ALE Application Partner Program
Inter-Working Report

Partner: **CyberData**
Application type: **SIP Phone**
Application name: **SIP Talk-Back speaker**
*Alcatel-Lucent Enterprise Platform***:**
OmniPCX Enterprise™

Certification overview

Date of the certification	June 2017

ALE representative	Frank Gadot
AAPP member representative	Paul Tuttle pttuttle@cyberdata.net

Alcatel-Lucent Enterprise Communication Platform	OmniPCX Enterprise
Alcatel-Lucent Enterprise Communication Platform release	R11.2.2 (L2.300.32.a)
AAPP member application release	V11.6.8b07
Application Category	Terminals

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Revision History

Edition 1: creation of the document – *May 2017*

Test results

☑ Passed ☐ Refused ☐ Postponed

☐ Passed with restrictions

Refer to the section 4 for a summary of the test results.

IWR validity extension

The validity of this IWR has been extended to the following software releases/products:

- OXE R12

Alcatel·Lucent Enterprise

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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 (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.

1.1 Glossary

AAPP	Alcatel-Lucent Application Partner Program	NTP	Network Time Protocol
CFB	Call Forward on Busy	OXE	OmniPCX Enterprise
CFNR	Call Forwarding on No Reply	PBX	private branch exchange
CFU	Call Forwarding Unconditional	PCS	Passive Communication Server
CLIR	Calling Line Identification Restriction	PSTN	Public Switched Telephone Network
DHCP	Dynamic Host Configuration Protocol	RTP	Real-time Transport Protocol
DND	Do Not Disturb	SEPLOS	SIP EndPoint Level Of Service
DNS	Domain Name System	SIP	Session Initiation Protocol
DTMF	Dual-tone multi-frequency	TCP	Transmission Control Protocol
IP	Internet Protocol	UDP	User Datagram Protocol
IWR	Inter-Working Report	VAD	Voice activity detection
MCDU	Multi-Purpose Control and Display Unit	VoIP	Voice over IP
MWI	Message Waiting Indicator		

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 (3^{rd} 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 Summary of test results

4.1 Summary of the main features tested

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

Feature	N/A	OK	OK But	NOK
Connectivity and Setup				
IP network connectivity	☐	☒	☐	☐
SIP Registration	☐	☒	☐	☐
Duplication and Robustness				
Erreur ! Source du renvoi introuvable.	☒	☐	☐	☐
Erreur ! Source du renvoi introuvable.	☒	☐	☐	☐
Erreur ! Source du renvoi introuvable.	☒	☐	☐	☐
CPU Redundancy (No special redundancy)	☐	☒	☐	☐
Audio codecs negotiations/ VAD / Framing				
G 711 A, G 711 µ support (Uncompressed codec)	☐	☒	☐	☐
Outgoing Calls				
Local/Network calls	☐	☒	☐	☐
Call to a forwarded user	☐	☒	☐	☐
OXE features (Call back, voice mail deposit)	☒	☐	☐	☐
Incoming Calls				
Local/Network calls	☐	☒	☐	☐
Features during Conversation				
Hold/resume	☐	☒	☐	☐
DTMF sending	☐	☒	☐	☐

4.2 Summary of problems

None.

4.3 Summary of limitations

CyberData speakerphone only supports G.711 and G.722 codecs.

4.4 Notes, remarks

SIP Session Timer:

The SIP Session Timer (RFC4028) periodically refreshes a SIP session by sending either a repeated INVITE request (RE_INVITE) or an UPDATE request. In one word, it is a kind of keep alive mechanism which allows proxies to retain a call state when needed.

CyberData speakerphone only supports RE_INVITE as a session timer method.
The default parameter within the OXE is to send an UPDATE message.

This option can be changed in the SIP GATEWAY settings (SESSION TIMER METHOD). Be careful, this setting will affect all SIP devices in your network.

IMPORTANT:

- Some interworking problems were found in previous releases of CyberData. Therefore, it is important to use at least the version mentioned in this document.
- It is important to know that if the Cyberdata speaker is setup with a button, and if it calls an extension with no Associated number, the call will stay in a setup mode (keep ringing)

5 Application information

Application family : SIP overhead speakerphone

Application commercial name: SIP Talk-back speaker

Application version: v11.6.8b07

Interface type: SIP

Brief application description:

CyberData's new SIP Talk-Back Speaker enables two-way conversations in settings such as classrooms, offices, medical facilities and clinics. By use of remote call button (sold separately, part# 011185), calls to a predetermined extension can be initiated from the room with the speaker. During the active calls, the LED light on the switch can be programmed to blink to show call activity. Alternatively, a call can be placed to the speaker to initiate either a page or two-way conversation.

Type of application/product:

6 Test environment

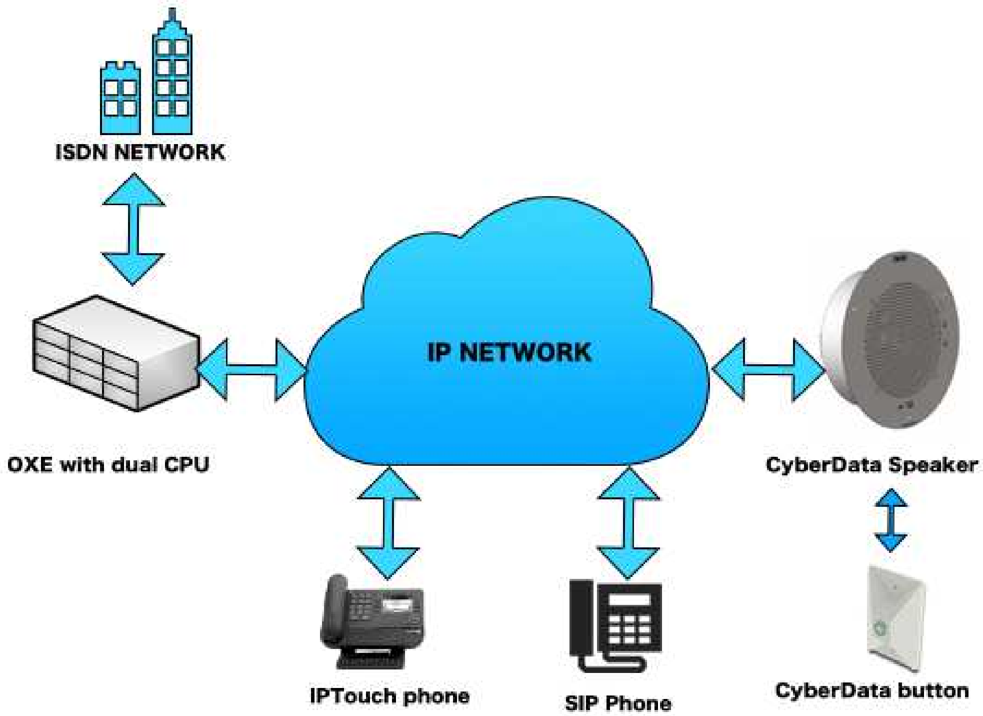


Figure 1 **Test environment**

6.1 Hardware configuration

- **Alcatel-Lucent Communication Platform**:

Node1:
  - o Duplicated call servers, no spatial redundancy

Setup Details

```
+------------------------------------------------------------------+
| Cr | cpl| cpl type   | hw type   | cpl  state  | coupler ID     |
|----|----|------------|-----------|-------------|----------------|
| 0 |  6 | App. Server|---------- |  IN SERVICE |   NO PCMS CODE |
| 0 | 10 | App. Server|---------- |  IN SERVICE |   NO PCMS CODE |
+------------------------------------------------------------------+
```

```
+------------------------------------------------------------------+
| Cr | cpl| cpl type   | hw type   | cpl  state  | coupler ID     |
|----|----|------------|-----------|-------------|----------------|
| 2 |  0 |        GD3|---------- |  IN SERVICE |  NO PCMS CODE |
| 2 |  3 |     PRA T1|---------- |  IN SERVICE |  NO PCMS CODE |
| 2 |  6 |     PRA T1|---------- |  IN SERVICE |  NO PCMS CODE |
+------------------------------------------------------------------+
```

Setup Information	
Module	**Details**
Main CPU	10.60.1.10
CPU A	10.60.1.11
CPU B	10.60.1.12
GD	10.60.1.13
Domain Name	lab.fg
IPTouch 4068 / Ext: 1000	10.60.1.21
IPTouch 4038 / Ext: 2000	10.60.1.20
X-LITE Sip Client / Ext: 7000	10.60.1.101
CyberData Speaker V3.1 / Ext: 8000	10.60.1.142

6.2 Software configuration

- **Alcatel-Lucent Communication Platform**:
  - o OmniPCX Enterprise R11.2.2 / L2.300.32a

- **Application platform**:
  - o CyberData Speaker V3.1/ V11.6.8b07

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 Alcatel-Lucent side or on Application Partner side
Comment: to be filled in with any relevant comment. Mandatory in case a test has failed especially the reference number of the issue.

8 Test Results

8.1 Connectivity and Setup

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 CyberData Speaker 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 CyberData Speaker with a dynamic IP address (given by a DHCP server)	☐	☒	☐	
2	SIP Registration				
A	**SIP registration, using OXE MAIN IP address without authentication** CyberData Speaker is configured to register with the main IP address.	☐	☒	☐	
B	**SIP registration, using OXE as DNS server without authentication** CyberData Speaker 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.	☐	☒	☐	
C	**SIP registration, using an External DNS server without authentication** CyberData Speaker is configured with an external DNS server and registers successfully using the MAIN cpu node name.	☐	☒	☐	
D	**Support of "423 Interval Too Brief" (1)** CyberData Speaker is configured with a value lower than OXE SIP Min Expiration Date.	☐	☒	☐	
E	**SIP registration with authentication** For this test, register on a Node with authentication enable (2) Configure the CyberData Speaker with main IP address as SIP registrar. 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	**Signaling TCP.** Configure your SIP set to use the protocol SIP over TCP Check the registration, and basic calls.	☒	☐	☐	CyberData speaker only supports UDP
B	**Signaling 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 CyberData Speaker configured to retrieve the date and time from the node1 primary main IP address. Check that CyberData Speaker retrieves the right date and time information and displays it.	☐	☒	☐	The time is only available on the web interface

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"

8.2 Duplication and Robustness

Check how the system will react in case of a CPU reboot, switchover or link failure etc.
The test system is configured without spatial redundancy (duplicate call servers on two different IP subnetworks).

For each configuration, check:
Can new calls to the CyberData Speaker be made immediately after switchover?
Are existing calls maintained after switchover?
Are outgoing calls accepted immediately after switchover?

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	Case of Dual CPU				
A	**CPU Reboot** Test that the CyberData Phone can receive phone calls right after a bascul, and can also make a phone call with the help of a button (Cyberdata part #011185 Remote call button)	☐	☒	☐	
B	**Call maintained during bascul** Check that a call is maintained during a bascul event	☐	☒	☐	
2	Partner SIP endpoint reboot				
A	**Partner SIP set reboot** Reboot CyberData phone. When it comes back in service, call an extension with the help of the Remote Call Button.	☐	☒	☐	
3	Network failure				
A	**Temporary Link down between OXE and the partner SIP set** Disconnect the link between CyberData phone and OXE. CyberData Phone becomes out of service. When the link comes back up, OXE extension should be able to call right away the phone,	☐	☒	☐	

8.3 Audio codecs negotiations/ VAD / Framing

CyberData Speakerphone can be setup to either ONE or ALL 3 codecs at the same time (there is no in between):

- PCMU/G.711u-law
- PCMA/G.711a-law
- G.722

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	G 711 A, G 711 μ support (Uncompressed codec)				
A	**Call from** CyberData **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** CyberData **(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, in this order** **Call from** CyberData **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** CyberData **(intra-domain)** Check that the call is established using direct RTP in G711 µ-law. Check audio quality	☐	☒	☐	
2	Voice Activity Detection				

Test Case Id	Test Case	N/A	OK	NOK	Comment
A	**Configure** CyberData Speaker **to use VAD** **Configure** OXEset-1 **NOT to use VAD** **Call from** CyberData Speaker **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** CyberData Speaker **(intra-domain)** Check that the call is established using direct RTP in G711 A-law. Check audio quality **Configure** CyberData Speaker **to use VAD** **Configure** OXEset-1 **to use VAD** Redo the same tests **Configure** CyberData Speaker **NOT to use VAD** **Configure** OXEset-1 **to use VAD** Redo the same tests	☒	☐	☐	CyberData speaker has no VAD option.
B	**Configure** CyberData Speaker **to use VAD** **Configure** OXEset-2 **NOT to use VAD** **Call from** CyberData Speaker **to** OXEset-2 **(extra-domain)** Check that the call is established using direct RTP in G729. Check audio quality **Call from** OXEset-2 **to** CyberData Speaker **(extra-domain)** Check that the call is established using direct RTP in G729. Check audio quality **Configure** CyberData Speaker **to use VAD** **Configure** OXEset-2 **to use VAD** Redo the same tests **Configure** CyberData Speaker **NOT to use VAD** **Configure** OXEset-2 **to use VAD** Redo the same tests	☒	☐	☐	With VAD activated on the OXE side, calls worked both way.
4	Packet framing				

Test Case Id	Test Case	N/A	OK	NOK	Comment
A	**Configure** CyberData Speaker **to use framing=30ms (G.711)** **Call from** CyberData Speaker **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** CyberData Speaker **(intra-domain)** Check that the call is established using direct RTP in G711 A-law. Check audio quality	☒	☐	☐	Packet framing not configurable on CyberData speakerphone
B	**Configure** CyberData Speaker **to use framing=30ms (G.729)** **Call from** CyberData Speaker **to** OXEset-2 **(extra-domain)** Check that the call is established using direct RTP in G729. Check audio quality **Call from** OXEset-2 **to** CyberData Speaker **(extra-domain)** Check that the call is established using direct RTP in G729. Check audio quality	☒	☐	☐	Packet framing not configurable on CyberData speakerphone

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.
By default, all phones are multiline set with two lines.

NOTE: It is only possible to call from the CyberData Speakerphone if the REMOTE CALL BUTTON is installed and programmed. (Part #011185)

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	Local/Network calls				
A	**Call to a local user** With CyberData Speaker call the OXE phone OXEset-1. Check that OXEset-1 is ringing. On CyberData Speaker check the ring back tone. On both sets check display (name and extension number) Answer the call and check audio and display.	☐	☒	☐	
D	**Call to another SIP set** With the CyberData Speaker 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 CyberData Speaker sip set configured with authentication. With CyberData Speaker call NwkSIPset-2. Answer the call, check audio and display.	☐	☒	☐	
F	**Call to external number (via T2 loopback)** (Check ring back tone, called party display) With CyberData Speaker dial OXEset-1 though DPT1 board Check that OXEset-1 is ringing. Answer the call and check audio, display and call release.	☐	☒	☐	
G	**SIP session timer expiration** Check if call is maintained after the session timer expiration: If possible, configure the "Session timer" on CyberData Speaker to 120 seconds. With CyberData Speaker 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 CyberData Speaker to the default value.	☒	☐	☐	SIP Session timer expiration is not configurable.

Test Case Id	Test Case	N/A	OK	NOK	Comment
H	**Call to wrong number** (SIP: "404 Not Found") With the CyberData Speaker call a wrong number which is not in the dialing plan. Check the ring back tone and display.	☐	☒	☐	
2	Local/Network calls – called party is not available				
A	**Call to local user with no answer** With CyberData Speaker 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	☐	☒	☐	With the button feature, it is possible to hang up during the call process as long as the binary is at least v11.6.8b07
B	**Call to busy user** (SIP: "486 Busy Here") With CyberData Speaker call OXEset-1, answer the call, and don't hang up. With SIPset-2 call OXEset-1, answer the call, and don't hang up. With SIPset-3 call OXEset-1 which is busy Check the ring back tone and display. With OXEset-1 call CyberData Speaker, answer the call, and don't hang up. With SIPset-2 call CyberData Speaker, answer the call, and don't hang up. With SIPset-3 call CyberData Speaker which is busy Check the ring back tone and display.	☐	☒	☐	OXE sends 183/SessionProgress/ is busy Then 487/RequestTerminated
C	**Call to user in "Out of Service" state** (SIP: "480 Temporarily Unavailable") Disconnect SIPset-2 and wait for SIP deregister With CyberData Speaker 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 CyberData Speaker call the OXEset-1. Check ring back tone and display. Redial 42 on OXEset-1 to cancel the DND	☐	☒	☐	CyberData speakerphone will hear twice "The party you are calling is unavailable, please call again later" then the OXE will hang up the communication.
3	Call release				

Test Case Id	Test Case	N/A	OK	NOK	Comment
A	**Call release during an outgoing call, release done by the partner SIP set** With CyberData Speaker call OXEset-1 and don't answer the call. With CyberData Speaker, release the call during the ringing period. Check that OXEset-1 plays a release tone and goes in idle mode after some seconds	☐	☒	☐	This is only possible
5	Call to a forwarded user				
A	**Call to local user, immediate forward (CFU).** (SIP: "302 Moved Temporarily")(1) On OXEset-1 dial Immediate forward prefix. After the voice guide, enter SIPset-2 <target MCDU number>) to activate the CFU. Wait for acknowledgement from OXE. With CyberData Speaker 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 CyberData Speaker 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.	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
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 CyberData Speaker 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.	☐	☒	☐	
6	OXE features (Call back, voice mail deposit)				
A	**Call Back on free set** From CyberData Speaker 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 CyberData Speaker is ringing, answer the call and check audio + display.	☒	☐	☐	
B	**Voice mail deposit** From CyberData Speaker 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.	☒	☐	☐	

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.

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	Local/ISDN calls				
A	**Local / ISDN call to free SIP terminal** Local: with OXEset-1 call CyberData Speaker. Check that CyberData Speaker is ringing and answer the call Check ring back tone, audio and called party display. PSTN: with OXEset-1 call CyberData Speaker by dialing through PSTN. Check that CyberData Speaker is ringing and answer the call. Check ring back tone, audio and called party display.	☐	☒	☐	
B	**Local / ISDN call to busy SIP terminal** Local: Make CyberData Speaker busy by making a call. With OXEset-1 call CyberData. PSTN: While CyberData Speaker is busy, call it though a PSTN line. Check ring back tone, and called party display.	☐	☒	☐	
C	**Local/network call to unplugged SIP terminal** Local: Unplug the CyberData Speaker SIP set and call it with OXEset-1. Check the ring back tone and display	☐	☒	☐	

Test Case Id	Test Case	N/A	OK	NOK	Comment
E	**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 CyberData Speaker. Answer the call on CyberData Speaker and 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 (3).	☐	☒	☐	OXE needs to be configured to send UPDATES instead of default RE_INVITE
3	Call release				
A	**Call release during an incoming call, release done by the partner SIP set** With OXEset-1 call CyberData Speaker and don't answer the call. With CyberData Speaker, reject the call during the ringing period. Check that OXEset-1 plays a release tone and goes in idle mode after some seconds	☐	☐	☒	CyberData Speakerphone will not reject an incoming call. (This is the way it is supposed to work)
B	**Call release during an incoming call, release done by the OXE set** With OXEset-1 call CyberData Speaker and don't answer the call. With OXEset-1, release the call during the ringing period. Check that CyberData Speaker plays a release tone and goes in idle mode after some seconds	☐	☒	☐	CyberData Speaker tries to picks up the call, but since a SIP leg transaction does not exist, it will hangs up shortly after.

Notes:

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

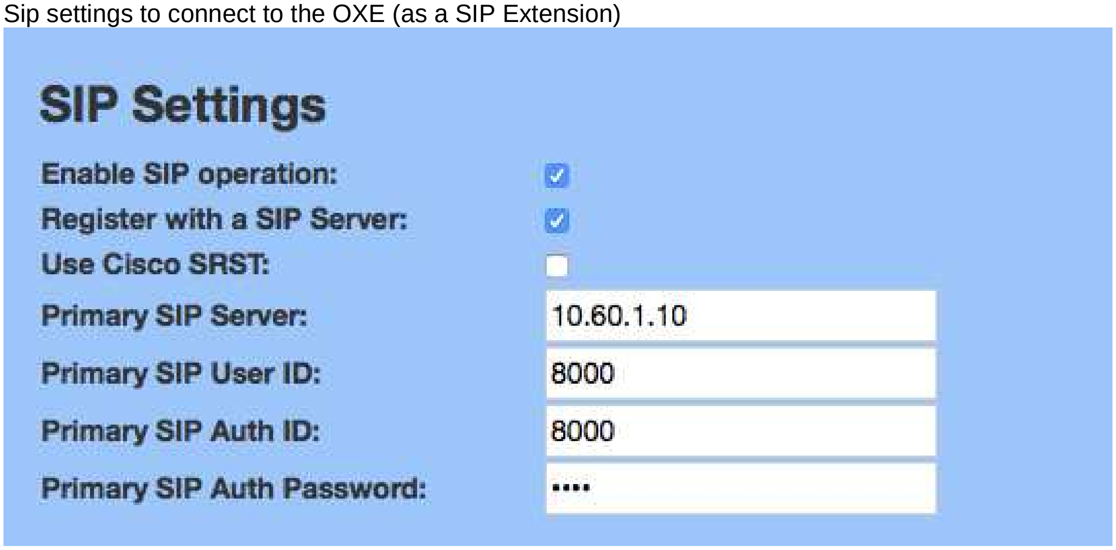
8.6 Features during Conversation

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.

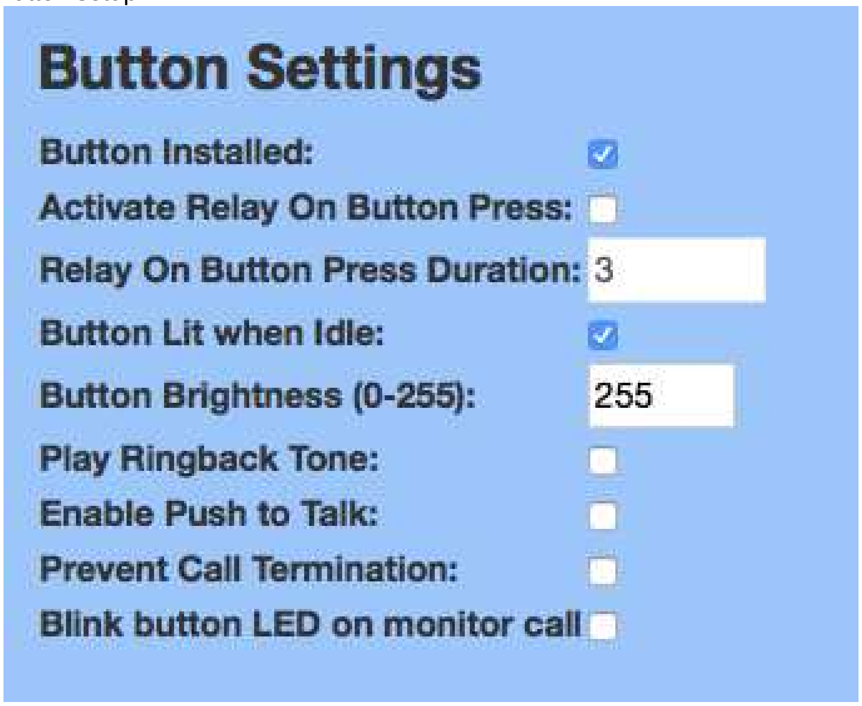
Test Case Id	Test Case	N/A	OK	NOK	Comment
1	Hold/resume				
A	**Hold and resume in case of a single call (by local feature if applicable)** With CyberData speaker call OXEset-1 Answer the call, check audio and display. With OXEset-1 put CyberData Speaker on hold with “Hold” key, check tones and display 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.	☐	☒	☐	
2	Call release				
A	**Call release during conversation, release done by the partner SIP set** With CyberData Speaker call OXEset-1 Answer the call, check audio and display. With CyberData Speaker, release the call. Check that OXEset-1 plays a release tone and goes in idle mode after some seconds	☐	☒	☐	
B	**Call release during conversation, release done by the OXE set** With CyberData Speaker call OXEset-1 Answer the call, check audio and display. With OXEset-1, release the call. Check that CyberData Speaker plays a release tone and goes in idle mode after some seconds	☐	☒	☐	
4	DTMF sending				
A	**Sending DTMF** Configure CyberData Speaker to receive DTMF using RFC 2833 From OXE, Call CyberData speakerphone and enter the 4 digits security code in order to be able to announce a message.	☐	☒	☐	

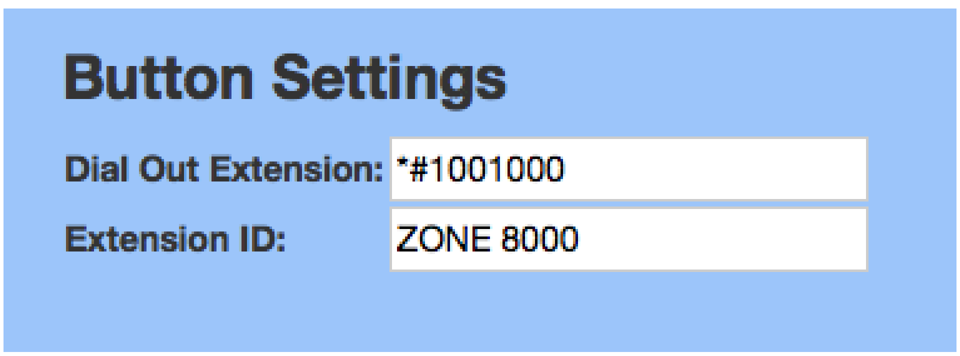
9 Appendix A : AAPP member's Application description

Sip settings to connect to the OXE (as a SIP Extension)



Button setup:





Auto-Answer incoming calls setting:

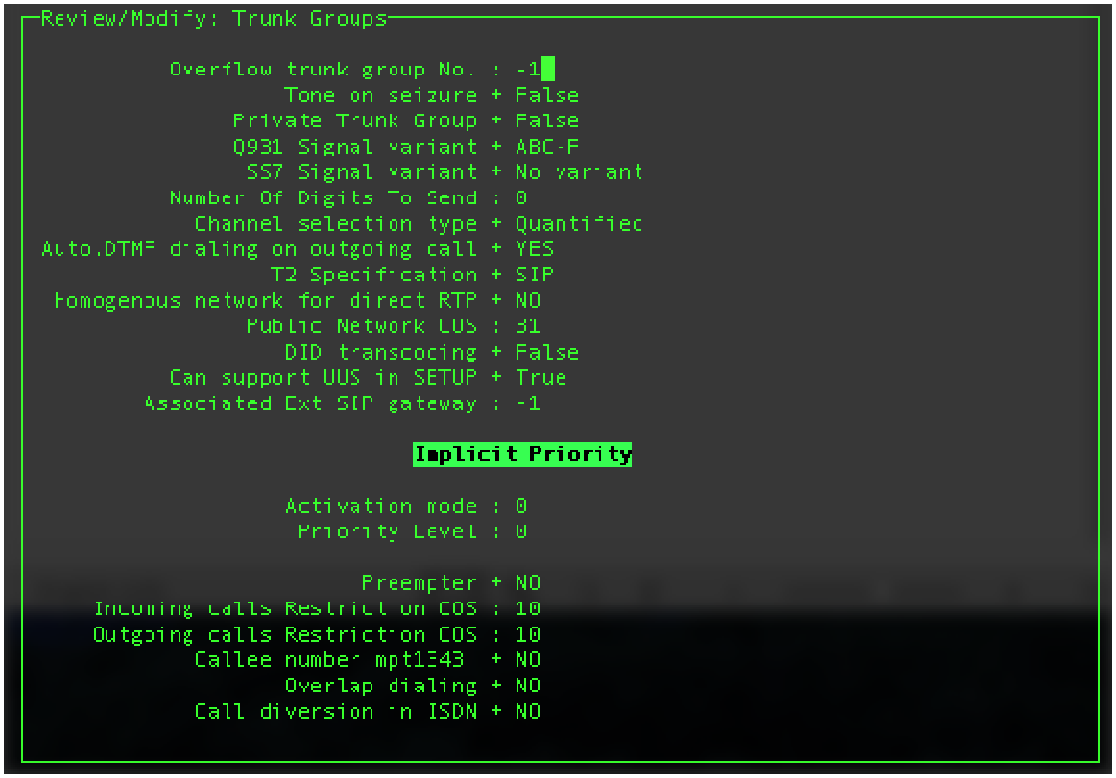
10 Appendix B: Configuration requirements of the AAPP member's application

No requierement.

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

<u>Configurations done in OXE:</u>

SIP TRUNK:



```
Review/Modify: Trunk Groups

              Node Number (reserved)   1
                      Trunk Group ID   300

                   Trunk Group Type  - T2
                    Trunk Group Name   SIP GW ABC
              UTF-8 Trunk Group Name   Sip Internal Gateway ABC-F
             Number Compatible With    -1
                      Remote Network   14
                  Shared Trunk Group - False
                    Special Services - Nothing
                         Node number   1
                Transcom Trunk Group - False
            Auto.reserv.by Attendant - False
            Overflow trunk group No.   -1
                      Tone on seizure - False
                 Private Trunk Group - False
                 Q931 Signal variant - ABC F
                  SS7 Signal variant - No variant
            Number Of Digits To Send   0
              Channel selection type - Quantified
 Auto.DTMF dialing on outgoing call - YES
                    T2 Specification   SIP
 Homogenous network for direct RTP - NO
                  Public Network COS   31
                     DID transcoding - False
            Can support UUS in SETUP - True
```

SIP GATEWAY:

```
Review/Modify: SIP Gateway

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

                    SIP Subnetwork : 14
                   SIP Trunk Group : 300
                        IP Address : 10.60.1.10
               Machine name - Host : node000000
             SIP Proxy Port Number : 5060
        SIP Subscribe Min Duration : 20
        SIP Subscribe Max Duration : 86400
                     Session Timer : 180
                 Min Session Timer : 90
              Session Timer Method + RE_INVITE
             DNS local domain name : lab.fg
                          DNS type + DNS A
               SIP DNS1 IP Address : 10.60.1.5
               SIP DNS2 IP Address : -----------------------------------------------
                        SDP in 18x + True
                       CAC SIP-SIP + False
  INFO method for remote extension + False
     Dynamic Payload type for DTMF : 97
```

12 Appendix D: AAPP member's escalation process

Contact:

CyberData Corporation
3 Justin Court
Monterey, CA 93940 USA
www.CyberData.net
Phone: 800-CYBERDATA (800-292-3732)
Fax: 831-373-4193

Sales:

Sales 831-373-2601, Extension 334

Technical Support

The fastest way to get technical support for your VoIP product is to submit a VoIP Technical Support From at the following website http://support.cyberdata.net/

The Support Form initiates a ticket which CyberData uses for tracking customer requests. Most importantly, the Support Form tells us which PBX system and software version that you are using, the make and model of the switch, and other important information. This information is essential for troubleshooting. Please also include as much detail as possible in the **Comments** section of the Support Form.

Phone: (831) 373-2601, Extension 333

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:

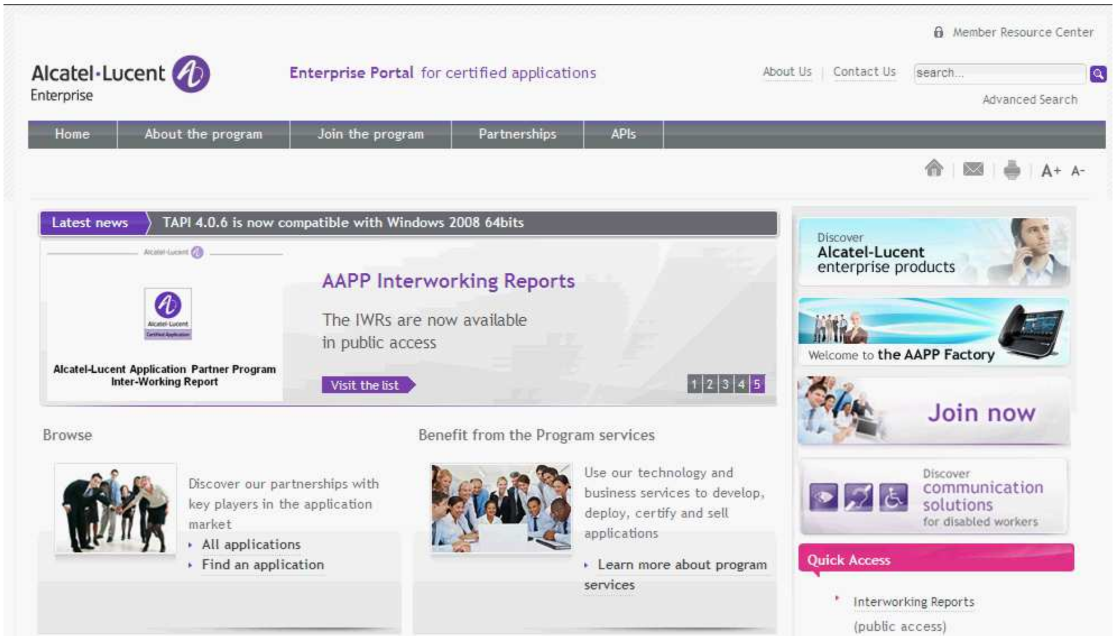
- **Provide easy interfacing for Alcatel-Lucent Enterprise communication products**: Alcatel-Lucent Enterprise's communication products for the enterprise market include infrastructure elements, platforms and software suites. To ensure easy integration, the AAPP provides a full array of standards-based 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 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/

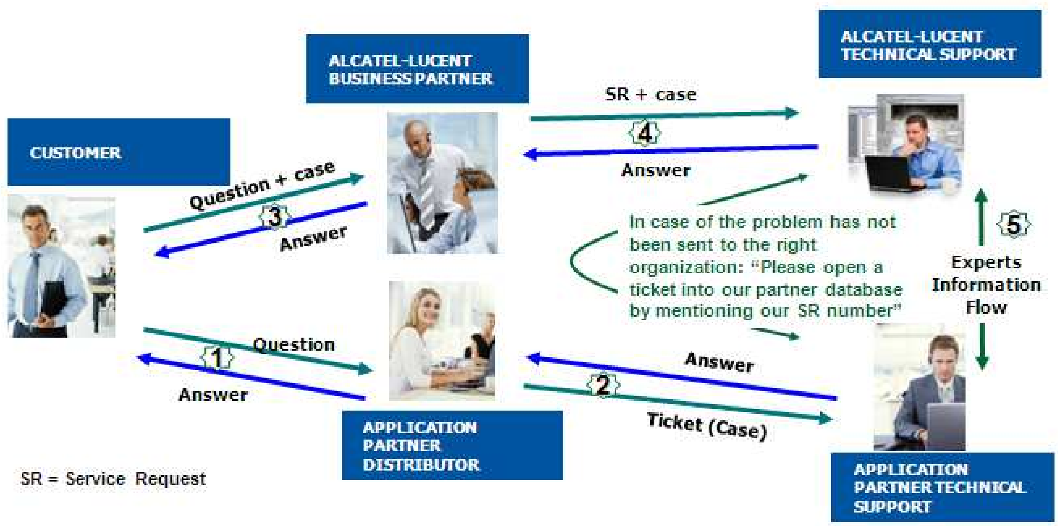
14 Appendix F: AAPP Escalation process

14.1 Introduction

The purpose of this appendix is to define the escalation process to be applied by the ALE 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 has demonstrated with traces a problem on the ALE side or if the Application Partner (not the Business Partner) needs the involvement of ALE

In that case**,** the ALE Business Partner must provide the reference of the Case Number on the Application Partner side. 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: https://applicationpartner.alcatel-lucent.com) or Enterprise Business Portal (Url: Enterprise Business Portal) 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): http://applicationpartner.alcatel-lucent.com
- e-Support from the ALE 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 Business Partners Support Center for countries:

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

For other countries:

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

END OF DOCUMENT