



ALE Application Partner Program Inter-Working Report

LANCOM Systems:
*Application type: **SBC***
SBC - Gateway:
Alcatel-Lucent Enterprise Platform:
OXO Connect & OXO Connect Evolution

LANCOM
Systems

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	October 1st, 2019
ALE representative	Benjamin LAY Juergen KOHLER
AAPP member representative	Wolfgang KRIEGISCH
Alcatel-Lucent Enterprise Communication Platform	OXO Connect Evolution OXO Connect
Alcatel-Lucent Enterprise Communication Platform release	R3.1/ 30.002
AAPP member application release	LANCOM LCOS Vers. 10.20 RU7
Application Category	SBC Gateway

Author(s):

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Revision History

Edition 1: October 2019 – First edition

Test results

- Passed
 Refused
 Postponed
 Passed with restrictions

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

IWR validity extension

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TABLE OF CONTENTS

1	INTRODUCTION	6
2	VALIDITY OF THE INTERWORKING REPORT	7
3	LIMITS OF THE TECHNICAL SUPPORT	8
3.1	CASE OF ADDITIONAL THIRD-PARTY APPLICATIONS	8
4	APPLICATION INFORMATION	9
5	TEST ENVIRONMENT	10
5.1	HARDWARE CONFIGURATION.....	10
5.2	SOFTWARE CONFIGURATION	10
6	SUMMARY OF TEST RESULTS	11
6.1	SUMMARY OF MAIN FUNCTIONS SUPPORTED.....	11
6.1.1	<i>OXO Connect with SBC</i>	11
6.2	SUMMARY OF PROBLEMS NO PROBLEMS COULD BE OBSERVED DURING THE TEST PLAN.....	11
6.3	SUMMARY OF LIMITATIONS DTMF SIP INFO IS NOT SUPPORTED IN OXOC	11
6.4	NOTES, REMARKS.....	11
7	TEST RESULT TEMPLATE	12
8	TEST RESULTS	13
8.1	ESBC TESTS	13
8.1.1	<i>Test Objectives</i>	13
8.2	TEST RESULTS.....	13
8.3	OUTGOING CALL.....	14
8.3.1	<i>Test Objectives</i>	14
8.3.2	<i>Test Results</i>	14
8.4	INCOMING CALL	17
8.4.1	<i>Test Objectives</i>	17
8.4.2	<i>Test Results</i>	17
8.5	FEATURES DURING CALL	18
8.5.1	<i>Test Objectives</i>	18
8.5.2	<i>Test Results</i>	18
8.6	CALL TRANSFER.....	19
8.6.1	<i>Test Objectives</i>	19
8.6.2	<i>Test Result</i>	20
APPENDIX A	AAPP MEMBER'S APPLICATION DESCRIPTION	21
APPENDIX B	CONFIGURATION REQUIREMENTS OF THE AAPP MEMBER'S APPLICATION	22
APPENDIX C	ALCATEL-LUCENT ENTERPRISE COMMUNICATION PLATFORM: CONFIGURATION REQUIREMENTS	31
C.1.	LICENSES.....	31
C.2.	SIP TRUNK CONFIGURATION	31
APPENDIX D	AAPP MEMBER'S ESCALATION PROCESS	33
APPENDIX E	AAPP PROGRAM	35
E.1.	ALCATEL-LUCENT APPLICATION PARTNER PROGRAM (AAPP)	35
E.2.	WEB SITE.....	35
E.3.	ENTERPRISE.ALCATEL-LUCENT.COM	36
APPENDIX F	AAPP ESCALATION PROCESS	37
F.1.	INTRODUCTION	37
F.2.	ESCALATION IN CASE OF A VALID INTER-WORKING REPORT	37

F.3.	ESCALATION IN ALL OTHER CASES	38
F.4.	TECHNICAL SUPPORT ACCESS	38

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://www.al-enterprise.com/en/partners/aapp>) with free access.

2 Validity of the InterWorking Report

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

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

A new release is identified as following:

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

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

Note 1: *The InterWorking report becomes automatically obsolete when the mentioned product releases are end of life.*

Note 2: *The renewal of the interoperability test (certification) is under the responsibility of the partner except if the certification fee is included in the program fee (e.g. “Application Partner” membership level) in this case ALE will schedule a new certification every two year*

3 Limits of the Technical support

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

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

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

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

3.1 Case of additional Third-party applications

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



4 Application information

Application commercial name:

LANCOM VoIP Routers with LCOS (LANCOM Operating System) : LANCOM 883 VoIP, LANCOM R 883+, LANCOM 884 VoIP, LANCOM 178X, LANCOM 179X, LANCOM 19XX, LANCOM ISG 1000 / 4000 all versions with SBC included

Application version:

LCOS 10.20 RU7

Interface type for tested R833+ : 4 x Gigabit Ethernet, 1 x xDSL, 4 x analog, 1 x ISDN TE/NT, 1 x ISDN NT

Brief application description:

This VoIP Router enables small businesses on single sites to securely and seamlessly migrate from ISDN/analog to the new All-IP network.

The LANCOM Voice Call Manager provide common functionalities of a Session Border Controller:

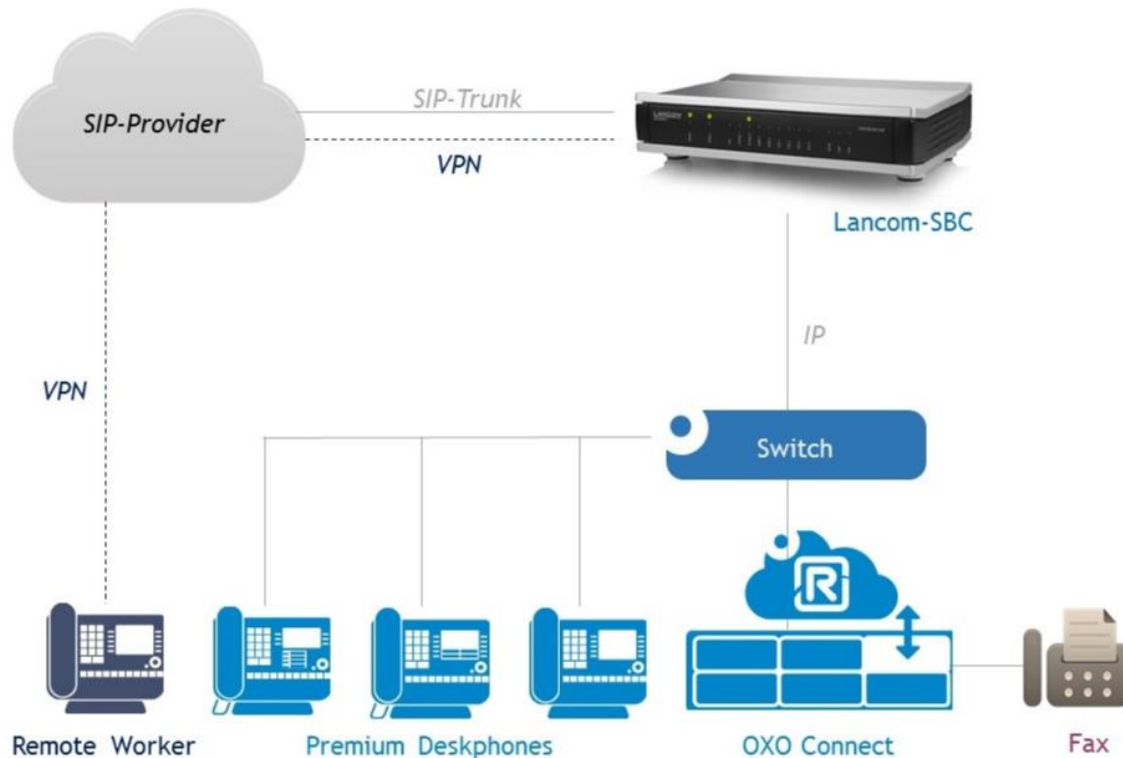
For instance, it enables the secure separation of external (insecure) and internal (secure) networks.

Ensuring a high voice quality, voice packets are preferred due to bandwidth reservation (QoS).

On top of that, the VCM as a SIP proxy enables the professional management of signaling and voice data for a high security during establishment, processing and termination of phone calls – including the necessary conversion of protocols via transcoding.

5 Test environment

Figure 1 Test environment



5.1 Hardware configuration

List main hardware equipments used for testing

- **OXO Connect:**
Premium Deskphones: 8028s,8058s,8068s, 8068s for Remote Worker
Fax: Canon MX300 on analog Port of OXO Connect
- **LANCOM R883+:**

5.2 Software configuration

List main softwares used for testing

- **Alcatel-Lucent OXO Connect:** ONEDE031/030.002
- **Partner Application:** LCOS 10.20RU7

6 Summary of test results

6.1 Summary of main functions supported

6.1.1 OXO Connect with SBC

Features	Results	Remarks
Initialisation and network configuration	OK	supported sip config for OXO reused with minimal changes for registrar name/IP (OXOC terminates in SBC)
SIP registration	OK	
SIP Authentication	OK	
VoIP and RTP support	OK	
TLS and SRTP support on carrier side	OK	Tested with Telekom DLAN SIP Trunk
Outgoing call	OK	
Incoming call	OK	
Features during conversation	OK	
Fax G711	OK	Fax T38 not supported in test environment on other side – Fax T38 on OXO is working with fallback to G711
Remote Homeworker VPN IKE V1	OK	LANCOM R883+ is supporting VPN server functionality
Remote Homeworker VPN IKE V2	OK	

6.2 Summary of problems

No problems could be observed during the test plan

6.3 Summary of limitations

DTMF SIP Info is not supported in OXOC

6.4 Notes, remarks

Tests have been done with a fully certified German SIP Trunk “Telekom DeutschlandLAN SIP Trunk” on OXO Connect / Evolution with Release 3.0 in direct connection mode without SBC. Main purpose of this test is to ensure additional SBC functions and SIP encryption by using existing SIP trunking profiles for OXO Connect & OXO Connect Evolution with minimal adaptations.

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 <ul style="list-style-type: none"> Action Expected result 	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	
2	Test case 2 <ul style="list-style-type: none"> Action Expected result 	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	The application waits for PBX timer or phone set hangs up
3	Test case 3 <ul style="list-style-type: none"> Action Expected result 	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	Relevant only if the CTI interface is a direct CSTA link
4	Test case 4 <ul style="list-style-type: none"> Action Expected result 	<input type="checkbox"/>	<input type="checkbox"/>	<input checked="" type="checkbox"/>	No indication, no error message
...	...	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	

Test Case Id: a feature testing may comprise multiple steps depending on its complexity. Each step must 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 side or on AAPP member side

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

8 TEST Results

8.1 eSBC Tests

8.1.1 Test Objectives

The eSBC Configuration:

eSBC is configured to use specific codec G.722, G.711 A-law, G.711 mu-law, G.729, G.723 in this order

Phone configuration:

Configure Premium Deskphone with codec G.722, G.711 A-law, G.711 mu-law, G.729, G.723 in this order and to NOT use VAD (unless otherwise stated).

Sip provider:

Configure Sip Provider to use G.722, G.711 A-law, G.711 mu-law, G.729, G.723 in this order

8.2 Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
8-1-1	<p>Codec G711 / G722</p> <p>Select G711 as first codec on Provider</p> <p>Select G711 as first codec on SBC</p> <p>Select G722 as first codec on Premium Deskphone And G.711 A-law, G.711 mu-law, G.729 as other priority</p> <p>Call from external phone (PSTN) to Premium Deskphone Check that call is correctly established</p> <p>In all Case check audio quality</p>	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	
8-1-2	<p>Codec G729 / G711</p> <p>Select G729 as first codec on Provider</p> <p>Select G729 as first codec on SBC</p> <p>Select G711 as first codec on Premium Deskphone And G.729 as other priority</p> <p>Call from external phone to Premium Deskphone Check that call is correctly established</p> <p>In all Case check audio quality</p>	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	Operator always uses G711 as first codec, if that is supported. LCOS supports all codecs by default hardcode. Call is established correctly.

8-1-3	<p>Codec G723 / G711</p> <p>Select G723 as first codec on Provider</p> <p>Select G723 as first codec on SBC</p> <p>Select G711 as first codec on Premium Deskphone</p> <p>Call from external phone to Premium Deskphone Check that call is correctly established</p> <p>In all Case check audio quality</p>	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<p>Operator always uses G711 as first codec, if that is supported. LCOS supports all codecs by default hardcode. Call is established correctly. Audio Quality is good in both directions.</p>
8-1-4	<p>Codec G723 / G711</p> <p>Select G723 as first codec on Provider</p> <p>Select G723 as first codec on SBC</p> <p>Select G711 as first codec on Premium Deskphone</p> <p>Call from external phone to Premium Deskphone Check that call is correctly established</p> <p>In all Case check audio quality</p>	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<p>Case is equal to case 8-1-3 ??</p>

8.3 Outgoing call

8.3.1 Test Objectives

Generate calls to External PSTN line to check SBC integrity

The outgoing call is generated on an external PSTN phone number

8.3.2 Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
8-2-1	<p>Outgoing call with DTMF RFC 2833</p> <p>Call to external attendant using DTMF RFC 2833 Test DTMF return. Call an IVR and navigate to the corresponding menu and verify that DTMF is working Hang-up the call</p>	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<p>Outgoing Call Codec setting on IP-Touch 8028s G722 – Setting in LANCOM: Fallback on SIP Info</p>
8-2-2	<p>Outgoing call with DTMF Sip Info</p> <p>Call to external attendant using DTMF Sip Info Test DTMF return.</p>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<p>Not supported in OXO</p>

	<p>Call an IVR and navigate to the corresponding menu and verify that DTMF is working</p> <p>Hang-up the call Then Hang-up</p>				
8-2-3	<p>Outgoing call with DTMF Inband</p> <p>Call to external attendant using DTMF Inband Test DTMF return. Call an IVR and navigate to the corresponding menu and verify that DTMF is working</p> <p>Hang-up the call Then Hang-up</p>	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	Setting in LANCOM: "Fallback on SIP Info"
8-2-4	<p>Call to External number from VPN connected Premium Deskphone</p> <p>Call external number from VPN Premium Deskphone Check audio, then hang-up</p>	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	VPN IKE1, VPN IKE2 Audio is ok in both cases
8-2-5	<p>Outgoing call with DTMF RFC 2833 with VPN Premium Deskphone</p> <p>Call to external attendant using DTMF RFC 2833 Test DTMF return. Call an IVR and navigate to the corresponding menu and verify that DTMF is working</p> <p>Hang-up the call</p>	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	VPN IKE1, VPN IKE 2 DTMF is working
8-2-6	<p>Outgoing call with DTMF Sip Info with VPN Premium Deskphone</p> <p>Call to external attendant using DTMF Sip Info Test DTMF return. Call an IVR and navigate to the corresponding menu and verify that DTMF is working</p> <p>Hang-up the call Then Hang-up</p>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	DTMF SIP Info not available in OXOC
8-2-7	<p>Outgoing call with DTMF Inband with VPN Premium Deskphone</p> <p>Call to external attendant using DTMF Inband Test DTMF return. Call an IVR and navigate to the corresponding menu and verify that DTMF is working</p>	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	VPN IKE1, VPN IKE 2, DTMF is working

	Hang-up the call Then Hang-up				
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8.4 Incoming call

8.4.1 Test Objectives

Generate calls from External PSTN line to check SBC integrity

Called party can be in different states: Free, Busy, Out of services, DND, etc...

8.4.2 Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
8-3-1	<p>ExtCall to Premium Deskphone Timeout</p> <p>Call from Ext-PSTN to the DID configure on Premium Deskphone 1 Answer the call and check audio. Stay online for 5 minutes Then Hang-up</p>	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	Audio is ok and keeping for min 5 mins.
8-3-2	<p>ExtCall to Premium Deskphone Display</p> <p>Call from Ext-PSTN to the DID configure on Premium Deskphone 1 Check display Answer the call and check audio. Then Hang-up</p>	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	Display OK, Audio OK
8-3-3	<p>ExtCall to Premium Deskphone Display multiline</p> <p>Call from Ext-PSTN to the DID configure on Premium Deskphone 1 Check display Answer the call and check audio. Keep the call Call from Ext-PSTN to the DID configure on Premium Deskphone 1 Check display Answer the 2d call and check audio. Then Hang-up</p>	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	Both Display OK, Both Audio OK. Music on Hold from OXO for both ext. parties
8-3-4	<p>Call from External number to VPN connected Premium Deskphone</p> <p>Make a call to External number Answer the call on VPN Premium Deskphone Check audio then hang-up</p>	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	IKE V1, IKE V2, Display OK, Audio OK
8-3-5	<p>Call from VPN Premium Deskphone to Premium Deskphone 1</p>	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	IKE V1, IKE V2, Display OK, Audio OK, Wireshark Trace does not show SIP

	Make a call to internal number allocated to Deskphone 1. Answer the call on Premium Deskphone 1. Check audio then hang-up				traffic, because of SIP-NOE Protocol
8-3-6	Ext call to unplugged VPN Premium Deskphone If Applicable Unplug VPN Premium Deskphone With Ext PSTN phone call VPN Premium Deskphone Check the ring back then hang-up	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	IKE V1, IKE V2, Call reaches operator of OXOC according config "incoming call handling – other cases- go to attendant". Audio OK
8-3-7	Ext call to DND VPN Premium Deskphone If Applicable Enable DND on VPN Premium Deskphone With Ext PSTN phone call VPN Premium Deskphone Check the ring back then hang-up Cancel the DND on VPN Premium Deskphone	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	IKE V1, IKE V2, Call reaches operator of OXOC according config "incoming call handling – other cases- go to attendant". Audio OK

8.5 Features during call

8.5.1 Test Objectives

The objective is to test Features between different users during conversation.

Before test we need to check that dtmf are generated correctly, and multiple sip line is available on devices.

8.5.2 Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
8-4-1	Hold and resume a current call From Ext PSTN call Premium Deskphone 1 Answer the call and check audio. On Premium Deskphone1 press hold. Check tones and display on both parts Resume the call Keep the call for next test	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	Tones and Display OK
8-4-2	Switch between calls With FXS-1 call Premium Deskphone 1 With IPtouch1 switch between FXS-1 and Ext PSTN Check tones and display	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	FXS is analog Port on OXOC, Tones and display ok

	Keep the calls for next test				
8-4-3	<p>Three party conferences initiated from OXO set</p> <p>With Ext PSTN call Premium Deskphone 1 Answer and keep the call</p> <p>With IPtouch1 call Premium Deskphone 2 Answer and keep the call</p> <p>With Premium Deskphone 1 start a conference</p> <p>Check audio, Display, then hang-up.</p>	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<p>Call from Case 8-4-2 is not kept, otherwise 3 party conference with Premium Deskphone 2 is not possible. Audio with Ext PSTN is ok, Display is ok, Audio with 3 party conf is ok</p>

8.6 Call Transfer

8.6.1 Test Objectives

Many sorts of transfer can be requested, the objective is to test several transfer services.

- Unattended transfer
- Semi-attended transfer
- Attended transfer

For each we need to test:

- Audio
- Tone
- Display

Actors:

- A- Transferee
- B- Transferor
- C- Transfer target

Unattended transfer or Blind transfer:

The Transferor provides the Transfer Target's contact to the Transferee. The Transferee attempts to establish a session using that contact and reports the results of that attempt to the Transferor

Semi-Attended Transfer or Transfer on ringing:

1. The transferee calls the Transferor
2. The transferor calls the transfer target. The transferee is on Hold. The transfer target is ringing.
3. The transferor executes the transfer. The transferor drops the call. The transfer target is already in ringing state, the transfer target answers the call. The Transferee and the Transfer target are now in communication.

Attended Transfer or Transfer on ringing:

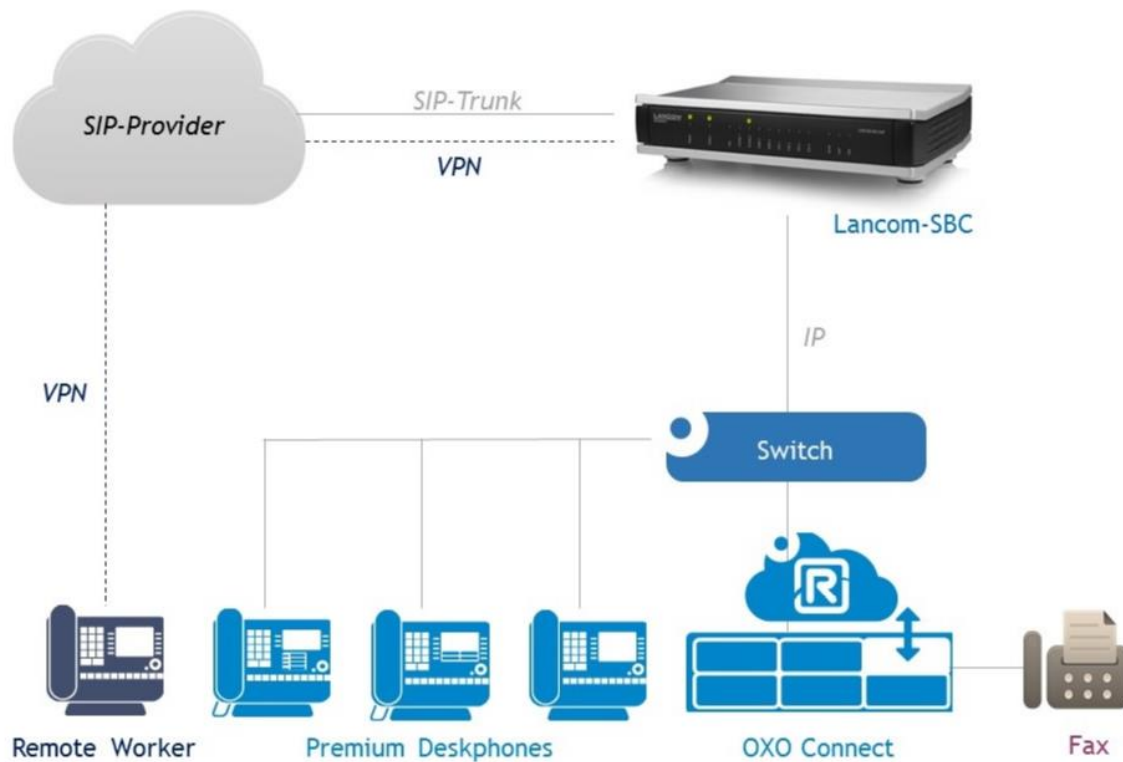
1. The transferee calls the Transferor
2. The transferor calls the transfer target. The transferee is on Hold. Transfer target picks up the call and call is established with the transferor
3. The transferor executes the transfer. The transferor drops the call. The transferee is now on line with the Transfer target.

8.6.2 Test Result

Test Case Id	Sort of transfer	Transferee	Transferor	Transfer Target	N/A	OK	NOK	Comment
8-5-1	Unattended	Premium Deskphone 2	Premium Deskphone 1	ExtNum	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	Shown number is the Premium Deskphone 1
8-5-2	Semi-attended	Premium Deskphone 2	Premium Deskphone 1	ExtNum	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	Shown number is still the Premium Deskphone 1
8-5-3	Attended	Premium Deskphone 2	Premium Deskphone 1	ExtNum	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	Shown number is still the Premium Deskphone 1

Appendix A AAPP member's Application description

Lancom routers with LANCOS operating system are customer premise Enterprise SBC solutions. They connect to the **OXO system** in the Enterprise's LAN and to an Internet telephony service provider (ITSP) and they can enable connectivity to Operators which do require SIP TLS and/or, SRTP functionality. Additionally, the Lancom Routers have own Telephony modules integrated, which do allow to connect ISDN or analog clients. This will allow the customer to connect also clients for emergency situations.



Appendix B Configuration requirements of the AAPP member's application

LANconfig > Configuration > Voice Call Manager > General - Local VoIP domain (e.g. local IP-Address of LANCOM)

R883Plus - V10.20 D2019-08-21 T1207_IKE2.lcf

QuickFinder

Configuration

- Management
- Wireless LAN
- Interfaces
- Date & Time
- Log & Trace
- Communication
 - IPv4
 - IPv6
 - IP Router
 - Routing protocols
 - Firewall/QoS
 - VPN
 - Certificates
 - COM Ports
 - NetBIOS
 - Public-Spot
 - RADIUS
 - Voice Call Manager
 - General**
 - Lines
 - Users
 - Call Router
 - Extended
 - Miscellaneous Services

Voice Call Manager (VCM) enabled

SIP parameters

To use the internal services on the VCM, a local VoIP domain must be configured for the router.

Local VoIP domain:

This domain may only be used on your end devices to register this router.

Messaging

Create a SYSLOG message for each call

Send an email for each call

Email target address:

WAN login lock

Lock configuration after: login failures

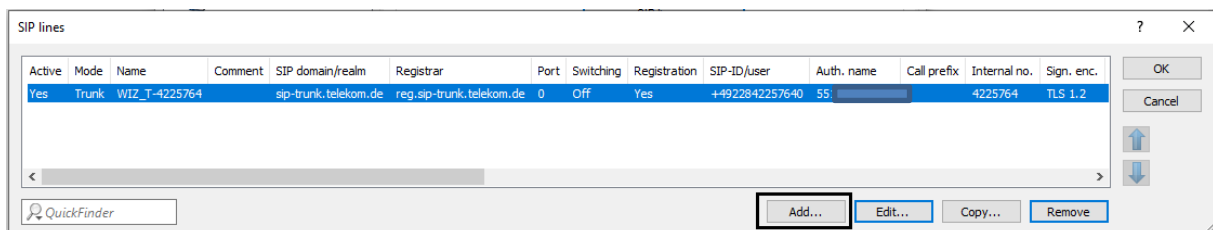
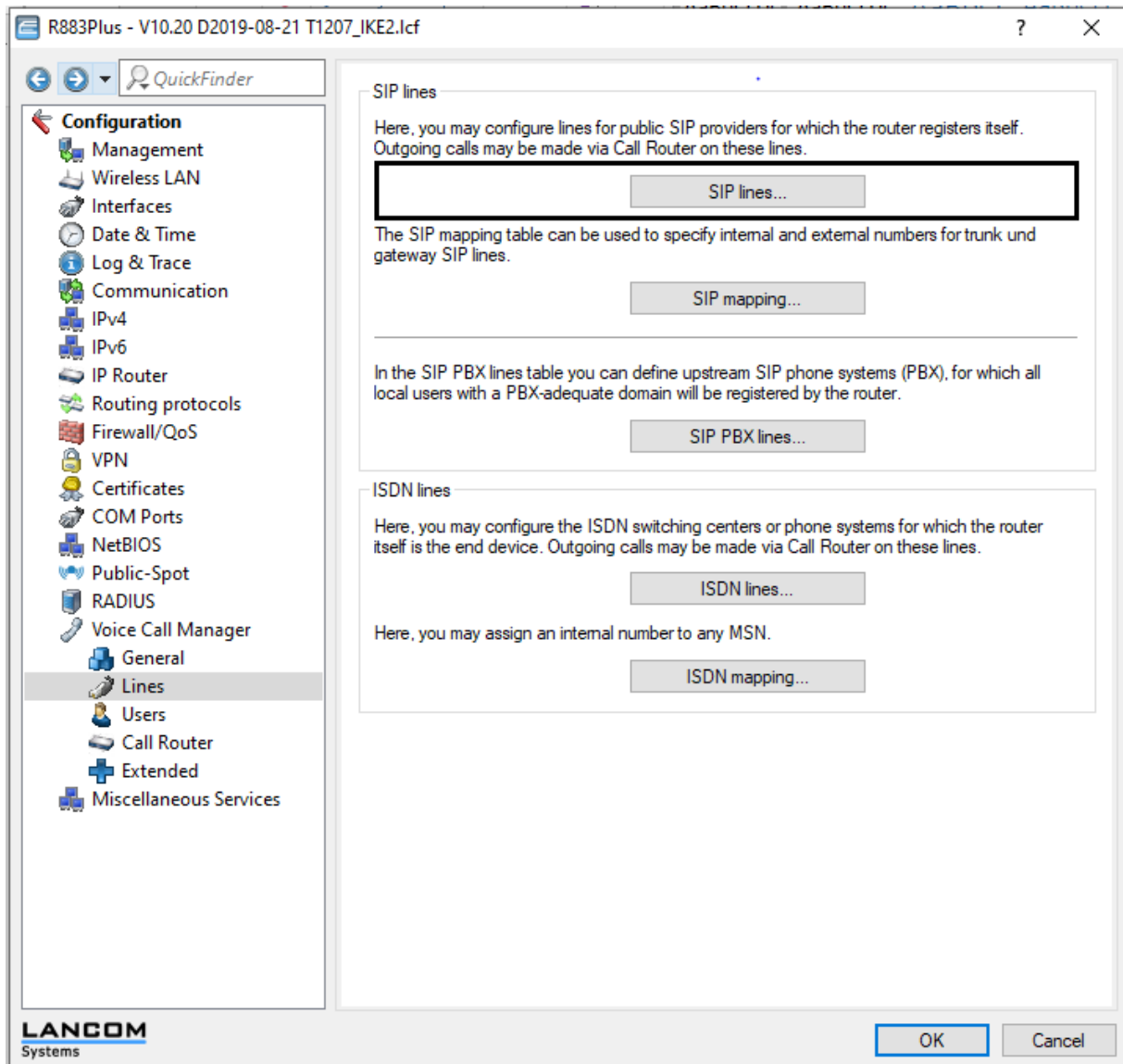
Lock configuration for: minutes

LANCOM Systems

OK Cancel

Create a SIP Line >(Trunk):

LANconfig > Configuration > Voice Call Manager > Line – SIP lines ...



General Security Advanced

Entry active

Mode: Trunk

Provider name: WIZ_T-4225764

Comment:

Provider data

SIP domain/realm: sip-trunk.telekom.de

Registrar (optional): reg.sip-trunk.telekom.de

Port: 0

Switching at provider active

Login data

(Re-)Registration

SIP-ID/user: +4922842257640

Display name (optional):

Authentication name: 55

Password: Show

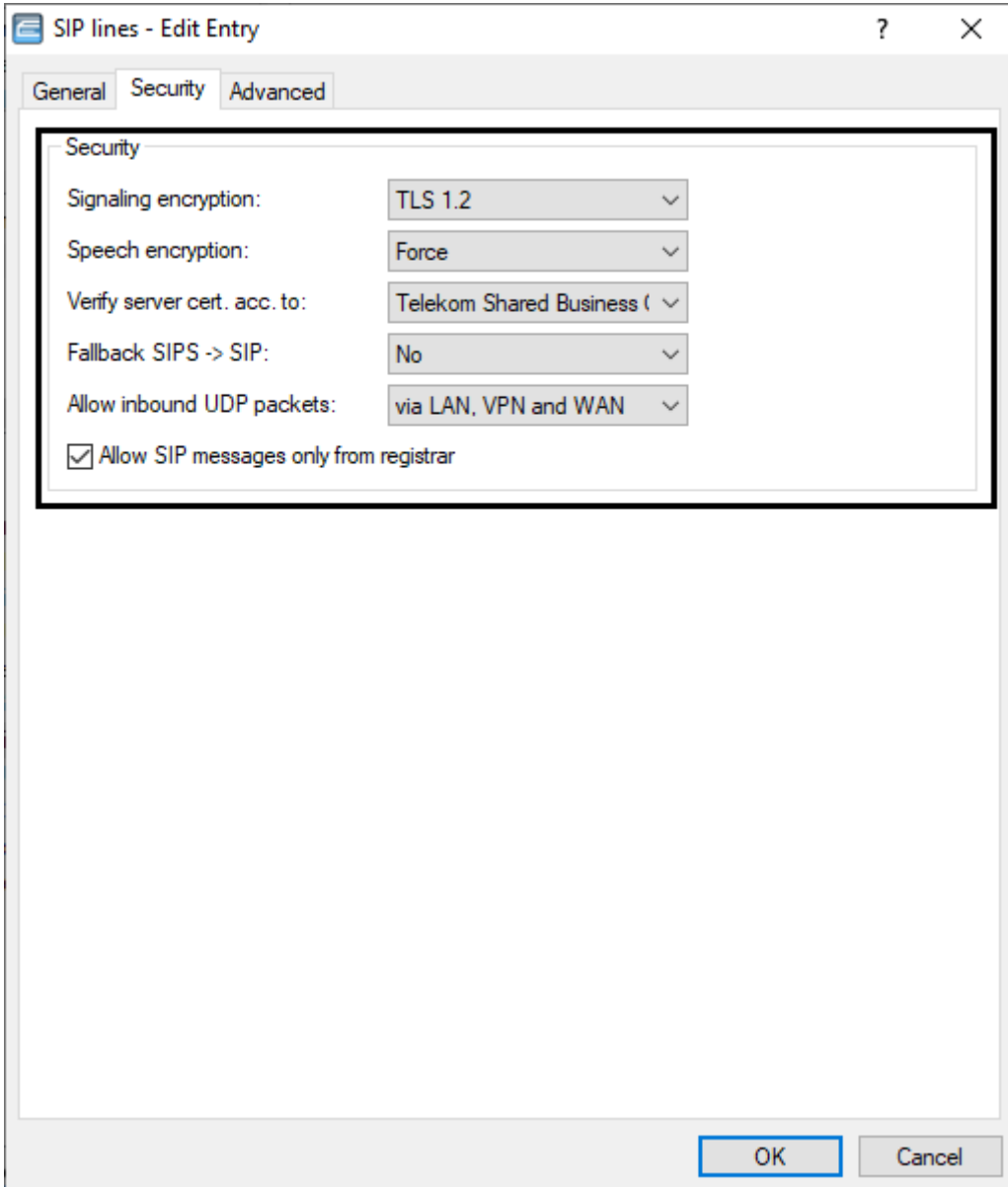
Generate password

Call prefix:

Internal dest. number: 4225764

OK Cancel

LANconfig > Configuration > Voice Call Manager > Line > SIP lines ... => Security Settings for Signaling and Speech encryption



The screenshot shows a configuration window titled "SIP lines - Edit Entry" with three tabs: "General", "Security", and "Advanced". The "Security" tab is active and contains the following settings:

Setting	Value
Signaling encryption:	TLS 1.2
Speech encryption:	Force
Verify server cert. acc. to:	Telekom Shared Business (
Fallback SIPS -> SIP:	No
Allow inbound UDP packets:	via LAN, VPN and WAN

There is also a checked checkbox labeled "Allow SIP messages only from registrar". At the bottom right of the window are "OK" and "Cancel" buttons.

SIP lines - Edit Entry

General Security **Advanced**

VoIP router

SIP proxy port:

Routing tag:

Line control

Control method:

Control interval: seconds

SIP privacy

Trusted Area activated

Transmission method:

Codec filter

DTMF signaling:

Dialing

Overlap-Dialing

Call forwarding using SIP302

SIP-ID Transmission:

OK Cancel

LANconfig > Configuration > Voice Call Manager > Line – SIP Mapping:

SIP mapping

Active	Name	Comment	External number	Length of called number	Internal number
Yes	WIZ_T-4225764	Ein-/Ausgehend Global	+492284225764#	0 digits	+492284225764#
Yes	WIZ_T-4225764	Eingehend Global var.	00492284225764#	0 digits	+492284225764#
Yes	WIZ_T-4225764	Eingehend mit Ortsvorwahl	02284225764#	0 digits	+492284225764#
Yes	WIZ_T-4225764	Eingehend Lokal	4225764#	0 digits	+492284225764#

QuickFinder

Add... Edit... Copy... Remove

OK Cancel

Active	Name	Comment	Auth. name	WAN	Device type	CLIR	DTMF signaling	Msg. Waiting (MWI) via	Transport protocols	Speech encryption
Yes	+492284225764#	Sip-Reg-Juergen-von OXO	55	denied	Phone and/or Fax	Off	Events or in-band		UDP+TCP+TLS	Ignore

Create a SIP User (SIP Trunk LAN):
LANconfig > Configuration > Users > SIP User ...

SIP users - Edit Entry

Entry active

Internal call number:

Comment:


Login data

Authentication name:

Password: Show

Access via WAN:

Device type:

 The rest of the settings (e.g. domain) must be made on the SIP end device or client.

Suppress transmission of own phone number to the remote site (CLIR)

DTMF signaling:

Msg. Waiting (MWI) via:

Security

Transport protocols:

Speech encryption:

SRTP cipher list

AES-CM-256 AES-CM-192
 AES-CM-128 F8-128

SRTP authentication

HMCA-SHA1-80 HMCA-SHA1-32

Create Call-Routen for outgoing and incoming Calls:
LANconfig > Configuration > Voice Call Manager > Call Router

Call routing

Usage	Prio	Clid. ID	Comment	Cln. ID (out)	Dest. ID	Dest. line	2. nr.	2. line	3. nr.	3. line	Clid. domain	Cln. ID (in)	Cln. domain	Src. line
On	0	00492284225764#			+492284225764#	USER								WIZ_T-4225764
On	0	#			#	WIZ_T-4225764								USER.#

QuickFinder Add... Edit... Copy... Remove

Call routing - Edit Entry

Entry active / default line: Active

Priority: 0

Called number: 00492284225764#

Comment:

Mapping

Calling number:

Destination number: +492284225764#

Destination line: USER Select

If the line is not available, you can define additional destinations here.

2. dest. number:

2. dest. line: Select

3. dest. number:

3. dest. line: Select

Filters

In addition to the called number you can define further filters for this entry:

Called domain: Select

Calling number:

Calling domain: Select

Source line: WIZ_T-4225764 Select

OK Cancel

Call routing

Usage	Prio	Cl. ID	Comment	Ch. ID (out)	Dest. ID	Dest. line	2. nr.	2. line	3. nr.	3. line	Cl. domain	Ch. ID (in)	Ch. domain	Src. line
On	0	00492284225764#			+492284225764#	USER								WIZ_T-4225764
On	0	#		#		WIZ_T-4225764								USER.#

QuickFinder Add... Edit... Copy... Remove

Call routing - Edit Entry

Entry active / default line: Active

Priority: 0

Called number: #

Comment:

Mapping

Calling number:

Destination number: #

Destination line: WIZ_T-4225764

If the line is not available, you can define additional destinations here.

2. dest. number:

2. dest. line:

3. dest. number:

3. dest. line:

Filters

In addition to the called number you can define further filters for this entry:

Called domain:

Calling number:

Calling domain:

Source line: USER.#

LANconfig > Configuration > Voice Call Manager > Extended:

R883Plus - V10.20 D2019-08-21 T1207_IKE2.Icf

QuickFinder

Configuration

- Management
- Wireless LAN
- Interfaces
- Date & Time
- Log & Trace
- Communication
- IPv4
- IPv6
- IP Router
- Routing protocols
- Firewall/QoS
- VPN
- Certificates
- COM Ports
- NetBIOS
- Public-Spot
- RADIUS
- Voice Call Manager
 - General
 - Lines
 - Users
 - Call Router
 - Extended**
 - Miscellaneous Services

Country specific profile for: Germany

Detect fax transmission and use the T.38 protocol if possible

SIP parameters

Echo cancelling from SIP to ISDN/Analog

Prefixes for displaying the calling number of incoming calls

From internal to SIP user:

From external to SIP user:

From internal to ISDN user:

From external to ISDN user:

From internal to analog user:

From external to analog user:

Quality of Service

Prioritize SIP packets by changing the other packets:

Prioritize outgoing packets: PMTU reduction & fragment

Prioritize incoming packets: No change

Reduced packet size (MTU): 576 byte

SIP DiffServ codepoint (DSCP): CS-6

RTP DiffServ codepoint (DSCP): EF

LANCOM Systems

OK Cancel

A detailed configuration guide can be found at:

<https://www.lancom-systems.com/service-support/instant-help/knowledge-base/>

Appendix C Alcatel-Lucent Enterprise Communication Platform: configuration requirements

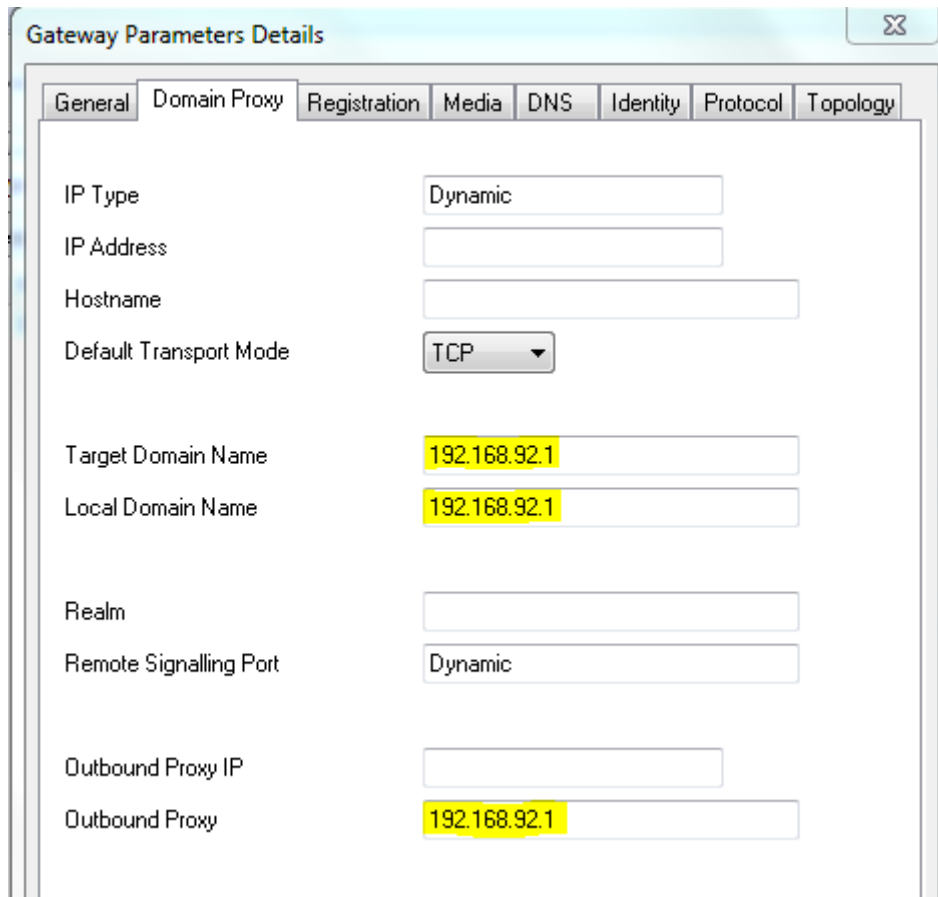
C.1. Licenses

only SIP Trunk licenses in OXOC as usual for SIP trunking.

C.2. Sip trunk configuration

Configuration can be retrieved via SIP trunk configuration file (spf file) for SIP Trunks in GA (general availability state) available on the Alcatel-Lucent Enterprise Partner Portal. Calls are terminated in the LANCOM SBC. The LANCOM SBC is handling TLS and SRTP encryption towards the public call server as well as registering in and call handling with the NGN server.

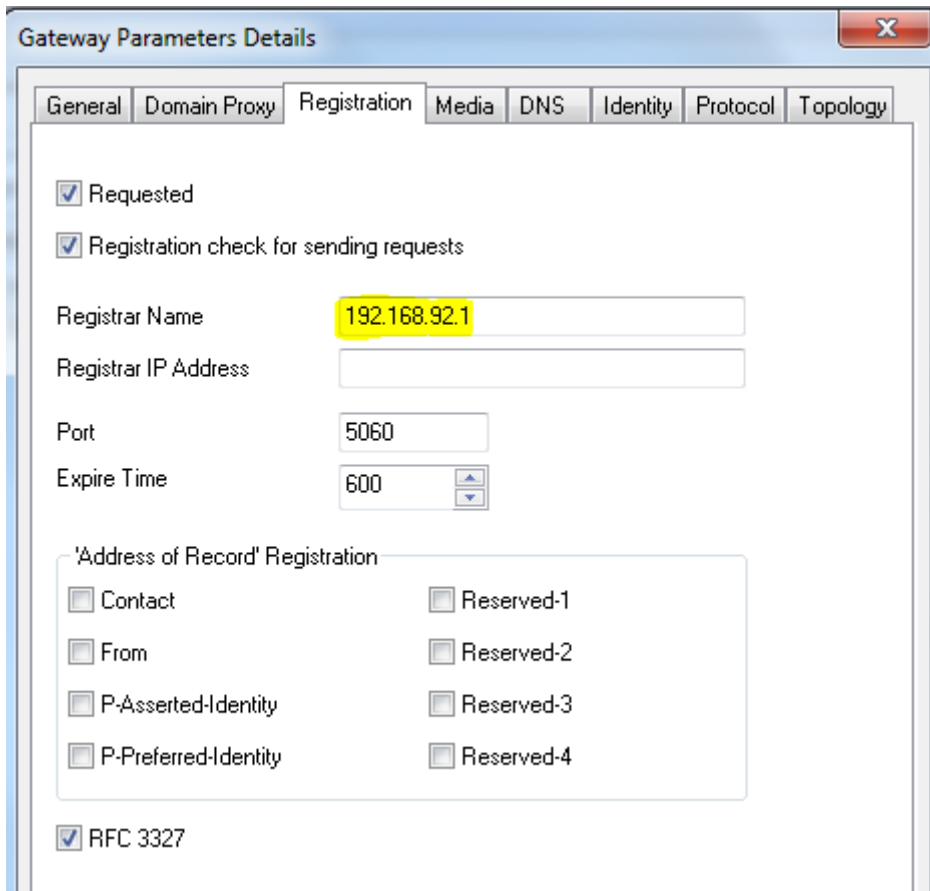
Adapt Target Domain,Local Domain and Outbound Proxy to LANCOM Router LAN IP address:



The screenshot shows a configuration window titled "Gateway Parameters Details" with a close button in the top right corner. The window has several tabs: "General", "Domain Proxy", "Registration", "Media", "DNS", "Identity", "Protocol", and "Topology". The "General" tab is selected. The configuration parameters are as follows:

Parameter	Value
IP Type	Dynamic
IP Address	
Hostname	
Default Transport Mode	TCP
Target Domain Name	192.168.92.1
Local Domain Name	192.168.92.1
Realm	
Remote Signalling Port	Dynamic
Outbound Proxy IP	
Outbound Proxy	192.168.92.1

Adapt Registrar Name to LANCOM Router LAN IP address:



Gateway Parameters Details

General | Domain Proxy | **Registration** | Media | DNS | Identity | Protocol | Topology

Requested

Registration check for sending requests

Registrar Name: 192.168.92.1

Registrar IP Address: [Empty]

Port: 5060

Expire Time: 600

'Address of Record' Registration

Contact Reserved-1

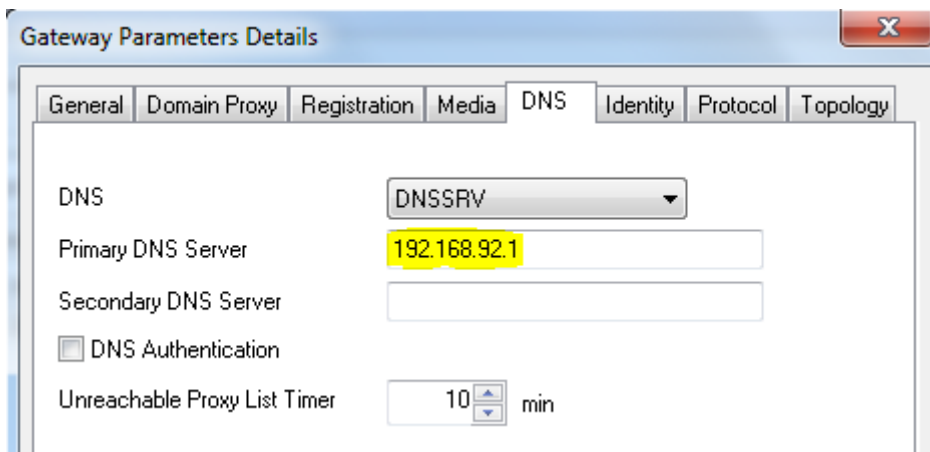
From Reserved-2

P-Asserted-Identity Reserved-3

P-Preferred-Identity Reserved-4

RFC 3327

Adapt DNS Server to LANCOM Router LAN IP address:



Gateway Parameters Details

General | Domain Proxy | Registration | Media | **DNS** | Identity | Protocol | Topology

DNS: DNSSRV

Primary DNS Server: 192.168.92.1

Secondary DNS Server: [Empty]

DNS Authentication

Unreachable Proxy List Timer: 10 min

Ensure correct Log and Authentication Parameters for OXOC at the LANCOM Router:

SIP Accounts					
Index	Login	Passw...	Registered Username	Gateway Parameters Ind...	RFC 6140
1	55 [Redacted]	*****	+492284221234	1 DT SIP TRUNK-SBC	Enabled

Appendix D AAPP member's escalation process

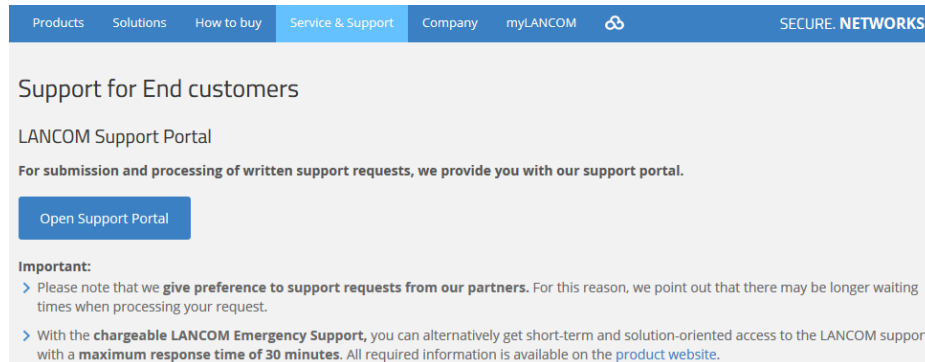
Installation is guided by Setup-Wizard
Specific configuration steps are explained by Screens Shots in the IWR

A lot of information and tricks are available at the LANCOM's Knowledge Base:
<https://www.lancom-systems.com/service-support/instant-help/knowledge-base/>

Escalation phase:

For End-Customers: >>> LANCOM Support Portal:

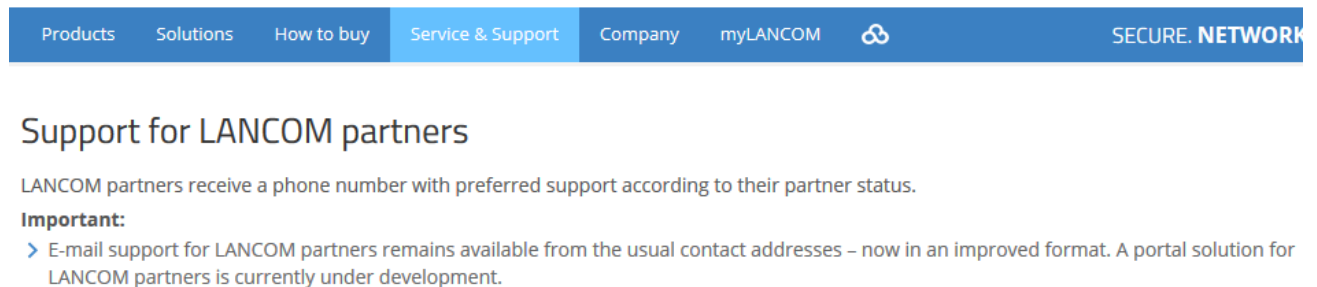
<https://www.lancom-systems.com/service-support/support-warranty/support-contact/>



The screenshot shows the top navigation bar of the LANCOM website with 'Service & Support' selected. Below the navigation bar, the page title is 'Support for End customers'. The main content area is titled 'LANCOM Support Portal' and contains the text: 'For submission and processing of written support requests, we provide you with our support portal.' Below this text is a blue button labeled 'Open Support Portal'. Underneath the button, there is an 'Important:' section with two bullet points: '> Please note that we give preference to support requests from our partners. For this reason, we point out that there may be longer waiting times when processing your request.' and '> With the chargeable LANCOM Emergency Support, you can alternatively get short-term and solution-oriented access to the LANCOM support with a maximum response time of 30 minutes. All required information is available on the product website.'

For LANCOM Partners: >>> LANCOM Support Portal:


<https://www.lancom-systems.com/service-support/support-warranty/support-contact/>



The screenshot shows the top navigation bar of the LANCOM website with 'Service & Support' selected. Below the navigation bar, the page title is 'Support for LANCOM partners'. The main content area contains the text: 'LANCOM partners receive a phone number with preferred support according to their partner status.' Below this text, there is an 'Important:' section with one bullet point: '> E-mail support for LANCOM partners remains available from the usual contact addresses – now in an improved format. A portal solution for LANCOM partners is currently under development.'

Repair Processing (warranty) >>>

<https://www.lancom-systems.com/service-support/support-warranty/repair-processing/>

Products Solutions How to buy **Service & Support** Company myLANCOM  SECURE. NETWORK

LANCOM > Service & Support > Support & Warranty > Repair processing



Repair processing

With your LANCOM network products, you not only enjoy the benefits of our many years of experience in data communication, but also our excellent service.

Under warranty ▾ Out of warranty Option transfer

Under warranty

You have booked an advance exchange?

Please contact us by telephone for defective devices with [LANCOM Warranty Advanced Option](#): +49 (0) 2405 499 36 - 210.
(Please have the serial number of your device as well as the license number of your option ready.)

Main escalation contact detail: **services@lancom.de**

Appendix E AAPP program

E.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.

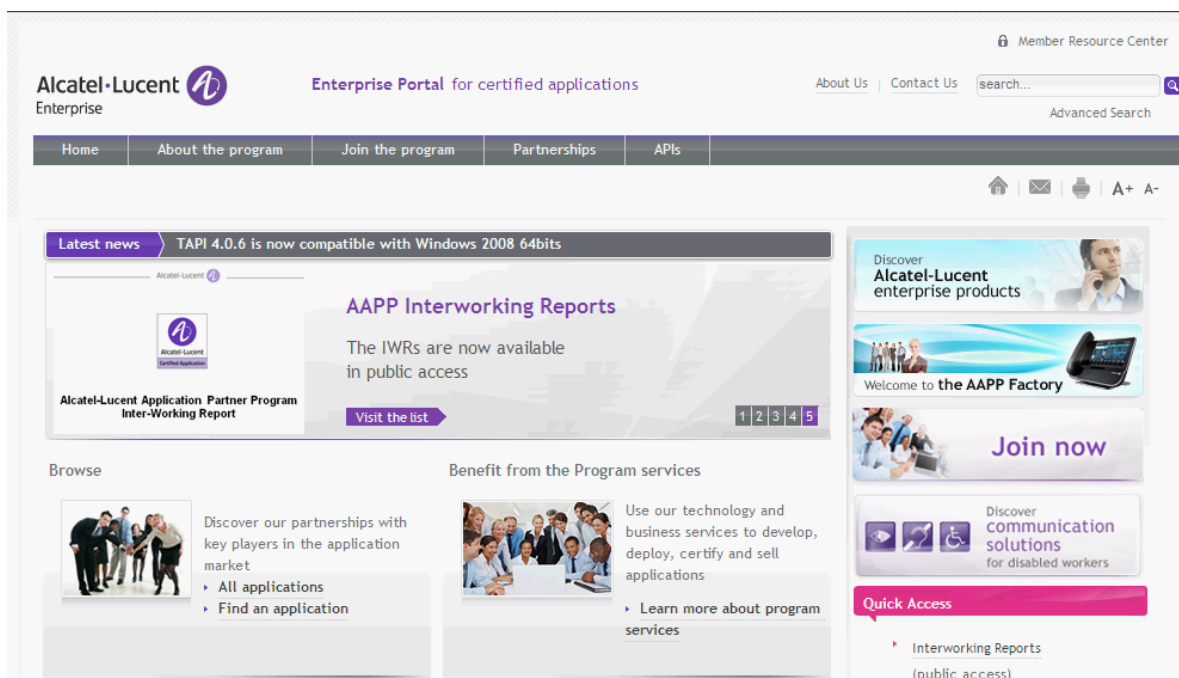
The Alcatel-Lucent Application Partner Program (AAPP) has two main objectives:

- 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.

E.2. Web site

The Application Partner Portal is a website dedicated to the AAPP program and where the InterWorking Reports can be consulted. Its access is free at <https://www.al-enterprise.com/en/partners/aapp>



The screenshot shows the Alcatel-Lucent Enterprise Portal for certified applications. The header includes the Alcatel-Lucent logo, the text 'Enterprise Portal for certified applications', and navigation links for 'About Us' and 'Contact Us'. A search bar is also present. The main navigation menu includes 'Home', 'About the program', 'Join the program', 'Partnerships', and 'APIs'. The main content area features a 'Latest news' section with a headline 'TAPI 4.0.6 is now compatible with Windows 2008 64bits'. Below this is a large banner for 'AAPP Interworking Reports' with the text 'The IWRs are now available in public access' and a 'Visit the list' button. To the right, there are several promotional tiles: 'Discover Alcatel-Lucent enterprise products', 'Welcome to the AAPP Factory', and 'Join now'. At the bottom, there are two 'Browse' sections: 'Discover our partnerships with key players in the application market' with links for 'All applications' and 'Find an application', and 'Benefit from the Program services' with a link for 'Learn more about program services'. A 'Quick Access' section at the bottom right highlights 'Interworking Reports (public access)'.

E.3. Enterprise.Alcatel-Lucent.com

You can access the Alcatel-Lucent Enterprise website at this URL: <https://www.al-enterprise.com>

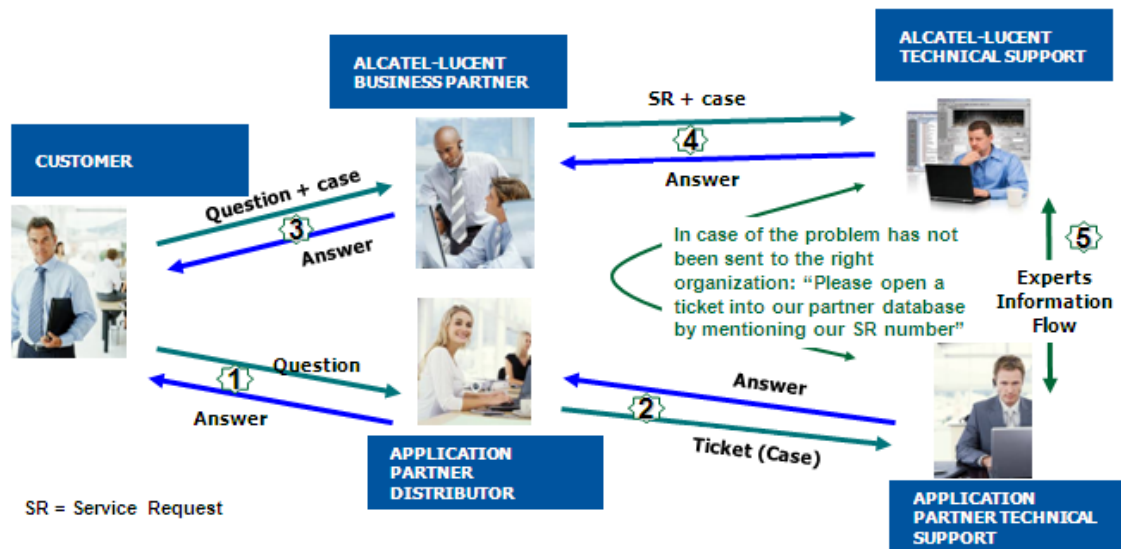
Appendix F AAPP Escalation process

F.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

F.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 cannot 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 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://www.al-enterprise.com/en/partners/aapp>) 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.

F.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 analyse.

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.

F.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): <https://www.al-enterprise.com/en/partners/aapp>
- 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: Ebq_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