ALE Application Partner Program
Inter-Working Report

Partner: **POLYCOM**
Application type: **SIP conference device**
Application name: **Trio 8500, 8800, SoundStation Duo, IP 5000, 6000, 7000**
Alcatel-Lucent Enterprise Platform: **OmniPCX Enterprise™**

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

Date of the certification	March 2018

ALE International representative	Sebastien EHRHARD
AAPP member representative	Dave Finney

Alcatel-Lucent Enterprise Communication Platform	OmniPCX Enterprise
Alcatel-Lucent Enterprise Communication Platform release	R12.1 (m2.300.18.d)
AAPP member application release	5.5.3.3441 (Trio 8500/8800) 4.1.1.0731 (Duo, IP5000) 4.0.13.1445 (IP6000/7000)
Application Category	Terminals Conferencing (audio & video)

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Revision History

Edition 1: creation of the document – *March 2018*

Test results

☐ Passed ☐ Refused ☐ Postponed

☑ Passed with restrictions

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

IWR validity extension

None

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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 International cannot guarantee accuracy of printed material after the date of certification nor can it accept responsibility for errors or omissions. Updates to this document can be viewed on:

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

- the Application Partner portal (https://applicationpartner.alcatel-lucent.com) with free access.

2 Validity of the InterWorking Report

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

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

A new release is identified as following:

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

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

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

3 Limits of the Technical support

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

The certification does not verify the functional achievement of the AAPP member's application as well as it does not cover load capacity checks, race conditions and generally speaking any real customer's site conditions.

Any possible issue will require first to be addressed and analyzed by the AAPP member before being escalated to ALE International. Access to technical support by the Business Partner requires a valid ALE maintenance contract

For details on all cases (3rd party application certified or not, request outside the scope of this IWR, etc.), please refer to Appendix F "AAPP Escalation Process".

3.1 Case of additional Third-party applications

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

4 Application information

Application family : SIP conference phones

Application commercial name:
Trio 8800, 8500
SoundStation Duo, IP 5000, IP 6000, IP 7000

Application version:

Trio 8800, 8500:	5.5.3.3441
SoundStation Duo, IP 5000:	4.1.1.0731
SoundStation IP 6000, IP 7000:	4.0.13.1445

Interface type: SIP

Interface version (if relevant): /

Trio 8800 and Trio 8500 share the same software.
SoundStation IP 6000 and SoundStation IP 7000 share the same software.
SoundStation Duo and SoundStation IP 5000 share the same software.

Trio 8800

Trio 8500

IP6000

Duo

5 Tests environment

5.1 General architecture

Figure 1 **Tests environment**

6 Summary of test results

6.1 Summary of main functions supported

The interoperability tests between the Polycom devices and Alcatel OmniPCX Enterprise have been pure functional tests. No tests have been performed regarding the respect of the SIP protocol itself.
Only subjective audio tests have been performed.
No video tests have been performed.

Feature	N/A	OK	OK But	NOK
Connectivity and Setup				
IP network connectivity	☐	☒	☐	☐
SIP Registration	☐	☒	☐	☐
Duplication and Robustness				
Spatial redundancy with alternate proxy method	☐	☐	☐	☒
Spatial redundancy with alternate DNS method	☐	☐	☐	☒
Switchover to PCS	☐	☐	☐	☐
Erreur ! Source du renvoi introuvable.				
G 711 A, G 711 µ support (Uncompressed codec)	☐	☒	☐	☐
G 723.1, G 729 support (Compressed codec)	☐	☐	☒	☐
Outgoing Calls				
Local/Network calls	☐	☒	☐	☐
Call to a forwarded user	☐	☒	☐	☐
OXE features (Call back, voice mail deposit)	☐	☐	☐	☒
Incoming Calls				
Local/Network calls	☐	☒	☐	☐
Forward (Immediate/On no reply/On busy)	☐	☐	☒	☐
Erreur ! Source du renvoi introuvable.	☐	☒	☐	☐
Features during Conversation				
Hold/resume	☐	☒	☐	☐
DTMF sending	☐	☒	☐	☐
Conference	☐	☒	☐	☐
Call Transfer				
Unattended transfers	☐	☒	☐	☐
Semi-Attended Transfer (on Ringing)	☐	☒	☐	☐
Attended Transfer (in Conversation)	☐	☒	☐	☐
Attendant				
Call to/from an attendant station	☐	☒	☐	☐
Call transfer from an attendant station	☐	☒	☐	☐
Voice Mail				
Message waiting indicator	☐	☒	☐	☐
Forward to voice mail	☐	☒	☐	☐
Local Telephonic Features				
Hold Key	☐	☒	☐	☐
Broker key	☐	☒	☐	☐
Transfer Key in Ringing / Conversation	☐	☒	☐	☐
Conference key	☐	☒	☐	☐

6.2 Summary of problems

- Spatial redundancy (OXE Call Servers in different subnetworks)

  After the switch over from one Call Server to the other, when a Polycom equipment calls another Polycom equipment, the call is not set up when the called one answer.
  The issue is the same whatever the configuration: OXE as DNS server, external DNS server (delegation) or multi proxies.
  There is no issue when the Polycom equipment calls an OXE device (8068 for example).

  Ongoing OXE ticket: 1-219970853

6.3 Summary of limitations

- G723 codec

  The Polycom equipment does not support G723 codec.

- Voice Activity Detection (VAD)

  The Polycom equipment does not offer VAD configuration possibility.

- Suffix during conversation: call back and voice mail deposit suffix

  The Polycom equipment does not offer to display the dial pad during an outgoing call (ringing phase). Thus, OXE call back and voice mail deposit suffix are not supported.

- Ongoing conversation features: Do not disturb

  The Polycom equipment does not offer to display the dial pad during an outgoing call (ringing phase). Thus, using some OXE prefix like DND does only work when dialing the complete sequence in one step. For example: 420000 to activated the DND (where 0000 is the user password). 42 – start the call and listen to the voice guide asking for the user password – dial 0000 does not work.

- Local Polycom equipment feature: forward on busy

  Configuring the Polycom equipment locally to forward the incoming call when the user is busy does not work. The caller is not forwarded to the end destination.
  The OXE forward on busy feature is supported.

- Twin set: missed call

  The Polycom equipment is declared as a twin set of another device.
  When calling the user and after answering the call on the twin device, the Polycom equipment notifies the user with a missed call.

- Transfers: unattended transfer from an OXE device

The OXE does not offer the unattended transfer feature from its proprietary phones. This feature is not supported when triggered for example from an 8068 phone.
It is working when the “transferor” is a Polycom equipment.

6.4 Notes, remarks

None.

7 Test Result Template

The results are presented as indicated in the example below:

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

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

The interoperability tests between the Polycom devices and Alcatel OmniPCX Enterprise have been pure functional tests. No tests have been performed regarding the respect of the SIP protocol itself.
Only subjective audio tests have been performed.
No video tests have been performed.

SIPset-x are the Polycom devices located on the main OXE node
OXEset-x are OXE IP phones (like 8068) located on the main OXE node
NwkSIPset-x are the Polycom devices located on another OXE node
NwkOXEset-x are OXE IP phones (like 8068) located on another OXE node

8.1 Connectivity and Setup

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	IP network connectivity				
A	**SIP set network setup with a static IP address** Configure the phone SIPset-1 with a static IP address Check the network connectivity by pinging the phone and display.	☐	☒	☐	
B	**SIP set network setup with a dynamic IP address** Configure the phone SIPset-1 with a dynamic IP address (given by a DHCP server) Check the network connectivity by pinging the phone and display.	☐	☒	☐	
2	SIP Registration				
A	**SIP registration, using OXE MAIN IP address(es) (without authentication)** The phone SIPset-1 is configured to register with the node1 primary main IP address. Check the phone registration and display.	☐	☒	☐	
B	**SIP registration, using OXE as DNS server (without authentication)** SIPset-1 DNS servers are configured with node1 primary main IP address as primary DNS server and with node1 secondary main IP address as secondary DNS server. SIPset-1 is configured to register with the node1 hostname. Check the phone registration and display.	☐	☒	☐	
C	**SIP registration, using an External DNS server (without authentication)** SIPset-1 DNS server is configured with the external IP address as primary DNS server There is no other DNS server address configured in SIPset-1. And SIP SIPset-1 is configured to register with the node1 hostname. Check the phone registration and display.	☐	☒	☐	
D	**Support of "423 Interval Too Brief" (1)** The phone SIPset-1 is configured with a value lower than OXE SIP Min Expiration Date. Check the phone registration and display.	☐	☒	☐	
E	**SIP registration with authentication** For this test, register on a Node with authentication enable (2) Configure the phone SIPset-1 with its authentication password . Check the phone registration and display. After make the same actions with a wrong password and check that the phone is rejected.	☐	☒	☐	
3	UDP/TCP signaling				

Test Case Id	Test Case	N/A	OK	NOK	Comment
A	**Signalling TCP.** Configure your SIP set to use the protocol SIP over TCP Check the registration, and basic calls.	☐	☐	☐	Not tested
B	**Signalling UDP.** Configure your SIP set to use the protocol SIP over UDP Check the registration, and basic calls. Note: all further tests to be made with UDP	☐	☒	☐	
4	Time synchronization				
A	**NTP registration (if applicable)** The SIP phone SIPset-1 is configured to retrieve the date and time from the node1 primary main IP address. Check that SIPset-1 retrieves the right date and time information and displays it.	☐	☒	☐	
5	Keep alive				
A	**Keep alive with SIP OPTIONS messages** Configure SIPset-1 to send SIP OPTIONS messages to OXE (3). Check that SIPset-1 stay "In service" (with "csipsets" OXE command) after SIPset-1 sends a SIP OPTIONS message and receives a response back from OXE. Disconnect SIPset-1 by removing the cable for a wired phone or the battery for a wireless phone. Check that SIPset-1 fails to "Out of service" state after a SIP OPTIONS period (by default 30 seconds).	☐	☐	☐	Not tested

Notes:

(1) On the SIP client, specify a default registration period inferior to that of OXE SIP registrar (configured via mgr under SIP/SIP Registrar/SIP Min Expiration Date). OXE will reject with error "423 Interval Too Brief". Check that SIP set increases registration period accordingly and the registration happens successfully.

(2) The SIP authentication is configured via mgr under : SIP/SIP Proxy/Minimal authentication method="SIP None" or "SIP Digest"

(3) The SIP keep-alive should be activated on OXE. The configuration is done with "Keep alive" parameter of the SIP phone classes of service. The SIP keep-alive period is configured in the IP Quality of Service COS.

8.2 Duplication and Robustness

8.2.1 Test results

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

Spatial redundancy can be configured in two ways:

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

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

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	Spatial redundancy with alternate proxy method				
A	**Spatial redundancy, using "Alternate Proxy method", two SIP sets in conversation** Configure SIPset-1, SIPset-2 and SIPset-3 to use two SIP proxies (OXE call server a and OXE call server b IP addresses). Configure SIPset-1, SIPset-2 and SIPset-3 to send SIP OPTIONS keep alive messages to both SIP proxies. With SIPset-1 call SIPset-2. Answer the call and check audio and display. Switchover to standby call server using OXE "bascul" command (check first the database replication using OXE "twin" command). Check that the existing call is maintained. Wait for a "session timer" expiration (this timer is negotiated between the INVITE and the OK message). Check that the call is maintained after this timer expiration. With SIPset-1 call SIPset-3. Answer the call and check that SIPset-2 is put on hold. With SIPset-1 transfer SIPset-3 to SIPset-2. Check that the transfer is correctly performed.	☐	☐	☒	After the bascul, when making a call from one SIP set to another, the called one can not answer. Caller continues to play the ring back melody. There is no issue when the SIP set calls an OXE set.

Test Case Id	Test Case	N/A	OK	NOK	Comment
B	**Spatial redundancy, using "Alternate Proxy method", one SIP set in conversation with a external party** Configure SIPset-1 and SIPset-2 to use two SIP proxies (OXE call server a and OXE call server b IP addresses). Configure SIPset-1 and SIPset-2 to send SIP OPTIONS keep alive messages to both SIP proxies. SIPset-1 call 72SIPset-2("prefix to take T2 loopback"+target MCDU number>). Answer the call and check audio and display. Switchover to standby call server using OXE "bascul" command (check first the database replication using OXE "twin" command). Check that the call is maintained. Wait for a "session timer" expiration (this timer is negotiated between the INVITE and the OK message). Check that the call is maintained after this timer expiration.	☐	☐	☒	After the bascul, when making a call from one SIP set to another, the called one can not answer. Caller continues to play the ring back melody. There is no issue when the SIP set calls an OXE set.
C	**Spatial redundancy, using "Alternate Proxy method", call after the switchover** Configure SIPset-1 and SIPset-2 to use two SIP proxies (OXE call server a and OXE call server b IP addresses). Configure SIPset-1 and SIPset-2 to send SIP OPTIONS keep alive messages to both SIP proxies. Switchover to standby call server using OXE "bascul" command (check first the database replication using OXE "twin" command). Just after the switchover, with SIPset-1 call SIPset-2. Answer the call and check audio and display.	☐	☐	☒	After the bascul, when making a call from one SIP set to another, the called one can not answer. Caller continues to play the ring back melody. There is no issue when the SIP set calls an OXE set.
D	**Spatial redundancy, using "Alternate Proxy method", call after the switchover and a registration timeout** Configure SIPset-1 and SIPset-2 to use two SIP proxies (OXE call server a and OXE call server b IP addresses). Configure SIPset-1 and SIPset-2 to send SIP OPTIONS keep alive messages to both SIP proxies. Switchover to standby call server using OXE "bascul" command (check first the database replication using OXE "twin" command). Wait for a Registration period timeout. After this period, with SIPset-1 call SIPset-2. Answer the call and check audio and display.	☐	☐	☒	After the bascul, when making a call from one SIP set to another, the called one can not answer. Caller continues to play the ring back melody. There is no issue when the SIP set calls an OXE set.

Test Case Id	Test Case	N/A	OK	NOK	Comment
2	Spatial redundancy with alternate DNS method				
A	**Spatial redundancy, using "DNS method" and OXE used as DNS server, two SIP sets in conversation** Configure SIPset-1, SIPset-2 and SIPset-3 to use one SIP proxies (OXE call server node name). Configure SIPset-1, SIPset-2 and SIPset-3 to use two DNS servers (OXE call server a and OXE call server b IP addresses). With SIPset-1 call SIPset-2. Answer the call and check audio and display. Switchover to standby call server using OXE "bascul" command (check first the database replication using OXE "twin" command). Check that the existing call is maintained. Wait for a "session timer" expiration (this timer is negotiated between the INVITE and the OK message). Check that the call is maintained after this timer expiration. With SIPset-1 call SIPset-3. Answer the call and check that SIPset-2 is put on hold. With SIPset-1 transfer SIPset-3 to SIPset-2. Check that the transfer is correctly performed.	☐	☐	☐	Not tested
B	**Spatial redundancy, using "DNS method" and OXE used as DNS server, one SIP set in conversation with a external party** Configure SIPset-1 and SIPset-2 to use one SIP proxies (OXE call server node name). Configure SIPset-1 and SIPset-2 to use two DNS servers (OXE call server a and OXE call server b IP addresses). SIPset-1 call 72SIPset-2("prefix to take T2 loopback"+target MCDU number>). Answer the call and check audio and display. Switchover to standby call server using OXE "bascul" command (check first the database replication using OXE "twin" command). Check that the call is maintained. Wait for a "session timer" expiration (this timer is negotiated between the INVITE and the OK message). Check that the call is maintained after this timer expiration.	☐	☐	☐	Not tested

Test Case Id	Test Case	N/A	OK	NOK	Comment
C	**Spatial redundancy, using "DNS method" and OXE used as DNS server, call after the switchover** Configure SIPset-1 and SIPset-2 to use one SIP proxies (OXE call server node name). Configure SIPset-1 and SIPset-2 to use two DNS servers (OXE call server a and OXE call server b IP addresses). Switchover to standby call server using OXE "bascul" command (check first the database replication using OXE "twin" command). Just after the switchover, with SIPset-1 call SIPset-2. Answer the call and check audio and display.	☐	☐	☐	Not tested
D	**Spatial redundancy, using "DNS method" and OXE used as DNS server, call after the switchover and a registration timeout** Configure SIPset-1 and SIPset-2 to use one SIP proxies (OXE call server node name). Configure SIPset-1 and SIPset-2 to use two DNS servers (OXE call server a and OXE call server b IP addresses). Switchover to standby call server using OXE "bascul" command (check first the database replication using OXE "twin" command). Wait for a Registration period timeout. After this period, with SIPset-1 call SIPset-2. Answer the call and check audio and display.	☐	☐	☐	Not tested
E	**Spatial redundancy, using "DNS method" with a delegation DNS server, two SIP sets in conversation** Configure SIPset-1, SIPset-2 and SIPset-3 to use one SIP proxies (OXE call server node name). Configure SIPset-1, SIPset-2 and SIPset-3 to use one DNS server (An external DNS server delegates the DNS request to OXE call server a and OXE call server b). With SIPset-1 call SIPset-2. Answer the call and check audio and display. Switchover to standby call server using OXE "bascul" command (check first the database replication using OXE "twin" command). Check that the existing call is maintained. Wait for a "session timer" expiration (this timer is negotiated between the INVITE and the OK message). Check that the call is maintained after this timer expiration. With SIPset-1 call SIPset-3. Answer the call and check that SIPset-2 is put on hold. With SIPset-1 transfer SIPset-3 to SIPset-2. Check that the transfer is correctly performed.	☐	☐	☒	After the bascul, when making a call from one SIP set to another, the called one can not answer. Caller continues to play the ring back melody. There is no issue when the SIP set calls an OXE set.

Test Case Id	Test Case	N/A	OK	NOK	Comment
F	**Spatial redundancy, using "DNS method" with a delegation DNS server, one SIP set in conversation with a external party** Configure SIPset-1 and SIPset-2 to use one SIP proxies (OXE call server node name). Configure SIPset-1 and SIPset-2 to use one DNS server (An external DNS server delegates the DNS request to OXE call server a and OXE call server b). SIPset-1 call 72SIPset-2("prefix to take T2 loopback"+target MCDU number>). Answer the call and check audio and display. Switchover to standby call server using OXE "bascul" command (check first the database replication using OXE "twin" command). Check that the call is maintained. Wait for a "session timer" expiration (this timer is negotiated between the INVITE and the OK message). Check that the call is maintained after this timer expiration.	☐	☐	☒	After the bascul, when making a call from one SIP set to another, the called one can not answer. Caller continues to play the ring back melody. There is no issue when the SIP set calls an OXE set.
G	**Spatial redundancy, using "DNS method" with a delegation DNS server, call after the switchover** Configure SIPset-1 and SIPset-2 to use one SIP proxies (OXE call server node name). Configure SIPset-1 and SIPset-2 to use one DNS server (An external DNS server delegates the DNS request to OXE call server a and OXE call server b). Switchover to standby call server using OXE "bascul" command (check first the database replication using OXE "twin" command). Just after the switchover, with SIPset-1 call SIPset-2. Answer the call and check audio and display.	☐	☐	☒	After the bascul, when making a call from one SIP set to another, the called one can not answer. Caller continues to play the ring back melody. There is no issue when the SIP set calls an OXE set.
H	**Spatial redundancy, using "DNS method" with a delegation DNS server, call after the switchover and a registration timeout** Configure SIPset-1 and SIPset-2 to use one SIP proxies (OXE call server node name). Configure SIPset-1 and SIPset-2 to use one DNS server (An external DNS server delegates the DNS request to OXE call server a and OXE call server b). Switchover to standby call server using OXE "bascul" command (check first the database replication using OXE "twin" command). Wait for a Registration period timeout. After this period, with SIPset-1 call SIPset-2. Answer the call and check audio and display.	☐	☐	☒	After the bascul, when making a call from one SIP set to another, the called one can not answer. Caller continues to play the ring back melody. There is no issue when the SIP set calls an OXE set.

Test Case Id	Test Case	N/A	OK	NOK	Comment
3	Switchover to PCS				
A	**Passive call server backup, using "Alternate Proxy method", two SIP sets in conversation** Configure SIPset-1, SIPset-2 and SIPset-3 to use two SIP proxies (OXE call server a and OXE call server PCS IP addresses). SIPset-1, SIPset-2 and SIPset-3 are part of IP domain with a PCS backup. Configure SIPset-1, SIPset-2 and SIPset-3 to send SIP OPTIONS keep alive messages to both SIP proxies. With SIPset-1 call SIPset-2. Answer the call and check audio and display. Stop OXE call server a and call server b. Check that the existing call is maintained. Wait for a "session timer" expiration (this timer is negotiated between the INVITE and the OK message). Check that the call is maintained after this timer expiration. With SIPset-1 call SIPset-3. Answer the call and check that SIPset-2 is put on hold. With SIPset-1 transfer SIPset-3 to SIPset-2. Check that the transfer is correctly performed.	☐	☐	☐	Not tested
B	**Passive call server backup, using "Alternate Proxy method", one SIP set in conversation with a external party** Configure SIPset-1 and SIPset-2 to use two SIP proxies (OXE call server a and OXE call server PCS IP addresses). SIPset-1 and SIPset-2 are part of IP domain with a PCS backup. Configure SIPset-1 and SIPset-2 to send SIP OPTIONS keep alive messages to both SIP proxies. SIPset-1 call 72SIPset-2("prefix to take T2 loopback"+target MCDU number>). Answer the call and check audio and display. Stop OXE call server a and call server b. Check that the call is maintained. Wait for a "session timer" expiration (this timer is negotiated between the INVITE and the OK message). Check that the call is maintained after this timer expiration.	☐	☐	☐	Not tested

Test Case Id	Test Case	N/A	OK	NOK	Comment
C	**Passive call server backup, using "Alternate Proxy method", call after the switchover** Configure SIPset-1 and SIPset-2 to use two SIP proxies (OXE call server a and OXE call server PCS IP addresses). SIPset-1 and SIPset-2 are part of IP domain with a PCS backup. Configure SIPset-1 and SIPset-2 to send SIP OPTIONS keep alive messages to both SIP proxies. Stop OXE call server a and call server b. Just after the switchover, with SIPset-1 call SIPset-2. Answer the call and check audio and display.	☐	☐	☐	Not tested
4	Partner SIP endpoint reboot				
A	**Partner SIP set reboot** Reboot SIPset-1. When SIPset-1 comes back in service, call SIPset-2. Check that SIPset-1 is registered and the call establishment.	☐	☒	☐	
5	Network failure				
A	**Temporary Link between OXE and the partner SIP set** Disconnect the link between SIPset-1 and OXE. Check that SIPset-1 becomes out of service after a keep-alive or a registration period. When SIPset-1 comes back in service, call SIPset-1. Reconnect the link between SIPset-1 and OXE. Check that SIPset-1 becomes in service registration period.	☐	☒	☐	

8.2.2 Recommendation

For the moment, spatial redundancy is not supported.
For single Call Server or simple redundancy (both Call Server in the same subnetwork), DNS (delegation or OXE) or proxy methods can be used.

8.3 Audio codecs negotiations/ VAD / Framing

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

Base Configuration :
Phones:
Configure the phones to use G.711 A-law, G.711 µ-law, G.729, G.723.1 in this order.
Configure the phones to use framing=20ms (G.711 and G.729) and 30ms (G.723.1).
Configure the phones to NOT use VAD.

OXE :
Manage 2 IP domains:
Domain1: intra=no compression extra=compression
Domain2: intra=no compression extra=compression
Assign SIPset-1 and OXEset-1 to domain 1.
Assign SIPset-2 and OXEset-2 to domain 2.
Set system law = A-law (1)
Set system compression type = G.729 (2)

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	G 711 A, G 711 µ support (Uncompressed codec)				
A	**Call from SIPset-1 to OXEset-1 (intra-domain)** Check that the call is established using direct RTP in G711 A-law. Check audio quality **Call from OXEset-1 to SIPset-1 (intra-domain)** Check that the call is established using direct RTP in G711 A-law. Check audio quality	☐	☒	☐	
B	**Set system law = µ-law** **Configure the phone to use G.711 µ-law, G.711 A-law, G.729, G.723.1 in this order** **Call from SIPset-1 to OXEset-1 (intra-domain)** Check that the call is established using direct RTP in G711 µ-law. Check audio quality **Call from OXEset-1 to SIPset-1 (intra-domain)** Check that the call is established using direct RTP in G711 µ-law. Check audio quality	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
2	G 723.1, G 729 support (Compressed codec)				
A	**Call from SIPset-1 to OXEset-2 (extra-domain)** Check that the call is established using direct RTP in G729. Check audio quality **Call from OXEset-2 to SIPset-1 (extra-domain)** Check that the call is established using direct RTP in G729. Check audio quality	☐	☒	☐	
B	**Set system compression type = G.723.1** **Call from SIPset-1 to OXEset-2 (extra-domain)** Check that the call is established using direct RTP in G723.1. Check audio quality **Call from OXEset-2 to SIPset-1 (extra-domain)** Check that the call is established using direct RTP in G723.1. Check audio quality	☐	☐	☒	Polycom equipment does not support G723 codec.
3	Voice Activity Detection				
A	**Configure SIPset-1 to use VAD** **Configure OXEset-1 NOT to use VAD** **Call from SIPset-1 to OXEset-1 (intra-domain)** Check that the call is established using direct RTP in G711 A-law. Check audio quality **Call from OXEset-1 to SIPset-1 (intra-domain)** Check that the call is established using direct RTP in G711 A-law. Check audio quality **Configure SIPset-1 to use VAD** **Configure OXEset-1 to use VAD** Redo the same tests **Configure SIPset-1 NOT to use VAD** **Configure OXEset-1 to use VAD** Redo the same tests	☒	☐	☐	Polycom equipment does not provide VAD configuration possibilities.

Test Case Id	Test Case	N/A	OK	NOK	Comment
B	**Configure** SIPset-1 **to use VAD** **Configure** OXEset-2 **NOT to use VAD** **Call from** SIPset-1 **to** OXEset-2 **(extra-domain)** Check that the call is established using direct RTP in G729. Check audio quality **Call from** OXEset-2 **to** SIPset-1 **(extra-domain)** Check that the call is established using direct RTP in G729. Check audio quality **Configure** SIPset-1 **to use VAD** **Configure** OXEset-2 **to use VAD** Redo the same tests **Configure** SIPset-1 **NOT to use VAD** **Configure** OXEset-2 **to use VAD** Redo the same tests	☒	☐	☐	Polycom equipment does not provide VAD configuration possibilities.
4	Packet framing				
A	**Configure** SIPset-1 **to use framing=30ms (G.711)** **Call from** SIPset-1 **to** OXEset-1 **(intra-domain)** Check that the call is established using direct RTP in G711 A-law. Check audio quality **Call from** OXEset-1 **to** SIPset-1 **(intra-domain)** Check that the call is established using direct RTP in G711 A-law. Check audio quality	☐	☒	☐	
B	**Configure** SIPset-1 **to use framing=30ms (G.729)** **Call from** SIPset-1 **to** OXEset-2 **(extra-domain)** Check that the call is established using direct RTP in G729. Check audio quality **Call from** OXEset-2 **to** SIPset-1 **(extra-domain)** Check that the call is established using direct RTP in G729. Check audio quality	☐	☒	☐	

Notes:

(1) The law choice is configured via mgr under : System/Other System Param./System Parameters/Law="A Law" or "Mu Law"

(2) The compression codec choice is configured via mgr under : System/Other System Param./Compression Parameters/Compression Type="G 723" or "G 729"

8.4 Outgoing Calls

Called party can be in different states: free, busy, out of service, do not disturb, etc.
Points to be checked: tones, voice during the conversation, display (name and extension number on caller and called party), hang-up phase.
OXE SEPLOS prefixes are mandatory for several tests of this section. For more information refer to the appendix.
By default, all phones are multiline set with two lines.

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

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	Local/Network calls				
A	**Call to a local user** With SIPset-1 call the OXE phone OXEset-1. Check that OXEset-1 is ringing. On SIPset-1 check the ring back tone. On both sets check display (name and extension number) Answer the call and check audio and display.	☐	☒	☐	
B	**Call to a local user with overlap dialing** With SIPset-1 call the OXE phone OXEset-1. **but**, Dial a first part of the number: call 33 (first part of OXEset-1 extension number), wait one second and dial 015. Check that call is transmitted to the OXEset-1which is ringing. When the OXEset-1 is ringing, hang-up Check release and display	☐	☒	☐	
C	**Call to a local user with overlap dialing, timeout** With SIPset-1 call the OXE phone OXEset-1. **but**, Dial a first part of the number: call 33 (first part of OXEset-1 extension number), and **never** dial the end of the number in order to have a timeout. Check time out and display	☐	☒	☐	After a timeout, the devices display the idle home screen.
D	**Call to another SIP set** With the SIPset-1 call the SIPset-2 Check the display and audio during all steps (dialing, ring back tone, conversation, and release).	☐	☒	☐	
E	**Call to a local user with SIP proxy Authentication** Check that SIPset-1 sip set configured with authentication. With SIPset-1 call SIPset-2. Answer the call, check audio and display.	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
F	**Call to external number (via T2 loopback)** (Check ring back tone, called party display) With SIPset-1 dial 72OXEset-1 ("prefix to take T2 loop"+target MCDU number>) Check that OXEset-1 is ringing. Answer the call and check audio, display and call release.	☐	☐	☐	Not tested
G	**SIP session timer expiration** Check if call is maintained after the session timer expiration: If possible, configure the "Session timer" on SIPset-1 to 120 seconds. With SIPset-1 call OXEset-1. Answer the call on OXEset-1and never hang up, wait for session timer expiration. Check that call is maintained. Configure the "Session timer" on SIPset-1 to the default value.	☐	☒	☐	
H	**Call to wrong number** (SIP: "404 Not Found") With the SIPset-1 call a wrong number which is not in the dialing plan. Check the ring back tone and display.	☐	☒	☐	
I	**Call rejected by call handling** (SIP: "183 Progress/487 Request Terminated") Try to enable OXE "do not disturb" feature on SIPset-1 with a **wrong** password calling 42 (Do not disturb prefix). After the voice guide, enter 1111 (<sip set **wrong** password>). Wait for error ring back tone from OXE. Check the call is rejected	☐	☒	☐	
2	Local/Network calls – called party is not available				
A	**Call to local user with no answer** With SIPset-1 call the OXE phone OXEset-1. And never answer the call. Check time out and display. Note that OXEset-1 don't have a Voice Mail	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
B	**Call to busy user** (SIP: "486 Busy Here") With SIPset-1 call OXEset-1, answer the call, and don't hang up. With SIPset-2 call OXEset-1 which is busy Check the ring back tone and display. With OXEset-1 call SIPset-1, answer the call, and don't hang up. With SIPset-2 call SIPset-1 which is busy Check the ring back tone and display.	☐	☒	☐	
C	**Call to user in "Out of Service" state** (SIP: "480 Temporarily Unavailable") Disconnect SIPset-2 and wait for SIP deregister With SIPset-1 call SIPset-2 which is in "Out of Service State" Check the display and ring back tone.	☐	☒	☐	
D	**Call to user in "Do not Disturb" (DND) state** (SIP: " 183 Session progress"): Dial "42" (Do not disturb prefix) on the OXEset-1 in order to enable the DND. Wait for acknowledgement from OXE With the SIPset-1 call the OXEset-1. Check ring back tone and display. Redial 42 on OXEset-1 to cancel the DND	☐	☒	☐	
3	Call release				
A	**Call release during an outgoing call, release done by the partner SIP set** With SIPset-1 call OXEset-1 and don't answer the call. With SIPset-1, release the call during the ringing period. Check that OXEset-1 plays a release tone and goes in idle mode after some seconds	☐	☒	☐	
4	Identity secrecy				
A	**Calling Line Identity Restriction (CLIR): Local call to SIP terminal with CLIR activated.** With SIPset-1 call OXEset-1 by dialing 409OXEset-1 (Secret identity prefix + <target MCDU number>) in order to hide SIPset-1 identity. Check that OXEset-1 is ringing, answer the call and check that SIPset-1 identity is hidden.	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
5	Call to a forwarded user				
A	**Call to local user, immediate forward (CFU).** (SIP: "302 Moved Temporarily")(1) On OXEset-1 dial the 51 (Immediate forward prefix). After the voice guide, enter SIPset-2 <target MCDU number>) to activate the CFU. Wait for acknowledgement from OXE. With SIPset-1 call the OXEset-1. Check that SIPset-2 is ringing and the display. Answer the call check audio and hung up. Dial 41 (Forward cancellation prefix) on OXEset-1 for forward cancellation.	☐	☒	☐	
B	**Call to local user, forward on no reply (CFNR). (1)** On OXEset-1 dial the 53 (Forward on no reply prefix) After the voice guide, enter SIPset-2 (<target MCDU number>) to activate the CFNR. Wait for acknowledgement from OXE. With SIPset-1 call the OXEset-1. Check that OXEset-1 is ringing but don't answer the call and wait the time out (about 30 sec). After time out check that SIPset-2 is ringing and answer the call. Check the audio and display. Dial 41 (Forward cancellation prefix) on OXEset-1 for forward cancellation.	☐	☒	☐	
C	**Call to local user, forward on busy (CFB). (1)** On OXEset-1 dial the 54 (Forward on busy prefix). After the voice guide, enter SIPset-2 (<target MCDU number>) to activate the CFB. Wait for acknowledgement from OXE. With SIPset-2 call OXEset-1 and answer the call. With SIPset-3 call OXEset-1 and answer the call to make it busy. With SIPset-1 call OXEset-1. Check that SIPset-2 is ringing and answer the call. Check the audio and display. Dial 41 (Forward cancellation prefix) on OXEset-1 for forward cancellation.	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
6	OXE features (Call back, voice mail deposit)				
A	**Call Back on free set** From SIPset-1 call OXEset-1 Dial "5" (Call Back suffix) while OXEset-1 is ringing and release the call. Activate the call back from OXEset-1. Check that SIPset-1 is ringing, answer the call and check audio + display.	☐	☐	☒	Dialpad is not available on the Polycom device while outgoing call is ringing.
B	**Voice mail deposit** From SIPset-1 call OXEset-1 Dial "6" (Voice Mail deposit suffix) while OXEset-1 is ringing. Leave a message when connected to the voice mail and release the call. Check the voice message on OXEset-1.	☐	☐	☒	Dialpad is not available on the Polycom device while outgoing call is ringing.

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.

8.5 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.
OXE SEPLOS prefixes are mandatory for several tests of this section. For more information refer to the appendix E.

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	Local/Network calls				
A	**Local /network call to free SIP terminal** Local: with OXEset-1 call SIPset-1. Check that SIPset-1 is ringing and answer the call Check ring back tone, audio and called party display. PSTN: with OXEset-1 call SIPset-1 by dialing 72SIPset-1 (prefix to take the T2 loopback+ <target MCDU number>). Check that SIPset-1 is ringing and answer the call. Check ring back tone, audio and called party display.	☐	☒	☐	
B	**Local/network call to busy SIP terminal** Local: With SIPset-2 call SIPset-1 and answer the call to make it partially busy, don't hang up. With SIPset-3 call SIPset-1 and answer the call to make it fully busy, don't hang up. With OXEset-1 call SIPset-1 which is busy Check the ring back tone and display	☐	☒	☐	
C	**Local/network call to SIP terminal in Do Not Disturb mode (DND), by local feature if applicable** Local: Enable DND on SIPset-1 and call it with OXEset-1. Check the ring back tone and display Cancel the DND on SIPset-1.	☐	☒	☐	
D	**Local/network call to SIP terminal in Do Not Disturb mode (DND), by system feature, secret code sent by DTMF (SEPLOS)** Local: Enable DND on SIPset-1 using 42 prefix (Do not disturb prefix). After the voice guide, enter 0000 (<sip set password>). Wait for acknowledgement from OXE. With OXEset-1 call SIPset-1 Check the ring back tone and display Cancel the DND on SIPset-1 using 42 (Do not disturb prefix).	☐	☐	☒	Dialpad is not displayed on the Polycom device after dialing 42 prefix.

Test Case Id	Test Case	N/A	OK	NOK	Comment
E	**Local call to SIP terminal in Do Not Disturb mode (DND), by system feature, prefix and secret code sent in the INVITE message (SEPLOS)** Local: Enable DND on SIPset-1 using 420000 (Do not disturb prefix + <sip set password>). for acknowledgement from OXE. With OXEset-1 call SIPset-1 Check the ring back tone and display Cancel the DND on SIPset-1 using 42 (Do not disturb prefix).	☐	☒	☐	
F	**SIP session timer expiration** Check if call is maintained after the session timer expiration: Configure the "Session timer" on OXE to 120 seconds (3). With OXEset-1 call SIPset-1. Answer the call on SIPset-1and never hang up, wait for session timer expiration. Check that call is maintained. Configure the "Session timer" on OXE to the default value : 1800 seconds (1).	☐	☒	☐	
G	**External call to SIP terminal.** Check that external call back number is shown correctly: With OXEset-1 dial 72SIPset-1 ("prefix to take T2 loop"+target MCDU number>) Check that SIPset-1 is ringing and the external call number is shown correctly Answer the call and check audio, display and call release.	☐	☒	☐	
2	Forward (Immediate/On no reply/On busy)				
A	**Local/network/SIP call to SIP terminal in immediate forward (CFU) to local user, by local feature if applicable** Local: On SIPset-1 enable CFU to OXEset-1 With SIPset-2 call SIPset-1. Check that OXEset-1 is ringing. Answer the call and check audio and display. Disable CFU on SIPset-1.	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
B	**Local/network/SIP call to SIP terminal in immediate forward (CFU) to local user, by system feature (SEPLOS)** Local: On SIPset-1 enable CFU to OXEset-1 using 51 prefix (Immediate forward prefix). After the voice guide, enter OXEset-1 (<target MCDU number>). Wait for acknowledgement from OXE. With SIPset-2 call SIPset-1. Check that OXEset-1 is ringing. Answer the call and check audio and display. Disable CFU on SIPset-1 using 41 (Forward cancellation prefix) prefix.	☐	☒	☐	
C	**Local/network/SIP call to SIP terminal in immediate forward (CFU) to a SIP user, by local feature if applicable** Local: On SIPset-2 enable CFU to SIPset-3 With SIPset-1 call SIPset-2. Check that SIPset-3 is ringing. Answer the call and check audio and display. Disable CFU on SIPset-2.	☐	☒	☐	
D	**Local/network/SIP call to SIP terminal in immediate forward (CFU) to a SIP user, by system feature (SEPLOS)** Local: On SIPset-2 enable CFU to SIPset-3 using 51 prefix (Immediate forward prefix). After the voice guide, enterSIPset-3 (<target MCDU number>). Wait for acknowledgement from OXE. With SIPset-1 call SIPset-2. Check that SIPset-3 is ringing. Answer the call and check audio and display. Disable CFU on SIPset-2 using 41 (Forward cancellation prefix).	☐	☒	☐	
E	**Local call to SIP terminal in "forward on busy" (CFB) state, by local feature if applicable** On SIPset-2 enable CFB to OXEset-1 With SIPset-2 call SIPset-3. With SIPset-2 call SIPset-4 to make it busy. With SIPset-1 call SIPset-2 which is busy. Check that OXEset-1 is ringing Answer the call and check audio and display. Disable CFU on SIPset-2.	☐	☐	☒	Outgoing call from SIPset-1 is not reaching OXEset-1. SIPset-1 plays the busy tone.

Test Case Id	Test Case	N/A	OK	NOK	Comment
F	**Local call to SIP terminal in "forward on busy" (CFB) state, by system feature (SEPLOS)** On SIPset-2 enable CFB to OXEset-1 using 52 prefix (Forward prefix on busy). After the voice guide, enter OXEset-1 (<target MCDU number>). Wait for acknowledgement from OXE. With SIPset-2 call SIPset-3. With SIPset-2 call SIPset-4 to make it busy. With SIPset-1 call SIPset-2 which is busy. Check that OXEset-1 is ringing Answer the call and check audio and display. Disable CFB on SIPset-2 using 41 (Forward cancellation prefix).	☐	☒	☐	
G	**Local call to SIP terminal in "forward on no reply" (CFNR), by local feature if applicable** On SIPset-2 enable CFNR to OXEset-1 With SIPset-1 call SIPset-2. Check that SIPset-2 is ringing and don't answer the call, wait for time out (about 30 seconds). After time out expiration check that SIPset-2 stops ringing and that the call is not displayed anymore. Check the OXEset-1 is ringing, answer the call and check audio and display. Disable CFNR on SIPset-2.	☐	☒	☐	
H	**Local call to SIP terminal in "forward on no reply" (CFNR), by system feature (SEPLOS)** On SIPset-2 enable CFNR to OXEset-1 using 53 prefix (Forward prefix on no reply). After the voice guide, enter OXEset-1 (<target MCDU number>). Wait for acknowledgement from OXE. With SIPset-1 call SIPset-2. Check that SIPset-2 is ringing and don't answer the call, wait for time out (about 30 seconds). After time out expiration the OXEset-1 is ringing, answer the call and check audio and display. Disable CFNR on SIPset-2 using 41 (Forward cancellation prefix).	☐	☒	☐	
3	Call release				

Test Case Id	Test Case	N/A	OK	NOK	Comment
A	**Call release during an incoming call, release done by the partner SIP set** With OXEset-1 call SIPset-1 and don't answer the call. With SIPset-1, reject the call during the ringing period. Check that OXEset-1 plays a release tone and goes in idle mode after some seconds	☐	☒	☐	
B	**Call release during an incoming call, release done by the OXE set** With OXEset-1 call SIPset-1 and don't answer the call. With OXEset-1, release the call during the ringing period. Check that SIPset-1 plays a release tone and goes in idle mode after some seconds	☐	☒	☐	
4	Calling name presentation				
A	**Calling Line Identity Restriction (CLIR): Local call to SIP terminal.** With OXEset-1 call SIPset-1 by dialing 409SIPset-1 (Secret identity prefix + <target MCDU number>) in order to hide OXEset-1 identity. Check that SIPset-1 is ringing, answer the call . Check that OXEset-1 identity is hidden on both ringing and conversation periods.	☐	☒	☐	
B	**Display: Call to free SIP terminal from user with a name containing non-ASCII characters.** Check caller display. With SIPset-2 call SIPset-1 (extension with a name containing non-ASCII characters). Check that SIPset-1 is ringing and check on its display the name "SIPset-2 éêêèè" is displayed. Check that non-ASCII characters (éêêèè) are correctly displayed.	☐	☒	☐	
C	**Display: Call to free SIP terminal from user with a UTF-8 name containing non-ASCII characters.** Check caller display. With SIPset-7 call SIPset-1 (extension with a name containing UTF-8 characters). Check that SIPset-1 is ringing and check on its display the name: "SIPset-7_&@#" is displayed. Check that UTF-8 characters (&@#) are correctly displayed.	☐	☒	☐	
5	Twin set				

Test Case Id	Test Case	N/A	OK	NOK	Comment
A	**SIP set is declared as a twin set (tandem).** Call to main set and see if twin set rings. Take call with twin set. With SIPset-1 call OXEset-2 which is in tandem with SIPset-4. Check that OXEset-2 and SIPset-4 are both ringing. Answer the call from OXEset-2 and check that SIPset-4 stop ringing. Check audio and display.	☐	☒	☐	SIPset-2 displays a missed call.

Notes:

(1) The SIP "Session timer" is configured via mgr under : SIP/SIP Gateway/ Session (value is in seconds)

8.6 Features during Conversation

Features during conversation between OXE 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.
OXE SEPLOS prefixes are mandatory for several tests of this section. For more information refer to the appendix E.

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	Hold/resume				
A	**Hold and resume in case of a single call (by local feature if applicable)** With SIPset-4 call OXEset-1 Answer the call, check audio and display. With SIPset-4 put OXEset-1 on hold with "Hold" key, check tones and display on both sets, then press again "Hold" key to resume the call (applicable if Hold Key is provided by the SIP set) On OXEset-1 put SIPset-4 on hold then resume.	☐	☒	☐	
B	**Hold and resume in case of a two calls** SIPset-4 (which is multi-lines) is in conversation with an external PSTN user. From SIPset-4 , call SIPset-1 and answers the call. Check that External PSTN user is put on Hold + tones and display. Check audio and display on SIPset-4 SIPset-1 On SIPset-4, toggle between the external PSTN user line and SIPset-1 line (via line key) Check hold tone, audio and display on the sets. Release the call from SIPset-4 and switch to the first line. Check that SIPset-1 and SIPset-4 are in conversation + display	☐	☒	☐	
2	Call release				
A	**Call release during conversation, release done by the partner SIP set** With SIPset-1 call OXEset-1 Answer the call, check audio and display. With SIPset-1, release the call. Check that OXEset-1 plays a release tone and goes in idle mode after some seconds	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
B	**Call release during conversation, release done by the OXE set** With SIPset-1 call OXEset-1 Answer the call, check audio and display. With OXEset-1, release the call. Check that SIPset-1 plays a release tone and goes in idle mode after some seconds	☐	☒	☐	
3	DTMF sending				
A	**Sending DTMF** Configure SIPset-1 to send DTMF using RFC 2833 From SIPset-1 call the node 1 voice mail directory number and try to navigate in its menu listed by the voice guide. Check that you can navigate in the menus.	☐	☒	☐	
4	Conference				
A	**Meet Me conference** With SIPset-1 call 509 (Meet me conference prefix) After the voice guide, enter SIPset-1 (<target extension>). After the voice guide, enter 0000 (conference password), don't release this call. With OXEset-1 call 509 (Meet me conference prefix) After the voice guide, enter SIPset-1 (<target extension>). After the voice guide, enter 0000 (conference password), don't release this call. Check that OXEset-1 and SIPset-1 are in conference. With SIPset-1 call 509 (Meet me conference prefix) After the voice guide, enter SIPset-1 (<target extension>). After the voice guide, enter 0000 (conference password), don't release this call. Check that OXEset-1, SIPset-1 and SIPset-2 are in conference. Check audio and display. Release the conference from OXEset-1. Check that SIPset-1 and SIPset-2 are in conference.	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
B	**Three party conference initiated from OXE set** With OXEset-1 call SIPset-1, answer the call and don't release it. With OXEset-1 call SIPset-2, answer the call and don't release it too. With OXEset-1, the conference bridge, start a conference. Check that OXEset-1, SIPset-1 and SIPset-2 are in conference. Check audio and display. Release the conference from OXEset-1. Check that SIPset-1 and SIPset-2 are in conversation.	☐	☒	☐	
C	**Three party conference initiated from SIP set (applicable to a SIP multi-line set only)** With SIPset-1 call an external PSTN user, answer the call and don't release it. With SIPset-1 call SIPset-2, answer the call and don't release it too. With SIPset-1, the conference bridge, start a conference by the local feature. Check that an external PSTN user, SIPset-1 and SIPset-2 are in conference. Check audio and display. Release the conference from SIPset-1. Check that the external PSTN user and SIPset-2 are in conversation. With SIPset-2 call SIPset-1, answer the call and don't release it. With SIPset-2, the conference bridge, start a conference by the local feature. Check that an external PSTN user, SIPset-1 and SIPset-2 are in conference. Check audio and display. Release the conference from SIPset-1. Check that the external PSTN user and SIPset-2 are in conversation.	☐	☒	☐	

8.7 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 kinds of transfers in the SIP world:

- ***Unattended Transfer*** or *Basic 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.
  Note: Unattended Transfer is not provided by the OXE, but might by supported by the SIP set.

- ***Semi-Attended Transfer*** or *Early Attended Transfer* or *Transfer on ringing*:
  1. A (Transferee) calls B (Transferor). A and B in conversation.
  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). A and B in conversation.
  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.

In the below tables, *SIP* means a partner SIP set, *OXE* means a proprietary OXE (Z/UA/IP) set, *Ext. Call* means an External Call, PSTN for example.

8.7.1 Unattended transfers

In VVX phones when there is an incoming call (in ringing state) we can see a forward button. On selecting the forward button we are able to enter the extension to which the call has to be transferred.

Test Case Id	Test Case						N/A	OK	NOK	Comment
	A		B		C					
	Transferee		Transferor		Transfer Target					
	Type of set (2)	MCDU Number	Type of set (2)	MCDU Number	Type of set (2)	MCDU Number				
1	OXE/Ext. Call	OXEset-1	SIP	SIPset-1	OXE/Ext. Call	OXEset-3	☐	☒	☐	
2	SIP	SIPset-1	OXE	OXEset-1	OXE/Ext. Call	OXEset-3	☒	☐	☐	Not available from an OXE set
3	OXE/Ext. Call	OXEset-1	OXE	OXEset-3	SIP	SIPset-1	☒	☐	☐	Not available from an OXE set
4	SIP	SIPset-1	OXE	OXEset-1	SIP	SIPset-2	☒	☐	☐	Not available from an OXE set
5	OXE/Ext. Call	OXEset-1	SIP	SIPset-1	SIP	SIPset-2	☐	☒	☐	
6	SIP	SIPset-1	SIP	SIPset-2	OXE/Ext. Call	OXEset-1	☐	☒	☐	
7	SIP	SIPset-1	SIP	SIPset-2	SIP	SIPset-3	☐	☒	☐	

8.7.2 Semi-Attended Transfer (on Ringing)

After answering the first call we can see a transfer button in the display of the phone. On selecting the transfer button we are able to enter extension to which the call has to be made.

When the desired extension is ringing we need to press the transfer button again to make successful semi attended transfer.

Test Case Id	Test Case						N/A	OK	NOK	Comment
	A Transferee		**B Transferor**		**C Transfer Target**					
	Type of set (2)	**MCDU Number**	**Type of set (2)**	**MCDU Number**	**Type of set (2)**	**MCDU Number**				
1	OXE/Ext. Call	OXEset-1	SIP	SIPset-1	OXE/Ext. Call	OXEset-3	☐	☒	☐	
2	SIP	SIPset-1	OXE	OXEset-1	OXE/Ext. Call	OXEset-3	☐	☒	☐	
3	OXE/Ext. Call	OXEset-1	OXE	OXEset-3	SIP	SIPset-1	☐	☒	☐	
4	SIP	SIPset-1	OXE	OXEset-1	SIP	SIPset-2	☐	☒	☐	
5	OXE/Ext. Call	OXEset-1	SIP	SIPset-1	SIP	SIPset-2	☐	☒	☐	
6	SIP	SIPset-1	SIP	SIPset-2	OXE/Ext. Call	OXEset-1	☐	☒	☐	
7	SIP	SIPset-1	SIP	SIPset-2	SIP	SIPset-3	☐	☒	☐	

8.7.3 Attended Transfer (in Conversation)

After answering the first call we can see a transfer button in the display of the phone. On selecting the transfer button we are able to enter extension to which the call has to be made.

When the desired extension is ringing we need to answer the call. After the call is answered in the VVX phone we can see two calls and on pressing the transfer button we can make successful attended transfer.

Test Case Id	Test Case						N/A	OK	NOK	Comment
	A Transferee		**B Transferor**		**C Transfer Target**					
	Type of set (2)	**MCDU Number**	**Type of set (2)**	**MCDU Number**	**Type of set (2)**	**MCDU Number**				
1	OXE/Ext. Call	OXEset-1	SIP	SIPset-1	OXE/Ext. Call	OXEset-3	☐	☒	☐	
2	SIP	SIPset-1	OXE	OXEset-1	OXE/Ext. Call	OXEset-3	☐	☒	☐	
3	OXE/Ext. Call	OXEset-1	OXE	OXEset-3	SIP	SIPset-1	☐	☒	☐	
4	SIP	SIPset-1	OXE	OXEset-1	SIP	SIPset-2	☐	☒	☐	
5	OXE/Ext. Call	OXEset-1	SIP	SIPset-1	SIP	SIPset-2	☐	☒	☐	
6	SIP	SIPset-1	SIP	SIPset-2	OXE/Ext. Call	OXEset-1	☐	☒	☐	
7	SIP	SIPset-1	SIP	SIPset-2	SIP	SIPset-3	☐	☒	☐	

8.8 Attendant

An attendant console (attendant set type : 4068) 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	Call to/from an attendant station				
A	**SIP set call to attendant** With SIPset-1, call attendant with prefix 9 (attendant call prefix), attendant answers. Check ringing back tone, display and audio.	☐	☒	☐	
B	**Attendant call to SIP set** With the attendant station, call SIPset-1, SIPset-1 answers. Check ringing back tone, display and audio.	☐	☒	☐	
C	**Outgoing call to SIP set while in conversation with attendant.** SIPset-1 being in conversation with the attendant. With SIPset-1 try to call SIPset-2, check that this call is not allowed. SIPset-1 should not be able to put the attendant on hold. Check that the SIPset-2 does not ring. Check that SIPset-1 stays in conversation with the attendant station (SIPset-1 can display an error message when it tries to put the attendant station on hold).	☐	☒	☐	Attendant is put on hold by SIPset-1. But call can be retrieved by pressing SIPset-1 “Resume” button.
E	**Outgoing call to SIP set in “Do not Disturb” (DND) state, by local feature if applicable** Enable DND on SIPset-1 and call it with the attendant station. Check that the call is not allowed, the ring back tone and display Cancel the DND on SIPset-1.	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
F	**Outgoing call to SIP set in "Do not Disturb" (DND) state, by system feature** Enable DND on SIPset-1 using 42 prefix (Do not disturb prefix). After the voice guide, enter 0000 (<sip set password>). Wait for acknowledgement from OXE. Call SIPset-1 with the attendant station. Check that the attendant displays that SIPset-1 is in "Do not disturb" state. Check that the attendant can override it by choosing to ring the SIPset-1 (by pressing attendant "Ring" softkey). Cancel the DND on SIPset-1 using 42 prefix (Do not disturb prefix).	☐	☒	☐	
2	Call transfer from an attendant station				
A	**SIP set call to attendant, attendant transfers to OXE set, semi-attended** With SIPset-1, call attendant with prefix 9 (attendant call prefix), attendant answers. From the attendant, call OXEset-1 and transfer semi-attended. Answer the call and check audio and display.	☐	☒	☐	
B	**SIP set call to attendant, attendant transfers to OXE set, attended** With SIPset-1, call attendant with prefix 9 (attendant call prefix), attendant answers. From the attendant, call OXEset-1 and transfer attended. Check audio and display.	☐	☒	☐	
C	**OXE set calls to attendant, attendant transfers to SIP set, semi-attended** With OXEset-1, call attendant with prefix 9 (attendant call prefix), attendant answers. From the attendant, call SIPset-1 and transfer semi-attended. Answer the call and check audio and display.	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
D	**OXE set calls to attendant (using attendant call prefix), attendant transfers to SIP set, attended** With OXEset-1, call attendant with prefix 9 (attendant call prefix), attendant answers. From the attendant, call SIPset-1 and transfer attended. Check audio and display.	☐	☒	☐	
E	**OXE set calls to attendant (using attendant call prefix), attendant transfers (semi attended transfer) to SIP set in "Do not Disturb" (DND) state (DND activated by local feature)** Enable DND on SIPset-1. With OXEset-1, call attendant with prefix 9 (attendant call prefix), attendant answers. From the attendant, call SIPset-1 and transfer semi-attended. Check that the transfer is not allowed Cancel the DND on SIPset-1.	☐	☒	☐	
F	**OXE set calls to attendant (using attendant call prefix), attendant transfers (semi attended transfer) to SIP set in "Do not Disturb" (DND) state (DND activated by system feature)** Enable DND on SIPset-1 using 42 prefix (Do not disturb prefix). After the voice guide, enter 0000 (<sip set password>). Wait for acknowledgement from OXE. With OXEset-1, call attendant with prefix 9 (attendant call prefix), attendant answers. From the attendant, call SIPset-1 and transfer semi-attended. Check that the attendant displays the SIPset-1 "Do not disturb" state. Check that the attendant can override it by pressing "Ring" softkey. Cancel the DND on SIPset-1 using 42 prefix (Do not disturb prefix).	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
G	**OXE set calls to attendant (using attendant call prefix), attendant transfers (attended transfer) to SIP set in "Do not Disturb" (DND) state (DND activated by local feature)** Enable DND on SIPset-1. With OXEset-1, call attendant with prefix 9 (attendant call prefix), attendant answers. From the attendant, call SIPset-1 and transfer attended. Check that the transfer is not allowed Cancel the DND on SIPset-1.	☐	☒	☐	
H	**OXE set calls to attendant (using attendant call prefix), attendant transfers (attended transfer) to SIP set in "Do not Disturb" (DND) state (DND activated by system feature)** Enable DND on SIPset-1 using 42 prefix (Do not disturb prefix). After the voice guide, enter 0000 (<sip set password>). Wait for acknowledgement from OXE. With OXEset-1, call attendant with prefix 9 (attendant call prefix), attendant answers. From the attendant, call SIPset-1 and transfer attended. Check that the attendant displays the SIPset-1 "Do not disturb" state. Check that the attendant can override it by pressing "Ring" softkey. And than, after SIPset-1 answered, can transfer the call. Check that the transfer is not allowed Cancel the DND on SIPset-1 using 42 prefix (Do not disturb prefix).	☐	☒	☐	

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8.9 Voice Mail

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

A voice mailbox is available for users SIPset-1, SIPset-2 and OXEset-1. The passwords and the voice mail directory number are given in section "3.1 Hardware Configuration".

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

Note 1: explicit subscription is required for RFC3842 MWI (ref 2.8.24.7 SIP Endpoint developers guide)

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	Message waiting indicator				
A	**Message display activation, MWI (1)** With SIPset-2 call the voice mail directory number. Follow the instructions in order to send a voice message in SIPset-1 box. Check that the MWI on SIPset-1 is activated.	☐	☒	☐	
2	Message listening				
A	**Message consultation** With SIPset-1 call the voice mail directory number. 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 SIPset-1 after message cancellation.	☐	☒	☐	
3	Voice mail management				
A	**Password modification** With SIPset-1 call the voice mail directory number 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.	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
4	Forward to voice mail				
A	**SIP call to a OXE user forwarded to Voice Mail** Forward the OXEset-1 to Voice Mail by dialing 51 prefix (Immediate forward prefix). After the voice guide, enter 31300 (<Voice Mail number>). With SIPset-1 call OXEset-1 and check that you are immediately forwarded to Voice Mail. Check that you can leave a message On OXEset-1 disable Voice Mail forwarding with 41 (Forward cancellation prefix).	☐	☒	☐	
B	**OXE call to a SIP user forwarded to Voice Mail** Forward the SIPset-1 to Voice Mail by dialing 51 prefix (Immediate forward prefix). After the voice guide, enter 31300 (<Voice Mail number>). With OXEset-1 call SIPset-1 and check that you are immediately forwarded to Voice Mail. Check that you can leave a message On SIPset-1 disable Voice Mail forwarding with 41 (Forward cancellation prefix).	☐	☒	☐	

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

9 Appendix A : AAPP member's Application description

Trio 8800

https://support.polycom.com/content/support/North_America/USA/en/support/voice/polycom-trio.html

http://www.polycom.com/content/dam/polycom/common/documents/data-sheets/realpresence-trio-data-sheet-enus.pdf

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

The Polycom equipments are configured through their web admin: https://<IP address>/index.htm
For example: https://10.1.18.154/index.htm

Default admin password is: 456

SIP configuration

SIP

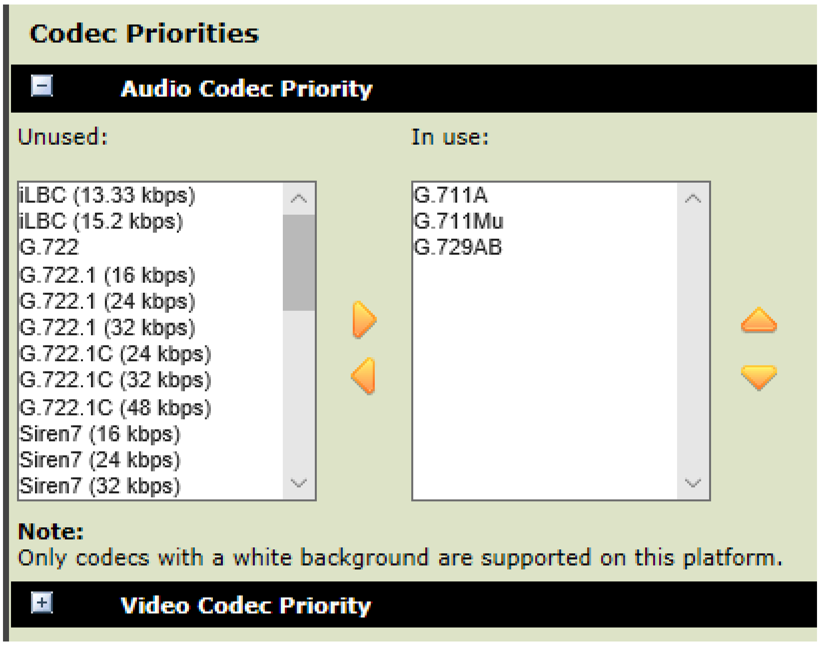
+ Local Settings	
+ Outbound Proxy	
− Server 1	
Special Interop	Standard
Address	10.1.2.1
Port	0
Transport	DNSnaptr
Expires (s)	180
Subscription Expires (s)	180
Register	(•) Yes ( ) No
Retry Timeout (ms)	0
Retry Maximum Count	3
Line Seize Timeout (s)	30
Re-Register on Failover	( ) Yes (•) No
Fail Back Mode	registration
Fail Back Timeout	60
− Server 2	
Special Interop	Standard
Address	10.10.10.1
Port	0
Transport	DNSnaptr
Expires (s)	180
Subscription Expires (s)	180
Register	(•) Yes ( ) No
Retry Timeout (ms)	0
Retry Maximum Count	3
Line Seize Timeout (s)	30
Re-Register on Failover	( ) Yes (•) No
Fail Back Mode	registration
Fail Back Timeout	60
+ Server 3	

Lines

Line 1

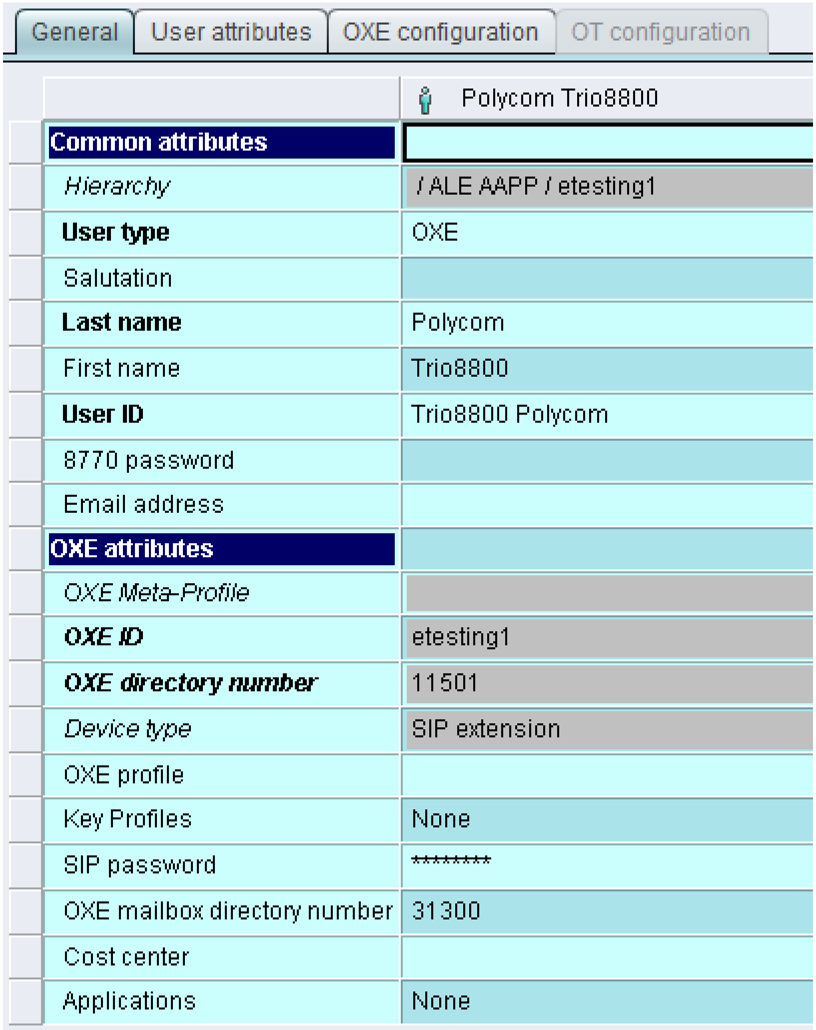
− Identification	
Display Name	11501
Address	11501
Label	11501
Type	(•) Private ( ) Shared
Third Party Name	
Number of Line Keys	1
Calls Per Line	24
Enable SRTP	(•) Yes ( ) No
Offer SRTP	( ) Yes (•) No
Require SRTP	( ) Yes (•) No
Server Auto Discovery	(•) Enable ( ) Disable
TCP Fast Failover	( ) Enable (•) Disable
+ Authentication	
+ Outbound Proxy	
− Server 1	
Special Interop	Standard
Address	
Port	0
Transport	DNSnaptr
Expires (s)	180
Subscription Expires (s)	180
Register	(•) Yes ( ) No
Retry Timeout (ms)	0
Retry Maximum Count	3
Line Seize Timeout (s)	30
Re-Register on Failover	( ) Yes (•) No
Fail Back Mode	registration
Fail Back Timeout	180
− Server 2	
Special Interop	Standard

Codecs


Codec Priorities
Audio Codec Priority
Unused:
In use:
iLBC (13.33 kbps)
iLBC (15.2 kbps)
G.722
G.722.1 (16 kbps)
G.722.1 (24 kbps)
G.722.1 (32 kbps)
G.722.1C (24 kbps)
G.722.1C (32 kbps)
G.722.1C (48 kbps)
Siren7 (16 kbps)
Siren7 (24 kbps)
Siren7 (32 kbps)
G.711A
G.711Mu
G.729AB
Note:
Only codecs with a white background are supported on this platform.
Video Codec Priority

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

The Polycom equipments are configured as SIP extension users :



12 Appendix D: AAPP member's escalation process

Polycom has Service Programs available that provide Customers with technical telephone support, advance parts replacement, software upgrades & updates, and access to Polycom's enhanced support portal.

These programs are available worldwide and are available through Polycom Sales.

13 Appendix E: AAPP program

13.1 Alcatel-Lucent Application Partner Program (AAPP)

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

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

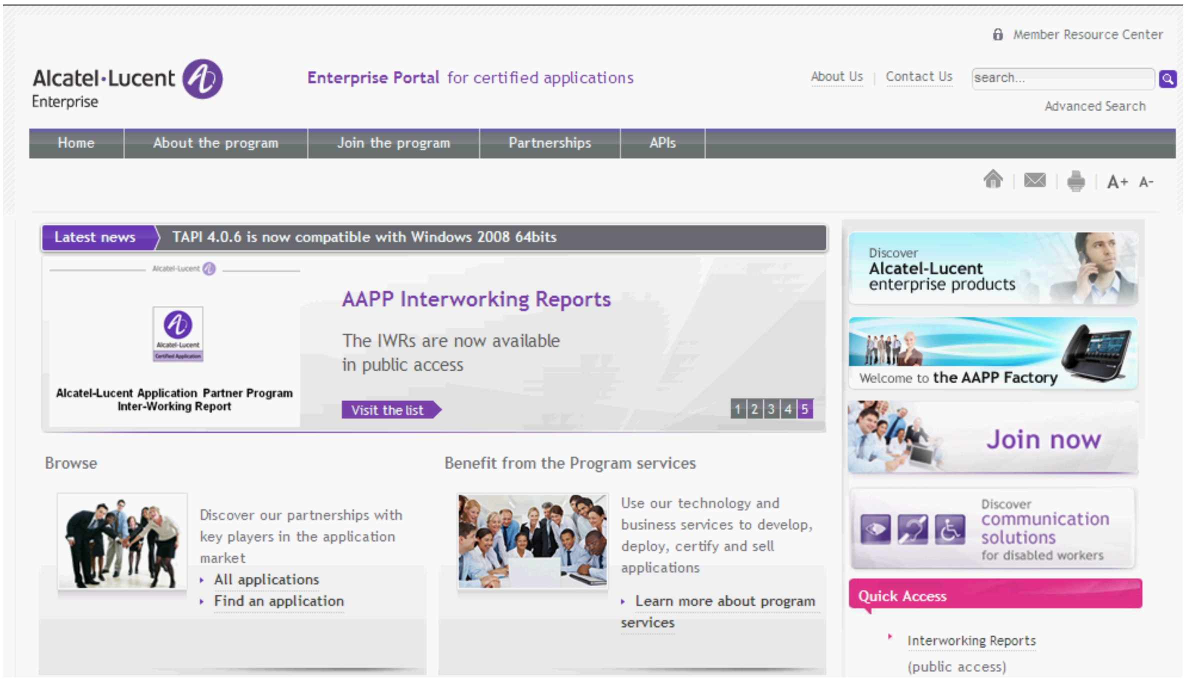
- **Provide easy interfacing for Alcatel-Lucent Enterprise communication products**: Alcatel-Lucent Enterprise's communication products for the enterprise market include infrastructure elements, platforms and software suites. To ensure easy integration, the AAPP provides a full array of standards-based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

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

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

Web site

The Application Partner Portal is a website dedicated to the AAPP program and where the InterWorking Reports can be consulted. Its access is free at http://applicationpartner.alcatel-lucent.com



13.2 Enterprise.Alcatel-Lucent.com

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

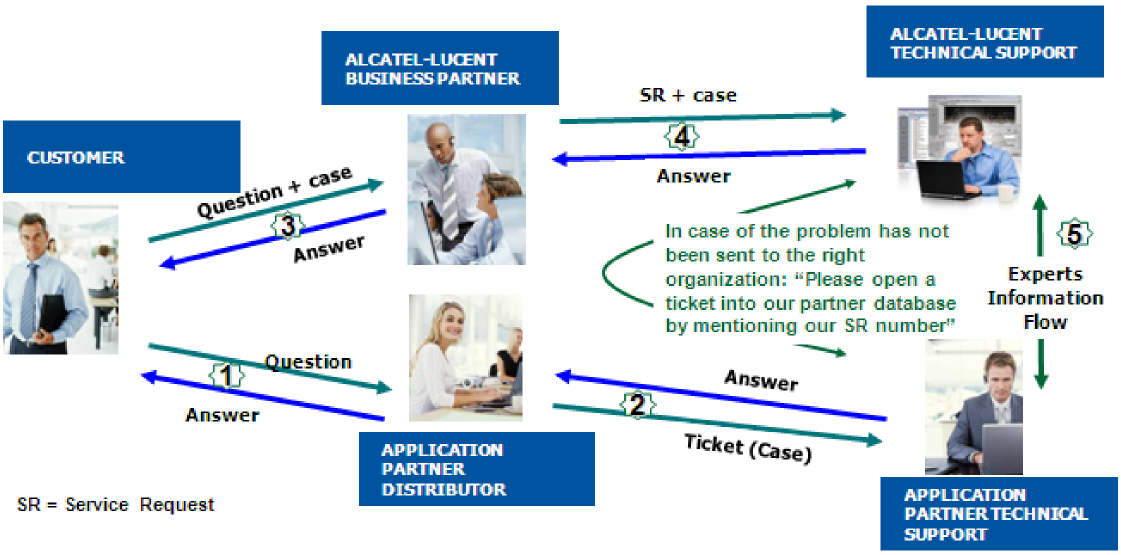
14 Appendix F: AAPP Escalation process

14.1 Introduction

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

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

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



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

14.2 Escalation in case of a valid Inter-Working Report

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

This defines the scope of what has been certified.

If the issue is in the scope of the IWR, both parties, ALE International and the Application Partner, are engaged:

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

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

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

- The Application Partner shall be contacted first by the Business Partner (responsible for the application, see figure in previous page) for an analysis of the problem.
- The ALE International Business Partner will escalate the problem to the ALE International Support Center only if the Application Partner has demonstrated with traces a problem on the ALE International side or if the Application Partner (not the Business Partner) needs the involvement of ALE International

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

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

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

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

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

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

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

14.3 Escalation in all other cases

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

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

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

14.4 Technical support access

The ALE International **Support Center** is open 24 hours a day; 7 days a week:

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

ALE International Business Partners Support Center for countries:

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

For other countries:

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

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