

ALE Application Partner Program Inter-Working Report

Partner: Audiocodes Application type: Analog Media Gateway Application name: MediaPack MP118 Alcatel-Lucent Enterprise Platform: OXO Connect™ OXO Connect Evolution™



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 issues a new major release of such Alcatel-Lucent Enterprise product (incorporating new features or functionalities), whichever first occurs.

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Certification overview

AAPP member representative

Date of the certification	August 2018			
ALE representative	Rachid Himmi			

Alcatel-Lucent Enterprise	OXO Connect			
Communication Platform				
	OXO Connect (PowerCPU EE)			
Alcatel-Lucent Enternrise	P3 0/045 001			
Communication Platform release	OXO Connect Evolution			
	R3.0/045.001			
AAPP member application release	6 60A 347 002			
	0.007 (0 11.002			
Application Cotogony	Gateway			
Application Category	· · · · · · · · · · · · · · · · · · ·			

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Revision History

Edition 1: creation of the document – August 2018 (OXO Connect -Power CPU) Edition 2: Additional tests with OXO connect evolution (IP BOX) – August 2018 (OXO Connect Evolution)

Test results

Passed

Refused

Postponed

Eran Battat

Passed with restrictions

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

IWR validity extension

Only the MP118 hardware has been tested. Behavior should be the same with all the MP11x family



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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 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 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 (<u>https://businessportal.alcatel-lucent.com</u>) in the Application Partner Interworking Reports corner (restricted to Business Partners)
- the Application Partner portal (<u>https://www.al-enterprise.com/en/partners/aapp</u>) 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 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 analyzed by the AAPP member before being escalated to ALE. 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 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 will consider that situation as to that where no IWR exists. ALE will handle this situation accordingly (for more details, please refer to Appendix F "AAPP Escalation Process").



4 Application information

Application commercial name:	Telephone Adapter / VoIP Gateway for Analog equipment
Application version:	6.60A.347.002
Interface type:	SIP/Ethernet

Brief application description:

AudioCodes' Media Pack 1xx series of analog VoIP gateways offer service providers and enterprises superior voice technology for connecting legacy telephones, fax machines and PBX systems with IP telephony networks and IP-PBX systems.

The Media Pack 1xx gateways are fully interoperable with leading soft switches and SIP servers.



- Leverage investment in legacy analog telephone, modem, and fax systems easing VoIP migration
- Secured zero-touch provisioning, useful for large-scale deployments
- Standalone Survivability (SAS) keeps your business running in the event of a network failure



5 Test environment

Figure 1 Test environment



5.1 Hardware configuration

List main hardware equipments used for testing

- OXO Connect:
 - OXO Connect Medium / PowerCPU EE (for OXO Connect)



> OXO IP box for OXO connect Evolution software.

Connect Connec

- Release: R3.0/045.001
- > OMC : 30.0_17.1a



Setup Details:

OXO Connect Power CPU EE

Setup Information OXO 1				
OXO 1 IP address	10.9.224.220			
Voicemail No	114 -121			
Attendant No	9			
OXO Extension Details used for test				
IP Touch extension numbers	IPset-1 : 130 IPset-2 : 129 IPset-3 : 131			
SIP extension numbers These are analog extension numbers connected behind MP118 Fax extension number connected behind MP118	SIP set-1 :120 SIP Set-2:121 GWFAXset-1 :122			

OXO Connect Evolution

Setup Information OXO 2				
OXO 1 IP address	10.9.223.140			
Voicemail No	114 -121			
Attendant No	9			
OXO Extension Details used for test				
IP Touch extension numbers	IPset-1 : 118 IPset-2 : 119			
SIP extension numbers These are analog extension numbers connected behind MP118 Fax extension number connected behind MP118	SIP set-1 :120 SIP Set-2:121 GWFAXset-1 :122			

5.2 Software configuration

List main softwares used for testing

- Alcatel-Lucent Communication Platform: OXO Connect Power CPU EE softwareR3.0/045.001 OXO Connect Evolution softwareR3.0/045.001
- Partner Application: Audio Codes Media pack 112 6.60A.347.002

Note: Analog phones are registered in the OXO Connect as "<u>Open SIP phone</u>". Fax phones are registered in the OXO Connect as "<u>Basic SIP phone</u>".





6 Summary of test results

6.1 Summary of main functions supported

6.1.1 OXO Connect R3.0 (power CPU EE)

Feature	Results	Remarks
Initialization including network	OK	
configuration		
SIP registration	OK	
SIP authentication	OK	
Voice over IP and RTP codec support	OK	
Outgoing Call	OK	
Incoming Call	OK	
Features During Conversation	OK	

6.1.2 OXO Connect Evolution R3.0

Feature	Results	Remarks
Initialization including network	OK	
configuration		
SIP registration	OK	
SIP authentication	OK	
Voice over IP and RTP codec support	OK	
Outgoing Call	OK	We used SIP trunk calls for external calling related scenarios
Incoming Call	OK	
Features During Conversation	OK	

6.2 Summary of problems

• None

6.3 Summary of limitations

None

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6.4 Notes, remarks

- Analog phones are registered in the OXO as "Open SIP phone".
- Fax extensions are suggested to be configured as "Basic SIP phone"
- We have tested fax only between two gateway devices that is behind the Audio Codes. We did not test scenarios involving external fax to gateway fax device. We need to confirm the validity of the tests involving external fax
- Only the MP118 hardware has been tested. Behavior should be the same with all the MP11x family
- When using OXO prefix to set a forward, Do not disturb, wake-up, there is no indication on the remote office analog sets. Only tones are heard when programming There is also no indication when an OXO forward is set,
- Message waiting indication has to be enabled separately in the configuration.
- Network calls are tested between OXO Connect power CPU EE and OXO Connect R3.0 Evolution.
- Extensions connected behind the AudioCodes are referred as SIP phone Z-set
- OXO connect evolution supports only 80X8 series phones. We tested with 8028 and 8068 phones during our testing.

When mentioning OXO in this document means system used for the tests (OXO Connect and/or OXO Connect Evolution)

7 Test Result Template

Test NOK Case **Test Case** N/A OK Comment ld Test case 1 Action 1 \square \square Expected result Test case 2 The application waits Action 2 \boxtimes \square for PBX timer or Expected result phone set hangs up Test case 3 Relevant only if the Action 3 \boxtimes CTI interface is a Expected result direct CSTA link Test case 4 Action No indication, no error \square \square 4 Expected result message •

The results are presented as indicated in the example below:

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</u> <u>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"</u> the reason for the failure and the reference number of the issue either on ALE side or on AAPP member side

Comment: to be filled in with any relevant comment. Mandatory in case a test has failed especially the reference number of the issue.



8 Test Results

8.1 Analog phones tests

8.1.1 Test Objectives

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)

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).

8.1.2 Test Results

Test Case Id	Test Case	N/A	ок	NOK	Comment
AN1	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.				
AN2	 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. 		X		
AN3	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				
AN4	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.				

8.2 Audio codec negotiations/ VAD / Framing

8.2.1 Test Objectives

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

Phone configuration: configure the analog gateway to use 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).

8.2.2 Test Results

Test Case Id	Test Case	N/A	ок	NOK	Comment
AU1	Select G711 A-law as 1 st 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				
AU2	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				
AU3	Select G723 as 1 st 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 supported in OXO Connect evolution 3.0

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AU4	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 IP Zset-2 NOT to use VAD Configure IP Touch IPset-2 to use VAD Redo the same tests			
AU5	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			
AU6	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			
AU7	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			
AU8	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			

8.3 Outgoing calls

8.3.1 Test Objectives

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	ок	NOK	Comment
OU1	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.				
OU2	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				
OU3	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).				
OU4	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				
OU5	Call to busy user (SIP: "182 Queued") 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				
OU6	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"				Instead of out of service message, the call is automatically forwarded with 181 forwarded from OXO

8.3.2 Test Results

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Case **Test Case** N/A OK NOK Comment ld Check the display and ring back tone 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 OU7 \square 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 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 **OU8** \bowtie 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. 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 OU9 \square (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. 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. OU10 With SIP phone Zset-2 call IPset-1 and take the call \square \square 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. Call to external number (Check ring back tone, called party display) **OXO Connect** We used ISDN T0 line **OU11** With SIP set Zset-2 dial 9 (9 prefix +external **OXO Connect evolution** \boxtimes \square We used SIP trunk for number) Take the call and check audio, display and call testing this scenario release. 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. OU12 \boxtimes 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.

8.4 Incoming calls

8.4.1 Test Objectives

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.

8.4.2 Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
IN1	Local /network call to free SIP terminalLocal: with IP Touch IPset-1 call SIP set Zset-2. Checkthat Zset-2 is ringing and take the callCheck ring back tone and called party display.Network: with IP Touch IPset-1 call SIP set NwkZset-2 onanother Node. Check that NwkZset-2 is ringing and takethe call.Check ring back tone and called party display.				
IN2	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. <u>Network</u> : 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.				
IN3	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 <u>Network</u> : Unplug the SIP set NwkZset-2 and call it with IPset-1 Check the ring back tone and display				

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Test **Test Case** N/A ΟΚ NOK Comment Case Id 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** 603 Declined IN4A Check the ring back tone and display \square message is sent 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. By system feature Local: 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. Call goes to IN4B \bowtie voicemail Network: 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. Local/network/SIP call to SIP terminal in immediate forward (CFU) to local user: By local feature if applicable: Local: 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 rinaina. Take the call and check audio and display. IN5A \bowtie Disable CFU on Zset-2. Network: 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.

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Test **Test Case** N/A ΟΚ NOK Comment Case Id By system feature: Local: 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. IN5B Disable CFU on Zset-2 using *60 prefix. \square Network: 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. 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. IN6A \square 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. 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. IN6B \boxtimes 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 rinaina. Take the call and check audio and display. Disable CFU on Zset-2 using *60 prefix.

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Test **Test Case** N/A ΟΚ NOK Comment Case Id 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-With Zset-3 call Zset-2. Check that NwkZset-1 is ringing. Take the call and check audio and display. IN7A \square 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. 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. IN7B Disable CFU on Zset-3 using *60 prefix. \square 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 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 IN8A With Zset-2 call the voice mail to make it busy. \square \bowtie 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. 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. IN8B With Zset-2 call the voice mail to make it busy. \square \bowtie 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.

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Test Case Id	Test Case	N/A	ок	NOK	Comment
IN9A	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		×		
IN9B	By system feature CNFR via prefix not available on OXO (dynamic routing has to be used)				
IN10	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
IN11	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.				
IN12	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.		X		
IN13	Display: Call to free SIP terminal from IP Touch user with a name containing non-ASCII characters (eg éëêèè). Check caller display. Check that SIP set is ringing and check on its display that the characters are correctly printed.				
IN14	Display: Call from IP Touch to SIP which has the name containing non-ASCII characters, eg &@(#?+)=. Check caller display. Check that SIP set is ringing and check that the characters are correctly printed.				

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Test **Test Case** N/A ΟΚ NOK Comment Case Id 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. IN15 And with IP Touch IPset-2 call the sequential hunt group \square 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. 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. IN16 And with IPset-1 call the cyclic hunt group MCDU number \boxtimes 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. 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. IN17 \bowtie 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.

8.5 Features during Conversation

8.5.1 Test Objectives

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.

8.5.2 Test Results

Test Case Id	Test Case	N/A	ОК	NOK	Comment
FE1A	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.				CPT file was generated using the Audiocodes CPTwizard and has to be loaded to the gateway to get the tones.
FE1B	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.				
FE1C	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.				
FE1D	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.				
FE2	Repeat the test 1C to 1D but using the call server feature		\boxtimes		
FE3	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.				

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Test **Test Case** N/A OK NOK Comment Case Id With IPset-1 start a conference. Check that IPset-1, IPset-2 and Zset-2 are in conference. Check audio and display. 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. With Zset-2 call IPset-2, take the call and don't release FE4A \square 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. Three party conferences initiated from SIP set with FE4B \bowtie OXO feature 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. FE5 \square 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.



8.6 Call Transfer

8.6.1 Test Objectives

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 OXO Connect.

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.

8.6.2 Test Results

Test case ID		Test case		N/A	ОК	NOK	Comment
	А	В	С				
	Transferee	Transferor	Transfer Target				
CT1	OXO	SIP	ОХО		\boxtimes		

		Alc Ente	catel·Lucent	Ð		
CT2	Ext Call	SIP	охо			OXO Connect We used ISDN T0 line OXO Connect evolution We used SIP trunk for testing this scenario
СТЗ	Ext Call	SIP	Ext Call			OXO Connect We used ISDN TO line OXO Connect evolution We used SIP trunk for testing this scenario
CT4	SIP	SIP	SIP		\square	
CT5	SIP	OXO	OXO		\square	
CT6	Ext Call	охо	SIP			OXO Connect We used ISDN T0 line OXO Connect evolution We used SIP trunk for testing this scenario
CT7	SIP	OXO	SIP		\boxtimes	

8.7 Attendant

8.7.1 Test Objectives

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

8.7.2 Test Results

Test Case Id	Test Case	N/A	ок	NOK	Comment
AT1	SIP set call to attendant From SIP set Zset-2 dial "9" (attendant call prefix) Check audio and display				
AT2	2 nd 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 Answer the call and check audio and display		X		Beep sound is heard when there is second incoming call.
AT3	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				
AT4	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				
AT5	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				
AT6	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				
AT7	SIP set call to attendant, attendant transfers to External From SIP set Zset-2, dial "9" (attendant call prefix) and answer				

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Test Case Id	Test Case	N/A	ок	NOK	Comment
	From the attendant, call an external ISDN destination and transfer semi-attended Answer and check audio and display.				



8.8 Voice Mail

8.8.1 Test Objectives

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.

8.8.2 Test Results

Test Case Id	Test Case	N/A	ок	NOK	Comment
	Password modification With SIP set Zset-3 call the Voice Mail and follow the Voice guide in order to modify the default password.				
VO1	When modification is accepted hang-up.				Password modification is successful
	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.				
VO2	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.	/I on Zset-3 is activated.			
VO3	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.				
	message cancellation.				
VO4	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.</voice>				Call forward can be enabled from the codes configured in the MP11x device
VO5	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</voice>				Call forward can be enabled from the codes configured in the MP11x device
	On Zset-3 disable Voice Mail forwarding with *60 prefix.				

8.9 Defence

8.9.1 Test Objectives

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

8.9.2 Test Results

Test Case Id	Test Case	N/A	ок	NOK	Comment
DE1	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.				OXO Connect We tested this scenario with OXO connect R3.0 Medium cabinet and mix board. OXO Connect evolution For OXO connect evolution we used SIP trunk for this scenario.
DE2	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.				OXO Connect This scenario is applicable only for OXO connect (Power CPU EE).
DE3	Power failure In OXO connect Evolution the 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.				OXO Connect evolution We used SIP trunk instead of ISDN for OXO connect evolution. OXO connect evolution is powered by PoE. So once the cable is disconnected, the evolution box is switched off. Once it is connected it boots up and calls are successful.

8.10 Fax tests

8.10.1 Test Objectives

In this section fax modules are connected as Basic SIP phone on OXO though the analog gateway. These fax modules are limited to G711 pass-through sending method.

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

8.10.2 Test Results

Basic Fax Tests

Test Case Id	Test Case	N/A	ОК	NOK	Comment
FA1	Register with no authentication On OXO and on the analog gateway, configure GWFAXset-1 and GWFAXset-2 registration with no authentication. Check registration.				Registration fax sets behind MP118
FA2	Register with authentication On OXO and on the analog gateway, configure GWFAXset-1 and GWFAXset-2 registration with digest authentication mode. Check registration.				Registration fax sets behind MP118
FA3	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				Registration fax sets behind MP118

Loop-back communication from GATEWAY Fax to GATEWAY Fax through OXO.

Test Case Id	Test Case	N/A	ОК	NOK	Comment
FA4	Fax sending to an between two gateway fax devices Send a fax from GWFAXset-1 to GWFAXset-2				We sent single fax between two MP118 phones.
FA5	Fax sending with no authentication Send a fax from an FAXset-1 to GWFAXset-1	\boxtimes			The scenario is not valid for OXO connect Evolution
FA6	Fax receiving with no authentication Send a fax from GWFAXset-1 to FAXset-1	\boxtimes			The scenario is not valid for OXO connect Evolution
FA7	Register with authentication On OXO and on the analog gateway, configure GWFAXset-1 and GWFAXset-2 registration with digest authentication mode. Check registration.				The scenario is not valid for OXO connect Evolution
FA8	Fax receiving with authentication Send a fax from an FAXset-1 to GWFAXset-1				The scenario is not valid for OXO connect Evolution

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FA9	Receiving fax with authentication Send a fax from GWFAXset-1 to FAXset-1		The scenario is not valid for OXO connect Evolution

Basic communication between Gateway fax and External fax

Test Case Id	Test Case	N/A	ок	NOK	Comment
FA10	Fax sending to an external fax Send a fax from an External fax to GWFAXset-1	\boxtimes			The scenario is not valid for OXO connect Evolution
FA11	Fax receiving to an external fax -0 Send a fax from GWFAXset-1 to an External fax				

Multiple pages exchanged between GATEWAY Fax and OXO.

Test Case Id	Test Case	N/A	ок	NOK	Comment
FA12	Fax sending with 5 pages to an between two gateway fax devices The test is to confirm proper working while sending multi page fax between two Send a fax (5 pages) from GWFAXset-1 to GWFAXset-2				
FA13	Fax receiving with 5 pages Send a fax (5pages) from FAXset-1 to GWFAXset-1	\boxtimes			The scenario is not valid for OXO connect Evolution
FA14	Fax sending with 5 pages Send a fax (5 pages) from GWFAXset-1 to FAXset-1				The scenario is not valid for OXO connect Evolution
FA15	Fax receiving with 5 pages from an external fax Send a fax (5pages) from an external fax device to GWFAXset-1	\boxtimes			The scenario is not valid for OXO connect Evolution
FA16	Fax sending with 5 pages to an external fax Send a fax (5pages) from GWFAXset-1 to an external fax device	\boxtimes			The scenario is not valid for OXO connect Evolution



8.11 Surveillance/Recovery

8.11.1 Test objectives

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

8.11.2 Perturbations

Description: Check the solution behaviors when network is perturb

Test Case Id	Test Case	N/A	ок	NOK	Comment
SU1	Fax receiving stop after the first page Send a fax from GWFAXset-1 to GWFAXset-2. Stop the transmission after sending the first page. Check the fax receiving is correctly stopped.				
SU2	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.	\boxtimes			The scenario is not valid for OXO connect Evolution
SU3	Fax receiving when busy Send one fax from GWFAXset-1 to GWFAXset-2 Send one fax from GWFAXset-3 to GWFAXset-2 Check the GWFAXset-3 receives a busy tone.				
SU4	Fax sending when no answer Send one fax from GWFAXset-1 to GWFAXset-2. Verify that the behavior is correct when there is no answer				

OXO system phones call GATEWAY Fax

Test Case Id	Test Case	N/A	ок	NOK	Comment
SU5	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

MediaPack 1xx

The Media Pack 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



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

To start with the configuration, we intially started with factory reset of the device.

1) Configure the IP interface table of the audiocodes.

Caudiocodes MP-118 FXS_FX0	Submit 🔘 Burn	Device Actions 💽 💼 Home 🔞 Help 😁 Log off	
Contiguration Maintenance Status Scenarios Search Besic Full System Vote Vote Vote DNS Bescurity Control Network Bescurity Control Network Control Network	 Single IP Settings Subnet Mask Default Gateway Address VoIP DNS Settings DNS Setondary Server IP DNS Secondary Server IP Multiple Interface Settings Multiple Interface Table 	2 10.9.223 3 255.255.0 10.9.223 180 10.9.224 225 10.9.224 225 10.9.224 225	

2) Check the IP routing table and add appropriate routes for your testing network

- a Dragnostics		- 10° N.U	liting lane					
enarios Search	#	Delete Row	Destination IP Address	Prefix Length	Gateway IP Address	Metric	Interface Name	Status
Basic 🖲 Full 🔿	1		10.9.223.0	24	10.9.223.3	0		Active
System	2		10.9.224.0	24	10.9.223.180	1		Active
VoIP	3		127.0.0.0	8	127.0.0.1	1		Active
Network	4 Enter th	e netwo	127.0.0.1 rk you want	32 Delete	127.0.0.1 Selected Entries	0		Active
QoS Settings		Add	a new table entry					
Media Services		Destir	nation IP Address Prefi 16	x Length	Gateway IP Address	1	Metric Inter	rface Name

 The registration can work even with default proxy settings. But for ease of understading We used IP group and a separate Proxy Set. We have given both configuration with default proxy and with separate IP group, proxy set.

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Configuration with IP group and separate proxy ID

Caudiocodes MP-118	FXS_FXO	Submit 🧕 Burn	Device Actions	Home	() Help
Configuration Maintenance Status & Diagnostics	IP Group Table				
Scenarios Search	Add +				
Basic • Full	Index Description				Local Host Na
B02.1x Settings		Common Gateway			
IPSec Proposal Table		Index	1		
IPSec Association Table		Description	IP GROUP	2 Enter th	e Proxy set ID
Media During Continue	l l	Proxy Set ID	1		
Eax/Modem/CID Settings		SIP Group Name		-	
RTP/RTCP Settings		Contact User]	-
IPMedia Settings		Local Host Name			_
General Media Settings		Media Realm Name		3 Ente	r the IP profile
Analog Settings		IP Profile ID	1		_
Media Security			1		
Media Quality of experience			🖻 S	ubmit × Cancel	
Bervices					- Ai
teast Cost Routing					
Applications Enabling					
Control Network					
IP Group Table					
Proxy Sets Table					

Click the gateway tab and select the SIP re-routing method. Select yes in the "always Use Route table"

Add your OXO IP address and port number in the proxy table 1. Select the same options as shown in the picture below. We are not using default proxy table 0 here as we specified proxy table 1 in the IP group.



Goto SIP definitions and select the Proxy & registration menu. Enter the Registrar IP address. We have added to screenshots of this menu as it is many options.



4) Remaining SIP parameters are

Constant System	ReRegister On Connection Failure Gateway Name	Disable v	
The Security	Gateway Registration Name		
±@Media	DNS Query Type	A-Record V	
* Services	Proxy DNS Query Type	A-Record T	4
Applications Enabling	Subscription Mode	Per Endpoint 🔻	
Control Network	Number of RTX Before Hot-Swap	3	
IP Group Table	Use Gateway Name for OPTIONS	No 🔻	
Proxy Sets Table	User Name		
SIP Definitions	Password	Default_Passwd	
Advanced Parameters	Cnonce	Default_Cnonce	5
Account Table	Registration Mode	Per Endpoint 🔻	
Proxy & Registration	Set Out-Of-Service On Registration Failure	Disable 🔻	_
RADIUS Accounting Settings	Challenge Caching Mode	None 🔻	
E Coders and Profiles	Mutual Authentication Mode	Optional 🔻	
± GW and IP to IP	Use Proxy IP as Host	Disable 🔻	
		Register Un-Register	
		Submit	

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5) Select the supported codecs in OXO.

Coudiocodes MP-118 FXS_F	exo 🖉 su	omit 🧕 E	Burn	Device Actio	ons 🔹 🧯	Home	, 🙆 Help
Configuration Maintenance Status & Diagnostics	Coders Table		2				
Search	Coder Name	ľ	Packetization	Time	Rate		Pavload Type
🛛 Basic 🔍 Full	G.711A-law	•	20	•	64	•	8
€@System	0.74111.000	_	20	-	64	-	0
C VoIP	G.7110-law	<u> </u>	20	<u> </u>	64	-	U
*@Network	G.729	•	20	•	8	•	18
	G.723.1	•	30	•	5.3	•	4
±// Media	G.726	T	20	•	32	•	2
The services		•		•		-	
Control Network				<u> </u>		<u> </u>	
IR Group Table		•		•		•	
Proxy Sets Table		•		•		•	
Caracteristics		•		•		•	
General Parameters		•		•		-	
Advanced Parameters		•				·	
Account Table							
Proxy & Registration							
RADIUS Accounting titings							
Coders and Proties							
Coders Group Settings							
Tel Profile Settings							

6) Select the IP profile declared in the IP group and check the approriate values are selected. For basic calling functionalities we use the default options.

asic 🖲 Full	Profile ID	1	v
System	Profile Name		
VoIP			
Network	Common Parameters		
Security	RTP IP DiffServ	46	
Services	Signaling DiffServ	40	
Applications Enabling	Disconnect on Broken Connection	Yes	T
Control Network	Dynamic Jitter Buffer Minimum Delay [msec](*)	10	
IP Group Table	Dynamic Jitter Buffer Optimization Factor(*)	10	
Proxy Sets Table	RTP Redundancy Depth(*)	0	•
SIP Definitions	Echo Canceler(*)	Enable	T
Advanced Parameters	Input Gain (-32 to 31 dB)(*)	0	
Account Table	Voice Volume (-32 to 31 dB)(*)	0	
Proxy & Registration	Symmetric MKI Negotiation	Disable	•
RADIUS Accounting Settings	MKI Size	0	
Coders and Profiles	Reset SRTP State Upon Re-key	Disable	T
Coders Group Settings 1 Tel Profile Settings IP Profile Settings GW and IP to IP	✓ Gateway Parameters		

asic 🖲 Full 🔿	- Cataway Davamatava		
Sustem	Gateway Parameters	C 711 Transport	3
VoIP	Pax Signaling Method	Disc	
Network	Play Ringback Tone to IP	Play	
Security	Early Media	Disable	
Media	Copy Destination Number to Redirect Number	Disable	•
Services	Media Security Behavior	Preferable	T
Applications Enabling	CNG Detector Mode	Disable	T
Control Network	Modems Transport Type	Enable Bypass	_
IP Group Table	NSE Mode	Disable	v
Proxy Sets Table	Number of Calls Limit	-1	_
SIP Definitions	Progress Indicator to IP	Not Configured	· / 4
General Parameters	Profile Preference	1	
Advanced Parameters	Coder Group	Default Coder Group	•
Account Table	Remote RTP Base UDP Port	0	
DADUS Accounting Settings	First Tx DTMF Option	RFC 2833	•
Coders and Profiles	Second Tx DTMF Option		•
Coders	Declare RFC 2833 in SDP	Yes	•
Coders Group Settings	Call Hold Service	Enable	¥
Tel Profile Settings		· · · · · · · · · · · · · · · · · · ·	

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7) Under hunt group settings endter the end point numbers. The end point numbers are declared in OXO as Open SIP numbers. Declare the channels and enter the telephone profile IP and hunt group ID.

Caudiocodes MP-118 FXS_	FXO	Submit (🗿 Bu	m Device Actions	•	💼 Home (🕘 Help	•	Log off
Configuration Maintenance Status & Diagnostics	Endpoint	Phone Number Table				2		
Scenarios Search		Channel(s)		Phone Number		Hunt Group ID	_	Tel Profile ID
Basic • Full	1	1		162		1		1
. ⊕@ Security ▲	2	2		163		1		1
■ @ Media	3	3	-	164	-	1		1
Services	-	0						
Control Network	4		_		_			
IP Group Table	5							
Proxy Sets Table	6							
Carl SIP Definitions	7							
General Parameters	8							
Advanced Parameters			_					
Proxy & Registration								
RADIUS Accounting Settings								
Coders and Profiles								
GW and IP to IP								
Hunt Group								
Hupt Group Settings								
* Manipulations								
*@Routing								
■ DTMF and Supplementary								
analog Gateway				Register	Cutur	Un-Register		
Advanced Applications					Subm	III		

DTMF and Dialling

Please enter maximum digits to be sent via the gateway. The default value is very less and make sure to change this maximum number of digits to a value that matches the Prefix + length of number to be dialled.

Caudiocodes MP-118 FXS	E_FX0 Submit i Burn	Device Actions 🔹 💼 Home 🔞 Help
Configuration Maintenance Status & Diagnostics Scenarios Search	DTMF & Dialing	2
Basic Full Basic Full System Network Security Media Control Network Control Network Coders and Profiles Coders Coders Group Settings	Max Digits In Phone Num Inter Digit Timeout [sec] Declare RFC 2833 in SDP 1st Tx DTMF Option 2nd Tx DTMF Option RFC 2833 Payload Type Hook-Flash Option Digit Mapping Rules Dial Plan Index Dial Tone Duration [sec] Hotline Dial Tone Duration [sec]	25 4 Yes ▼ RFC 2833 ▼ 96 Not Supported ▼ -1 16 16
☐Tel Profile Settings ☐IP Profile Settings □@@GW and IP to IP	Enable Special Digits Default Destination Number Special Digit Representation	Disable ▼ 1000 Special ▼
Hunt Group Manipulations Acount of the second supplementary DTMF & Dialing Supplementary Services Analog Gateway Advanced Applications Charging		



8) Hunt group settings select the registration mode.

Caudiocodes MP-118 FXS_FX	0		Submit 🧿 Burn (Device Actions 🔻	Home Home	() Help
Configuration Maintenance Status & Diagnostics	unt Gro	up Settings				
Scenarios Search						
Basic 🖲 Full	-	Index			1-12 🔻	
⊕ Becurity ▲ ⊕ Media					2	
	_	Hunt Group ID	Channel Select Mode	Registration Mode	Serving I	Gatew
Control Network	1	1	By Dest Phone Number	Per Endpoint	1 🔻	
IP Group Table	2		T	T		
Proxy Sets Table	3		T	T	T	
Ceneral Parameters	4		· · · ·			
Advanced Parameters	-		· · ·			
Account Table	5		· · · ·			
Proxy & Registration	6		•	•	•	
RADIUS Accounting Settings	7		▼	•	T	
Coders and Profiles	8		T	T	T	
G GW and IP to IP				· · · ·		
Endpoint Phone Number	2		· ·			
Hunt Group Settings	10		T	•	•	
* Manipulations	11		▼	•	T	
Routing Analog Gateway Advanced Applications Charging			Register	Jn-Register	· · · · ·	

9) Under the analog gateway please enter password for OXO declared open SIP extensions.

Configuration Maintenance Status & Diagnostics	Authentication		
Scenarios Search			
Paris 9 Full	Gateway Port	User Name	Password
	Port1 FXS	162	****
Dest Number IP->Tel	Port 2 FXS	163	****
Dest Number Tel->IP	Port 3 FXS	164	*****
Calling Name IP->Tel	Port 4 FXS		1
Source Number IP->Tel	Port 5 FXO		
Source Number Tel->IP	Port 6 FXO		
Phone Context	Port 7 FXO		3
Routing 1	Port 8 FXO		
DTMF and Supprementary Analog Gateway			
Keypad Features			
Metering Tones 2			
Authentication			
Automatic Dialing			
Caller Display Information			
Call Forward			

Password can be found in the IP/SIP options authentication shows below.

			nt 🕖
_	Subscriber		
:5	Phy. Add.	94-005-01	Keys
	Name		Features
	Dir. Numbers		Meterina
ık	Int. No.	Li 162 More	
	Secondary sets	IP/SIP Parameters	
9	Associated set	IP Parameters SIP Parameters	
36	Terminal	SIP password 3	
36 36	Original Type	37537661 Reset	ļ
36	Temporary Type	4	
36 36	Mode	SIP authentication	
36	Language		
эн Эң	Software Version		
€ 20	BootLoader Version		
34	Data Version		
34 34	Hardware Number		
	Serial Number		
.u	Localization Version		
	Customization Version	OK Cancel	
	Virtual terminal	media	

10) Keypad features need to be configured to enable forward for the analog phones behind MP11x

		_
Configuration Maintenance Status & Diagnostics	Keypad Features	2
Sceparios Search		
	- Forward	
OBasic 🖲 Full	Forward Unconditional	51
Coders Group Settings	Forward No Answer	52
Tel Profile Settings	Forward On Busy	53
IP Profile Settings	Forward On Busy or No Answer	54
GW and IP to IP	Do Not Disturb	55
Endpoint Rhope Number	Forward Deactivate	56
Hunt Group Settings	4	
Manipulations	 Caller ID Restriction 	
	Restricted Caller ID Activate	
Composition of the second state of the second	Restricted Caller ID Deactivate	
DTMF & Dialing		
Supplementary Services		
💷 Analog Gateway	Hot-line Activate	
Keypad Features	Hot-line Deactivate	
Metering Tones		
FXO Settings		
Authentication	Call Waiting Activate	
Automatic Dialing	Call Waiting Deactivate	
Caller Display Information		



11) For calling to work we need to manage the IP to tel routing in routing menu. Give the source IP and huntgoup ID proeperly.



12) Same way we need to manage routing from the analog extensions behind MP11x to OXO .

Caudiocodes MP-118 F	XS_FXO	Submit 🧕 Bu	Device Action	ns 🔻	💼 Home 🔞	Help 💽	Log off		
Configuration Maintenance Status & Diagnostics	Tel to IP Routing)							
Scenarios Search			_						Basic Par
Basic 🖲 Full			Routing Index			1-	10 🔻		
Coders Group Settings			Tel To IP Routing Mode			R	oute calls before manip	ulation 🔻	
Tel Profile Settings									
GW and IP to IP	Src. Hunt Group ID	Dest. Phone Prefix	Source Phone Prefix	- >	Dest. IP Address	Port	Transport Type	Dest. IP Group ID	IP Profile ID
Endpoint Phone Number	1 1	*	*	10.9	224.220	5059	Not Configured V	1	1
Hunt Group Settings	2	7					Not Configured •	-1	
Manipulations	3						Not Configured V	-1	
Routing General Parame	4						Not Configured V	-1	
Tel to IP Routing	5						Not Configured V	-1	
IP to Hunt Group Routing	6	2					Not Configured		
Alternative Routing Reasons	2						Not Configured *	•	
Compared on Basy name	/						Not Configured •	•1	
DTMF & Dialing	8						Not Configured ¥	-1	
Supplementary Services	9						Not Configured V	-1	
Contemporary Cateway	4		11	1.1			hist Configured 💌	1	

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Once the configuration is done the enpoint numbers should be registered properly.

onfiguration Maintenance Status	Registration Status			
Scenarios Search				
Basic 🖲 Full 🖆 🚺	Registered Per Gateway			NO
System Status				
VoIP Status				
IP Interface Status	 Ports Registration St 	atus		
Performance Statistics	Gateway Port	S	itatus	
IP to Tel Calls Count	Port 1 FXS	R	EGISTERED	
Tol to ID Calls Count	Port 3 EVS	K	EGISTERED	
De lie d'ans count	Port 4 EXS	N	IOT REGISTERED	
Call Routing Status	Port 5 FXO	N	IOT REGISTERED	
Registration Status	Port 6 FXO	N	IOT REGISTERED	
IP Connectivity	Port 7 FXO	N	IOT REGISTERED	
	3 Port 8 FXO	N	IOT REGISTERED	
	 Accounts Registration 	n Status		
	Index	Group Type	Group Name	Status
		1 21	· · · · · · · · · · · · · · · · · · ·	
	 Phone Numbers State 	us		
	Pho	one Number	Gateway Port	Status

13) Message waiting Indication coniguration.

To enable MWI support (Message Waiting Indicator) go to the below mentioned link. Configuration->VoIP->GW and IP to IP->DTMF and Supplementary ->Supplementary Services

Caudiocodes MP-118 FXS_FX0	Submit i Burn Device Acti	ons 🔹 💼 Home 🌘	🙆 Help 🛛 🐑 Log off	
Configuration Maintenance Status Supp	lementary Services			
Scenarios Search	Send All Coders on Retrieve	Disable	T	
Basic Full	 Message Waiting Indication (MWI) Parameters 			
Coders	Enable MWI	Enable	•	
Tel Profile Settings	MWI Analog Lamp	Enable	• • 3	
IP Profile Settings	MWI Display	Enable	• • •	
GW and IP to IP	Subscribe to MWI	Yes	T	
E Hunt Group	MWI Server IP Address	10.9.224.220:5059	< 4	
Endpoint Phone Number	MWI Server Transport Type	UDP	· ·	
Hunt Group Settings	MWI Subscribe Expiration Time	7200	OXO IP addre	ess
Manipulations	Stutter Tone Duration	2000		
B DTME and Supplementary	MWI Subscribe Retry Time	120		
DTME & Dialing				
Supplementary Services	▼ MLPP	1	1	
Cateway	Call Priority Mode	Disable		
Keypad Features	Reminder Ring	Enable		
Metering Tones	MLPP Diffserv	50		
FXO Settings	Precedence Ringing Type	-1		

14) Conference Feature.

Caudiocodes MP-118 FXS_FXO	Submit 🧕 Burn	Device Actions Home	🕑 Help 🛛 🐑 Log off
guration Maintenance Status	centary Services		
arios Search			
	RTP DSCP for MLPP Routine	-1	_
sic • Full	RTP DSCP for MLPP Priority	-1	_
System	RTP DSCP for MLPP Immediate	-1	
oIP	RTP DSCP for MLPP Flash	-1	
Security	RTP DSCP for MLPP Flash-Override	-1	
Media	RTP DSCP for MLPP Flash-Override-Override	-1	
arvices	MLPP Default Service Domain	000000	
Applications Enabling	MLPP Normalized Service Domain	000000	_
Control Network			2
P Definitions	 Conference 		
oders and Profiles	Enable 3-Way Conference	Enable	•
W and IP to IP	Establish Conference Code	3	
Appinulations	Conference ID	conf	
Routing	3-Way Conference Mode	On Board	▼ 3
DTMF and Supplementary /1 5	Max. 3-Way Conference	2	
DTMF & Dialing 🖌 🗖 🥳	Non Allocatable Ports	0	
Supplementary Services			
Analog Gateway	 Transfer 		
Advanced Applications	Blind Transfer		



Configuration with default proxy

If you are using the above configuration then please ignore this section. The endpoint management and extension number declaration is same desribed above.

The other general features can be declared

Default Proxy settings

Prox	y Set ID		0	•	
		Proxy Add	ress	Transport Type	
	1	10.9.224.220:5059		UDP 🔻	
	2			•	
	3			•	
	4			•	
	5			T	_
-					
Enab	le Proxy i	Keep Alive	Using Register	•	2
Prox	y Keep Al	ive Time	60		
Prox	y Load Ba	lancing Method	Disable	v	
Is Pr	oxy Hot S	Swap	No	•	



Procedure to load the tones.

The tones can be loaded in two ways.

1) With firmware and other tones using the Software configuration wizard.

Launch the software upgrade the wizard.





2) Or using the load auxillary files option in the maintainance tab as show below.





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

The tests were performed with both OXO connect (Power CPU EE) Software and OXO connect Evolution software.

Both softwares were of same version R3.0/045.001

License requirment.

OPEN SIP licenses are needed only if all call features are needed.

0×0 Model Type	Standard		
Software License Compatibility level	default		
	Authorized by software key	Really activated	
Universal telephony	300	300	
Open SIP Phone users	20	20	
VoIP channels			
My IC Mobile users (OTCV)	20	20	
My IC Web users	50	50	
Hot Desking users	52	52	

The configuration is same for both the versions of OXO software (OXO connect and OXO connection evolution)

SIP Set Configuration

Normal calls & FAX calls

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

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



For Normal calling functionalities we used Open SIP extension type

Automatic Routing: Preri	(a. 1. m)				
- 🔤 Trunk Groups Lists	Subscribers/Ba:	sestations List			
- Hours	Phy. Add.	No.	🔘 Terminal/Basestat.	🔘 Name	Add
- E Day Groups	94-009-01	166	IP Enabler	FAX	
- Providers/Destinations			8068 Premium DeskPhone		Delete
- Authorization Codes	96-008-01	153	8068s Premium DeskPhone	ACD 🔺	14 - 450
Tone/Dause-ME	96-009-01	154	8078s Premium DeskPhone	ACD	Modily
	96-010-01	155	8088 Smart DeskPhone	ACD	Details
ARS Miscellaneous	96-011-01	156	Advanced/IP	ACD	Distant
Collective Speed Dialing	96-012-01	157	Basic SIP Phone	ALD	Сору
💼 🐳 Emergency 💫 📶 🗉	96-013-01	158	Easy/IP First/IP	ACD	
	96-014-01	159	IP Deskton Softphone	ALD	More
	96-010-01	160	IP Enabler	ACD	
LOAP CONNECTOR	04.005.01	101	IPT ouch 4008/IP	ACD	Profiles
 Subscribers/Basestations List 	94 005 01	162	IPT ouch 4018/IP IPT ouch 4029/IP		Fill
🖶 💋 Voice Processing	94-007-01	163	IPTouch 4028G/IP		
- 🕐 Time Ranges	94-008-01	165	IPTouch 4038/IP	Polu	GAP Reg.
Attendant Groups	94-009-01	166	IPT ouch 4038G/IP IPT ouch 4068/IP	FAX T	Del MaiBox
- 🚱 Hunting Groups			IPTouch 4068G/IP		Dermaibox
Broadcast Groups	Automatic provis	sioning for IP pho	MIPT 300		Auto Provision
Pickup Groups			- MIPT 600 - 5		
	Beturn		MIPT 610		
Manager-Secretary Relations			MIPT 8118		
🕀 🧑 Subscribers Misc			MIPT 8128 Open SIP Phone		
🖶 🏤 External Lines			PC Multimedia		
List of Accesses			Premium/IP SIP Phone (8001)		
			SIP Phone (8001G)		
SID Gatavarue					

For fax purpose we need to use BASIC SIP type.

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Subscribers/Base	estations List			×
Phy. Add. 94-016-01	◯ No. 185	○ Terminal/Basestat.	© Name	Add
 Phy. Add. 94-016-01 96-001-01 96-002-01 96-002-01 96-003-01 96-005-01 96-006-01 96-007-01 96-009-01 96-009-01 96-010-01 96-012-01 96-013-01 Automatic provisi 	No. 185 146 147 148 149 150 151 152 153 154 155 156 157 158 oning for IP phor	Terminal/Basestat. IP Enabler S058s Premium DeskPhone 8068 Premium DeskPhone 8068 Premium DeskPhone 8078s Premium DeskPhone 8078s Premium DeskPhone 8082 My IC phone 8088 Smart DeskPhone 90 SIP Phone 90 Milpr 8128 0 pen SIP Phone 90 Milpr edia	ACD ACD ACD ACD ACD ACD ACD ACD ACD ACD	Add Delete Modify Details Copy More Profiles Fill GAP Reg. Del MailBox Auto Provision
		Premium/IP SIP Phone (8001) SIP Phone (8001G)		

	Enterprise			
ubscriber				23
Phy. Add.	94-009-01		Keys	V 24
Name	FAX		Features	Password
Dir. Numbers			Metering	ISDN
Int. No.	166	More	Pers. SPD.	Services
Secondary sets			Spd Dial	Misc.
IP/SIP Parameters			Barring	Diversion
Termin Origina	Parameters		Dvn. Rout.	Sel.Divers
			DECT/PWT	Hotel
Mode 53468904	Reset		IP/SIP	Appoint
Langu			Comb Sort	Maileau
Softwa 🛛 👽 SIP authenticatio	n			
BootLo			Mobility	Heset
Data V				
Hardw				
Serial I				7
Localiz 2		Physic	al out of service	1
Custon		Set N	at Connected	
Virtual				
Entity OK	Cancel			
Hot Desking set		Ou	it of Service (log	ically)

Network OXO Configuration.

This configuration was needed for the setup that we performed above.

The network OXO setup was done for executing certain specific test cases and can be used for checking external calling facility as OXO connect evolution does not have ISDN option.

Basically network OXO can be considered for external calling scenarios.

1) Check the VOIP channel availability.

	Enterprise	
/oIP: Parameters		— ×
General Gateway SIP Trunk SIP P	none	
VoIP Channels VoIP Channels mode	Multi-codecs [16]	•
Number of VoIP Channels		16
VoIP Channels for trunks with reserva	tion	3
VoIP Channels for IP phones and trun	ks without reservation	13
Number of Trunk Channels for trunks	without reservation	0
IP Quality of Service	10111000 DIFFSERV_PHB_E	•

2) Add the gateway under the SIP setting in external lines.

- 🤍 Directory	SIF	P Gatew	ays					- ×
LDAP Connector		SIP Gat	eways List					
Subscribers/Basestations List		Index	Index Label	IP Tupe	IP Address	Hostname	Domain Name	
Voice Processing		1	Index Edber	Statio	10.9.224.50	mostriane	Domain Name	
🖳 🕐 Time Ranges		2		Static	10.9.224.30			Currh
🕂 🍂 Attendant Groups		-		01000	10.0.221.210			Lreate
- 🍄 Hunting Groups					1			Details
Pickup Groups								Delete
	=				3			
Subscribers Misc					_			
😤 External Lines								Сору
								Paste
								1 4660
🖕 💼 SIP								
SIP Gateways								
- SIP Accounts 🔨								
SIP Public Numbering								
Traffic Counters 2								
Protocols								
- 📰 Analog Protocols Selection								
🔤 Incoming Call Handling								
		04						
- 🔤 Main Cabinet 🛛	- 11	UK	Lan	icei				

3) Add the VOIP channel to the list of access in external lines.

	— Alcatel·L	Ucent 💋)		
	Enterprise				
Attendant Groups	List of Accesse	5			_
Broadcast Groups	Phy. Add.	C Acc. Type	Identifier	No of Chan.	Add VolP
Pickup Groups Manager-Secretary Relations	05-005-01 05-010-01 05-011-01	TO TO TO	N000 N002 N003	2 4	Delete
General Lines List of Accesses	95-001-01 95-002-01	TU VolP VolP	NUU4 V001 V002	2 2 1	
List of Trunk Groups					
1 Remote Substitution]
	Return]			
Incoming Call Handling					
uran Hardware and Limits 					

.

. 0

ĺ	VoIP-Trunk	×
	Phy. Add. Type Identifier Trunk Channels 95-001-01 VoIP V001 2	l Dial Dist.
	Metering Counters Meter part. 0 Link- Meter total 0 3	Cat.
	Reserve VoIP Channels for the Trunk Channels Out of Service (logical) Public trunk	
	SIP Gateway Gateway Index 2- Gateway Alive Status 2-	
	Alternative CLIP/COLP Number	
	OK Cancel	4

4) Select the gateway in the list of access as shown below.

5) In the ARS manage the numbering range of the Other OXO so that it can be reached.

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пэсапасіон сурісаі **Vodification** typical Numbering Plans ж Vumbering Internal Numbering Plan Public Numbering Plan Restricted Public Numbering Plan Private Numbering Plan Installation Numbers 📰 Default Configuration Function Start End Base NMT Priv SIP Acc.Index Fax Add 399 📰 Numbering Plans Secondary Trunk Group 💌 300 ARS Yes 🔻 Кеер Ŧ . Delete 📰 Features in Contersation Resend Last Number Cancel Mail Booking Mail Booking Call Forwarding Activate Meet Me Join Meet Me Drop Drop # # *#6 *6 *7 *8 No No No No No No * DDI Number Multi Number Modification *#6 **6 *1 *7 *8 0 100 Modify ication T = Drop Drop Drop 0 0 9 100 on Table Up 📰 Splitting Table Down Drop 📰 End of Dialing Table Attendant Call Subscriber 0 199 Drop 📄 Automatic Routing Selectior ubscrib 🔤 Automatic Routing: Prefi Call Forwarding Secondary Trunk Group 📼 Trunk Groups Lists Drop No No 40 500 40 534 - 📰 Hours 🔤 Day Groups 2 Cancel Providers/Destinations ΟK - Authorization Codes 📰 Tone/Pause-MF DS Missellaneeu

Manage the Automatic routing prefixes





12 Appendix D: AAPP member's escalation process

In case you would need technical assistance, please contact the reseller/distributor where you purchased your AudioCodes products. They have been trained on the products to give you 1st and 2nd levels of support. They are in plus in direct relation with 3rd level AudioCodes support in case an escalation would be needed.



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 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 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 https://www.al-enterprise.com/en/partners/aapp



13.2 Enterprise.Alcatel-Lucent.com

You can access the Alcatel-Lucent Enterprise website at this URL: https://www.al-enterprise.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 Business Partners when facing a problem with the solution certified in this document.

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

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



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

- Case 1: the responsibility can be established 100% on ALE side. In that case, the problem must be escalated by the ALE Business Partner to the ALE Support Center using the standard process: open a ticket (eService Request –eSR)
- Case 2: the responsibility can be established 100% on 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 Business Partner will escalate the problem to the ALE Support Center only if the Application Partner <u>has demonstrated with traces a problem on the ALE side</u> or if the Application Partner (not the Business Partner) <u>needs the involvement of ALE</u>

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

ALE reserves the right to close the case opened on his side if the investigations made on the 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 offers the "On Demand Diagnostic" service where ALE will provide 8 hours assistance against payment .

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

Please check the availability of the Inter-Working Report on the AAPP (URL: <u>https://www.al-enterprise.com/en/partners/aapp</u>) or Enterprise Business Portal (Url: <u>Enterprise Business Portal</u>) web sites.

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



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 Support and shares all trouble shooting information and conclusions that shows a need for ALE 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 offers an "On Demand Diagnostic" service where assistance will be provided for a fee.



14.4 Technical support access

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

ALE Business Partners Support Center for countries:

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

For other countries:

+ 1	650 385 2193	3
+ 1	650 385 2196	3
+ 1	650 385 2197	7

+ 1 650 385 2198

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