ALE Application Partner Program
Inter-Working Report

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

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

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Certification overview

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):
Reviewer(s): Rachid Himmi

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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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:

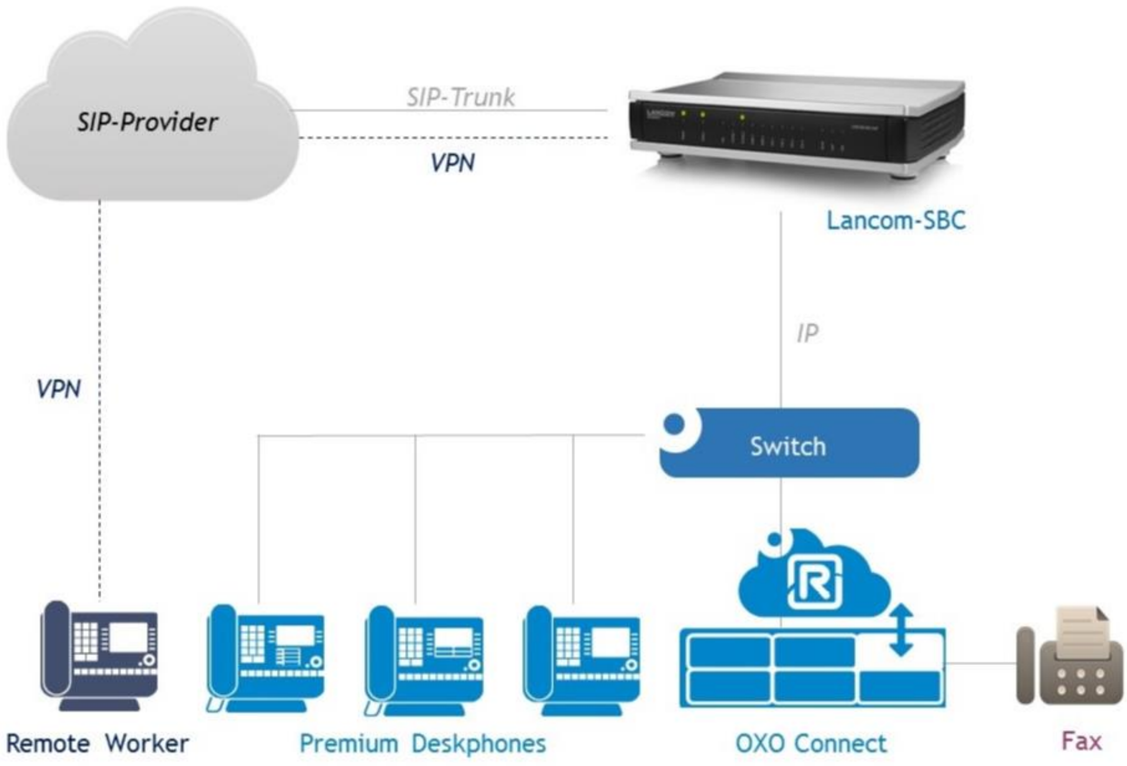
For instance, it enables the secure separation of external (insecure) and internal (secure) networks.

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 adaptions.

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 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	**Codec G711 / G722** Select G711 as first codec on Provider Select G711 as first codec on SBC Select G722 as first codec on Premium Deskphone And G.711 A-law, G.711 mu-law, G.729 as other priority Call from external phone (PSTN) to Premium Deskphone Check that call is correctly established In all Case check audio quality	☐	☒	☐	
8-1-2	**Codec G729 / G711** Select G729 as first codec on Provider Select G729 as first codec on SBC Select G711 as first codec on Premium Deskphone And G.729 as other priority Call from external phone to Premium Deskphone Check that call is correctly established In all Case check audio quality	☐	☒	☐	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	**Codec G723 / G711** Select G723 as first codec on Provider Select G723 as first codec on SBC Select G711 as first codec on Premium Deskphone Call from external phone to Premium Deskphone Check that call is correctly established In all Case check audio quality	☐	☒	☐	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.
8-1-4	**Codec G723 / G711** Select G723 as first codec on Provider Select G723 as first codec on SBC Select G711 as first codec on Premium Deskphone Call from external phone to Premium Deskphone Check that call is correctly established In all Case check audio quality	☐	☒	☐	Case is equal to case 8-1-3 ??

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	**Outgoing call with DTMF RFC 2833** 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	☐	☒	☐	Outgoing Call Codec setting on IP-Touch 8028s G722 – Setting in LANCOM: Fallback on SIP Info
8-2-2	**Outgoing call with DTMF Sip Info** Call to external attendant using DTMF Sip Info Test DTMF return.	☒	☐	☐	Not supported in OXO

	Call an IVR and navigate to the corresponding menu and verify that DTMF is working Hang-up the call Then Hang-up				
8-2-3	**Outgoing call with DTMF Inband** 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 Hang-up the call Then Hang-up	☐	☒	☐	Setting in LANCOM: “Fallback on SIP Info”
8-2-4	**Call to External number from VPN connected Premium Deskphone** Call external number from VPN Premium Deskphone Check audio, then hang-up	☐	☒	☐	VPN IKE1, VPN IKE2 Audio is ok in both cases
8-2-5	**Outgoing call with DTMF RFC 2833 with VPN Premium Deskphone** 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	☐	☒	☐	VPN IKE1, VPN IKE 2 DTMF is working
8-2-6	**Outgoing call with DTMF Sip Info with VPN Premium Deskphone** 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 Hang-up the call Then Hang-up	☒	☐	☐	**DTMF SIP Info not available in OXOC**
8-2-7	**Outgoing call with DTMF Inband with VPN Premium Deskphone** 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	☐	☒	☐	VPN IKE1, VPN IKE 2, DTMF is working

	Hang-up the call Then Hang-up				

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	**ExtCall to Premium Deskphone Timeout** 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	☐	☒	☐	Audio is ok and keeping for min 5 mins.
8-3-2	**ExtCall to Premium Deskphone Display** Call from Ext-PSTN to the DID configure on Premium Deskphone 1 Check display Answer the call and check audio. Then Hang-up	☐	☒	☐	Display OK, Audio OK
8-3-3	**ExtCall to Premium Deskphone Display multiline** 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	☐	☒	☐	Both Display OK, Both Audio OK. Music on Hold from OXO for both ext. parties
8-3-4	**Call from External number to VPN connected Premium Deskphone** Make a call to External number Answer the call on VPN Premium Deskphone Check audio then hang-up	☐	☒	☐	IKE V1, IKE V2, Display OK, Audio OK
8-3-5	**Call from VPN Premium Deskphone to Premium Deskphone 1**	☐	☒	☐	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	☐	☒	☐	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	☐	☒	☐	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	☐	☒	☐	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	☐	☒	☐	FXS is analog Port on OXOC, Tones and display ok

	Keep the calls for next test				
8-4-3	**Three party conferences initiated from OXO set** With Ext PSTN call Premium Deskphone 1 Answer and keep the call With IPtouch1 call Premium Deskphone 2 Answer and keep the call With Premium Deskphone 1 start a conference Check audio, Display, then hang-up.	☐	☒	☐	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

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:

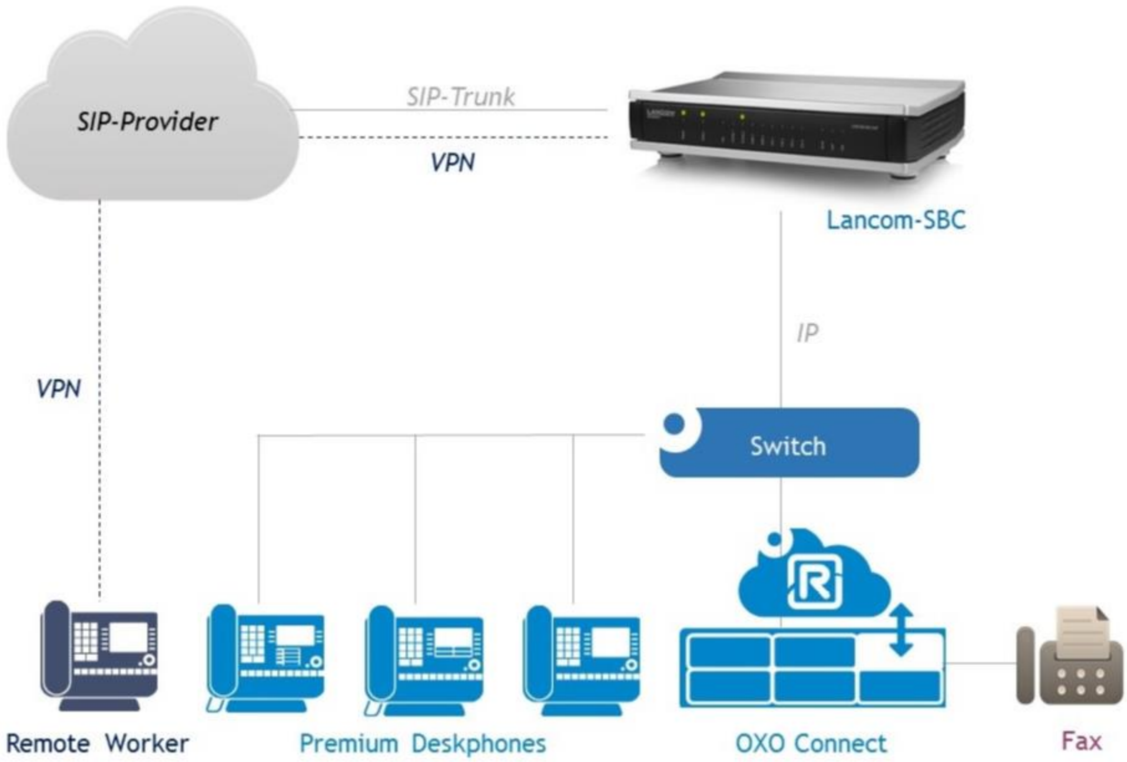
1. The transferee calls the Transferor
2. The transferor calls the transfer target. The transferee is on Hold. Transfer target picks up the call and call is established with the transferor
3. The transferor executes the transfer. The transferor drops the call. The transferee is now on line with the Transfer 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	☐	☒	☐	Shown number is the Premium Deskphone 1
8-5-2	Semi-attended	Premium Deskphone 2	Premium Deskphone 1	ExtNum	☐	☒	☐	Shown number is still the Premium Deskphone 1
8-5-3	Attended	Premium Deskphone 2	Premium Deskphone 1	ExtNum	☐	☒	☐	Shown number is still the Premium Deskphone 1

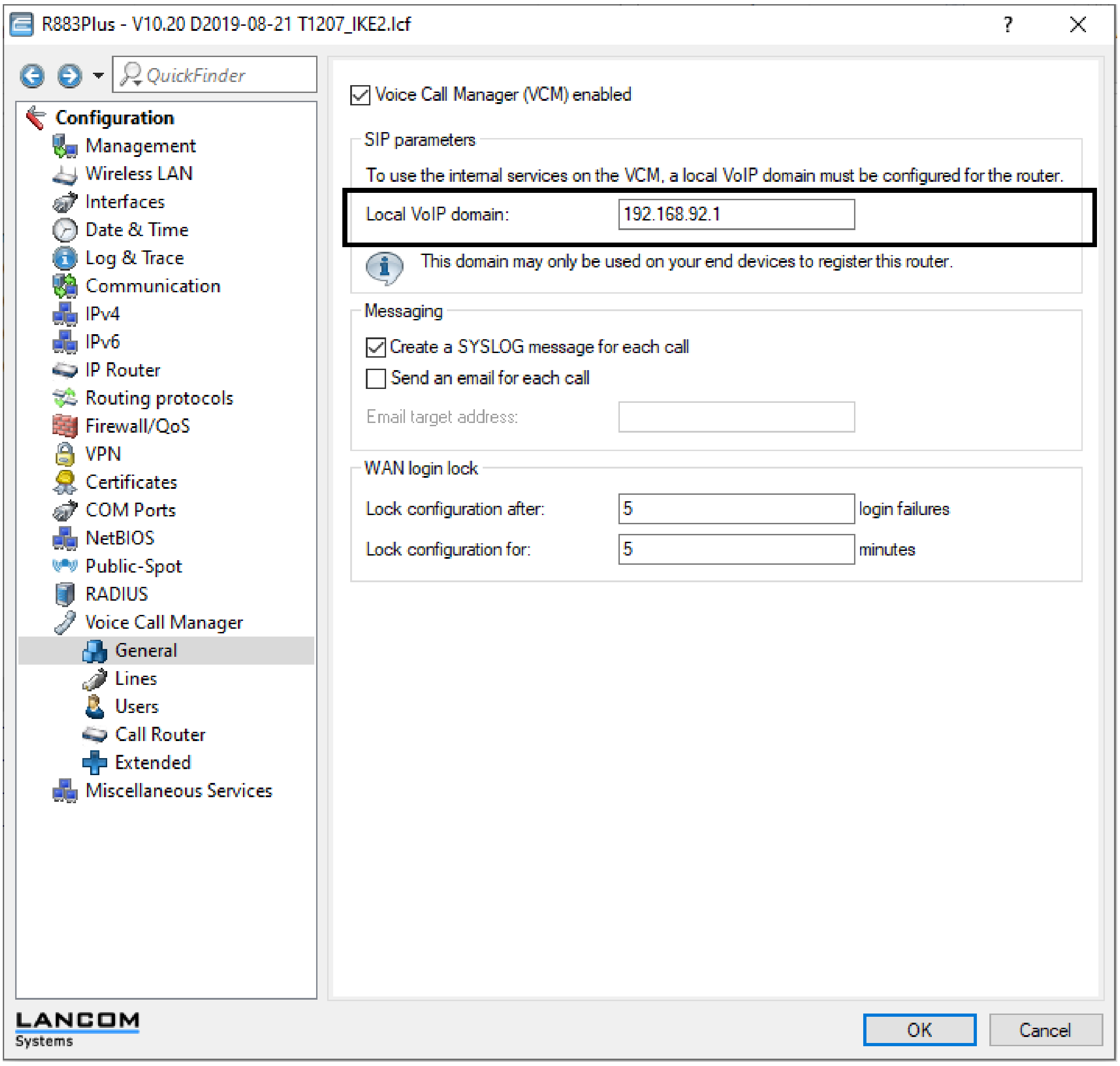
Appendix A AAPP member's Application description

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.

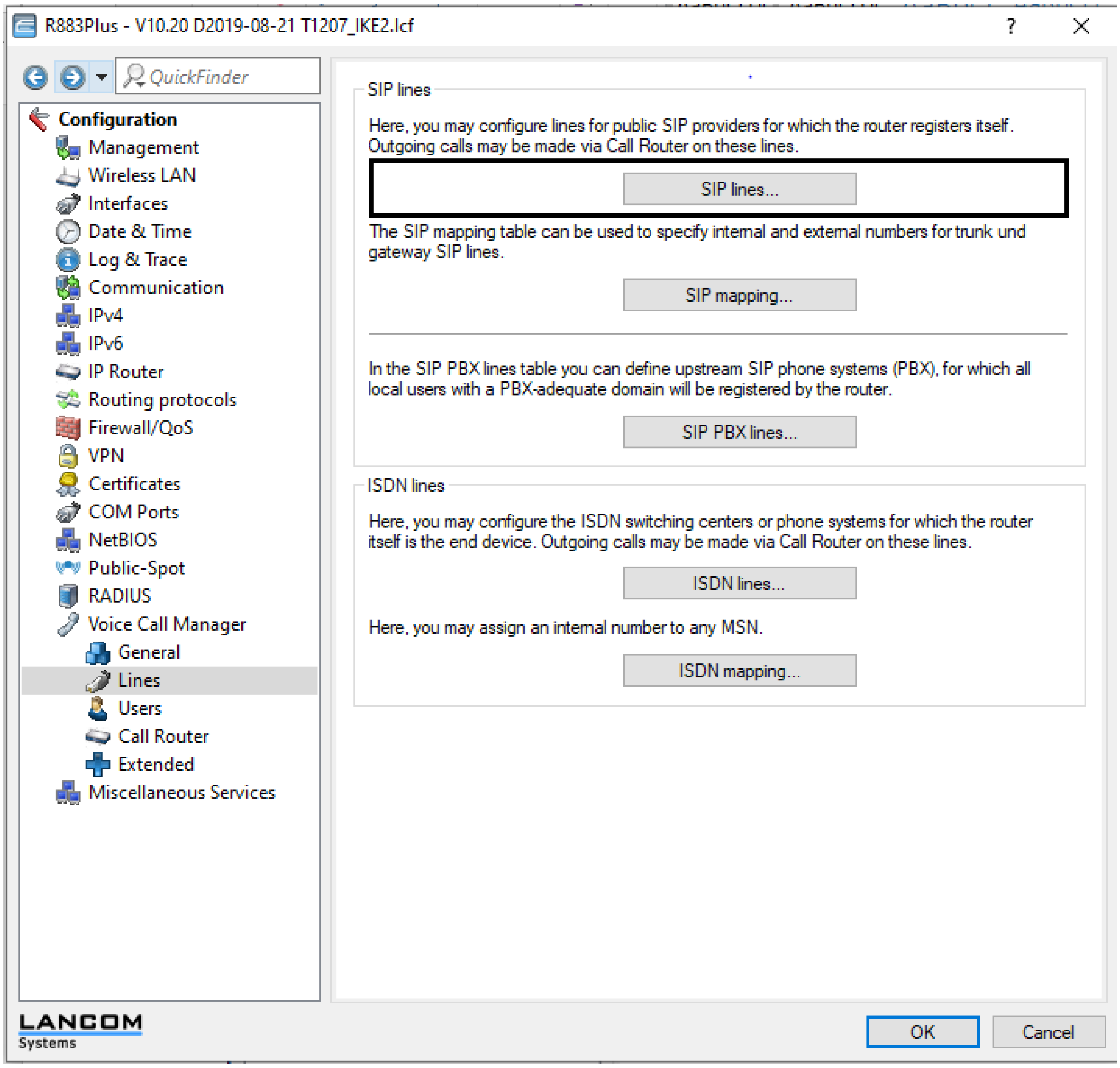


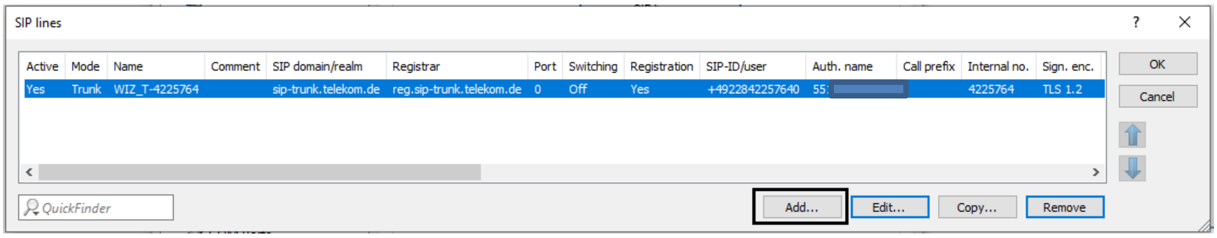
Appendix B Configuration requirements of the AAPP member's application

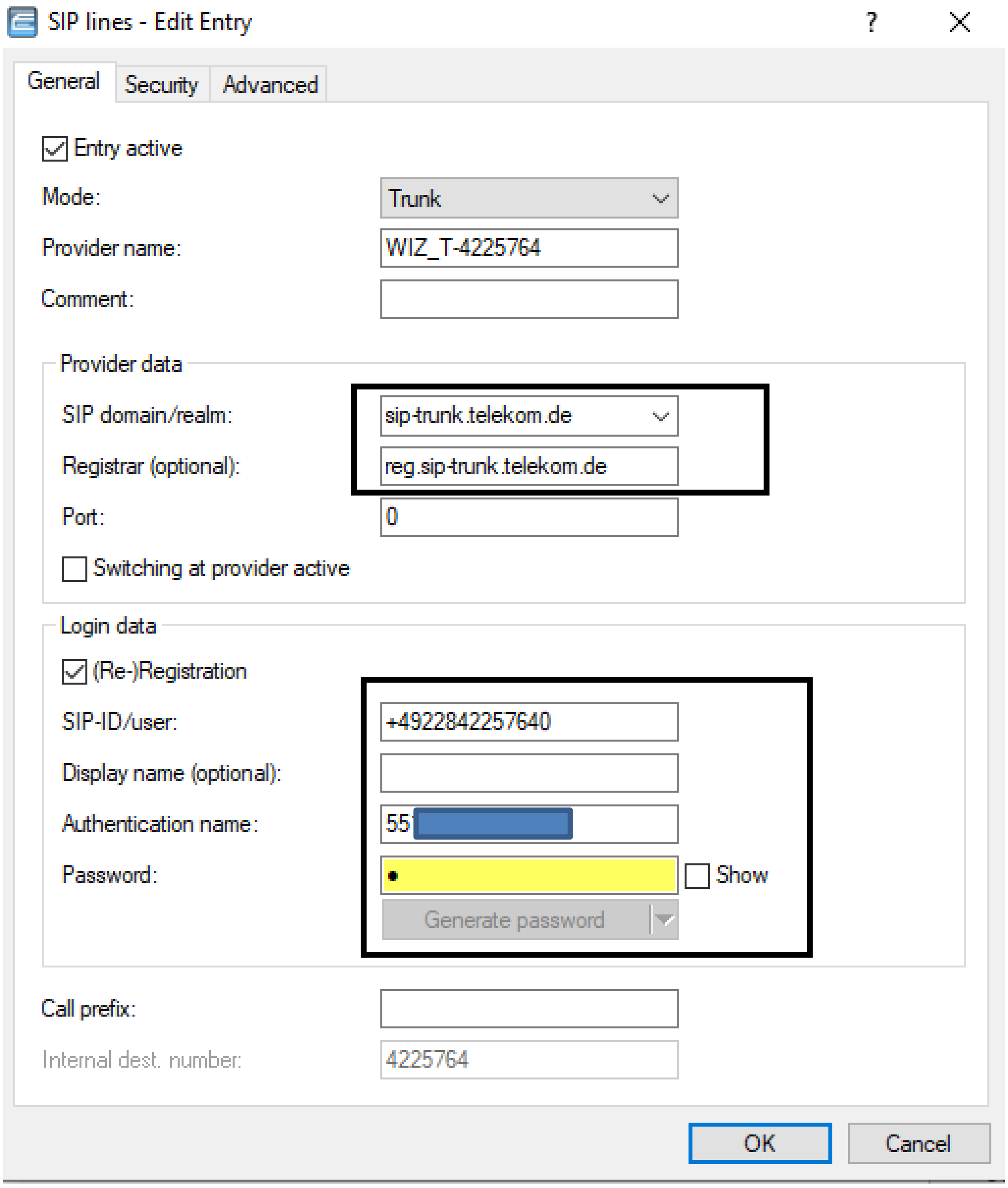
LANconfig > Configuration > Voice Call Manager > General - Local VoIP domain (e.g. local IP-Address of LANCOM)



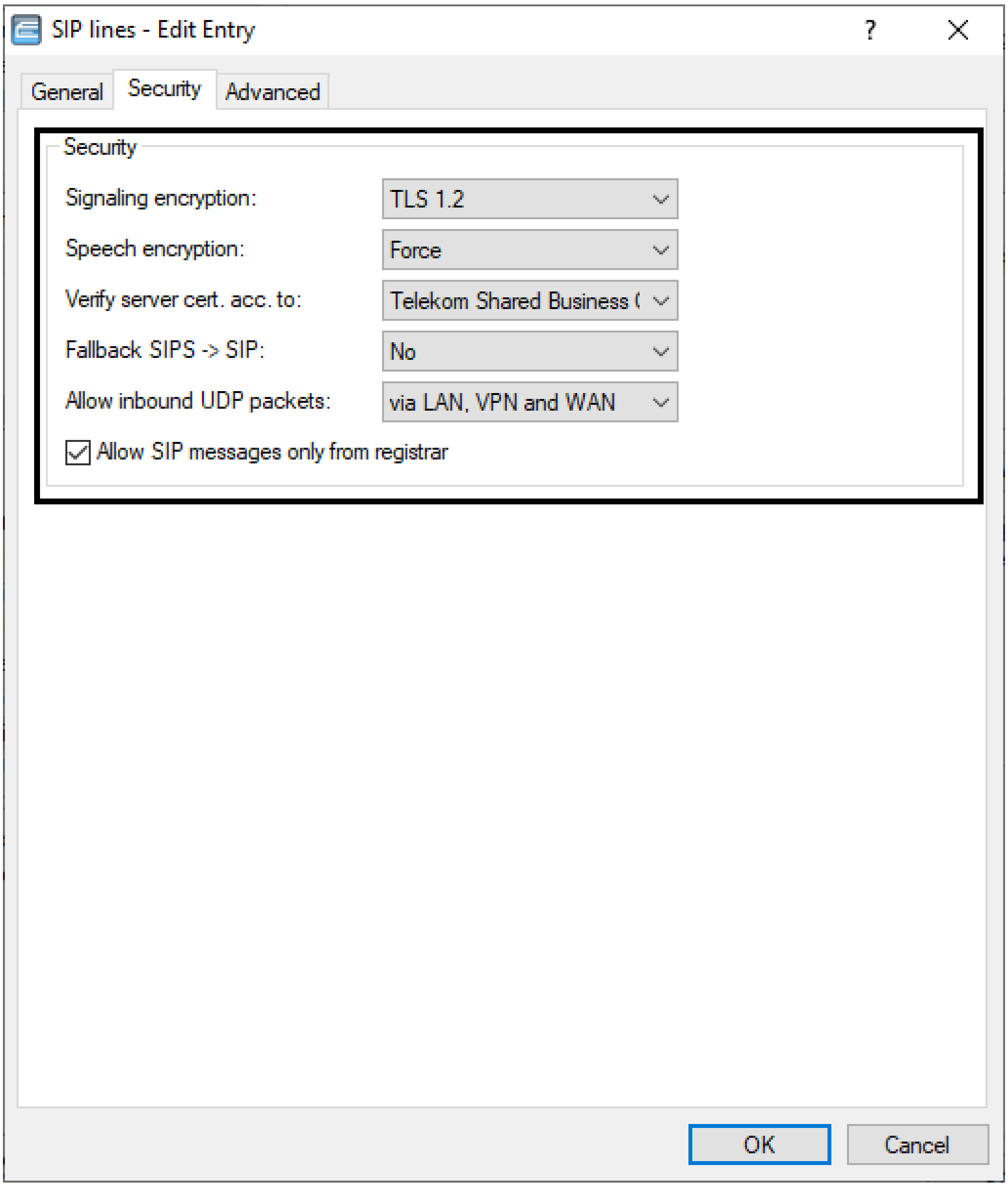
Create a SIP Line >(Trunk):
LANconfig > Configuration > Voice Call Manager > Line – SIP lines …

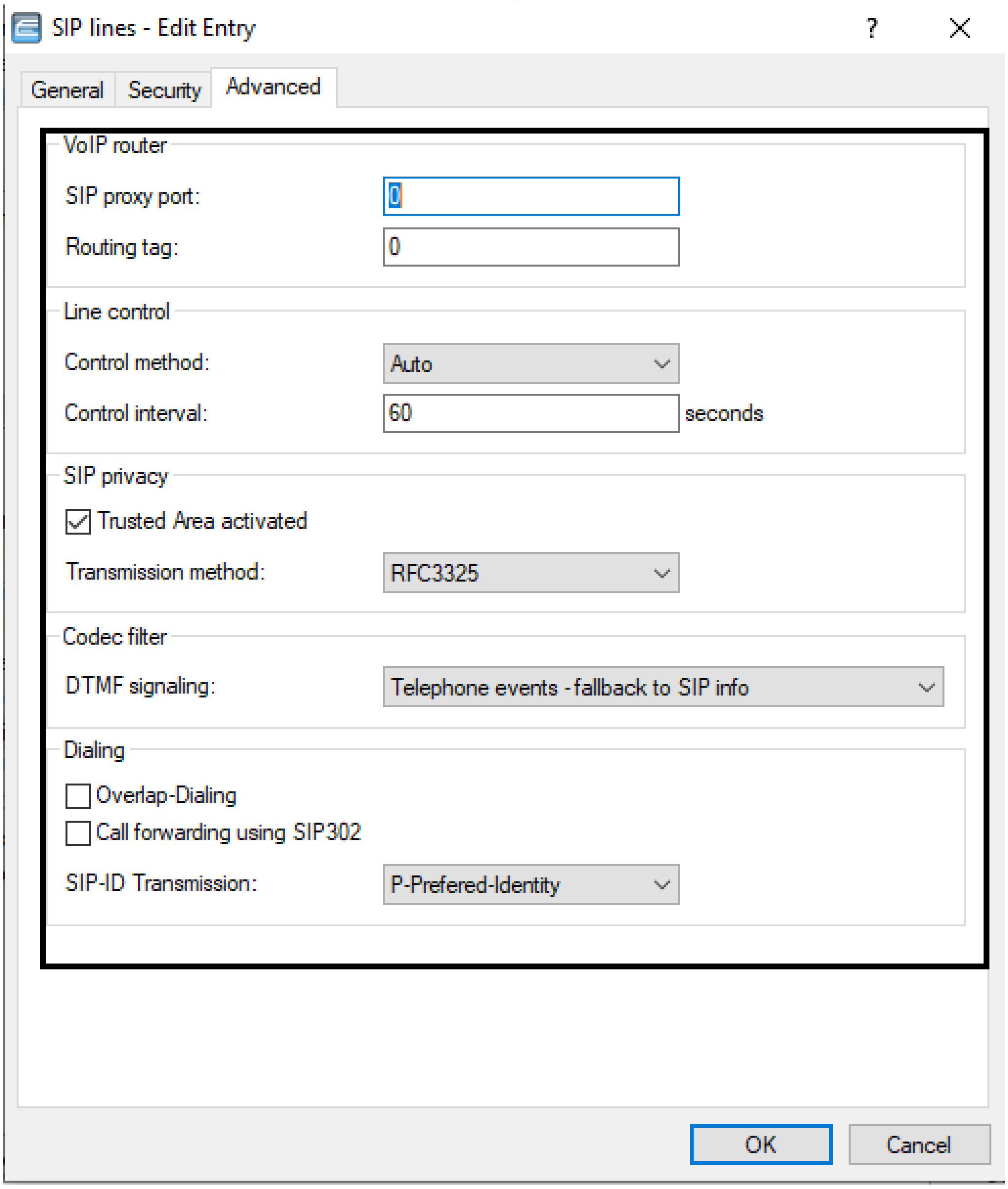




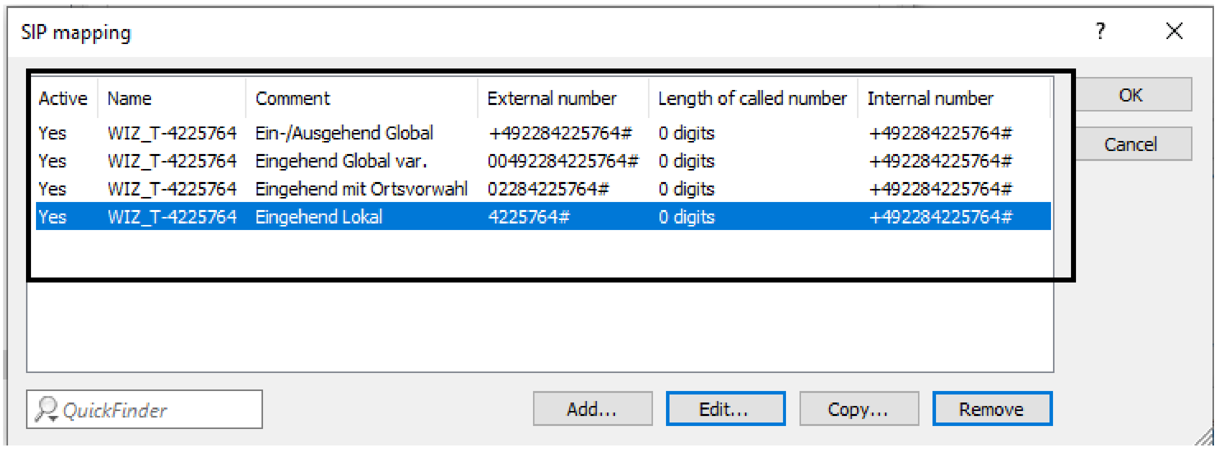

SIP lines - Edit Entry
General
Security
Advanced
Entry active
Mode:
Trunk
Provider name:
WIZ_T-4225764
Comment:
Provider data
SIP domain/realm:
sip-trunk.telekom.de
Registrar (optional):
reg.sip-trunk.telekom.de
Port:
0
Switching at provider active
Login data
(Re-)Registration
SIP-ID/user:
+4922842257640
Display name (optional):
Authentication name:
55[redacted]
Password:
Show
Generate password
Call prefix:
Internal dest. number:
4225764
OK
Cancel

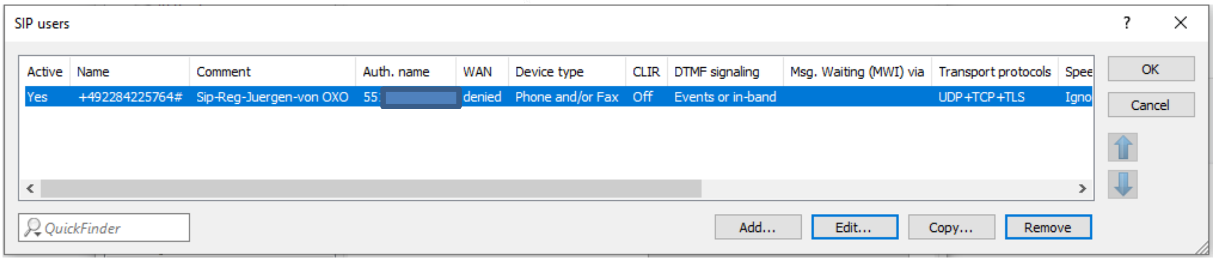
LANconfig > Configuration > Voice Call Manager > Line > SIP lines … => Security Settings for Signaling and Speech encryption



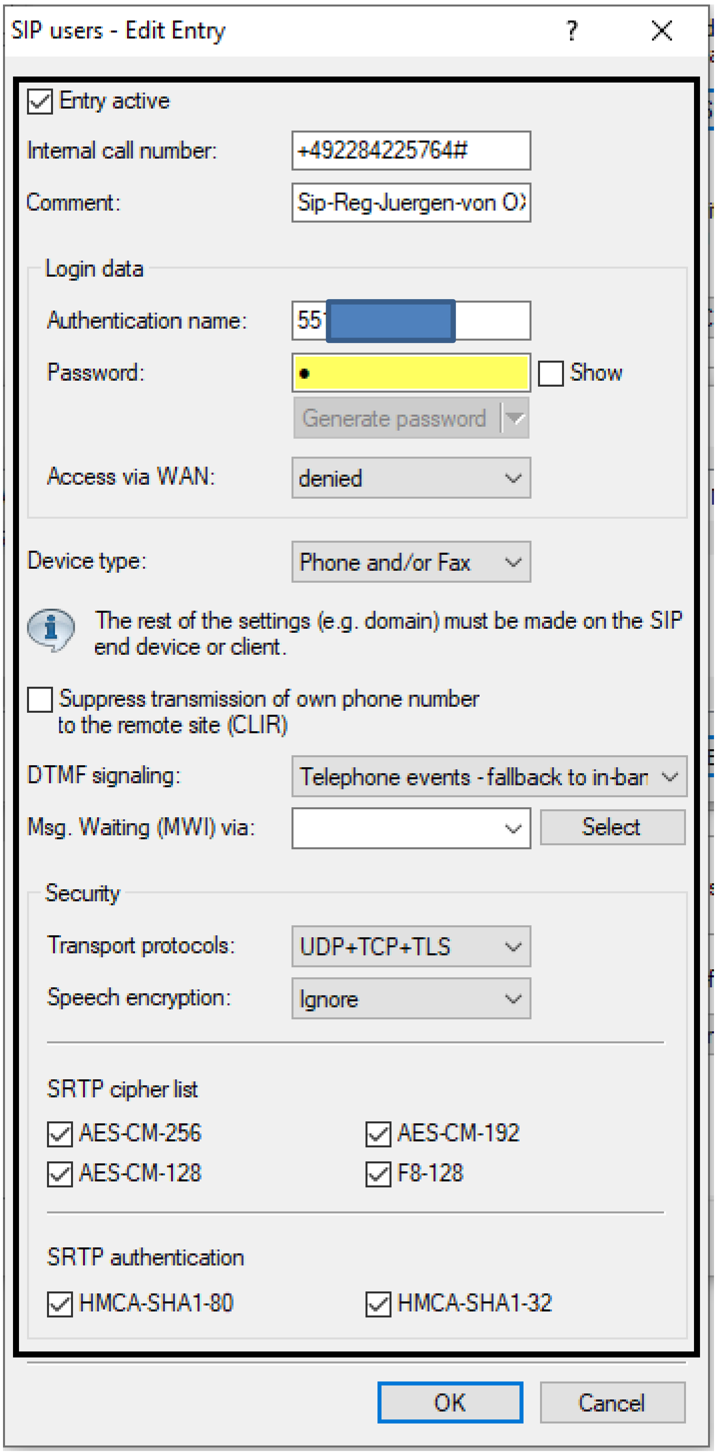


LANconfig > Configuration > Voice Call Manager > Line – SIP Mapping:

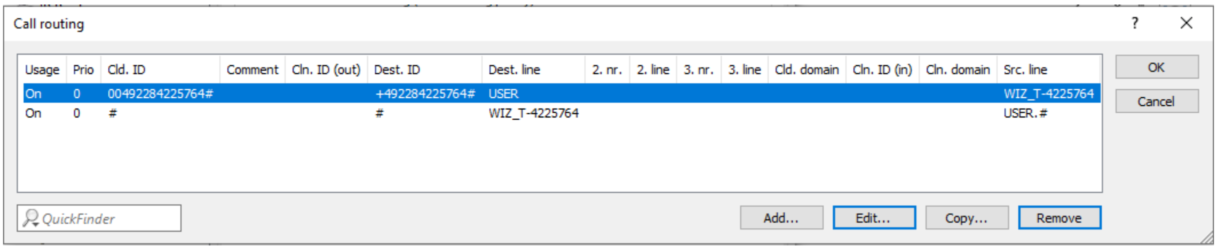


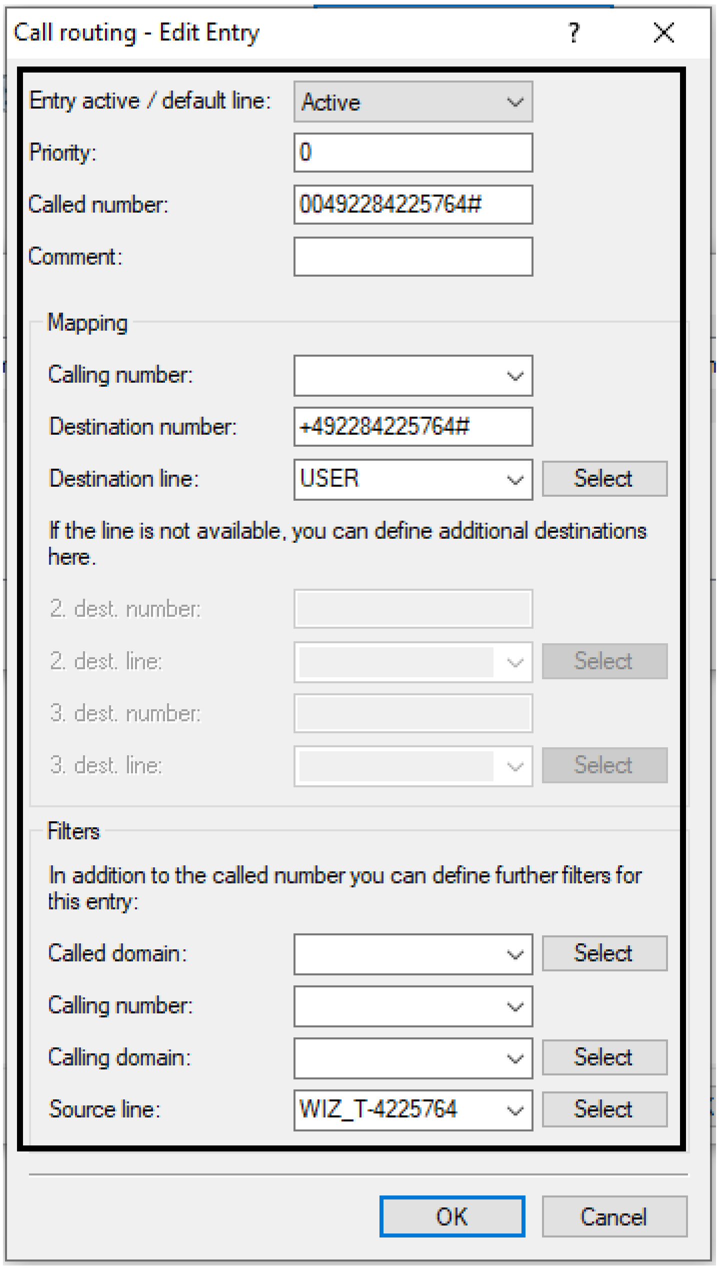


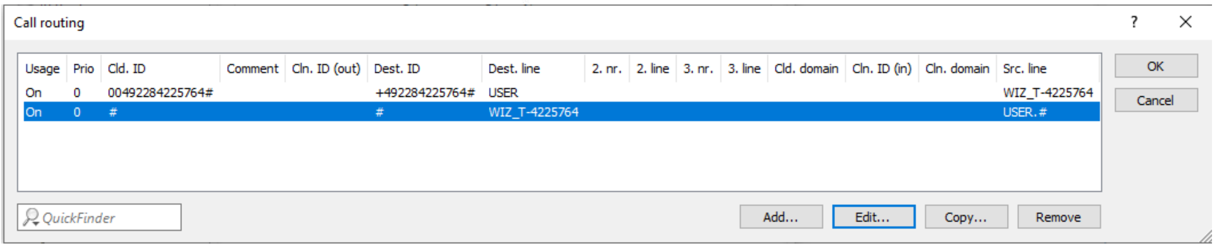
Create a SIP User (SIP Trunk LAN):
LANconfig > Configuration > Users > SIP User …

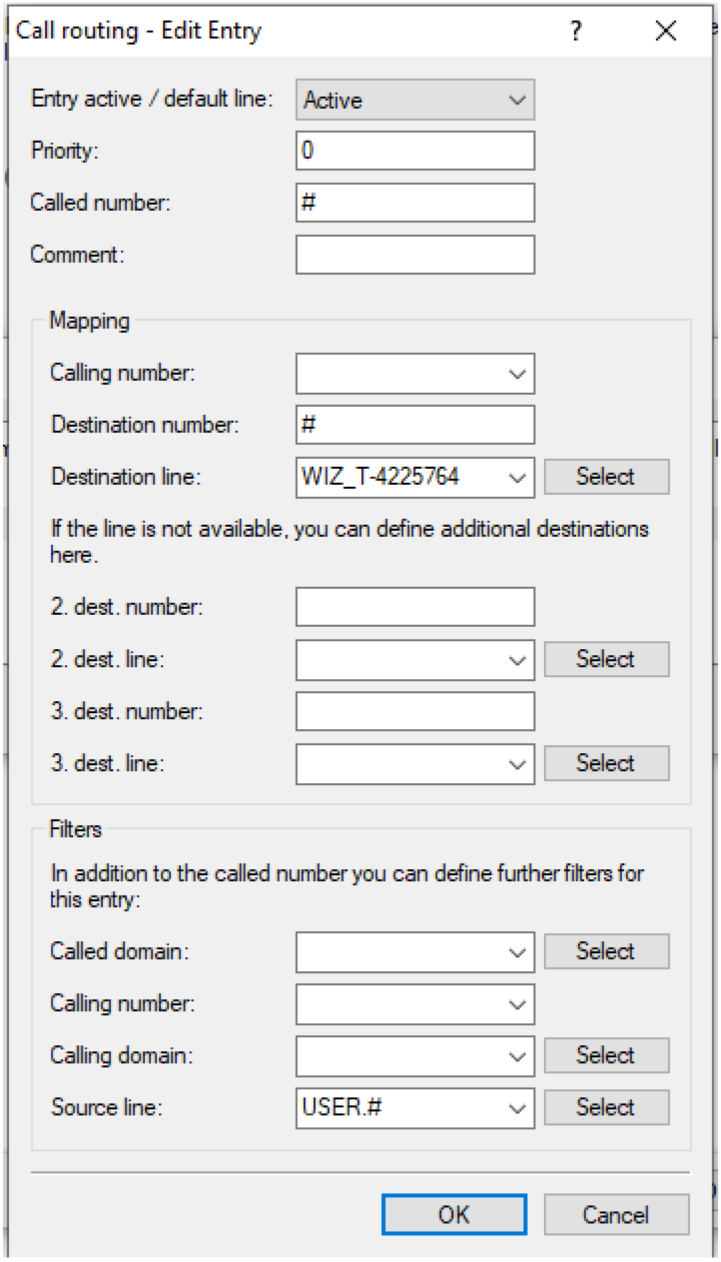


Create Call-Routen for outgoing and incoming Calls:
LANconfig > Configuration > Voice Call Manager > Call Router

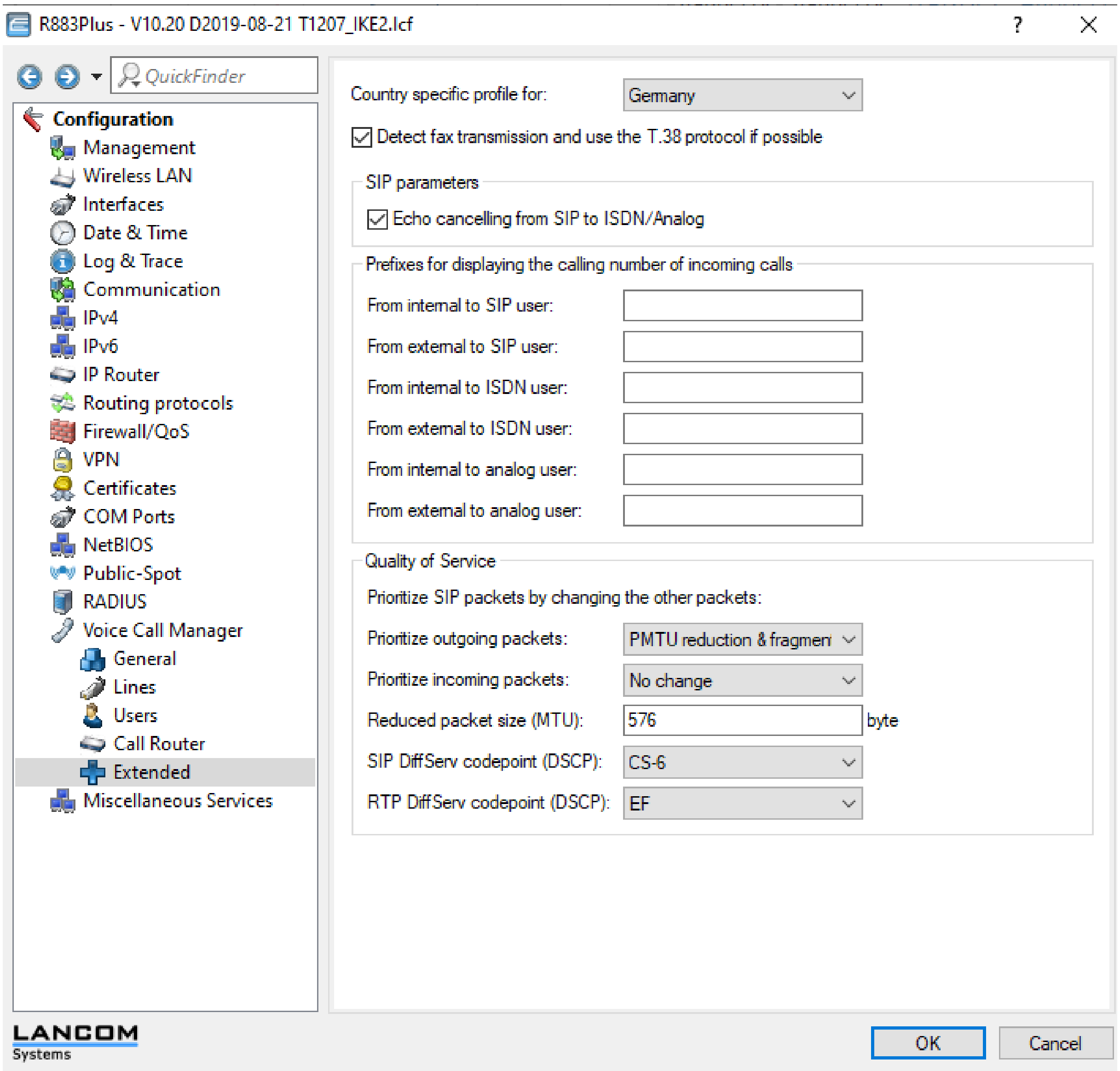





Call routing
?
Usage Prio Cld. ID Comment Cln. ID (out) Dest. ID Dest. line 2. nr. 2. line 3. nr. 3. line Cld. domain Cln. ID (in) Cln. domain Src. line
On 0 00492284225764# +492284225764# USER WIZ_T-4225764
On 0 # # WIZ_T-4225764 USER.#
OK
Cancel
QuickFinder
Add... Edit... Copy... Remove


Call routing - Edit Entry
?
Entry active / default line: Active
Priority: 0
Called number: #
Comment:
Mapping
Calling number:
Destination number: #
Destination line: WIZ_T-4225764 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: USER.# Select
OK
Cancel

LANconfig > Configuration > Voice Call Manager > Extended:



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

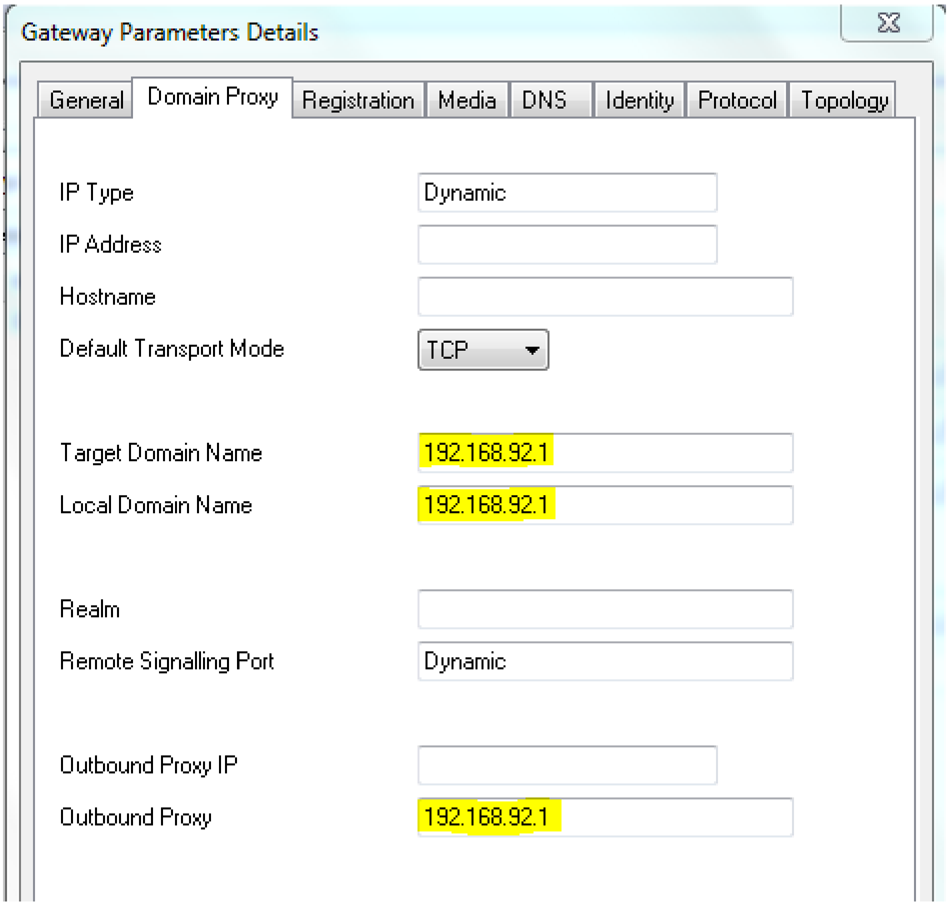
C.1. Licenses

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

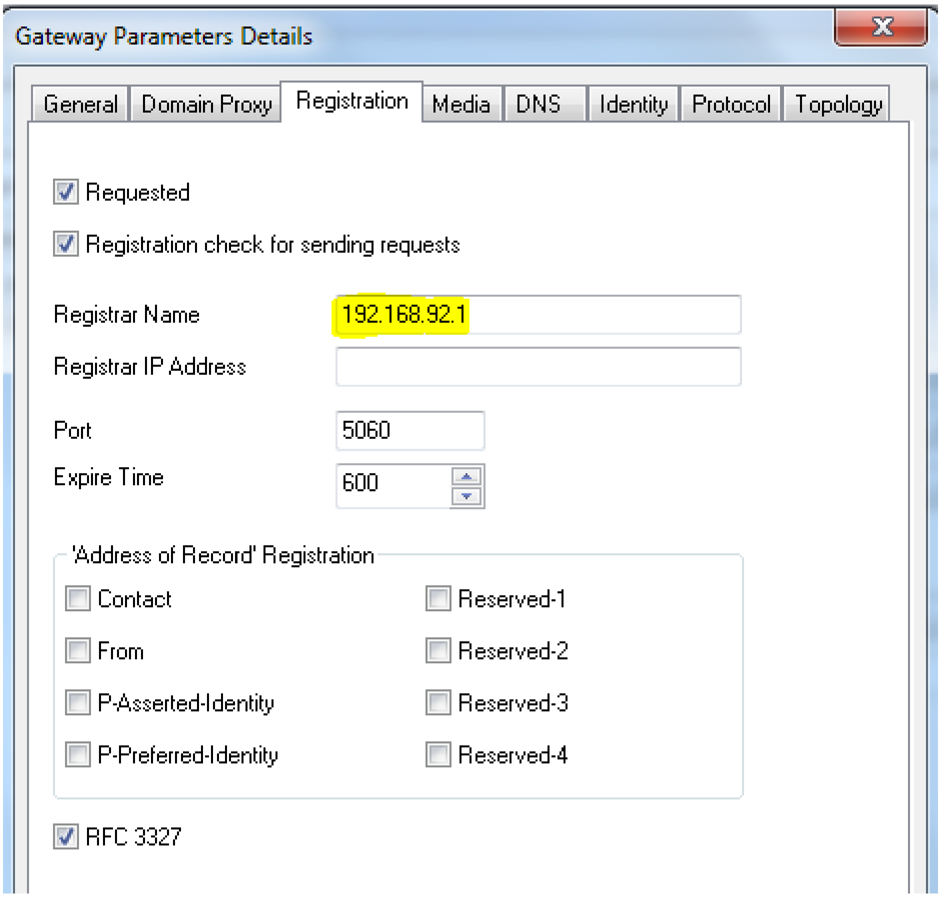
C.2. Sip trunk configuration

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.

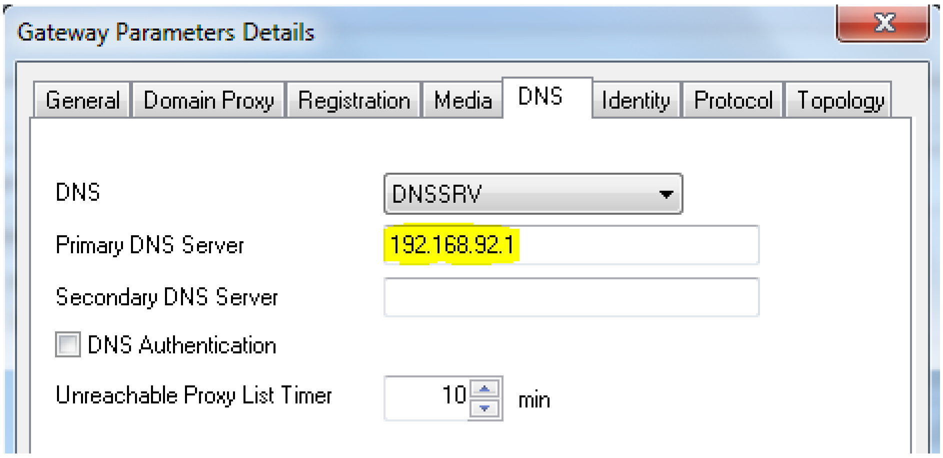
Adapt Target Domain,Local Domain and Outbound Proxy to LANCOM Router LAN IP address:



Adapt Registrar Name to LANCOM Router LAN IP address:



Adapt DNS Server to LANCOM Router LAN IP address:



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

SIP Accounts

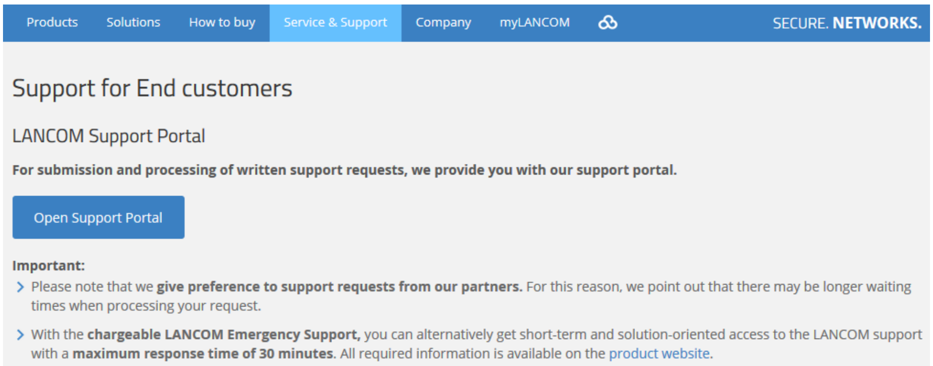
Index	Login	Passw...	Registered Username	Gateway Parameters Ind...	RFC 6140
1	55[redacted]	********	+492284221234	1 DT SIP TRUNK-SBC	Enabled

Appendix D AAPP member's escalation process

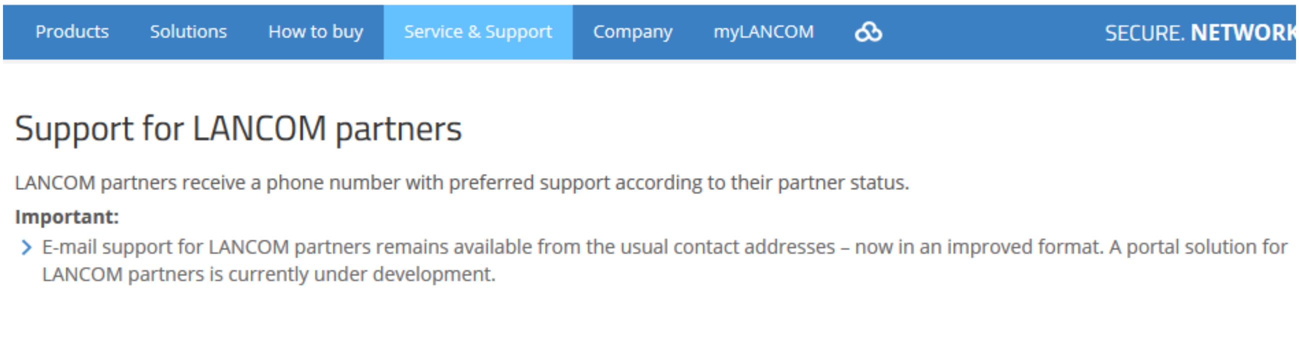
Installation is guided by Setup-Wizard
Specific configuration steps are explained by Screens Shots in the IWR

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/

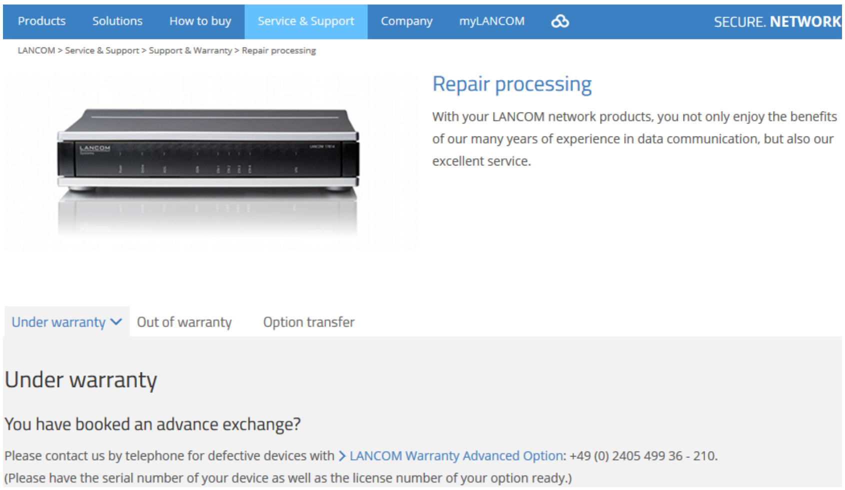
Escalation phase:
For End-Customers: >>> LANCOM Support Portal:
https://www.lancom-systems.com/service-support/support-warranty/support-contact/



For LANCOM Partners: >>> LANCOM Support Portal:
https://www.lancom-systems.com/service-support/support-warranty/support-contact/



Repair Processing (warranty) >>>
https://www.lancom-systems.com/service-support/support-warranty/repair-processing/



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.

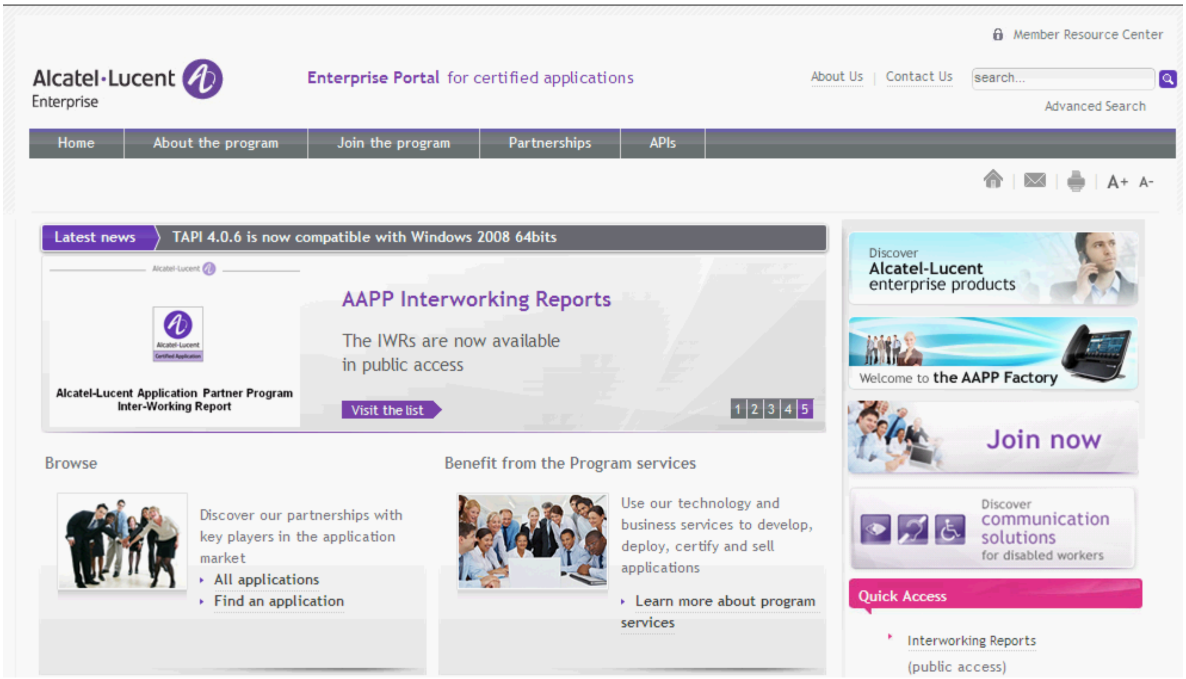
The Alcatel-Lucent Application Partner Program (AAPP) has two main objectives:

- **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



E.3. Enterprise.Alcatel-Lucent.com

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

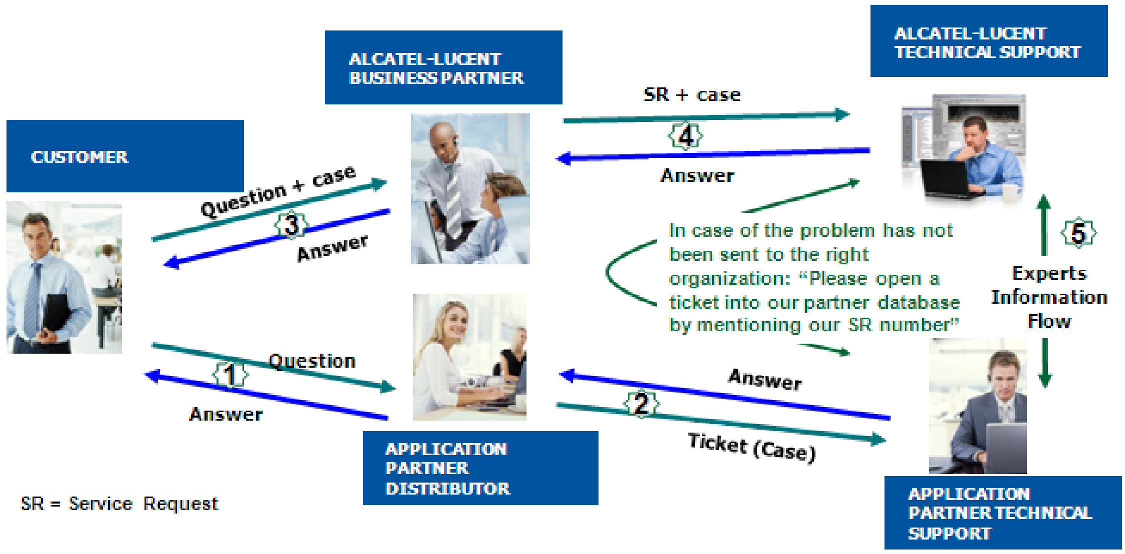
Appendix F AAPP Escalation process

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: 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

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