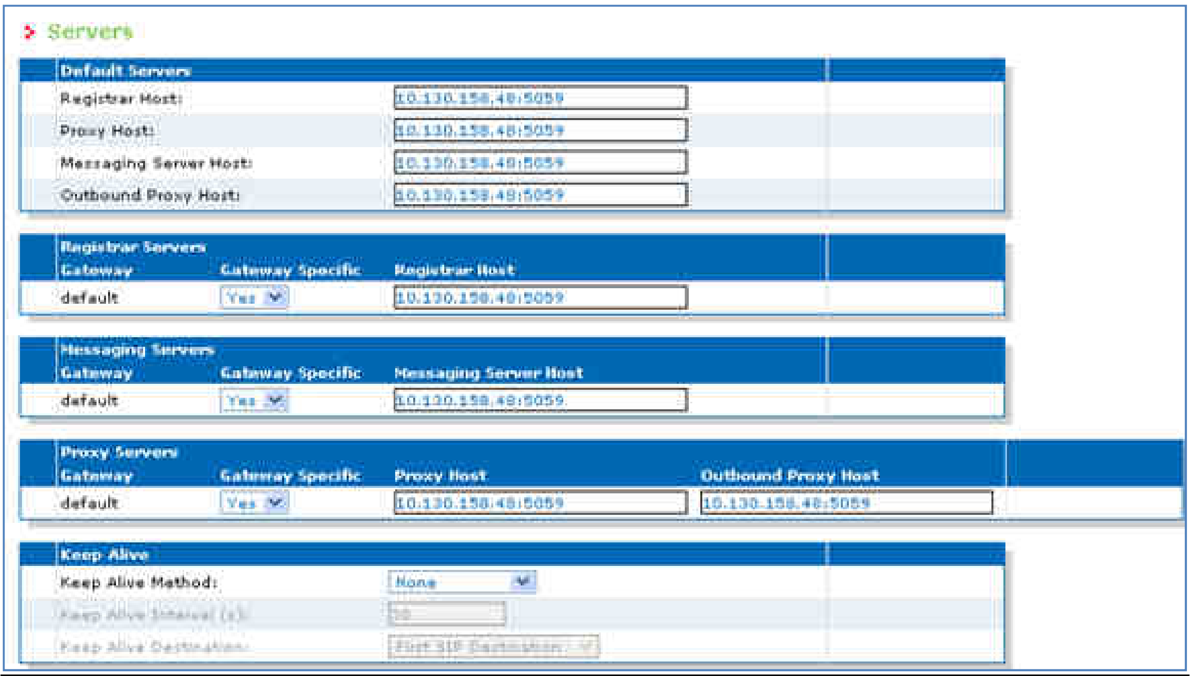
Advanced IP Routes				
Priority	Source Address	Source Link	Forward To Network	Status
No custom route currently applied in the routing table				

IPv4 Routes			
Link	Destination	Gateway	Protocol
lan	10.130.158.0/24	0.0.0.0	Kernel
lan	0.0.0.0/0	10.130.158.100	Static

8. User Services of Mediatrix 4102 & 4116

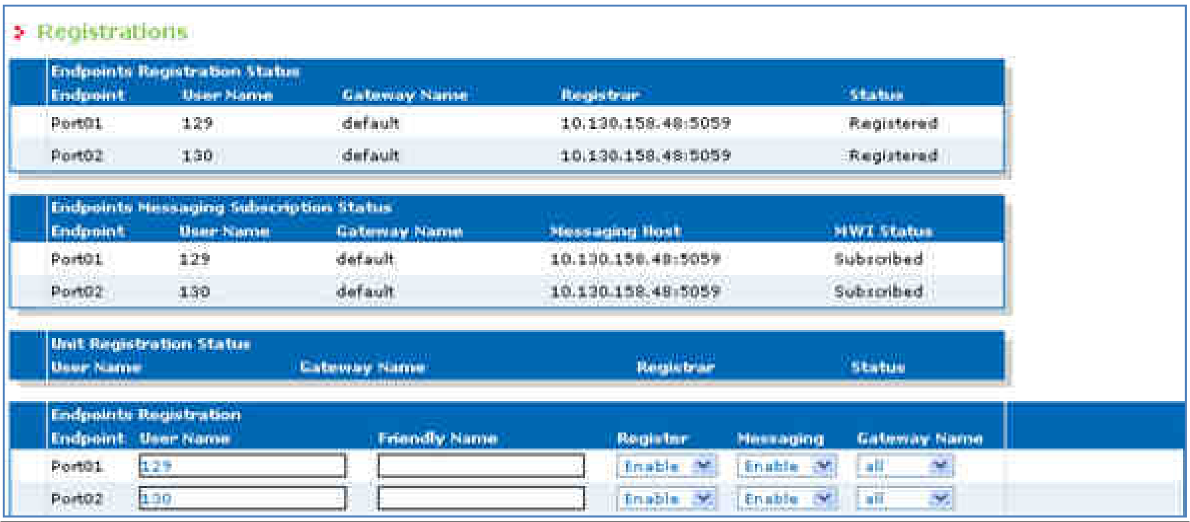
User Service	Status	Startup Type	Action	Comment
Basic Network Interface (BNI):	Started	Auto		
Call Routing (CROUT):	Started	Auto		
Call Detail Record (CDR):	Stopped	Manual		
Command Line Interface (CLI):	Started	Auto		
DHCP Server (DHCP):	Stopped	Manual		
Endpoint Administration (EPADM):	Started	Auto		
Endpoint Services (EPSERV):	Started	Auto		
IP Routing (IPROUTING):	Started	Auto		
Local Firewall (LFW):	Started	Auto		
Link Layer Discovery Protocol (LLDP):	Stopped	Manual		
Media IP Transport (MIPT):	Started	Auto		
Music On Hold (MOH):	Started	Auto		
Network Address Translation (NAT):	Stopped	Manual		
Network Firewall (NFW):	Stopped	Manual		
Notifications and Logging Manager (NLM):	Started	Auto		
Network Traffic Control (NTC):	Stopped	Manual		
Plain Old Telephony System (POTS):	Started	Auto		
SIP Endpoint (SIPEP):	Started	Auto		
Simple Network Management Protocol (SNMP):	Started	Auto		
Telephony Interface (TELIF):	Started	Auto		
Web (WEB):	Started	Auto		

9. SIP Servers Configuration Page of Mediatrix 4102 & 4116



10. SIP Registrations

4102 :



4116 :

Registrations

Endpoints Registration Status

Endpoint	User Name	Gateway Name	Registrar	Status
Port01	129	default	10.130.158.48:5059	Registered
Port02	130	default	10.130.158.48:5059	Registered
Port03	131	default	10.130.158.48:5059	Registered

Endpoints Messaging Subscription Status

Endpoint	User Name	Gateway Name	Messaging Host	MWI Status
Port01	129	default	10.130.158.48:5059	Subscribed
Port02	130	default	10.130.158.48:5059	Subscribed
Port03	131	default	10.130.158.48:5059	Subscribed

Unit Registration Status

User Name	Gateway Name	Registrar	Status

Endpoints Registration

Endpoint	User Name	Friendly Name	Register	Messaging	Gateway Name
Port01	129	129	Enable	Enable	all
Port02	130	130	Enable	Enable	all
Port03	131	131	Enable	Enable	all
Port04			Disable	Disable	all
Port05			Disable	Disable	all

11. Codecs Selection in Mediatrix 4102 & 4116 web page

Codecs

Select Endpoint: Default

Codec	Voice	Data	Advanced
G.711 a-Law	Enable	Enable	Edit
G.711 u-Law	Enable	Enable	Edit
G.726 16Kbps	Enable		Edit
G.726 24Kbps	Enable		Edit
G.726 32Kbps	Enable	Enable	Edit
G.726 40Kbps	Enable	Enable	Edit
G.729	Enable		Edit
T.38		Disable	Edit
Clear Mode	Disable	Disable	Edit
Clear Channel	Disable	Disable	Edit
X CCD	Disable	Disable	Edit

Generic Voice Activity Detection (VAD)

Enable (G.711 and G.726): Conservative

Su

Codecs

Select Endpoint: Default

Jitter Buffer	
Level:	Normal
Voice Call	
Minimum:	30
Maximum:	125
Data Call	
Playout Type:	Fixed
Minimum:	0
Nominal:	67
Maximum:	135

DTMF Transport	
Transport Method:	Out-of-Band using RTP
SIP Transport Method:	Draft Choudhuri SIP Info Digit 00
Payload Type:	101

Machine Detection	
CNG Tone Detection:	Enable
CED Tone Detection:	Enable
V.21 Modulation Detection:	Enable
Behavior On CED Tone Detection:	Passthrough

Base Ports	
RTP:	5004
SRTP:	5004
T.38:	6004

Submit

12. Country Settings in Mediatrix 4102 & 4116 web page

Misc

Country	
Country Selection:	UK1

Submit

13. DTMF Maps in Mediatrix 4102 & 4116 web page

DTMF Maps

DTMF Map: Allowed

General Configuration				
First DTMF Timeout	Inter DTMF Timeout	Completion Timeout	Digit Detection (FXS/FXO)	
20000	3000	60000	When Pressed	Edit Endpoints

Allowed DTMF Map

Index	Enable	Apply To	Endpoints	Suggestions	DTMF Map	Transformation	Target	Emergency
1	Enable	Unit		--- Suggestion ---	x.#	x{#}		Disable
2	Enable	Unit		--- Suggestion ---	*xx	x		Disable
3	Enable	Unit		--- Suggestion ---	x.T	x		Disable
4	Enable	Unit		--- Suggestion ---	x.T	x		Disable
5	Enable	Unit		--- Suggestion ---	x.T	x		Disable
6	Enable	Unit		--- Suggestion ---	x.T	x		Disable
7	Enable	Unit		--- Suggestion ---	x.T	x		Disable
8	Enable	Unit		--- Suggestion ---	x.T	x		Disable
9	Enable	Unit		--- Suggestion ---	x.T	x		Disable
10	Enable	Unit		--- Suggestion ---	x.T	x		Disable

Submit

14. Call Forward options in Mediatrix 4102 & 4116

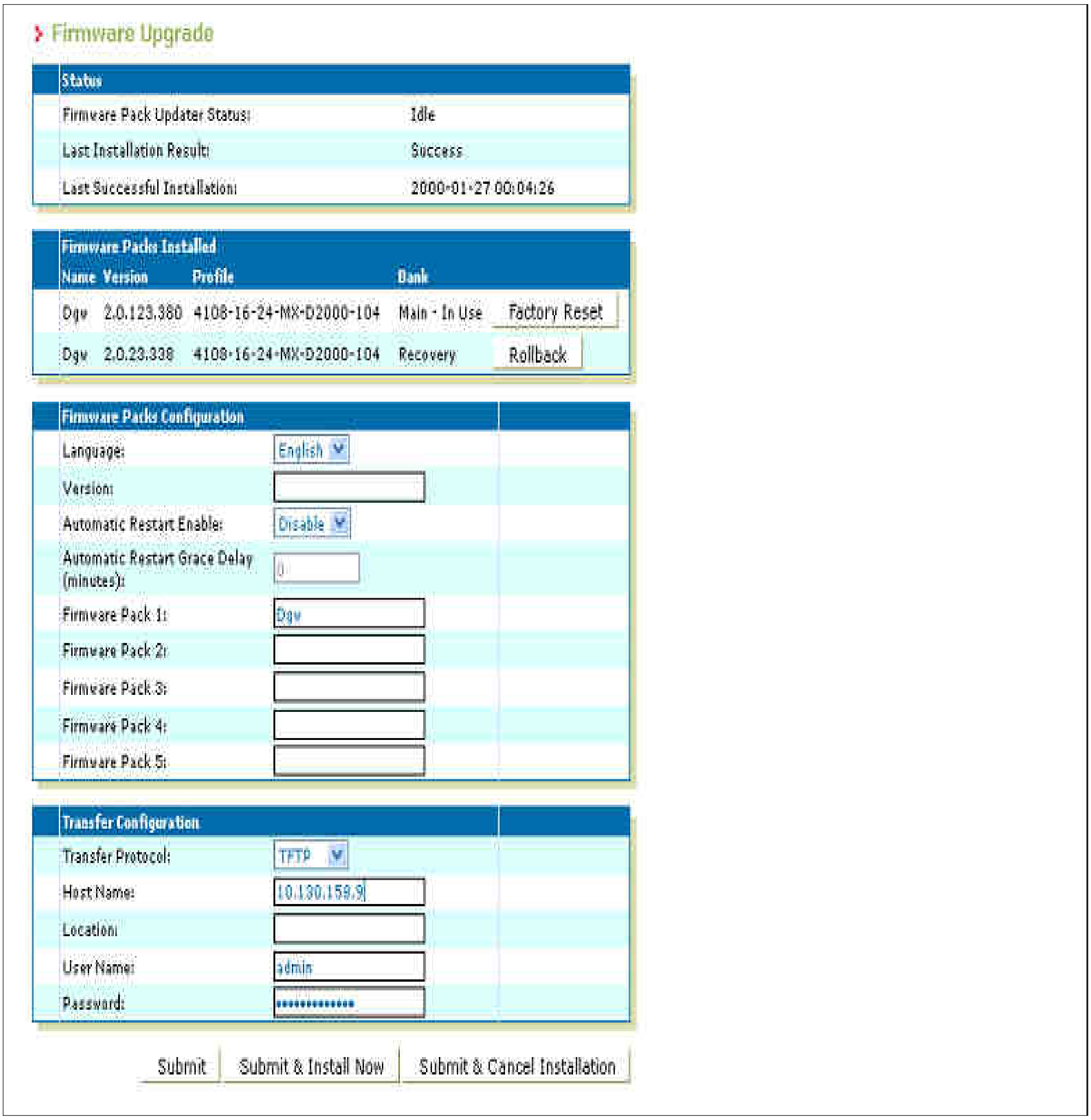
:

Call Transfer	
Blind Transfer Activation:	Enable
Attended Transfer Activation:	Enable
Call Waiting	
Call Waiting Activation:	Enable
Cancel DTMF Map:	
Conference	
Conference Activation:	Enable
Delayed Hotline	
Delayed Hotline Activation:	Disable
Delayed Hotline Condition:	FirstDtmfTimeout
Delayed Hotline Target:	
Direct IP Address Call	
Direct IP Address Call Activation:	Disable
Hold	
Hold Activation:	Enable
Second Call	
Second Call Activation:	Enable

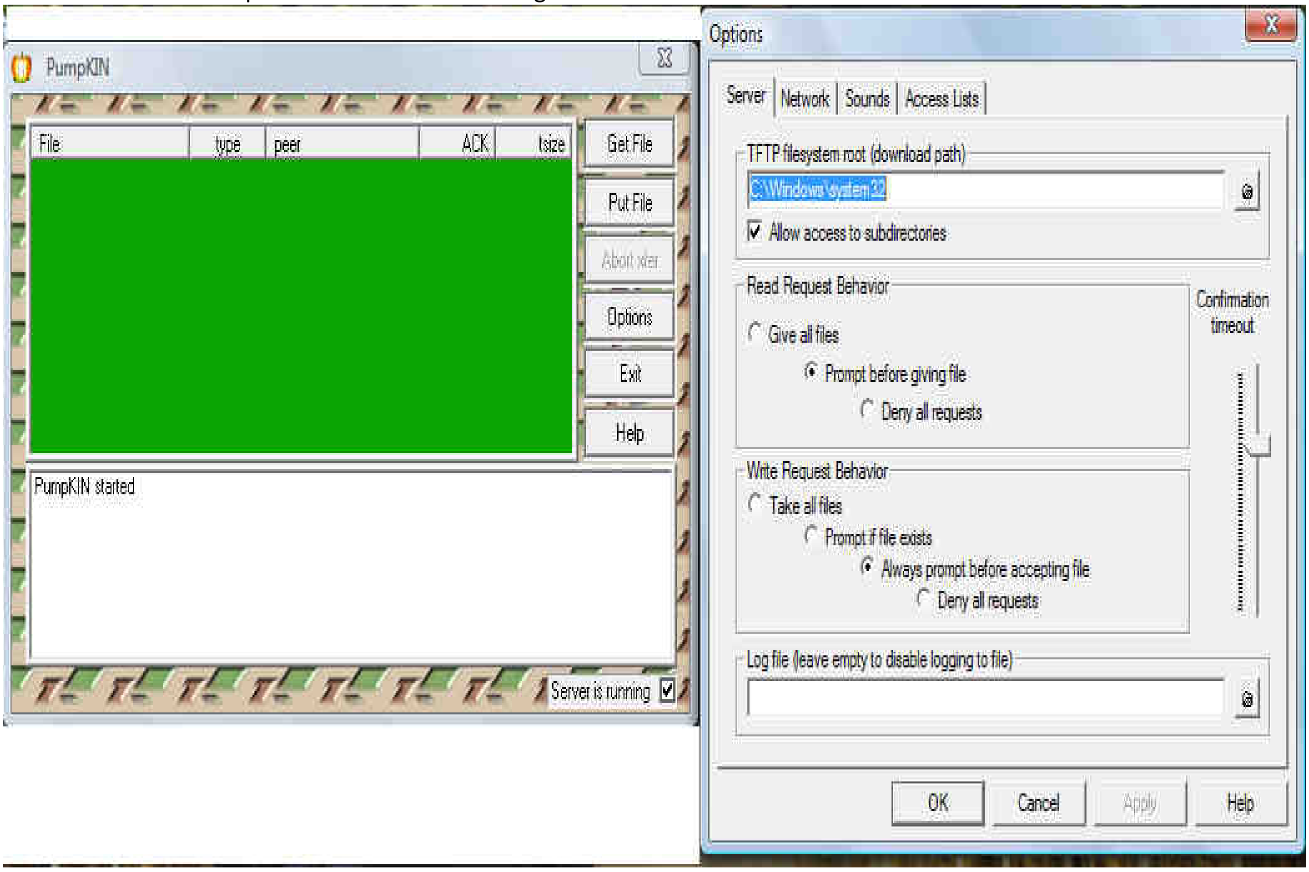
Firmware Upgrade

Firmware upgrade procedure

Management >> Firmware Upgrade



- Select the transfer protocol as TFTP as in the above picture
- Provide the IP address of the TFTP server in which the Firmware file is loaded.
- We used Pumpkin TFTP server for loading the Firmware file.



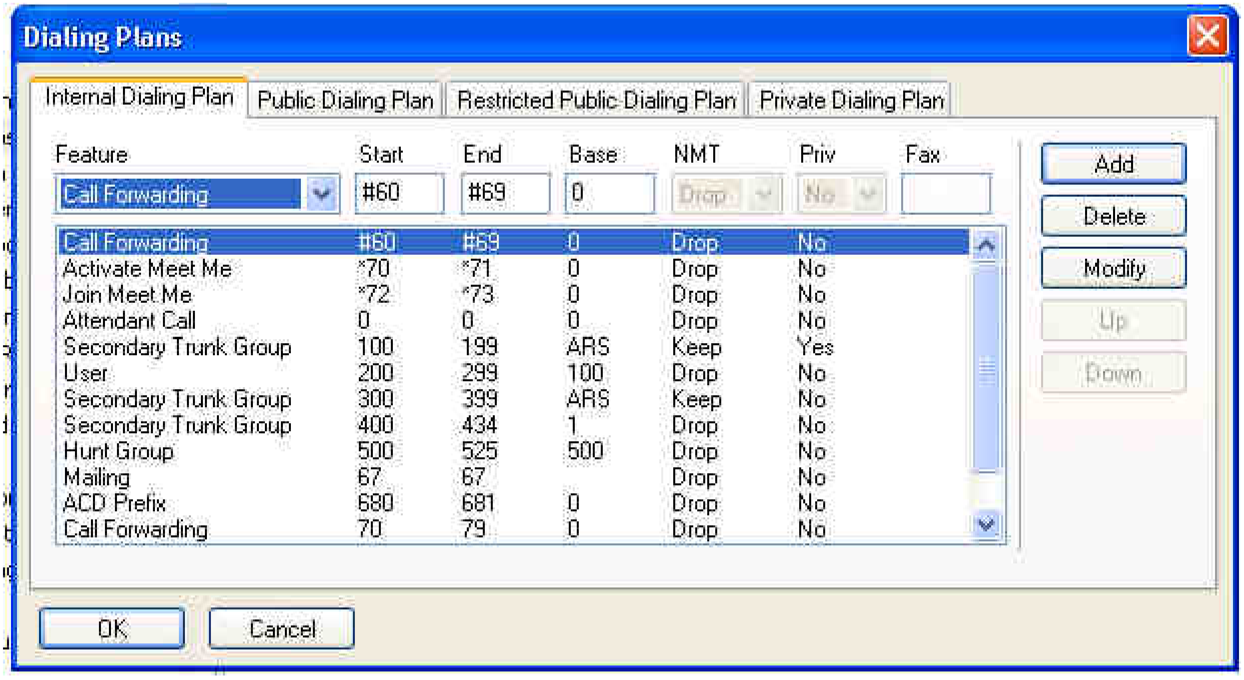
We need to copy the Firmware file to the TFTP server download path and start the server.
After that we need to click on submit & install in the GUI page of 4102 or 4116.

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

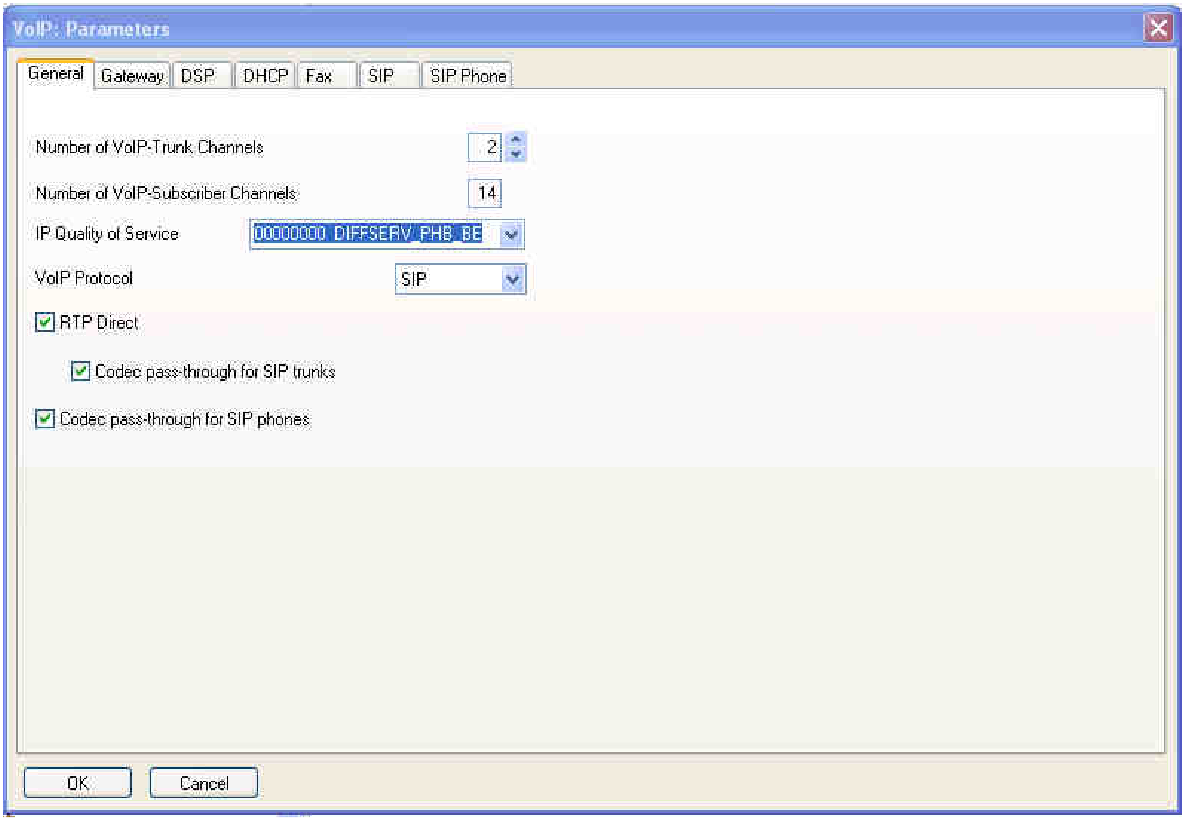
OXO Specific Configuration

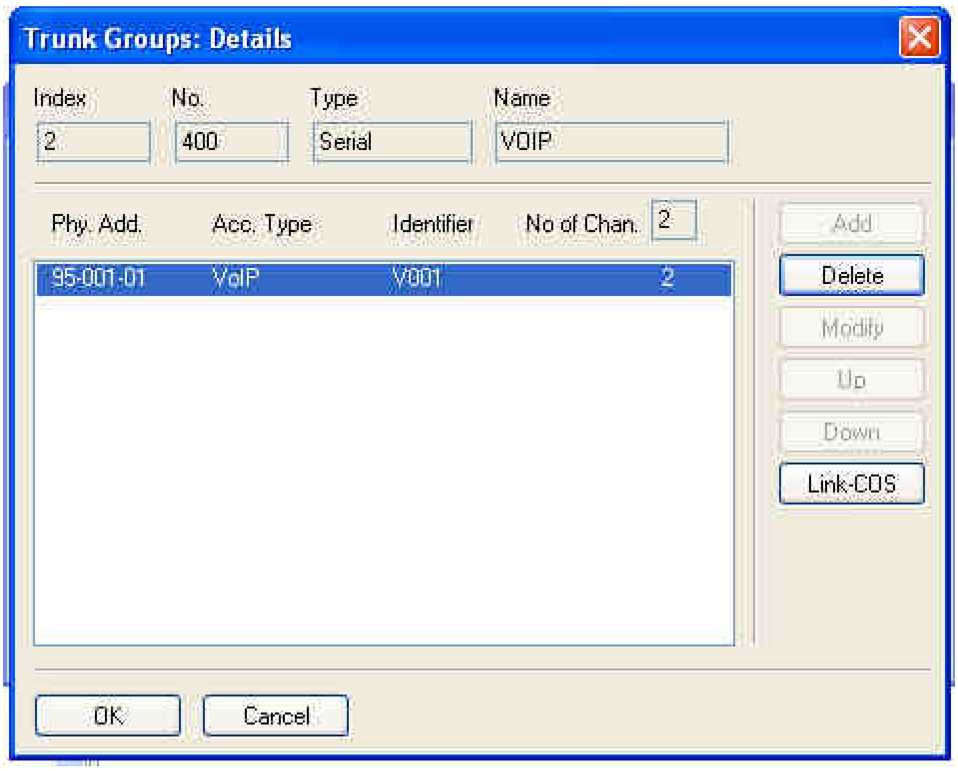
These configurations are not specific to the gateway device. The settings given below are common OXO configuration (Trunk configuration is required for network calls)

Dialing Plan

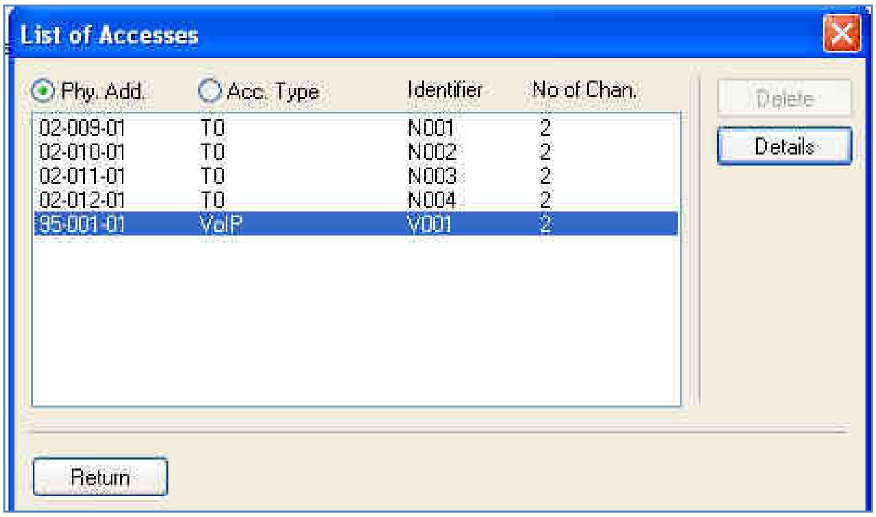


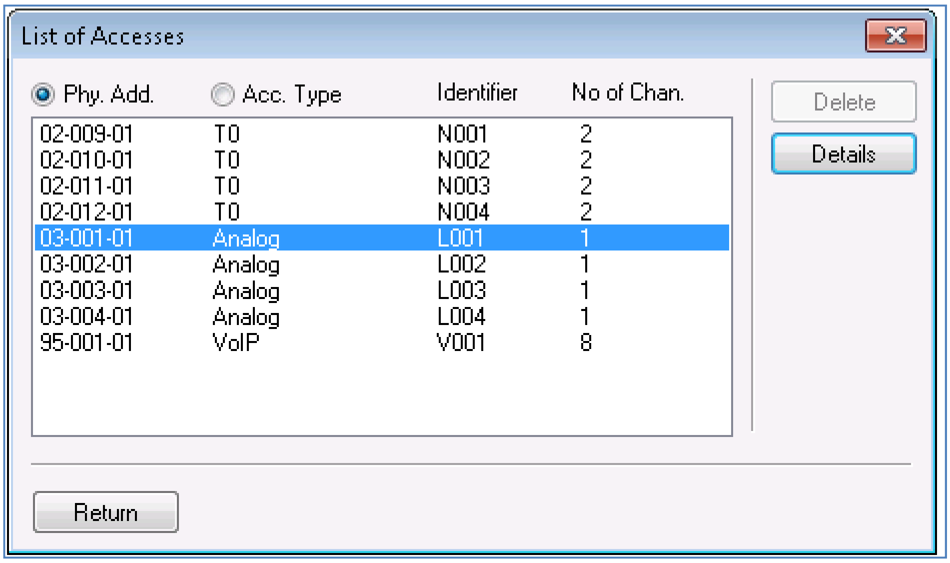
Trunk Configuration:





Trunk Access:





Network Call Configuration:

Automatic Routing: Prefixes

Activation	Network	Prefix	Ranges	Substitute	TrGpList	Called(ISVPN/H450)	Use...	Destination	IP T...	IP Address	Hostname
Yes	priv	1	00-99	1	1	hom	OXO2	SIP Gateway	Static	10.130.158.88	

Fax calls

For Fax calls we need to configure the sip user as Basic SIP phone.

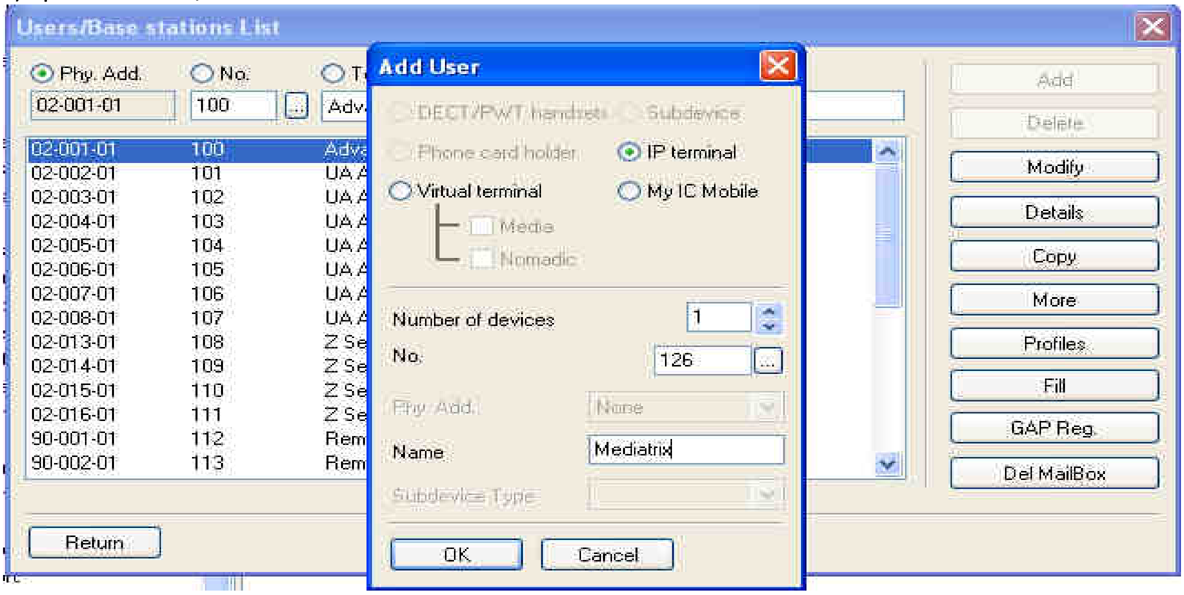
Normal calls

For normal calls we need to configure the sip users as Open SIP phone.

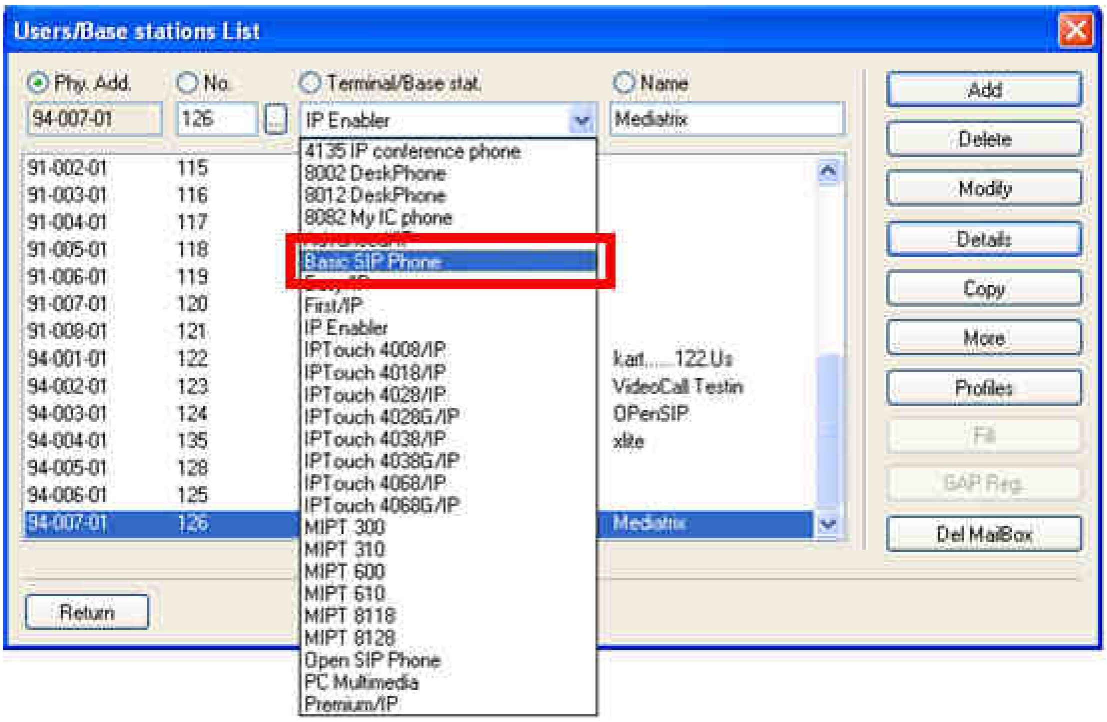
SIP Gateway Specific Configuration

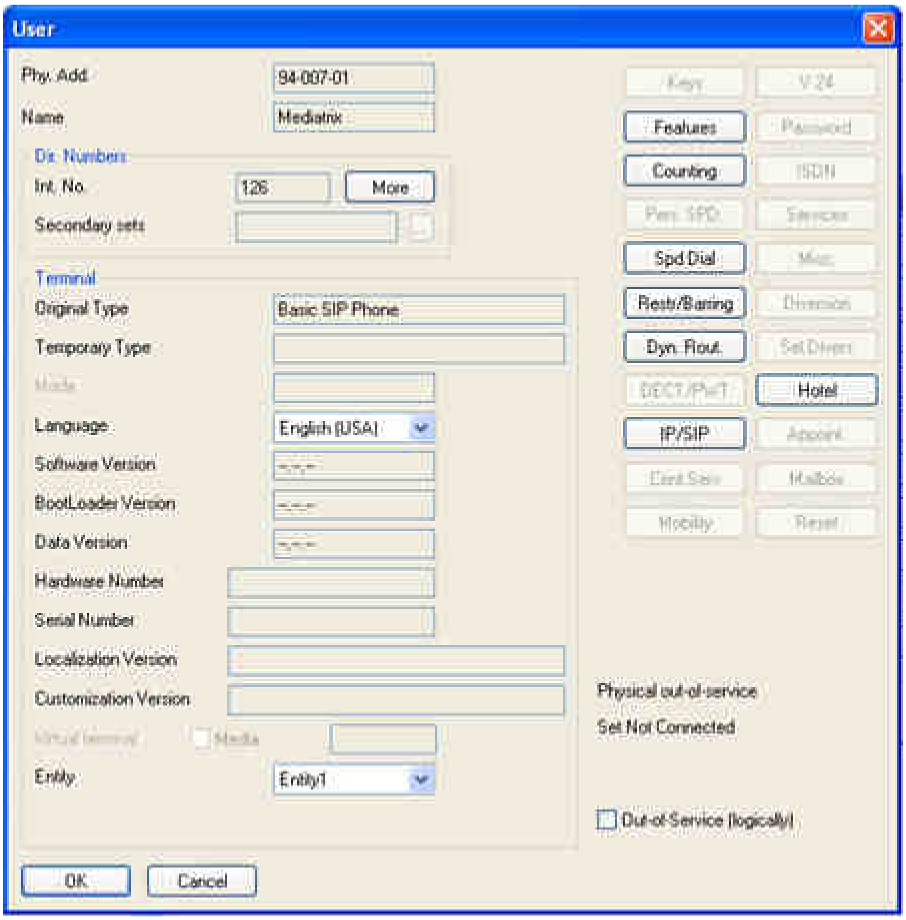
Below configuration is required for registering the analog devices behind the gateway with the OXO.

1)Open the User/Base stations List in the OMC. And click on Add

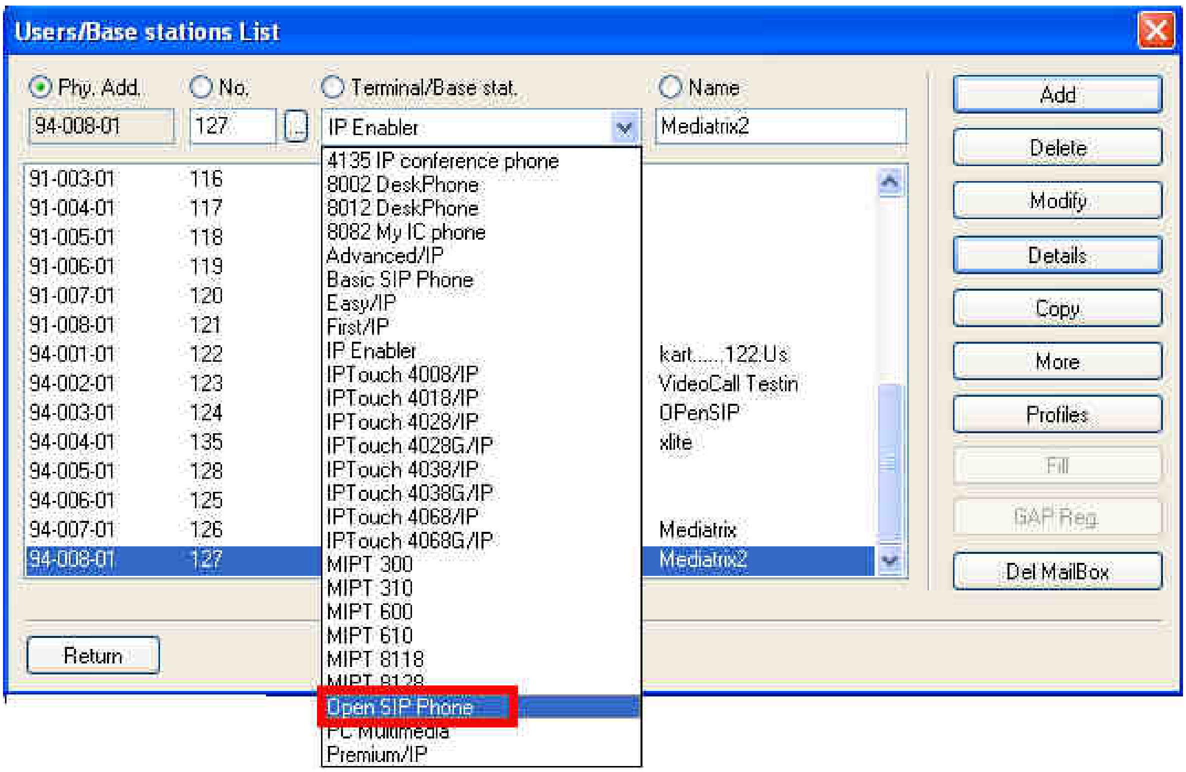


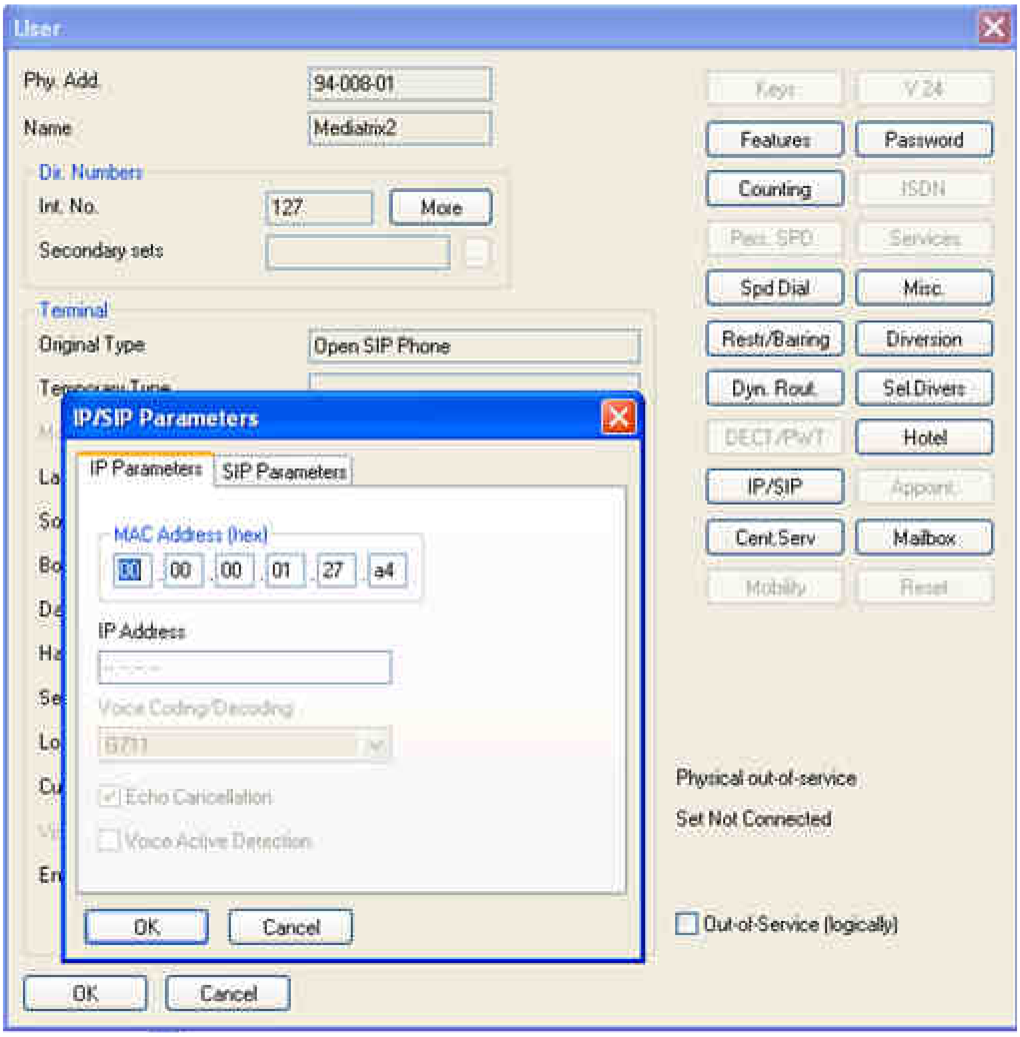
2) After creation of the user modify the (extension number) IP enabler to Basic sip phone from for the fax calls.





3) For other tests configure the user (extension number) to Open SIP phone as below.




User
Phy. Add.
94-008-01
Name
Mediatrix2
Dir. Numbers
Int. No.
127
More
Secondary sets
Terminal
Original Type
Open SIP Phone
IP/SIP Parameters
IP Parameters
SIP Parameters
MAC Address (hex)
00
00
00
01
27
a4
IP Address
Voice Coding/Decoding
G711
Echo Cancellation
Voice Active Detection
OK
Cancel
Keys
V 24
Features
Password
Counting
ISDN
Pers. SPD
Services
Spd Dial
Misc.
Restr./Barring
Diversion
Dyn. Rout.
Sel.Divers
DECT/PWT
Hotel
IP/SIP
Appoint.
Cent.Serv
Mailbox
Mobility
Reset
Physical out-of-service
Set Not Connected
Out-of-Service (logically)
OK
Cancel

12 Appendix D: AAPP member's Escalation Process

12.1.1 Technical Support Process

The following steps must be followed prior to escalating any incident to Mediatrix

- Mediatrix Business Partner and/or customers shall subscribe to a Service Level Agreement (SLA) with Mediatrix in order to benefit from escalation privilege.
- Business Partner will provide 1st and 2nd Level support to their customer base. When a problem is reported by an end user, Business Partner will take steps to isolate the source of the problem.
- Once Business Partner has determined that the problem is related to a Mediatrix unit, they will report the incident through Mediatrix Support Portal (Web interface to Mediatrix ticket management system).
- Mediatrix will ensure that all required information is obtained from Business Partner and will proceed with the analysis.
- Mediatrix will work with Business Partner directly to resolve the issue.
- Escalation process shall be used to promote incident priority if resolution time does not meet agreed or expected Service Level Agreement.
- When escalating by phone incident number must be mentioned.
- In order to trigger the escalation by email the subject field must begin with the word "**Escalation (Incident P14-"followed** by Mediatrix incident number.

12.1.2 Escalation Table

Step	Action	Prime	Contact
1	Report Incident in Mediatrix Support Portal	Primary Support Engineer	www.mediatrix.com/support_MSP.php
2	Initial Escalation Request	Primary Support Engineer	Tel: +1-819 829-8749 ext: 5 tac@mediatrix.com
3	R&D Escalation	Technical Support Leader	Sylvie Dubois (sdubois@mediatrix.com)
4	Management Level Escalation	Director Support & Services	Stephane Blais (sblais@mediatrix.com)
5	Business & Product Level Escalation	Product Line Manager	MLA_PLM_Mtx@mediatrix.com

13 Appendix E: AAPP program

13.1 Alcatel-Lucent Application Partner Program (AAPP)

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

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

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

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

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

Web site

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

13.2 Alcatel-Lucent.com

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

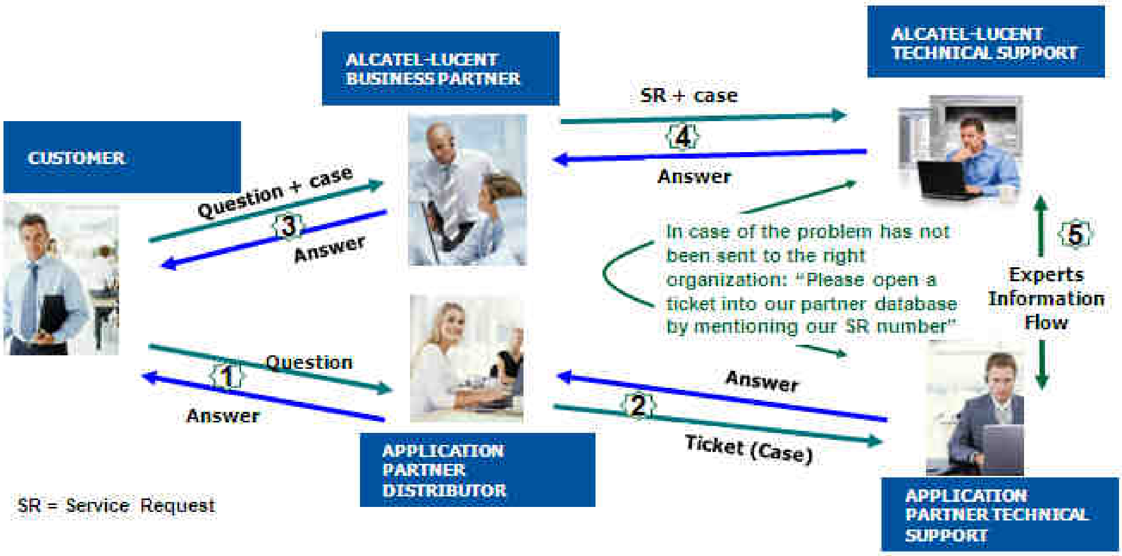
14 Appendix F: AAPP Escalation process

14.1 Introduction

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

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

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



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

14.2 Escalation in case of a valid Inter-Working Report

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

This defines the scope of what has been certified.

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

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

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

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

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

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

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

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

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

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

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

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

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

14.3 Escalation in all other cases

These cases can cover following situations:

1. An InterWorking Report exist but is not valid (see Chap **Erreur ! Source du renvoi introuvable.** "Validity of an Interworking Report")

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

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

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

14.4 Technical support access

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

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

Alcatel-Lucent Business Partners Support Center for countries:

<table>
<tr><th>Country</th><th>Supported language</th><th>Toll free number</th></tr>
<tr><td>France</td><td rowspan="3">French</td><td rowspan="23">+800-00200100</td></tr>
<tr><td>Belgium</td></tr>
<tr><td>Luxembourg</td></tr>
<tr><td>Germany</td><td rowspan="3">German</td></tr>
<tr><td>Austria</td></tr>
<tr><td>Switzerland</td></tr>
<tr><td>United Kingdom</td><td rowspan="16">English</td></tr>
<tr><td>Italy</td></tr>
<tr><td>Australia</td></tr>
<tr><td>Denmark</td></tr>
<tr><td>Ireland</td></tr>
<tr><td>Netherlands</td></tr>
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<tr><td>Sweden</td></tr>
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<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