ALE Application Partner Program
Inter-Working Report

Partner: **AudioCodes**
Application type: **Session Border Controler**
Application name:
CloudBond365™ Enterprise Box Edition
Alcatel-Lucent Enterprise Platform:
OmniPCX Enterprise™ and **OpenTouch™**

Certification overview

Date of the certification	June 2016

ALE International representative	Claire Dechristé
AAPP member representative	Serge Leclercq

Alcatel-Lucent Enterprise Communication Platform	OmniPCX Enterprise OpenTouch BE/MS
Alcatel-Lucent Enterprise Communication Platform release	OXE R11.2 OTMS R2.2
AAPP member application release	AudioCodes CloudBond 365 V5.0
Application Category	SBC Collaboration & UC

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Reviewer(s): Jean-Paul SPARD, Serge LECLERCQ

Revision History

Edition 1: creation of the document – *June2016*

Test results

☐ Passed ☐ Refused ☐ Postponed

☑ Passed with restrictions

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

IWR validity extension

None

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1 Introduction

This document is the result of the certification tests performed between the AAPP member's application and Alcatel-Lucent Enterprise's platform.

It certifies proper inter-working with the AAPP member's application.

Information contained in this document is believed to be accurate and reliable at the time of printing. However, due to ongoing product improvements and revisions, ALE International cannot guarantee accuracy of printed material after the date of certification nor can it accept responsibility for errors or omissions. Updates to this document can be viewed on:

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

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

1.1 Glossary

Acronym	Meaning
OXE	OmniPCX Enterprise
OT	OpenTouch
Transferee	The party being transferred to the transfer target
Transferor	The party initiating the transfer
Transfer target	The new party being introduced into a call with the transferee
Blind or semi-attended transfer	The transferor having a session in hold state with the transferee and initiating the transfer by a consultation call to the target performs the transfer while the target is in ringing state
Attended transfer or transfer on conversation	The transferor waits to be in conversation state with the target before completing the transfer
MoH	Music On Hold
PSTN(analogique card on the OXE)	Public Switched Telephone Network
SBC	Session Border Controller
FE	Lync Front End server

2 Validity of the InterWorking Report

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

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

A new release is identified as following:
a “Major Release” is any x. enumerated release. Example Product 1.0 is a major product release.
a “Minor Release” is any x.y enumerated release. Example Product 1.1 is a minor product release

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

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

3 Limits of the Technical support

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

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

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

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

3.1 Case of additional Third party applications

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

4 Application information

Application commercial name: Microsoft Lync 2013 (5.0.8308)

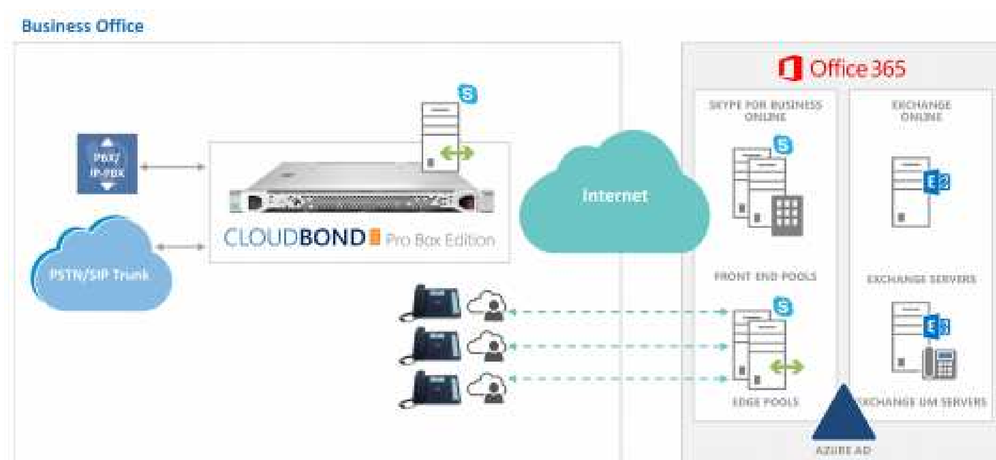
Audiocodes SBC 7.00A.029.005

Interface type: SIP

Brief application description:

AudioCodes CloudBond 365 is a complete Skype for Business enterprise voice solution for Office 365 customers. It provides connectivity to the cloud on the one hand and to the PSTN on the other.

As Office 365 Skype for Business Online does not currently provide Enterprise Voice , PSTN access or PBX replacement features, a user who need these capabilities must be registered into an On-premises Skype for Business server (like CloudBond 365)



CloudBond 365:

- Enables the migration to Skype for Business enterprise voice providing the option to have full PBX features or be homed in Cloud PBX
- Offers a complete Skype for Business solution that integrates the connectivity and management tools in one package
- Comes in different box edition sizes, as a virtualized appliance, or as a management pack and can adapt to different architectures and business models
- Delivers special management interfaces for Office 365 and the corporate Active Directory, which automate the Hybrid or Cloud PBX connection
- Supports new user management capability to simplify user policy management
- Offers backup and restore support
- Includes desk phone management capabilites

A Skype for Business on-premises deployment, such as CloudBond 365, can take advantage of several features of Office 365:

1. Office 365 can provide the Exchange Unified Messaging component to Skype for Business, allowing voicemail facilities, and some Automated Attendant facilities.
2. Office 365 can provide the Outlook Client for Skype for Business, showing Skype for Business presence information for contacts, for calendar items, and allowing the scheduling of Conferences.
3. Skype for Business Online and Skype for Business On-premises can share a SIP domain, allowing users who do not require Enterprise Voice features to be hosted entirely in the cloud, while still being part of your larger Skype for Business environment.

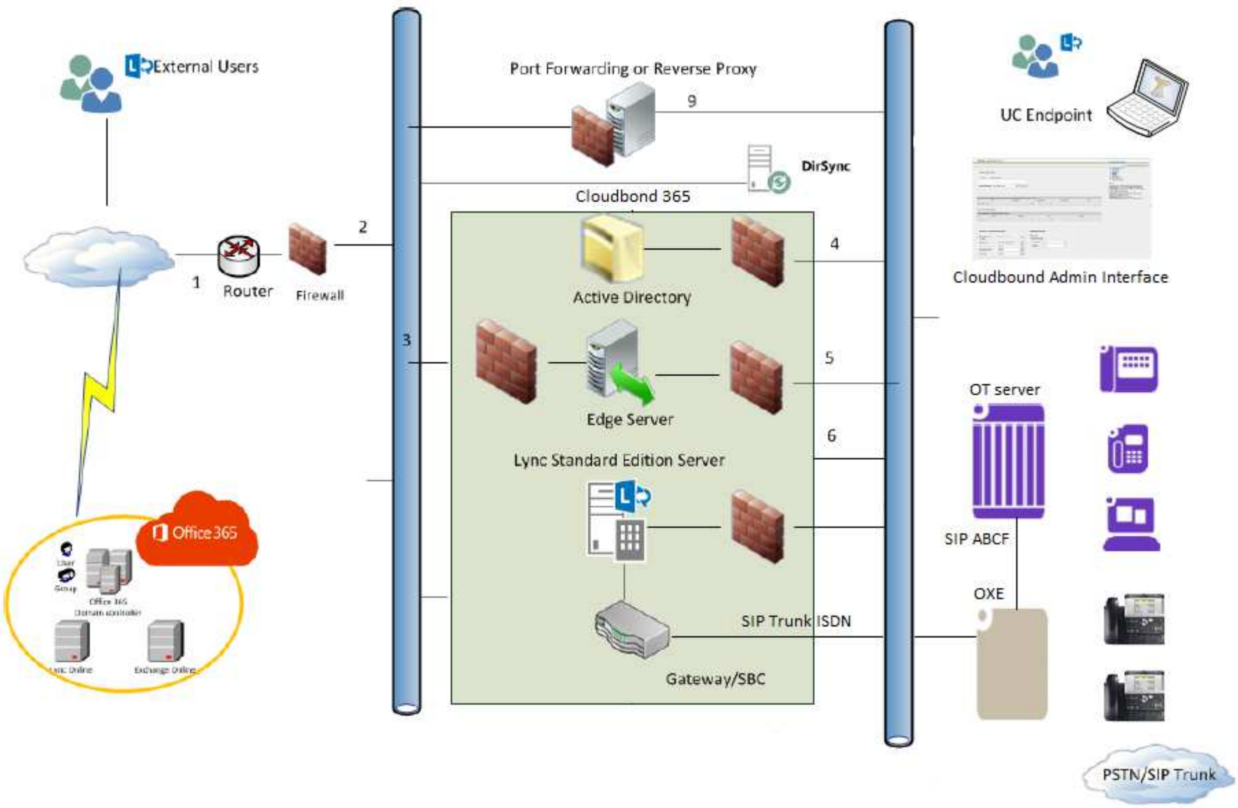
O365 licences:

OFFICE 365 ENTERPRISE SUITES

Features *Office 365 Services*	E1 $8	E3 $20	E5 $35
Business Class Email and Calendars *Exchange Online*	50 GB	Unlimited	Unlimited
Social, Video, Sites, Task Management *Yammer, O365 Video, SharePoint Online, Planner*	New ●	New ●	●
IM, Online Meetings, Meeting Broadcast *Skype for Business*	New ●	New ●	●
File Storage, Sharing, Information Discovery *OneDrive for Business, Delve*	●	●	●
Office Online	●	●	●
Office Client Apps *Office 365 ProPlus*		●	●
Archiving, Rights Management, Data Loss Prevention, Encryption		New ●	●
Equivio Analytics for eDiscovery, Secure Attachments and URLs, Access Control			●
End User and Organizational Analytics *Power BI Pro, Delve Analytics*			●
Cloud PBX *Skype for Business*			●
PSTN Conferencing* *Skype for Business*			●
Enterprise Plan Add-ons			
PSTN Calling** *Skype for Business*			+$24
CRM Online Professional *Dynamics*			+$50

<u>For more information, refer to Audiocodes Cloudbond documentation.</u>

Global architecture including OT solution:



Microsoft Lync 2013 server connects to the SBC which is connected to the OXE system via an ISDN SIP Trunk.

5 Test environment

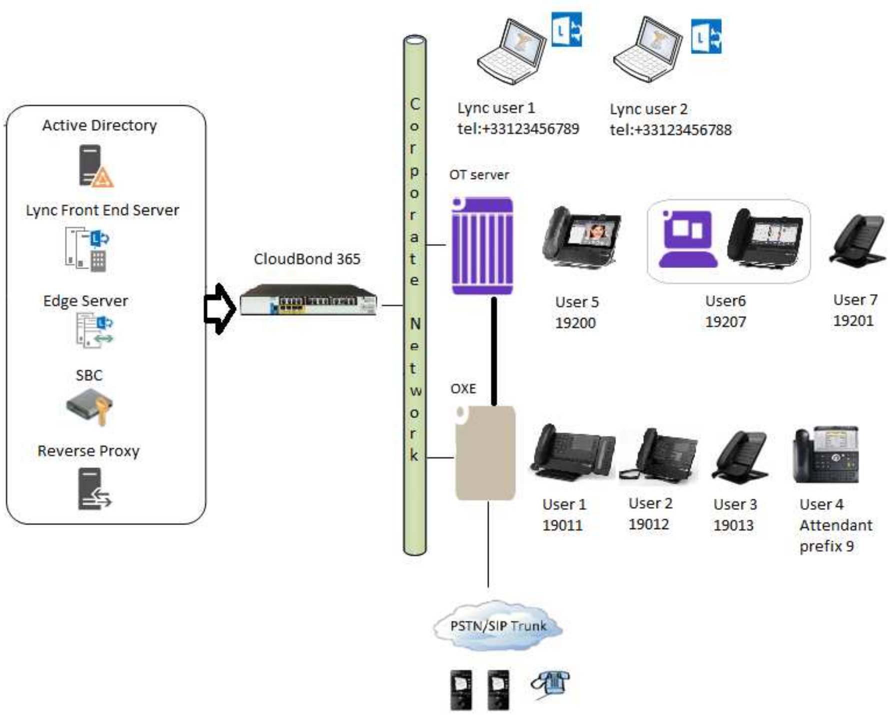
5.1 General architecture

The tests are performed on the Alcatel-Lucent Etesting platform in the following environment:

Alcatel-Lucent Communication Platform:
name: etesting9.etesting.lab
IP address: CPU A: 10.1.20.1

AudioCodes CloudBond 365:
domain: ac-onebox.com
Domain Controller IP address: 10.1.2.60
SBC IP address: 10.1.2.63

Microsoft Lync 2013:
Domain: ac-onebox.com
Default SIP domain: aapp-etesting.com
Simple's URL: https://meet.aapp-etesting.com/dialin, https://meet.aapp-etesting.com/meet
Lync Front End FQDN: UC-FE.ac-onebox.com
Lync Front End IP address: 10.1.2.61
FE External web services: ewslync.aapp-etesting.com
Lync Edge FQDN: UC-Edge.ac-onebox.com
Lync Edge internal IP address: 10.1.2.62
Edge Pool: sip.aapp-etesting.com



The Mediant 800 from the CloudBond box is used in SBC mode as a SIP/SIP gateway between the OmniPCX Enterprise and the Microsoft Lync server. It is needed to adapt OXE SIP implementation to Lync SIP specificities and vice-versa. It is declared on the OmniPCX Enterprise as an external SIP Gateway. An ISDN SIP trunk connects the OXE and the Mediant E-SBC (see section Appendix C : Lync 2013 Configuration).

On Microsoft Lync 2013, the SBC is seen as a PSTN gateway and the connection is done through a public SIP trunk (see Appendix B :)

5.2 Hardware configuration

Alcatel-Lucent Communication Platform:

Opentouch : HP Proliant DL380p Gen8
OXE : HP Proliant DL120 G6

AudioCodes Cloudbond (formerly Onebox 365) Server hosting Windows Lync server:

	 ONE BOX 365 PRO EDITION
Max. OneBox capacity	Up to **500** Users
Topology	Mid scale server-based appliance with Software SBC

Including preinstalled:	• Lync Server **Standard/Enterprise*** Edition:
	  • Front End Server
	  • Mediation Server
	  • Monitoring Server
	  • Edge Server
	• Dedicated Connection Tools
	  • Active Directory Connector
	  • Office 365 Connector
	• Windows Server 2012 R2 Emb **(x5)**
	• SQL Server Std 2012 Emb
	• **Reverse Proxy**
	• Software Session Border Controller (SBC)
	• Management and Configuration application
	• **Additional virtual machines**

Telephony Connectivity Options	• SBC Only (up to 150 sessions can be ordered) • External Media Gateway can be added (not included)
Server Spec	32GB RAM, 6 Core Processor, Dual Power Supply, 2HDD with RAID 1

5.3 Software configuration

Alcatel-Lucent Communication Platform:

OmniPCX Enterprise R11.2 l2.300.29.a
OTMS 11.0.017.004

AudioCodes Plateform : Mediant 800 SBC version 7.00A.029.005

Server application: Windows Server 2012 R2 Standard edition
Application platform: Microsoft Lync 2013 5.0.8308.726

6 Summary of test results

6.1 Summary of main functions supported for OXE

Feature	N/A	OK	NOK
Outgoing call			
Call to free user	☐	☒	☐
Call to forwarded user	☐	☒	☐
Call to busy user	☒	☐	☐
Call to user in dnd	☐	☒	☐
Missed call feature	☒	☐	☐
Secret identity	☐	☒	☐
Incoming call			
Call to free user	☐	☒	☐
Call to forwarded user	☐	☒	☐
Call to busy user	☐	☒	☐
Call to user in dnd	☐	☒	☐
Missed call feature	☐	☒	☐
Features during conversation			
Hold/Resume	☐	☒	☐
Consultation call	☐	☒	☐
Broker call	☐	☒	☐
Transfer			
Transfer unattended	☒	☐	☐
Transfer on ringing	☐	☒	☐
Transfer on conversation	☐	☒	☐
Conference			
Three party conference	☐	☒	☐
Voicemail			
Lync user forwarded to Exchange voicemail	☒	☐	☐
Call to OXE user forwarded to voicemail	☐	☒	☐
Attendant			
Lync call to Attendant	☐	☒	☐
Attendant transfers lync user to OXE user	☐	☒	☐
Attendant transfers OXE user to lync user	☐	☒	☐
Attendant transfers external user (T2) to Lync user	☒	☐	☐
Attendant transfers lync user to External user (T2)	☒	☐	☐
OXE switchover			
Call continuity during switchover (OXE spatial redundancy)	☐	☒	☐
Call state change after switchover	☐	☐	☒
New call after switchover	☐	☒	☐

6.2 Summary of main functions supported for OT

Feature	N/A	OK	NOK
Outgoing call			
Call to free user	☐	☒	☐
Call to forwarded user	☐	☒	☐
Call to busy user	☒	☐	☐
Call to user in dnd	☐	☒	☐
Missed call feature	☒	☐	☐
Secret identity	☐	☒	☐
Incoming call			
Call to free user	☐	☒	☐
Call to forwarded user	☐	☒	☐
Call to forwarded user (->to Lync number)	☒	☐	☐
Call to busy user	☐	☒	☐
Call to user in dnd	☒	☐	☐
Missed call feature	☐	☒	☐
Features during conversation			
Hold/Resume	☐	☒	☐
Consultation call	☐	☒	☐
Broker call	☐	☒	☐
Transfer			
Transfer unattended	☒	☐	☐
Transfer on ringing	☐	☐	☒
Transfer on conversation	☐	☒	☐
Conference			
Three party conference	☐	☒	☐
Voicemail			
Lync user forwarded to Exchange voicemail	☒	☐	☐
Call to OXE user forwarded to voicemail	☐	☒	☐
Attendant			
Lync call to Attendant	☒	☐	☐
Attendant transfers lync user to OT user	☒	☐	☐
Attendant transfers OT user to lync user	☒	☐	☐
Attendant transfers external user (T2) to Lync user	☒	☐	☐
Attendant transfers lync user to External user (T2)	☒	☐	☐

6.3 Summary of problems

6.3.1 OXE

None

6.3.2 Audiocodes/Lync

None

6.3.3 OT

- SR 1-192855375/crqms00199160: AAPP: semi-attended transfer OT-OT-Lync does not work.
OXE sends a 404 not found when the transfer is done by OT user

6.4 Summary of limitations

6.4.1 OXE

None

6.4.2 Lync

- Calls from Lync to OXE/OT users:
Lync doesn’t display the calling name, only the number despite the name of OXE user is available in the “from” or PAI field.

- Calls from OXE/OT users to Lync:
The called Lync number is displayed, not the called name because Lync does not send the user name to AudioCodes SBC (only the number).

- Transfers from Lync: Lync doesn’t update transferee information after a transfer (semi attended or attended transfer).

- Semi attended or attended transfers from OXE: OXE sends the user information to AudioCodes gateway (in REFER or REINVITE messages). However Lync display is not updated. Transfer is OK.

- In Lync client, missed call list and conversation history tabs are empty.

- After an OXE CPU swichover to the standby CPU, an existing call cannot evolve (state changed to “on hold”, “transferred”…)

6.4.3 OT

- Forward to Lync phone is forbidden on conversation users (OT): “Error: User destination is not valid in the routing profile” on 8088

6.4.4 Audiocodes

After an OXE CPU swichover to a standby CPU resolved by spatial redundancy mechanism, a call cannot be established with the new main call server before the next DNS request. These DNS requests are configured to be sent every 10 seconds on the SBC (parameter PROXYIPLISTREFRESHTIME=10).

DNS requests are not issued by Audiocodes E-SBC for every message when TTL value is set to 0. This can lead to communication troubles during the time of the CPU switchover. As a workaround, it is advised to set the parameter Proxy IP List Refresh Time to 10s on SBC to refresh dns cache every 10s.

6.5 Notes, remarks

- Media anchoring is used on SBC (SBC direct Media disabled), all media streams are going through the SBC
- Media bypass is not enabled on lync server, all media streams are going through lync server
- On Lync server side, G729 cannot be used, only G711
- Case of Exchange voicemail (from O365 or on premises) for Lync users has not been tested
- No Lync on-line (O365) user have been used in this interworking report

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
In yellow, there is a small limitation found during the test
In green 100% OK
NOK: when checked, means the test case has failed. In that case, describe in the field "Comment" the reason for the failure and the reference number of the issue either on ALE International side or on AAPP member side
Comment: to be filled in with any relevant comment. Mandatory in case a test has failed especially the reference number of the issue.

8 Test Results

In next sections, users A, B, C are OXE or OT users. Test devices are iptouch 40x8/80x8 phones or 8012 SIP phones, but could be replaced by analog 40x9/80x8 series, SIP or DECT phones.

8.1 Outgoing calls: OXE/OT users to Lync

8.1.1 Test Objectives

The calls are generated to several numbers corresponding to users on the Lync platform.
Called party can be in different states: free, busy, Out of service, do not disturb.
Points to be checked: tones, voice during the conversation, display (on caller and called party), hang-up phase.

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

8.1.2 OXE Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	Call to free user	☐	☒	☐	
2	Call to wrong number	☐	☒	☐	403 forbidden sent by Lync
3	Call in conversation; DTMF reception ; Calling / caller line identity ; display	☐	☒	☐	Outgoing calls: On OXE phone, the called number is displayed, not the called name. Lync does not send the user name to AudioCodes SBC (only the number). DTMF sent using RFC2833 OK
4	Call to busy user (mono-line / multi-line)	☒	☐	☐	Lync user is never in busy mode
5	Call to user in “Out of Service” state	☐	☒	☐	“no answer” display on caller: SIP 480 ”Temporary Unavailable” sent by Lync
6	Call to user in “Do not Disturb” state	☐	☒	☐	“no answer” display on caller: SIP 480 ”Temporary Unavailable” sent by Lync
7	Call to forwarded user (locally) (immediate forward) OXE user1 call Lync user1 who is in immediate forward to Lync user2.	☐	☒	☐	Same display issue as step 3 And there is no display update on OXE set after the forward. Lync user 1 number is still displayed.
8	Missed call feature : OXE user1 call Lync user1 and hung up. Lync user1 can press call back to call OXE user1 from the missed call list	☒	☐	☐	No missed calls on Lync client Test needs to be done using the missed call details of the outlook client

9	No answer of the called party (Forward no reply) OXE phone calls Lync user1 and Lync user does not take the call and after some minute the call is forwarded to another Lync user2	☐	☒	☐	Same display issue as step 3 And there is no display update on OXE set after the forward. Lync user 1 number is still displayed. (Configured via “Call forwarding” > “Unanswered call will go to” option)
10	Call from UA/TDM to Lync user	☒	☐	☐	Test done with a digital phone connected
11	Oxe Phone calls Lync user after activating secret identity feature (prefix 409)	☐	☒	☐	Anonymous is displayed on the Lync phone

8.1.3 OT Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	Call to free user	☐	☒	☐	
2	Call to wrong number	☐	☒	☐	403 forbidden sent by Lync
3	Call in conversation; DTMF reception ; Calling / caller line identity ; display	☐	☒	☐	Outgoing calls: On OT phone, the called number is displayed, not the called name. Lync does not send the user name to AudioCodes SBC (only the number). DTMF sent using RFC2833 OK
4	Call to busy user (mono-line / multi-line)	☒	☐	☐	Lync user is never in busy mode
5	Call to user in “Out of Service” state	☐	☒	☐	“Callee temporarily unavailable” is displayed on caller: SIP 480 ”Temporary Unavailable” in the trace
6	Call to user in “Do not Disturb” state	☐	☒	☐	“Callee temporarily unavailable” is displayed on caller: SIP 480 ”Temporary Unavailable” in the trace
7	Call to forwarded user (locally) (immediate forward) OT user1 call Lync user1 who is in immediate forward to Lync user2.	☐	☒	☐	Same display issue as step 3 And there is no display update on OT set after the forward. Lync user 1 number is still displayed.
8	Missed call feature : OT user1 call Lync user1 and hung up. Lync user1 can press call back to call OT user1 from the missed call list	☒	☐	☐	No missed calls on Lync client Test needs to be done using the missed call details of the outlook client
9	No answer of the called party (Forward no reply) OT phone calls Lync user1 and Lync user does not take the call and after some minute the call is forwarded to another Lync user2	☐	☒	☐	Same display issue as step 3 And there is no display update on OT set after the forward. Lync user 1 number is still displayed. (Configured via “Call forwarding” > “Unanswered call will go to” option)

10	Call from UA/TDM to Lync user	☒	☐	☐	
11	OT Phone calls Lync user after activating secret identity feature	☐	☒	☐	

8.2 Incoming calls: Lync => OXE/OT

8.2.1 Test Objectives

The calls are generated to several numbers corresponding to OXE or OT users.
Called party can be in different states: free, busy, Out of service, do not disturb.
Points to be checked: tones, voice during the conversation, display (on caller and called party), hang-up phase.
Call to unknown numbers must be rejected.

8.2.2 OXE Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	Call to free user (check Calling / caller line identity ; display, DTMF)	☐	☒	☐	Both name and numbers are displayed on OXE and Lync users DTMF sent using RFC2833.
2	Call to wrong number	☐	☒	☐	
3	Call to busy user (mono-line / multi-line)(Local/Network)	☐	☒	☐	“user is in another call” is displayed on Lync client
4	Call to user in “Out of Service” state (Local/Network)	☐	☒	☐	“user is unavailable or may be offline” is displayed on Lync client
5	Call to user in “Do not Disturb” state (prefix 42)	☐	☒	☐	“user is unavailable or may be offline” is displayed on Lync client
6	Call to forwarded user (immediate forward) Lync user1 call OXE user1 who is in immediate forward to OXE user2. (Local/Network)	☐	☒	☐	Lync phone displays the forwarded OXE user information, not final OT destination. AudioCodes let the updated PAI (with final destination information) in 200 OK/SDP but it is not interpreted by Lync phone
7	Missed call feature Lync user1 call OXE user1 and hung up. OXE user1 can use the missed call list to call Lync user1	☐	☒	☐	
8	Lync user 1 calls IP Phone which is on forward on busy to Lync user 2 (prefix 52) (Local/Network)	☐	☒	☐	Lync user 1 displays the forwarded OXE user information, not the Lync user 2 destination. AudioCodes let the updated PAI (with final destination information) in 200 OK/SDP but it is not interpreted by Lync phone Lync client 2 displays the forwarded OXE user information, not Lync user 1 info.

					OXE does not change the PAI or the from in the INVITE. Forwarded OXE user number is still sent.
9	Lync user 1 calls IP Phone which is on forward on no reply to Lync user 2	☐	☒	☐	Lync user 1 displays the forwarded OXE user information, not the Lync user 2 destination. AudioCodes let the updated PAI (with final destination information) in 200 OK/SDP but it is not interpreted by Lync phone Lync client 2 displays the forwarded OXE user information, not Lync user 1 info. OXE does not change the PAI or the from in the INVITE. Forwarded OXE user number is still sent.
10	Lync user1 calls IP Phone which is in immediate forward mode to Lync user2 (51) (Local/Network)	☐	☒	☐	Lync user 1 displays the forwarded OXE user information, not the Lync user 2 destination. AudioCodes let the updated PAI (with final destination information) in 200 OK/SDP but it is not interpreted by Lync phone Lync client 2 displays the forwarded OXE user information, not Lync user 1 info. OXE does not change the PAI or the from in the INVITE. Forwarded OXE user number is still sent.
11	Lync user put OXE phone on hold	☐	☒	☐	Local MoH or beep played from Lync client

8.2.3 OT Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	Call to free user (check Calling / caller line identity ; display, DTMF)	☐	☒	☐	Both name and numbers are displayed on OT user and Lync client DTMF sent using RFC2833.
2	Call to wrong number	☐	☒	☐	
3	Call to busy user (mono-line / multi-line)(Local/Network)	☐	☒	☐	“user is in another call” is displayed on Lync client
4	Call to user in “Out of Service” state (Local/Network)	☐	☒	☐	“user is unavailable or may be offline” is displayed on Lync client
5	Call to user in “Do not Disturb” state	☒	☐	☐	Feature not available on 8088 business
6	Call to forwarded user (immediate forward)	☐	☒	☐	Lync phone displays the forwarded OT user information, not final OT destination.

	Lync user1 call OT user1 who is in immediate forward to OT user2.				AudioCodes let the updated PAI (with final destination information) in 200 OK/SDP but it is not interpreted by Lync phone There is a missed call on OT user 1
7	Missed call feature Lync user1 call OT user1 and hung up. OXE user1 can use the misssed call list to call Lync user1	☐	☒	☐	
8	Lync user 1 calls IP Phone which is on forward on busy to Lync user 2 (prefix 52)	☒	☐	☐	It is not possible on OpenTouch to modify the routing profile of OT user with an external Lync number. Error: User destination is not valid in the routing profile
9	Lync user 1 calls IP Phone which is on forward on no reply to Lync user 2	☒	☐	☐	It is not possible on OpenTouch to modify the routing profile of OT user with an external Lync number. Error: User destination is not valid in the routing profile
10	Lync user1 calls IP Phone which is in immediate forward mode to Lync user 2 (51)	☒	☐	☐	It is not possible on OpenTouch to modify the routing profile of OT user with an external Lync number. Error: User destination is not valid in the routing profile
11	Lync user put OXE phone on hold	☐	☒	☐	Local MoH or beep played from Lync client

8.3 Features during conversation

8.3.1 Hold, Consultation call and broker call

8.3.1.1 Test objectives

During conversation, waiting and consultation call are provided and must be checked.
In addition, a second call must be generated in order to check that right tones are generated on Lync user.

8.3.1.2 OXE Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	A=Lync user, B=IPPhone1, C=IPPhone2				
1.1	Hold state request Lync user->IPPhone1, Lync user on hold	☐	☒	☐	
1.2	Consultation call request IPPhone1->IPPhone2	☐	☒	☐	
1.3	Broker request IPPhone1->Lync user, IPPhone2 on hold	☐	☒	☐	
2	A=PSTN, B=IPPhone1, C=Lync user				
2.1	Hold state request PSTN->IPPhone1, PSTN on hold	☐	☒	☐	
2.2	Consultation call request IPPhone1->Lync user	☐	☒	☐	
2.3	Broker request IPPhone1->PSTN, Lync user on hold	☐	☒	☐	
3	A=IPPhone1, B=IPPhone2, C=Lync user				
3.1	Hold state request IPPhone1->IPPhone2, IPPhone1 on hold	☐	☒	☐	
3.2	Consultation call request IPPhone2->Lync user	☐	☒	☐	
3.3	Broker request IPPhone2->IPPhone1, Lync user on hold	☐	☒	☐	
4	A=IPPhone1, B=Lync user, C=IPPhone2				
4.1	Hold state request IPPhone1->Lync user, IPPhone1 on hold	☐	☒	☐	

4.2	Consultation call request Lync user->IPPhone2	☐	☒	☐	
4.3	Broker request Lync user->IPPhone1, IPPhone2 on hold	☐	☒	☐	
5	A=Lync user1, B=IPPhone, C= Lync user2				
5.1	Hold state request Lync user1->IPPhone, Lync user1 on hold	☐	☒	☐	
5.2	Consultation call request IPPhone->Lync user2	☐	☒	☐	
5.3	Broker request IPPhone->Lync user1, Lync user2 on hold	☐	☒	☐	If IPPhone hooks on, Lync user 1 and Lync user 2 are in conversation If IPPhone ends the call with the soft key, it switches automatically to the other call
6	A=IPPhone, B=Lync user1, C=Lync user2				
6.1	Hold state request IPPhone->Lync user1, IPPhone on hold	☐	☒	☐	
6.2	Consultation call request Lync user1->Lync user2	☐	☒	☐	
6.3	Broker request Lync user1->IPPhone, Lync user2 on hold	☐	☒	☐	

8.3.1.3 OT Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	A=Lync user, B=IPPhone1, C=IPPhone2				
1.1	Hold state request Lync user->IPPhone1, Lync user on hold	☐	☒	☐	
1.2	Consultation call request IPPhone1->IPPhone2	☐	☒	☐	
1.3	Broker request IPPhone1->Lync user, IPPhone2 on hold	☐	☒	☐	
2	A=PSTN, B=IPPhone1, C=Lync user				
2.1	Hold state request PSTN->IPPhone1, PSTN on hold	☐	☒	☐	
2.2	Consultation call request IPPhone1->Lync user	☐	☒	☐	

2.3	Broker request IPPhone1->PSTN, Lync user on hold	☐	☒	☐	
3	A=IPPhone1, B=IPPhone2, C=Lync user				
3.1	Hold state request IPPhone1->IPPhone2, IPPhone1 on hold	☐	☒	☐	
3.2	Consultation call request IPPhone2->Lync user	☐	☒	☐	
3.3	Broker request IPPhone2->IPPhone1, Lync user on hold	☐	☒	☐	
4	A=IPPhone1, B=Lync user, C=IPPhone2				
4.1	Hold state request IPPhone1->Lync user, IPPhone1 on hold	☐	☒	☐	
4.2	Consultation call request Lync user->IPPhone2	☐	☒	☐	
4.3	Broker request Lync user->IPPhone1, IPPhone2 on hold	☐	☒	☐	
5	A=Lync user1, B=IPPhone, C= Lync user2				
5.1	Hold state request Lync user1->IPPhone, Lync user1 on hold	☐	☒	☐	
5.2	Consultation call request IPPhone->Lync user2	☐	☒	☐	
5.3	Broker request IPPhone->Lync user1, Lync user2 on hold	☐	☒	☐	
6	A=IPPhone, B=Lync user1, C=Lync user2				
6.1	Hold state request IPPhone->Lync user1, IPPhone on hold	☐	☒	☐	
6.2	Consultation call request Lync user1->Lync user2	☐	☒	☐	
6.3	Broker request Lync user1->IPPhone, Lync user2 on hold	☐	☒	☐	

8.3.2 Transfer call

8.3.2.1 Test objectives

During the consultation call step, the transfer service can be requested and must be tested.
Several transfer services exist: blind transfer, supervised transfer and busy transfer.
Audio, tones and display must be checked.

Tests are performed using all possible combinations of legacy (IPPHONE) and Lync sets.

8.3.2.2 Test Procedure

During the consultation call step, the transfer service can be requested and must be tested.
Several transfer services exist: Unattended Transfer, Semi-Attended Transfer and Attended Transfer.
Audio, tones and display must be checked.

We use the following scenario, terminology and notation:

There are three actors in a given transfer event:
A – *Transferee:* the party being transferred to the Transfer Target.
B – *Transferor:* the party doing the transfer.
C – *Transfer Target:* the new party being introduced into a call with the Transferee.

There are three sorts of transfers in the SIP world:
Unattended Transfer or *Blind Transfer*: The Transferor provides the Transfer Target's contact to the Transferee. The Transferee attempts to establish a session using that contact and reports the results of that attempt to the Transferor.

Semi-Attended Transfer or Early Attended Transfer or Transfer on ringing:
A (Transferee) calls B (Transferor).
B (Transferor) calls C (Transfer Target). A is on hold during this phase. C is in ringing state (does not pick up the call).
B executes the transfer. B drops out of the communication. A is now in contact with C, in ringing state. When C picks up the call it is in conversation with A.

Attended Transfer or Consultative Transfer or Transfer in conversation:
A (Transferee) calls B (Transferor).
B (Transferor) calls C (Transfer Target). A is on hold during this phase. C picks up the call and goes in conversation with B.
B executes the transfer. B drops out of the communication. A is now in conversation with C.

Check the transfer for two configuration possibilities on Lync (with or without REFER).
For blind transfer check that the transferred call can be taken back from the transferee in case of no answer or wrong number dialed.

8.3.2.3 OXE Test Results

Unattended Transfer (Blind)

Test	Action			Result	Comment
	A Transferee	B Transferor	C Transfer Target		
1	LYNC	OXE	OXE	OK_but	No display update after the transfer on Lync client A

2	OXE	LYNC	OXE	N/A	Unattended transfers not available from the Lync client.
3	OXE	OXE	LYNC	N/A	Unattended transfers not available from an OXE user to an external Lync user
4	OXE	LYNC	LYNC	N/A	Unattended transfers not available from the Lync client.
5	LYNC	OXE	LYNC	N/A	Unattended transfers not available from an OXE user to an external Lync user
6	LYNC	LYNC	OXE	N/A	Unattended transfers not available from the Lync client.

Semi attended Transfer (On Ringing)

Test	Action			Result	Comment
	A Transferee	B Transferor	C Transfer Target		
1	LYNC	OXE	OXE	OK_but	No display update after the transfer on Lync client A
2	OXE	LYNC	OXE	OK_but	No display update after the transfer on the transferee (A) Need to unset SENDING REFER TO GATEWAY in Lync server
3	OXE	OXE	LYNC	OK_but	No display update after the transfer on Lync client C. The number is displayed on OXE set A.
4	OXE	LYNC	LYNC	OK_but	No display update after the transfer on the transferee (A)
5	LYNC	OXE	LYNC	OK_but	No display update after the transfer on the transferee (A) and transfer target C
6	LYNC	LYNC	OXE	OK	The number is displayed on Lync client A

Attended Transfer (in conversation)

Test	Action			Result	Comment
	A Transferee	B Transferor	C Transfer Target		
1	LYNC	OXE	OXE	OK_but	No display update after the transfer on Lync client A
2	OXE	LYNC	OXE	OK_but	No display update after the transfer on the transferee (A) and transfer target C. Lync number is still displayed on both sets.
3	OXE	OXE	LYNC	OK_but	No display update after the transfer on Lync client C. The number is displayed on OXE set A.
4	OXE	LYNC	LYNC	OK_but	No display update after the transfer on the transferee (A). Lync client only display the number, not the name.
5	LYNC	OXE	LYNC	OK_but	No display update after the transfer on Lync clients (A and C)
6	LYNC	LYNC	OXE	OK_but	No display update after the transfer on OXE set. Lync client only display the number, not the name.

8.3.2.4 OT Test Results

Unattended Transfer (Blind)

Test	Action			Result	Comment
	A Transferee	B Transferor	C Transfer Target		
1	LYNC	OT	OT	N/A	Unattended transfers not available from the OT client.
2	OT	LYNC	OT	N/A	Unattended transfers not available from the Lync client.
3	OT	OT	LYNC	N/A	Unattended transfers not available from the OT client.
4	OXE	LYNC	LYNC	N/A	Unattended transfers not available from the Lync client.
5	LYNC	OT	LYNC	N/A	Unattended transfers not available from the OT client.
6	LYNC	LYNC	OT	N/A	Unattended transfers not available from the Lync client.

Semi attended Transfer (On Ringing)

Test	Action			Result	Comment
	A Transferee	B Transferor	C Transfer Target		
1	LYNC	OT	OT	OK_but	No display update after the transfer on Lync client A
2	OT	LYNC	OT	OK_but	No display update after the transfer on the transferee (A)
3	OT	OT	LYNC	NOK	« Wrong number » displayed on transferee OT issue: SR 1-192855375 crqms00199160
4	OT	LYNC	LYNC	OK_but	No display update after the transfer on the transferee (A)
5	LYNC	OT	LYNC	OK_but	No display update after the transfer on transferee (A) and transfer target C OXE number is displayed on both lync clients
6	LYNC	LYNC	OT	OK	The number is displayed on Lync client A

Attended Transfer (in conversation)

Test	Action			Result	Comment
	A Transferee	B Transferor	C Transfer Target		
1	LYNC	OT	OT	OK_but	No display update after the transfer on Lync client A Only the number is displayed on user C
2	OT	LYNC	OT	OK_but	No display update after the transfer on OT user A and user C. Lync number is displayed on both sets.
3	OT	OT	LYNC	OK_but	No display update after the transfer on Lync client (C)

4	OT	LYNC	LYNC	OK_but	No display update after the transfer on the transferee (A).
5	LYNC	OT	LYNC	OK_but	No display update after the transfer on Lync clients (A and C)
6	LYNC	LYNC	OT	OK_but	No display update after the transfer on OXE set. Lync client only display the number, not the name.

8.3.3 Conference

8.3.3.1 Test objectives

During the consultation call step, the conference is provided and must be tested.
Programmed conference and 3 steps conferences have to be checked by analyzing the audio and display on each user.

8.3.3.2 Test procedure

We use the following scenario, terminology and notation:
We start with A in conversation with B. (A->B)
A places B on hold. B should hear hold tone.
A calls C while B is on hold. C rings and goes off-hook.
A activates conference.
A, B, C should be in communication now.

8.3.3.3 OXE Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	3 steps conference request PSTN ->IPPhone->Lync user				The conference is initiated by the IPPhone Only the trunk name is displayed on IPPhone
1.1	PSTN leaves the conference	☐	☒	☐	
1.2	Lync user leaves the conference	☐	☒	☐	
1.3	IPPhone leaves the conference	☐	☒	☐	PSTN and Lync are still in communication
2	3 steps conference request Lync user 1->IPPhone-> Lync user 2				The conference is initiated by the IPPhone Only the trunk name is displayed on IPPhone instead of Lync user 2 Only the number is displayed on Lync user 1
2.1	Lync user 1 leaves the conference	☐	☒	☐	
2.2	Lync user 2 leaves the conference	☐	☒	☐	
2.3	IPPhone leaves the conference	☐	☒	☐	
2.4	Stay in a conference for long period to check keep alive mechanisms.				
3	3 steps conference request IPPhone1 ->Lync->IPPhone2				Conference is set up from lync client (add a participant) Only the number of Lync user and trunk name is displayed on IPPhone1. Only the

					trunk name is displayed on IPPhone 2 On Lync only the number is displayed for IPPhone 2 The last user in the conference must end the call
3.1	IPPhone1 leaves the conference	☐	☒	☐	
3.2	Lync user leaves the conference	☐	☒	☐	
3.3	IPPhone2 leaves the conference	☐	☒	☐	

8.3.3.4 OT Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	3 steps conference request PSTN ->IPPhone->Lync user				The conference is initiated by the IPPhone Only the number is displayed on IPPhone (lync user 2 number) and Lync (IPPhone number)
1.1	PSTN leaves the conference	☐	☒	☐	
1.2	Lync user leaves the conference	☐	☒	☐	
1.3	IPPhone leaves the conference	☐	☒	☐	
2	3 steps conference request Lync user 1->IPPhone-> Lync user 2				The conference is initiated by the IPPhone Only the number is displayed on IPPhone (lync user 2 number) and Lync (IPPhone number)
2.1	Lync user 1 leaves the conference	☐	☒	☐	
2.2	Lync user 2 leaves the conference	☐	☒	☐	
2.3	IPPhone leaves the conference	☐	☒	☐	
2.4	Stay in a conference for long period to check keep alive mechanisms.				
3	3 steps conference request IPPhone1 ->Lync->IPPhone2				Conference is set up from lync client (add a participant) Only the trunk name is displayed on IPPhone2. Only the number is displayed on IPPhone 1 Only the number is displayed on lync client for IPPhone 2

					The last user in the conference must end the call
3.1	IPPhone1 leaves the conference	☐	☒	☐	
3.2	Lync user leaves the conference	☐	☒	☐	
3.3	IPPhone2 leaves the conference	☐	☒	☐	

8.3.3.5 Mixed scenario OT/OXE/Lync Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	3 steps conference request Lync ->OT user->OXE user				
1.1	Lync leaves the conference	☐	☒	☐	
1.2	OXE user leaves the conference	☐	☒	☐	
1.3	OT leaves the conference	☐	☒	☐	
2	3 steps conference request OXE user ->OT user->Lync user				
2.1	OXE leaves the conference	☐	☒	☐	
2.2	Lync user leaves the conference	☐	☒	☐	
2.3	OT leaves the conference	☐	☒	☐	
3	3 steps conference request Lync ->OXE user->OT user				
3.1	Lync leaves the conference	☐	☒	☐	
3.2	OT user leaves the conference	☐	☒	☐	
3.3	OXE leaves the conference	☐	☒	☐	
3	3 steps conference request OT user ->OXE user->Lync user				
3.1	OT user leaves the conference	☐	☒	☐	
3.2	Lync user leaves the conference	☐	☒	☐	
3.3	OXE leaves the conference	☐	☒	☐	
3	3 steps conference request OT user ->Lync user->OXE user				

3.1	OT user leaves the conference	☐	☒	☐	
3.2	OXE user leaves the conference	☐	☒	☐	
3.3	Lync user leaves the conference	☐	☒	☐	
3	3 steps conference request OXE user ->Lync user->OT user				
3.1	OXE user leaves the conference	☐	☒	☐	
3.2	OT user leaves the conference	☐	☒	☐	
3.3	Lync user leaves the conference	☐	☒	☐	

8.3.4 Voicemail

Test objectives

Note: It has to be defined which Voicemail system is used for the interoperability tests:

Option 1: Voice mail is Exchange 2O13 for all users: Not tested
Option 2: Each system is served by its own Voice Mail: OK

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

8.3.4.1 OXE Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	Lync user forwarded to VoiceMail				No Exchange voicemail available
1.1	IPTouch phone leaves a voice message for the Lync user. Check that MWI is OK on Lync user				No Exchange voicemail available
1.2	Message consultation by Lync user	☒	☐	☐	No Exchange voicemail available
1.3	Password modification by Lync user	☒	☐	☐	No Exchange voicemail available
2	OXE user forwarded to VoiceMail				
2.1	Lync user call to a OXE user forwarded to Voice Mail	☐	☒	☐	
2.2	Message consultation by OXE user	☐	☒	☐	

8.3.4.2 OT Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	Lync user forwarded to VoiceMail				No Exchange voicemail available
1.1	OT phone leaves a voice message for the Lync user. Check that MWI is OK on Lync user				No Exchange voicemail available
1.2	Message consultation by Lync user	☒	☐	☐	No Exchange voicemail available
1.3	Password modification by Lync user	☒	☐	☐	No Exchange voicemail available
2	OT user forwarded to VoiceMail				
2.1	Lync user call to a OT user forwarded to Voice Mail	☐	☒	☐	
2.2	Message consultation by OT user	☐	☒	☐	

8.4 Attendant

8.4.1 Test Objectives

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

8.4.2 OXE Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	Lync user calls attendant number	☐	☒	☐	
2	Attendant calls Lync user	☐	☒	☐	
3	Attendant calls Lync user then Attendant calls OXE user Attendant transfer Lync user To OXE user	☐	☒	☐	the display on Lync user is not updated after the transfer
4	Lync user calls attendant , attendant transfers on ringing to OXE set.	☐	☒	☐	the display on Lync user is not updated after the transfer
5	Lync user calls attendant ,attendant transfers in conversation to OXE set,	☐	☒	☐	the display on Lync user is not updated after the transfer.
6	OXE set calls to attendant (using attendant call prefix “9”), attendant transfers during ringing to Lync user.	☐	☒	☐	On Lync user, only the number is displayed, not the name

7	OXE set calls to attendant (using attendant call prefix “9”), attendant transfers in conversation to Lync user.	☐	☒	☐	the display on Lync user is not updated after the transfer.
8	External user(T2) calls Attendant Attendant calls Lync user Attendant transfer on ringing External user(T2) to Lync user	☒	☐	☐	
9	Lync user calls Attendant Attendant calls External user(T2) Attendant transfer on conversation Lync user to External user(T2)	☒	☐	☐	

8.5 Defense / Recovery

8.5.1 Test Objectives

Test the robustness in case of a PBX reboot, switch-over or link failure.

8.5.2 Test Results

Test Case Id	Test Case	N/A	OK	NOK	Comment
1	Temporary Data Network Link down with the PBX and Mediant SBC	☐	☒	☐	Existing calls are stopped. Establishing new call is possible when the link is reestablished.
2	Spatial redundancy IP Method : CPU switchover with SIP communication	☒	☐	☐	
3	Spatial redundancy DNS method (delegation on a third party DNS server) : CPU switchover without SIP communication	☐	☒	☐	A call cannot be established after the switchover with the new main call server before the next DNS request. These DNS requests are configured to be sent every 10 seconds on the SBC (parameter PROXYIPLISTREFRESHTIME).
4	Spatial redundancy DNS method : CPU switchover with SIP communication	☐	☒	☐	Current call is OK, but it cannot evolve (state changed to on hold, transferred…). If this is done, the communication is cut on Lync client. A second call cannot be established after the switchover with the new main call server before the next DNS request. These DNS request are configured to be sent every 10 seconds on the SBC (parameter PROXYIPLISTREFRESHTIME). See note
5	Switchover to Passive Call Server (PCS). (IP link to main/stdby OXE call servers down)	☒	☐	☐	
6	Switchover to a backup AudioCodes gateway. Stop the main AudioCodes gateway; verify that a call is possible with the backup AudioCodes gateway.	☒	☐	☐	

Note: DNS requests are not issued by Audiocodes SBC for every message when TTL value is set to 0. This can lead to communication troubles during the time of the CPU switchover. As a workaround, it is advised to set the parameter Proxy IP List Refresh Time to 10s on SBC to refresh dns cache every 10s.

9 Appendix A : AudioCodes SBC Configuration

9.1 Getting Started

This section describes how to navigate in the Mediant E-SBC Web server navigation tree.

When navigating in the Navigation tree, you can view listed menus and submenus in either an expanded or contracted view. This is relevant when using the configuration tabs (**Configuration**, **Maintenance**, and **Status & Diagnostics**) on the Navigation bar.

The Navigation tree menu can be displayed in one of two views:

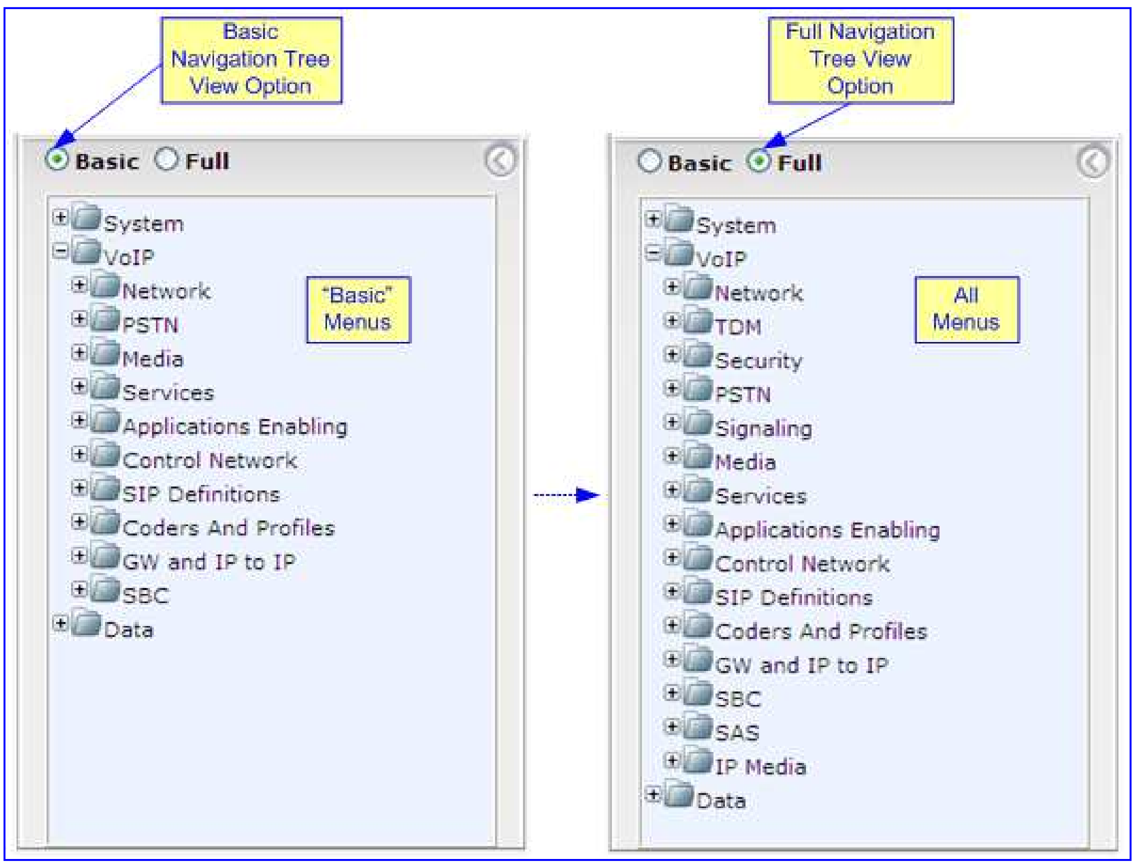
- **Basic:** displays only commonly used menus
- **Full:** displays all the menus pertaining to a configuration tab.

The advantage of the Basic view is that it prevents "cluttering" the Navigation tree with menus that may not be required. Therefore, a Basic view allows you to easily locate the required menus.

➢ **To toggle between Full and Basic view:**

- Select the **Basic** option (located below the Navigation bar) to display a reduced menu tree; select the **Full** option to display all the menus. By default, the **Basic** option is selected.

Figure 10-1: Navigation Tree in Basic and Full View



For more information, see the Mediant E-SBC User's manual.

9.2 Configuration Procedure

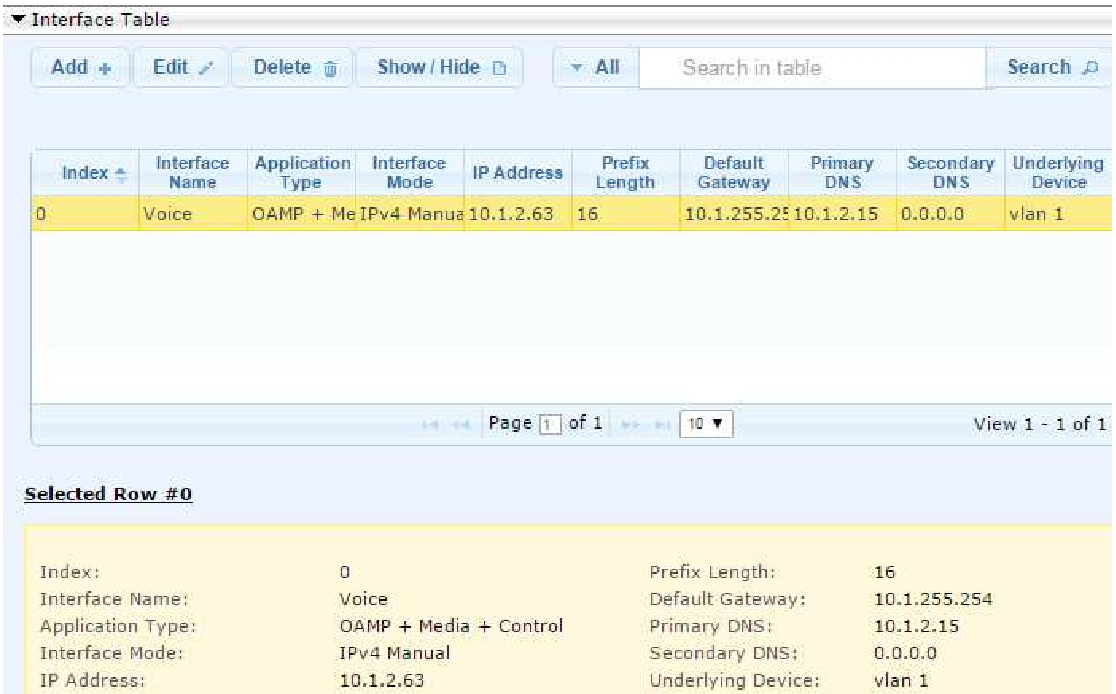
This section describes the Mediant SBC Configuration procedure.

9.2.1 Configure IP Address

➢ **To configure IP-Address:**

Open the 'IP Settings' page (Configuration **tab >** VoIP **menu >** Network **>** IP Interfaces Table**).**

Figure 10-2: IP Settings



Set the following parameters:

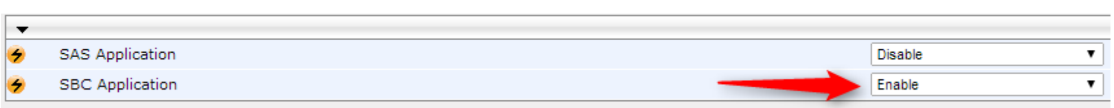
IP-Address: <Gateway IP-Address> (e.g., 10.1.2.63).

Prefix Length: The Subnet Mask in bits (e.g., 16 for 255.255.254.0).

Gateway: <Gateway Default Gateway> (e.g., 10.1.2.254).

DNS server : <Primary DNS Server IP Address> (e.g 10.1.2.15)

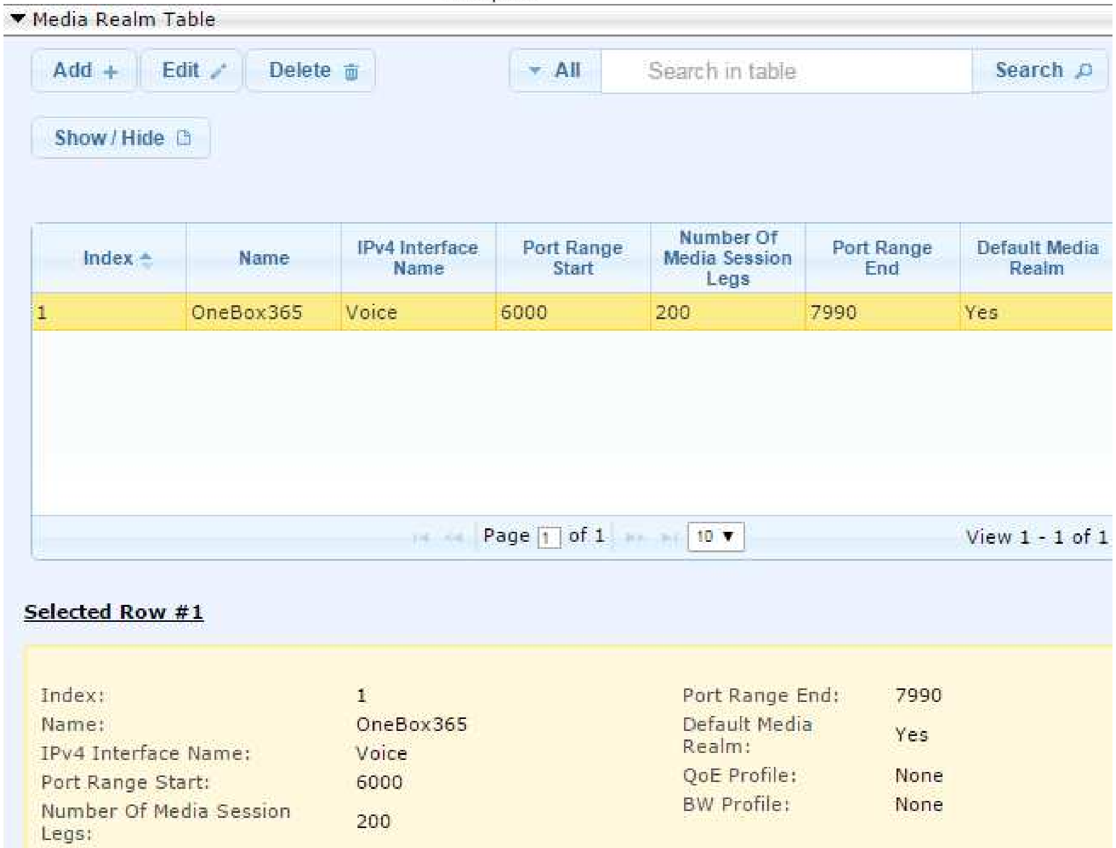
9.2.2 Enable the SBC Application



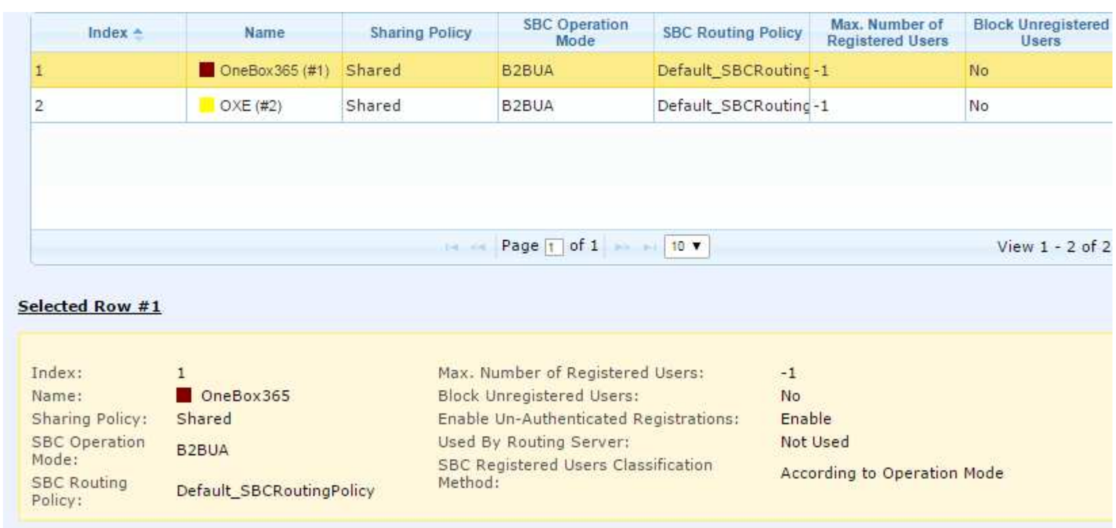
SBC Application: 'Enable'.

Note: Reset with BURN to FLASH is Required.

9.2.3 Media Realm



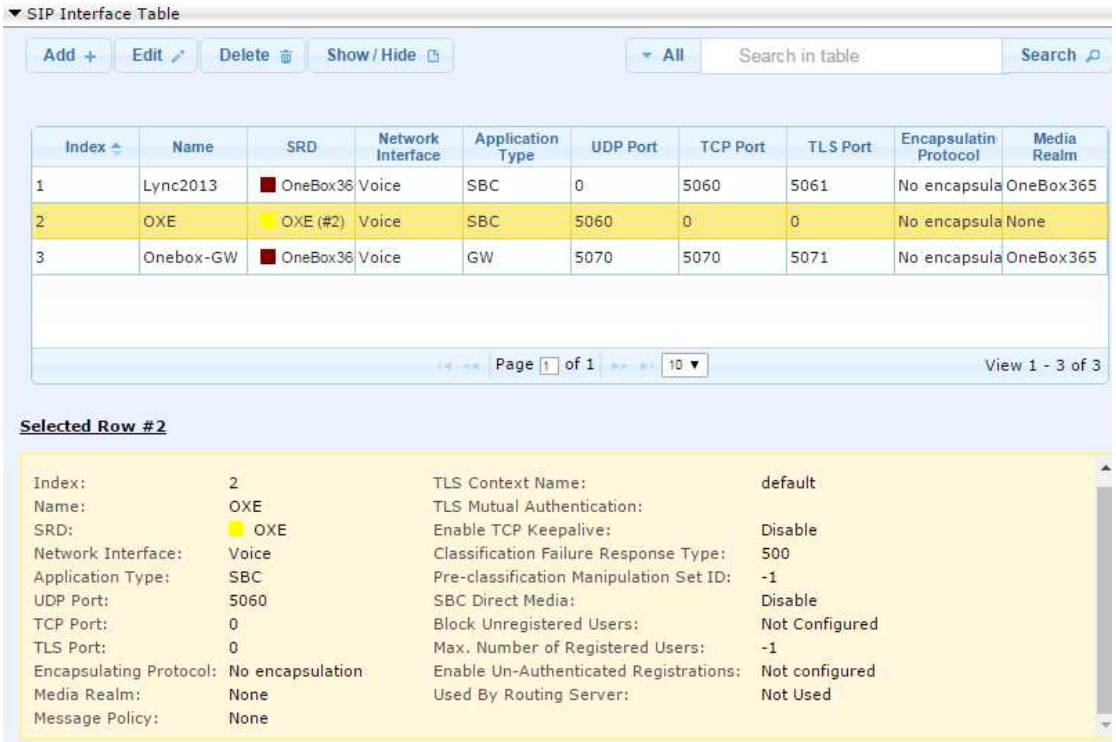
9.2.4 SRD Tables



9.2.5 SIP Interface Table

Open the ‘Sip Interface Table page (**Configuration** tab > **VoIP** menu > **VoIP Network** >**SIP Interface Table**).

Add a SIP Interface for OXE



▼ SIP Interface Table

Add + | Edit | Delete | Show / Hide | All | Search in table | Search

Index	Name	SRD	Network Interface	Application Type	UDP Port	TCP Port	TLS Port	Encapsulatin Protocol	Media Realm
1	Lync2013	OneBox36	Voice	SBC	0	5060	5061	No encapsula	OneBox365
2	OXE	OXE (#2)	Voice	SBC	5060	0	0	No encapsula	None
3	Onebox-GW	OneBox36	Voice	GW	5070	5070	5071	No encapsula	OneBox365

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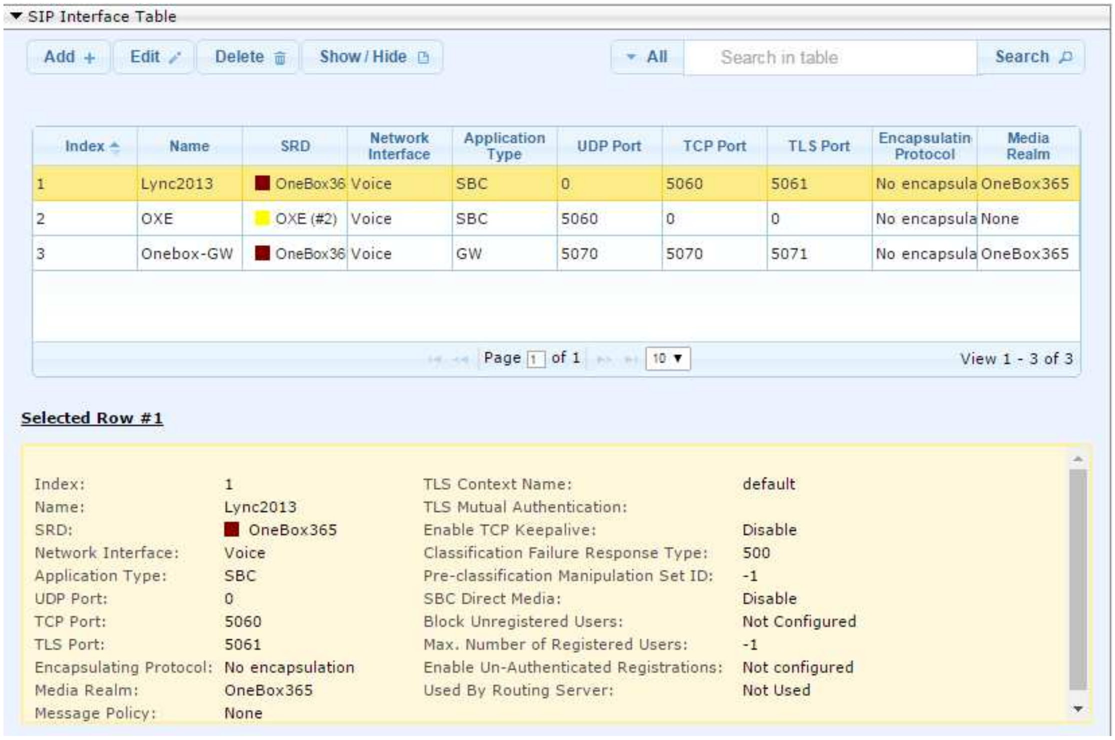
Selected Row #2

Index:	2	TLS Context Name:	default
Name:	OXE	TLS Mutual Authentication:	
SRD:	OXE	Enable TCP Keepalive:	Disable
Network Interface:	Voice	Classification Failure Response Type:	500
Application Type:	SBC	Pre-classification Manipulation Set ID:	-1
UDP Port:	5060	SBC Direct Media:	Disable
TCP Port:	0	Block Unregistered Users:	Not Configured
TLS Port:	0	Max. Number of Registered Users:	-1
Encapsulating Protocol:	No encapsulation	Enable Un-Authenticated Registrations:	Not configured
Media Realm:	None	Used By Routing Server:	Not Used
Message Policy:	None		

SBC Direct Media : Disable

- ➔ All media streams go through the SBC

Add a SIP Interface for Lync:



Index	Name	SRD	Network Interface	Application Type	UDP Port	TCP Port	TLS Port	Encapsulatin Protocol	Media Realm
1	Lync2013	OneBox36	Voice	SBC	0	5060	5061	No encapsula	OneBox365
2	OXE	OXE (#2)	Voice	SBC	5060	0	0	No encapsula	None
3	Onebox-GW	OneBox36	Voice	GW	5070	5070	5071	No encapsula	OneBox365

Index:	1	TLS Context Name:	default
Name:	Lync2013	TLS Mutual Authentication:	
SRD:	OneBox365	Enable TCP Keepalive:	Disable
Network Interface:	Voice	Classification Failure Response Type:	500
Application Type:	SBC	Pre-classification Manipulation Set ID:	-1
UDP Port:	0	SBC Direct Media:	Disable
TCP Port:	5060	Block Unregistered Users:	Not Configured
TLS Port:	5061	Max. Number of Registered Users:	-1
Encapsulating Protocol:	No encapsulation	Enable Un-Authenticated Registrations:	Not configured
Media Realm:	OneBox365	Used By Routing Server:	Not Used
Message Policy:	None		

SBC Direct Media : Disable

➔ All media streams go through the SBC

9.2.6 Proxy Sets Table

Open the ‘Proxy Sets Table’ page (**Configuration** tab > **VoIP** menu > **VoIP Network** >**Proxy Sets Table**).

Create 2 entries in the table one for Lync FE server, the other for OXE :

Lync FE proxy:

▼ Proxy Sets Table

Add + | Edit | Delete | Show / Hide | ▾ All | Search in table

Index	Name	SRD	Gateway IPv4 SIP Interface	SBC IPv4 SIP Interface	Proxy Keep-Alive Time [sec]	Redundancy Mode
1	Onebox	OneBox365 (#1	None	Lync2013	60	
2	OXE	OXE (#2)	None	OXE	60	
3	Onebox-GW	OneBox365 (#1	Onebox-GW	Lync2013	60	

Page 1 of 1 | 10 ▾

Selected Row #1

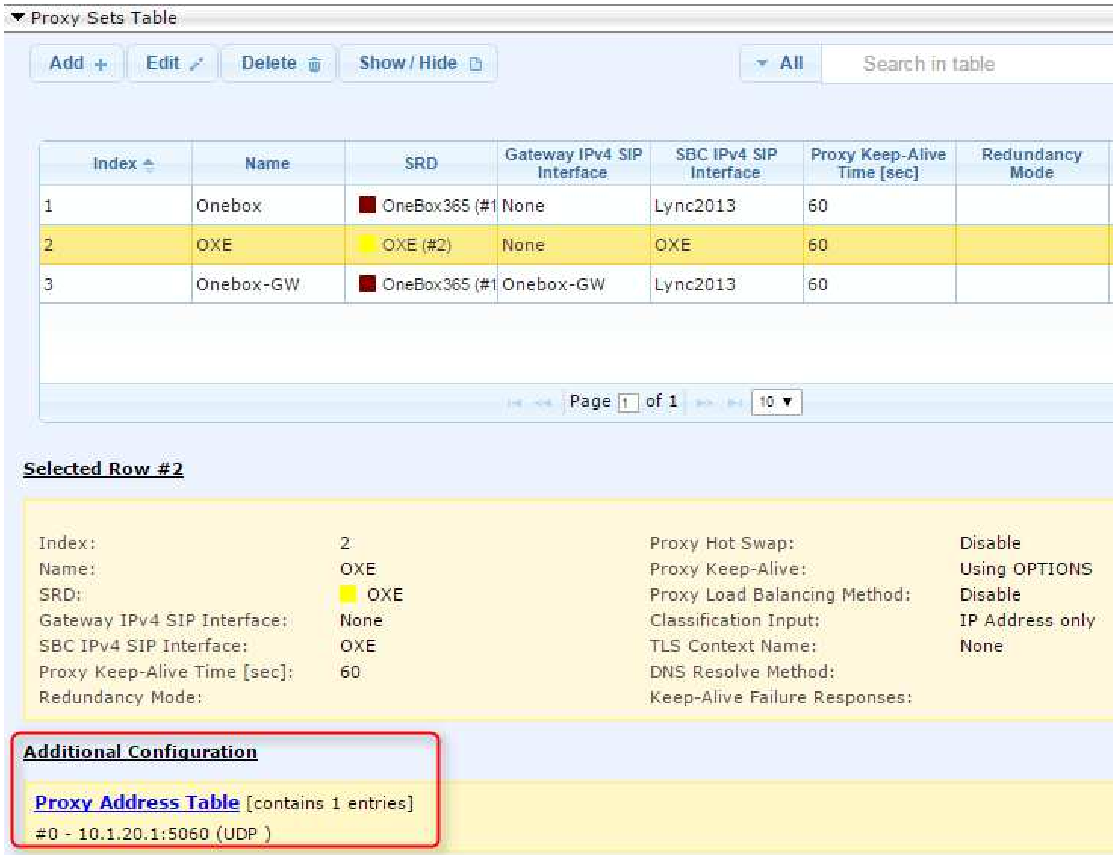
Index:	1	Proxy Hot Swap:	Enable
Name:	Onebox	Proxy Keep-Alive:	Using OPTIONS
SRD:	OneBox365	Proxy Load Balancing Method:	Disable
Gateway IPv4 SIP Interface:	None	Classification Input:	IP Address only
SBC IPv4 SIP Interface:	Lync2013	TLS Context Name:	None
Proxy Keep-Alive Time [sec]:	60	DNS Resolve Method:	
Redundancy Mode:		Keep-Alive Failure Responses:	

Additional Configuration

Proxy Address Table [contains 1 entries]

#0 - uc-fe.ac-onebox.com:5060 (TCP)

OXE Proxy:



▼ Proxy Sets Table

Add + | Edit | Delete | Show / Hide | ▼ All | Search in table

Index	Name	SRD	Gateway IPv4 SIP Interface	SBC IPv4 SIP Interface	Proxy Keep-Alive Time [sec]	Redundancy Mode
1	Onebox	OneBox365 (#1	None	Lync2013	60	
2	OXE	OXE (#2)	None	OXE	60	
3	Onebox-GW	OneBox365 (#1	Onebox-GW	Lync2013	60	

Page 1 of 1 | 10

Selected Row #2

Index:	2	Proxy Hot Swap:	Disable
Name:	OXE	Proxy Keep-Alive:	Using OPTIONS
SRD:	OXE	Proxy Load Balancing Method:	Disable
Gateway IPv4 SIP Interface:	None	Classification Input:	IP Address only
SBC IPv4 SIP Interface:	OXE	TLS Context Name:	None
Proxy Keep-Alive Time [sec]:	60	DNS Resolve Method:	
Redundancy Mode:		Keep-Alive Failure Responses:	

Additional Configuration

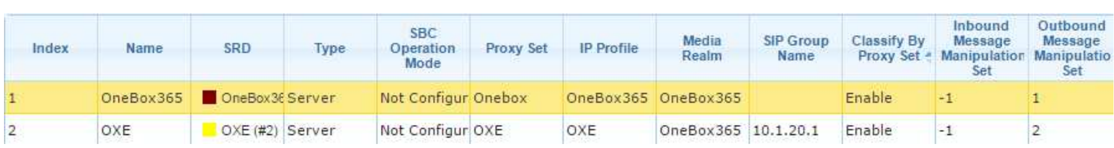
Proxy Address Table [contains 1 entries]

#0 - 10.1.20.1:5060 (UDP)

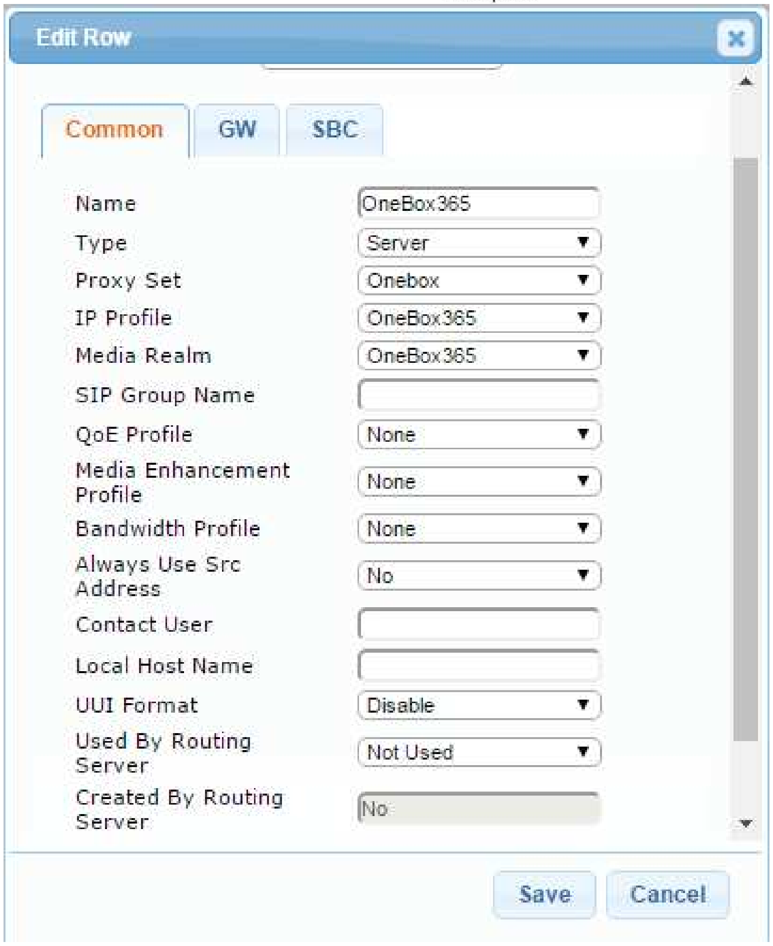
9.2.7 IP Group Table

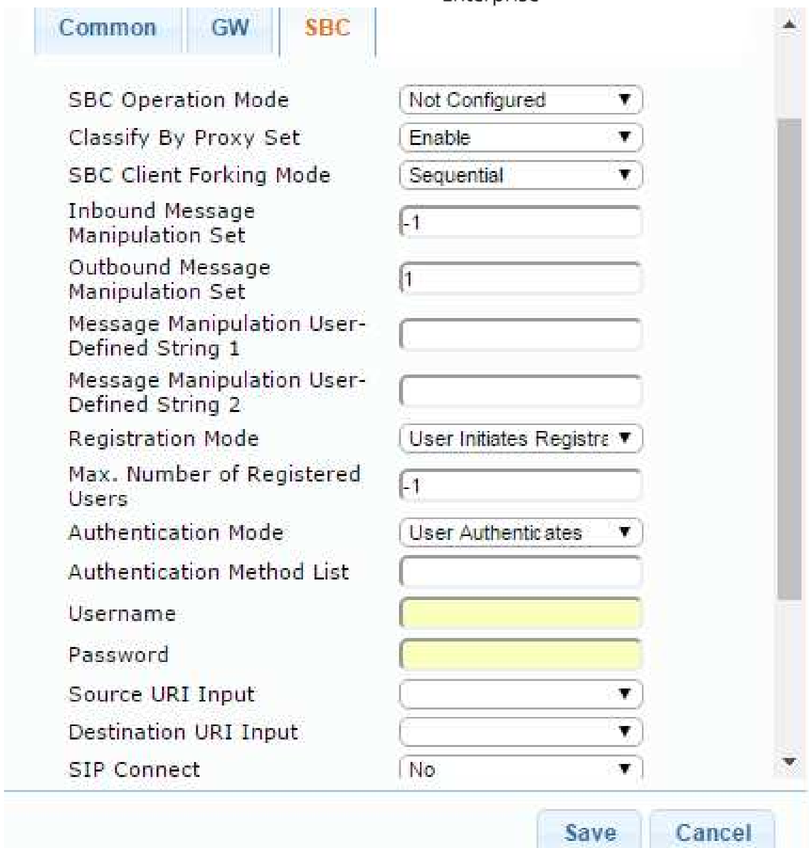
Open the 'IP Group Table' page (**Configuration** tab > **VoIP** menu > **VoIP Network**> **IP Group Table**)

Configure IP Group Table for Lync as below:

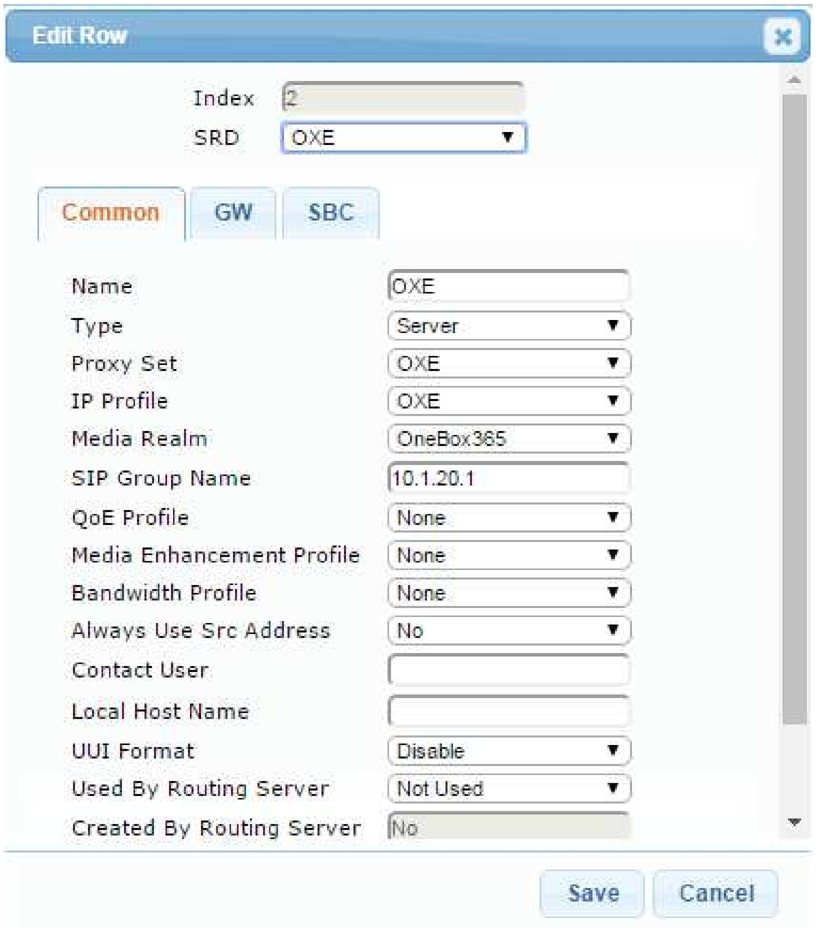


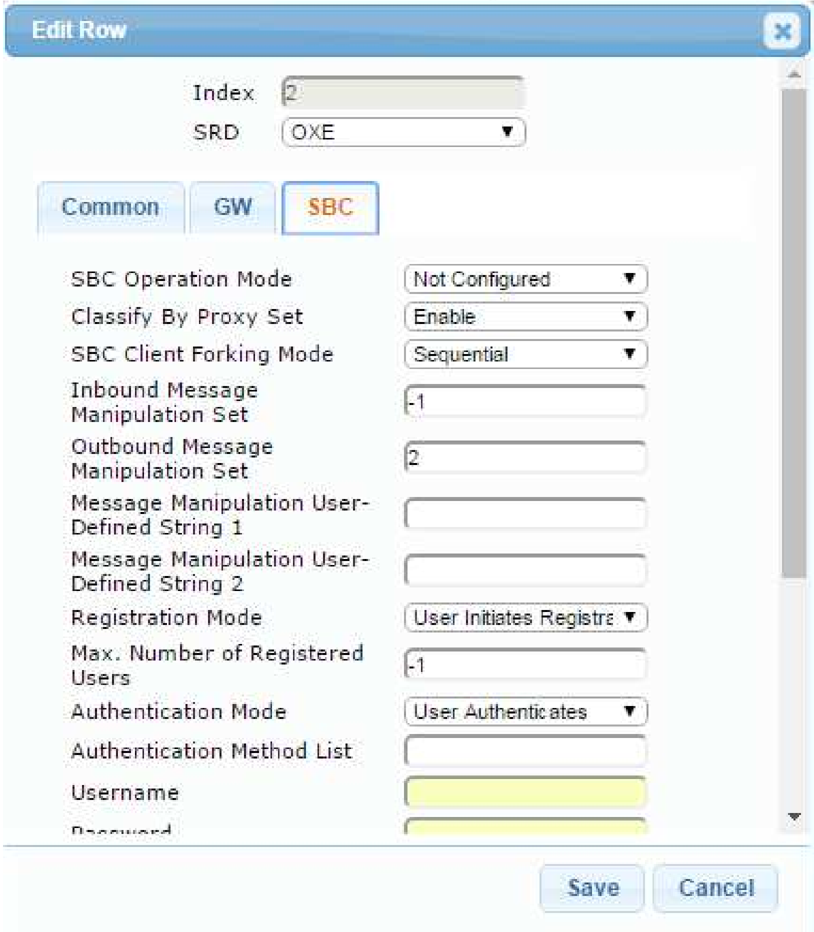
Index	Name	SRD	Type	SBC Operation Mode	Proxy Set	IP Profile	Media Realm	SIP Group Name	Classify By Proxy Set	Inbound Message Manipulation Set	Outbound Message Manipulatio Set
1	OneBox365	OneBox36	Server	Not Configur	Onebox	OneBox365	OneBox365		Enable	-1	1
2	OXE	OXE (#2)	Server	Not Configur	OXE	OXE	OneBox365	10.1.20.1	Enable	-1	2


Edit Row
Common
GW
SBC
Name
OneBox365
Type
Server
Proxy Set
Onebox
IP Profile
OneBox365
Media Realm
OneBox365
SIP Group Name
QoE Profile
None
Media Enhancement Profile
None
Bandwidth Profile
None
Always Use Src Address
No
Contact User
Local Host Name
UUI Format
Disable
Used By Routing Server
Not Used
Created By Routing Server
No
Save
Cancel


Common
GW
SBC
SBC Operation Mode
Not Configured
Classify By Proxy Set
Enable
SBC Client Forking Mode
Sequential
Inbound Message Manipulation Set
-1
Outbound Message Manipulation Set
1
Message Manipulation User-Defined String 1
Message Manipulation User-Defined String 2
Registration Mode
User Initiates Registra
Max. Number of Registered Users
-1
Authentication Mode
User Authenticates
Authentication Method List
Username
Password
Source URI Input
Destination URI Input
SIP Connect
No
Save
Cancel

Configure IP Group Table for OXE as below:

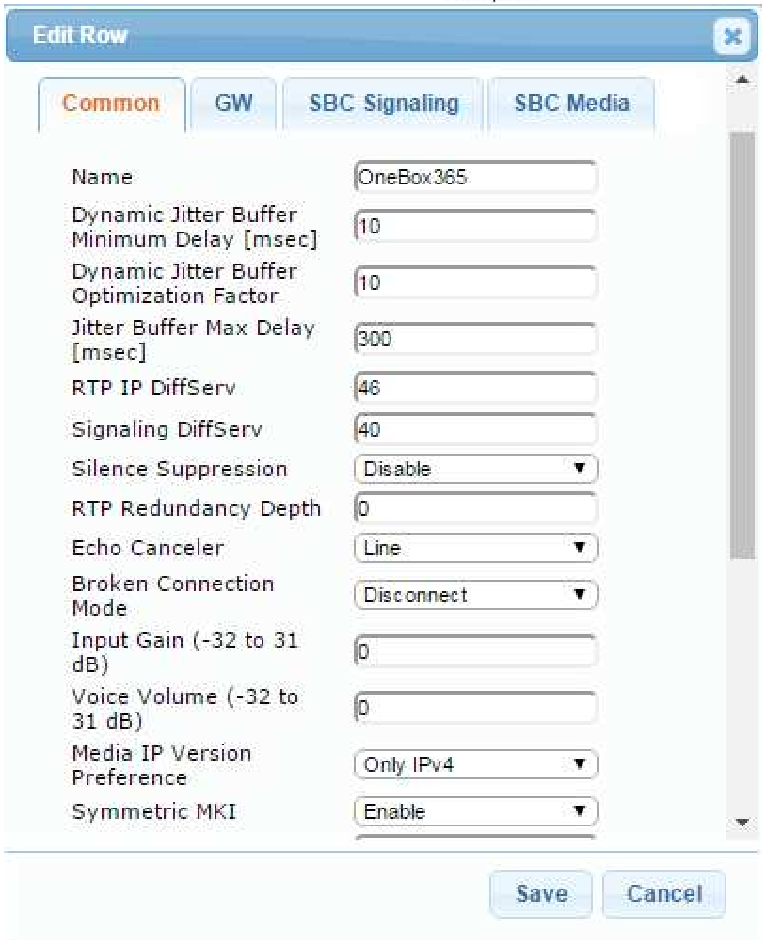




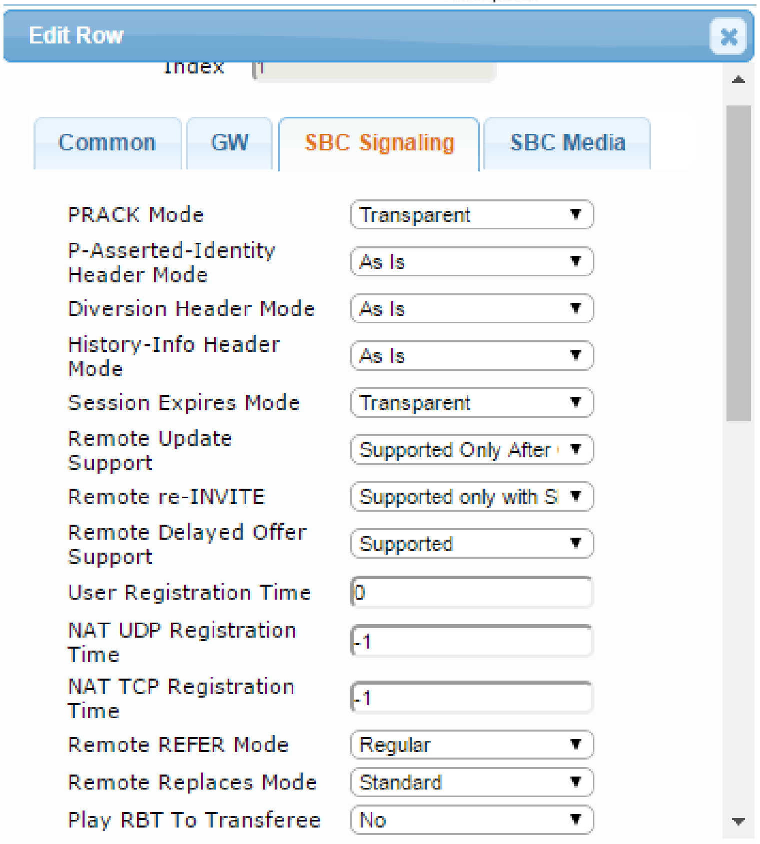
9.2.8 IP Profile Definition

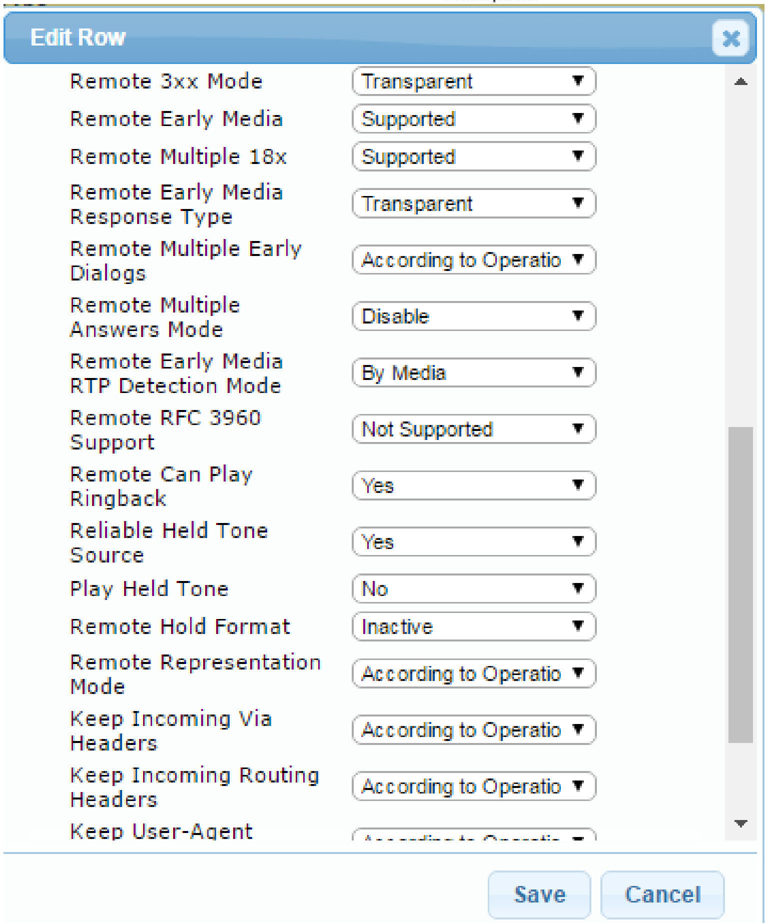
Open the IP Profile Settings page (**Configuration** tab > **Coders and Profiles** > **IP Profiles Settings**)

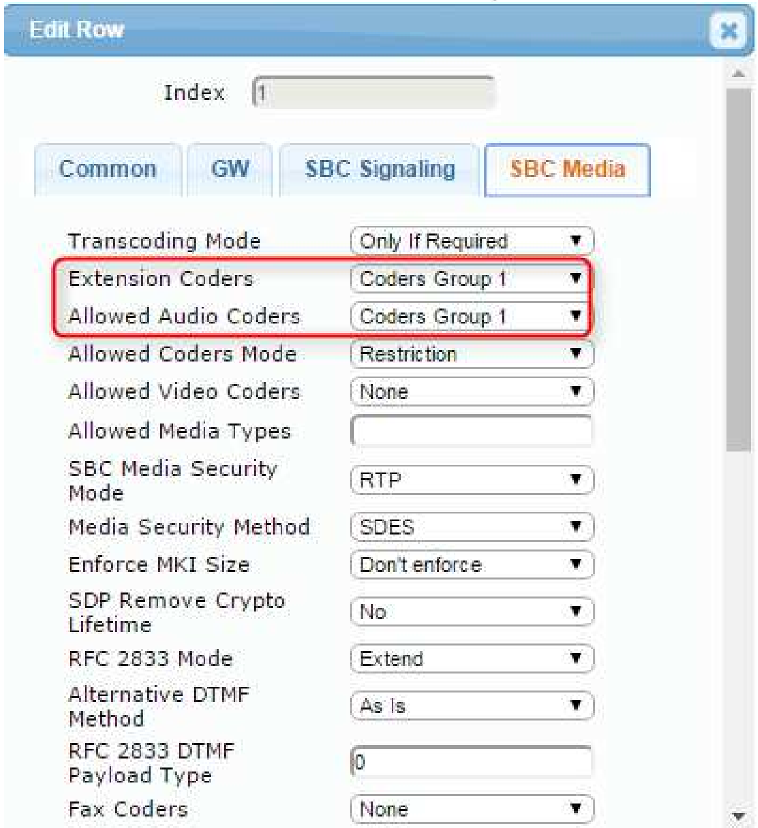
Configure IP Profile for Lync:


Edit Row
Common
GW
SBC Signaling
SBC Media
Name
OneBox365
Dynamic Jitter Buffer Minimum Delay [msec]
10
Dynamic Jitter Buffer Optimization Factor
10
Jitter Buffer Max Delay [msec]
300
RTP IP DiffServ
46
Signaling DiffServ
40
Silence Suppression
Disable
RTP Redundancy Depth
0
Echo Canceler
Line
Broken Connection Mode
Disconnect
Input Gain (-32 to 31 dB)
0
Voice Volume (-32 to 31 dB)
0
Media IP Version Preference
Only IPv4
Symmetric MKI
Enable
Save
Cancel

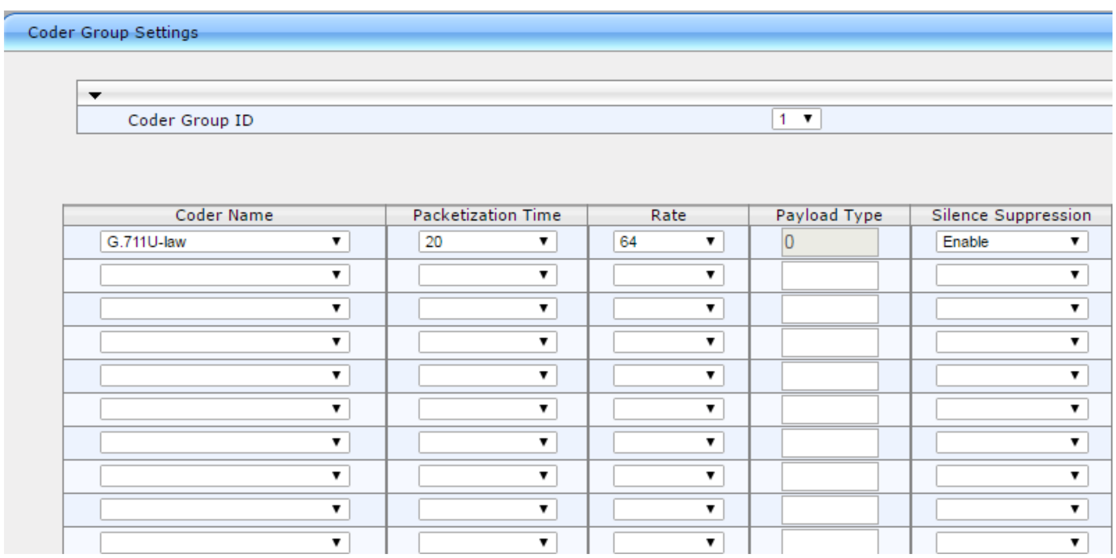

Edit Row
Index
1
Common
GW
SBC Signaling
SBC Media
Profile Preference
1
Coders
Default Coders group
Media Security Mode
Preferable - Single me
Is DTMF Used
Disable
First Tx DTMF Option
RFC 2833
Second Tx DTMF Option
Rx DTMF Option
Supported
Fax Signaling Method
No Fax
CNG Detector Mode
Disable
Vxx Modem Transport Type
Disable
NSE Mode
Disable
Play RB Tone to IP
Disable
Early Media
Enable
Progress Indicator to IP
Early 183
Disable
Early Answer Timeout
Save
Cancel

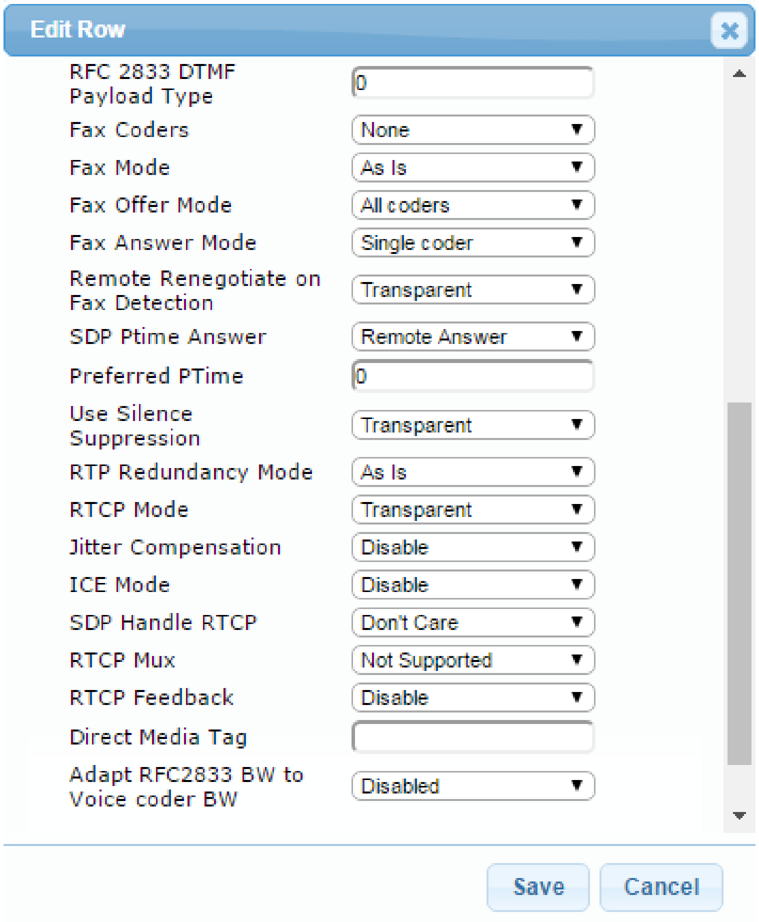

Edit Row
Index
Common
GW
SBC Signaling
SBC Media
PRACK Mode
Transparent
P-Asserted-Identity Header Mode
As Is
Diversion Header Mode
As Is
History-Info Header Mode
As Is
Session Expires Mode
Transparent
Remote Update Support
Supported Only After
Remote re-INVITE
Supported only with S
Remote Delayed Offer Support
Supported
User Registration Time
0
NAT UDP Registration Time
-1
NAT TCP Registration Time
-1
Remote REFER Mode
Regular
Remote Replaces Mode
Standard
Play RBT To Transferee
No


Edit Row
Remote 3xx Mode
Transparent
Remote Early Media
Supported
Remote Multiple 18x
Supported
Remote Early Media Response Type
Transparent
Remote Multiple Early Dialogs
According to Operatio
Remote Multiple Answers Mode
Disable
Remote Early Media RTP Detection Mode
By Media
Remote RFC 3960 Support
Not Supported
Remote Can Play Ringback
Yes
Reliable Held Tone Source
Yes
Play Held Tone
No
Remote Hold Format
Inactive
Remote Representation Mode
According to Operatio
Keep Incoming Via Headers
According to Operatio
Keep Incoming Routing Headers
According to Operatio
Keep User-Agent
Save
Cancel

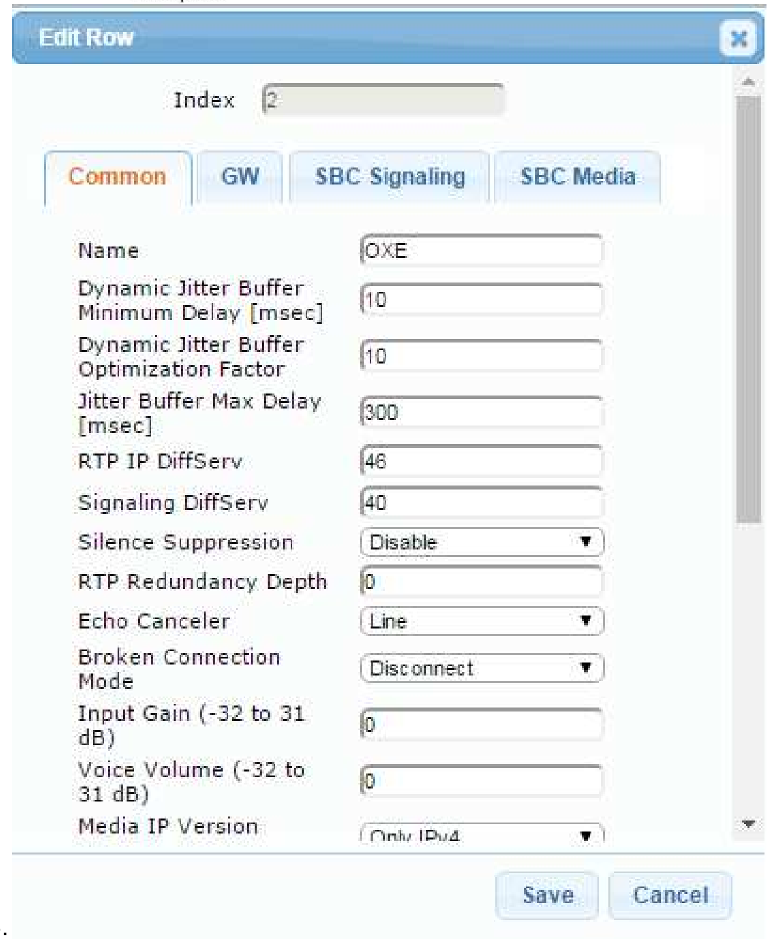


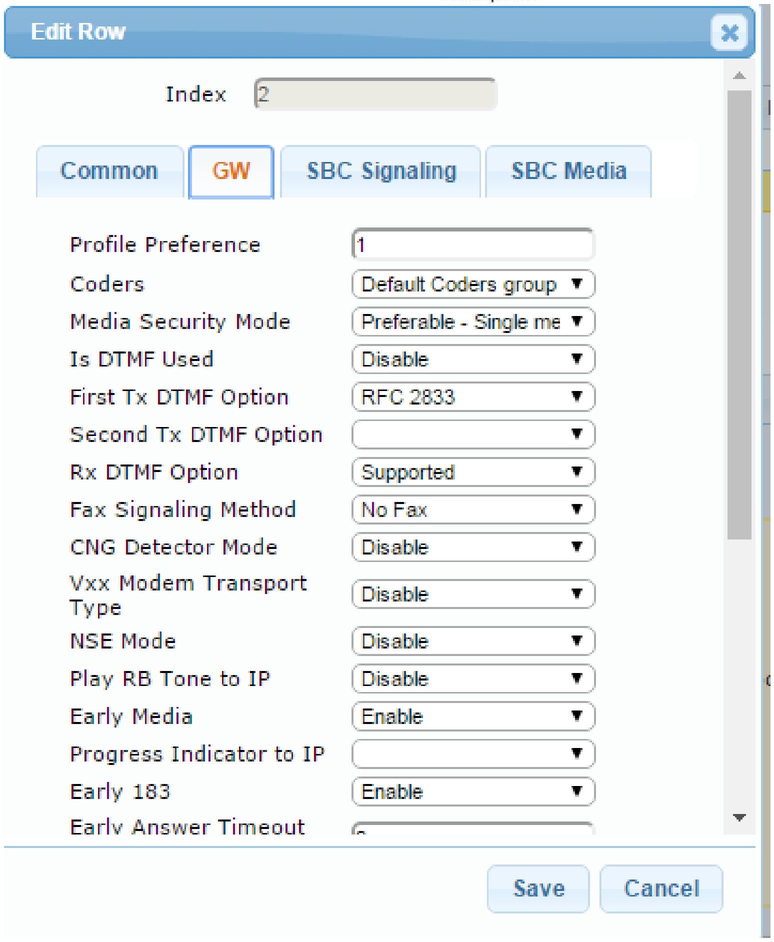
Coder Group 1 is set to G711 A law:



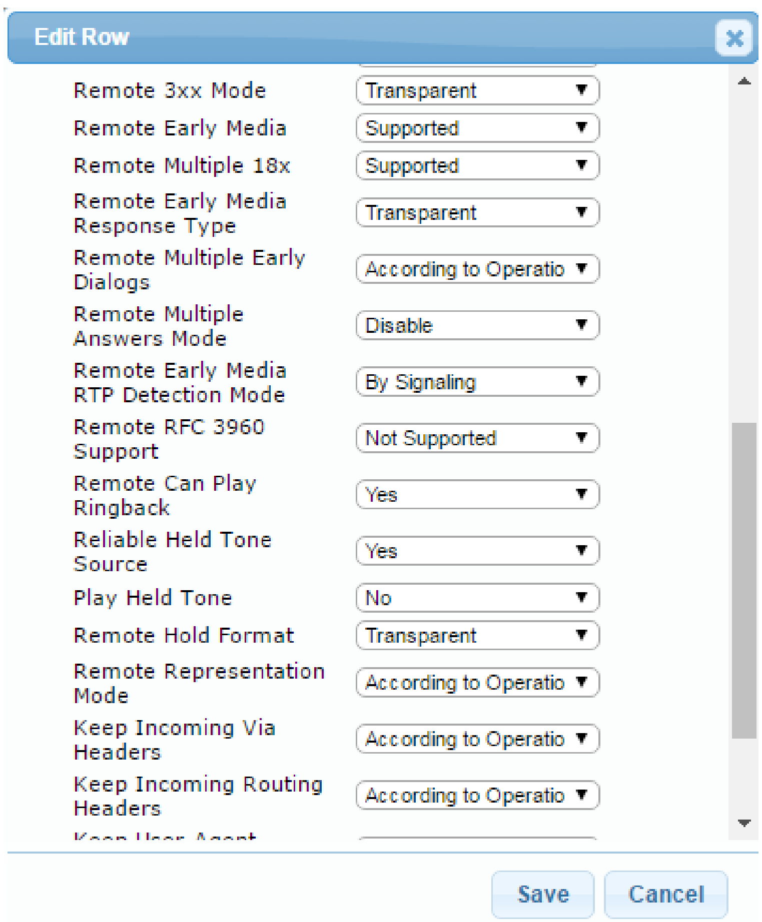

Edit Row
RFC 2833 DTMF Payload Type
0
Fax Coders
None
Fax Mode
As Is
Fax Offer Mode
All coders
Fax Answer Mode
Single coder
Remote Renegotiate on Fax Detection
Transparent
SDP Ptime Answer
Remote Answer
Preferred PTime
0
Use Silence Suppression
Transparent
RTP Redundancy Mode
As Is
RTCP Mode
Transparent
Jitter Compensation
Disable
ICE Mode
Disable
SDP Handle RTCP
Don't Care
RTCP Mux
Not Supported
RTCP Feedback
Disable
Direct Media Tag
Adapt RFC2833 BW to Voice coder BW
Disabled
Save
Cancel

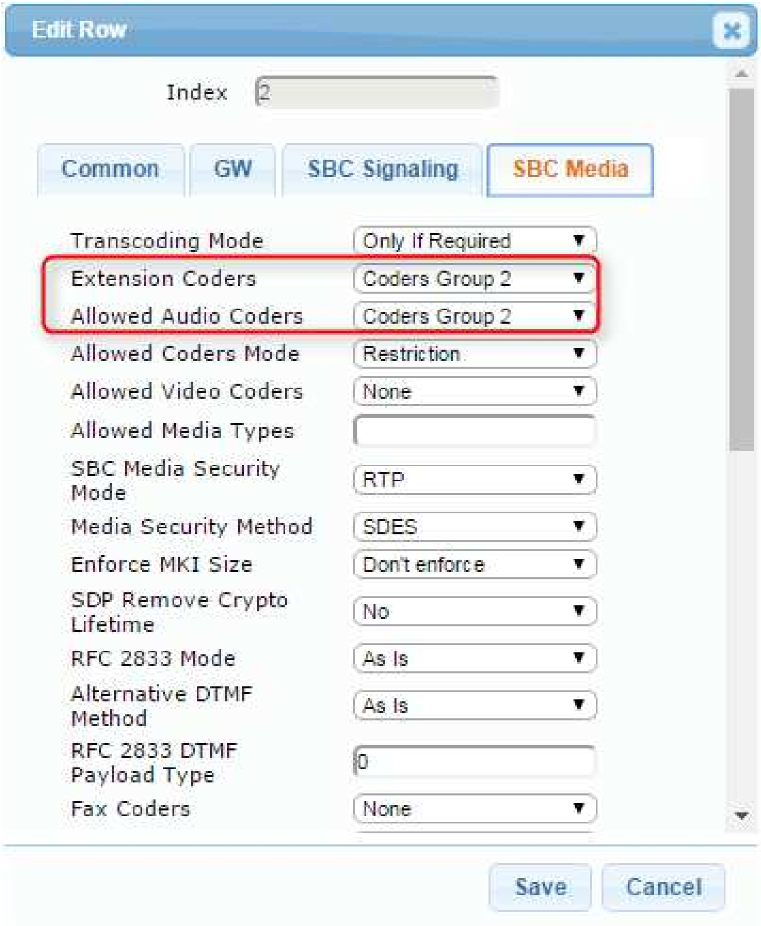
Configure IP Profile for OXE:




Edit Row
Index
2
Common
GW
SBC Signaling
SBC Media
Profile Preference
1
Coders
Default Coders group
Media Security Mode
Preferable - Single me
Is DTMF Used
Disable
First Tx DTMF Option
RFC 2833
Second Tx DTMF Option
Rx DTMF Option
Supported
Fax Signaling Method
No Fax
CNG Detector Mode
Disable
Vxx Modem Transport Type
Disable
NSE Mode
Disable
Play RB Tone to IP
Disable
Early Media
Enable
Progress Indicator to IP
Early 183
Enable
Early Answer Timeout
Save
Cancel


Edit Row
Index 2
Common
GW
SBC Signaling
SBC Media
PRACK Mode Transparent
P-Asserted-Identity Header Mode As Is
Diversion Header Mode As Is
History-Info Header Mode As Is
Session Expires Mode Transparent
Remote Update Support Supported
Remote re-INVITE Supported
Remote Delayed Offer Support Supported
User Registration Time 0
NAT UDP Registration Time -1
NAT TCP Registration Time -1
Remote REFER Mode Regular
Remote Replaces Mode Standard


Edit Row
Remote 3xx Mode Transparent
Remote Early Media Supported
Remote Multiple 18x Supported
Remote Early Media Response Type Transparent
Remote Multiple Early Dialogs According to Operatio
Remote Multiple Answers Mode Disable
Remote Early Media RTP Detection Mode By Signaling
Remote RFC 3960 Support Not Supported
Remote Can Play Ringback Yes
Reliable Held Tone Source Yes
Play Held Tone No
Remote Hold Format Transparent
Remote Representation Mode According to Operatio
Keep Incoming Via Headers According to Operatio
Keep Incoming Routing Headers According to Operatio
Save
Cancel



Coder Group 2 is set to G711 U law.

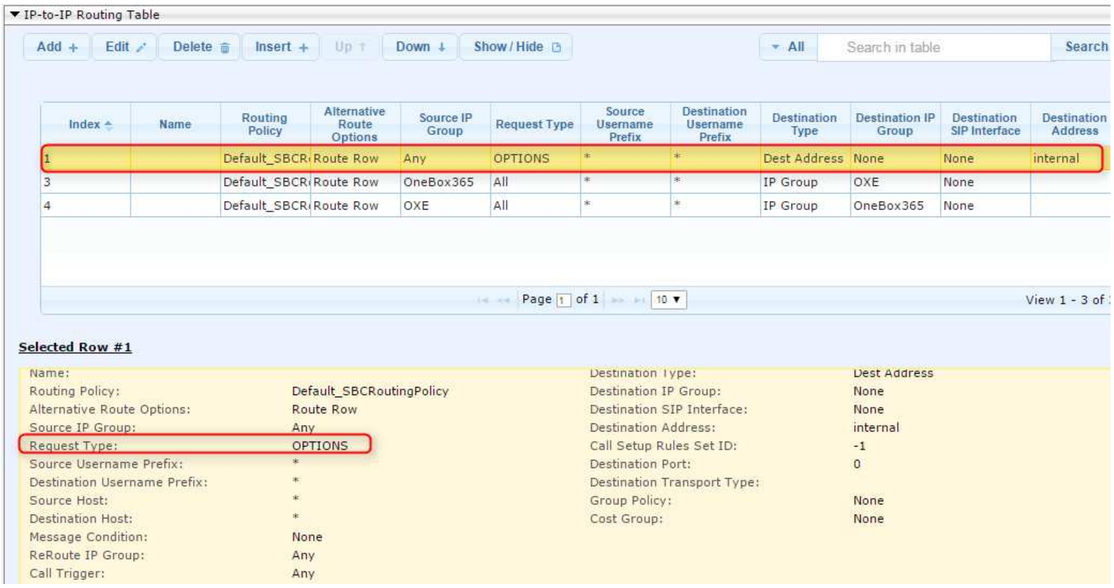
9.2.9 Configure Proxy 1 Registration:

Open the **'Proxy & Registration'** page (**Configuration** tab > **VoIP** menu > SIP Definitions > **Proxy & Registration**)

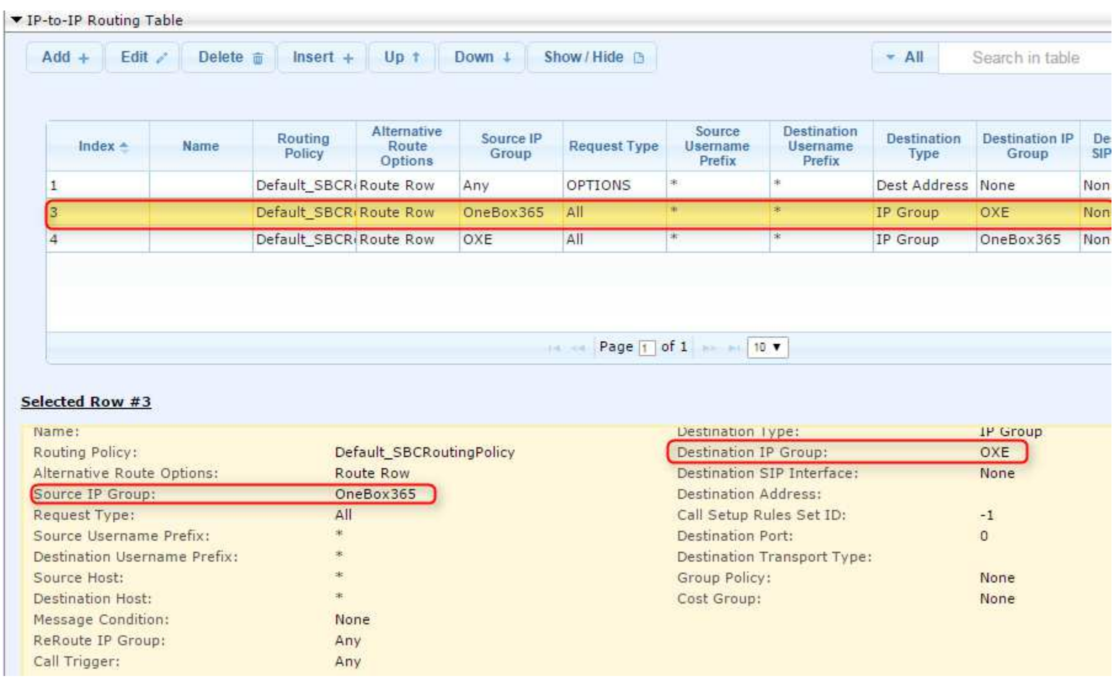
9.2.10 Configure Routing

Open the 'IP to Trunk Group Routing' page (**Configuration** tab > SBC > Routing SBC > **IP to IP Routing Table**).

Routing rule for OPTIONS :



Routing rule Lync to OXE:



Routing rule OXE to Lync :

▼ IP-to-IP Routing Table

Add + | Edit | Delete | Insert + | Up ↑ | Down ↓ | Show / Hide | ▾ All | Search in table

Index	Name	Routing Policy	Alternative Route Options	Source IP Group	Request Type	Source Username Prefix	Destination Username Prefix	Destination Type	Destination IP Group	Destinati SIP Interf
1		Default_SBCR	Route Row	Any	OPTIONS	*	*	Dest Address	None	None
3		Default_SBCR	Route Row	OneBox365	All	*	*	IP Group	OXE	None
4		Default_SBCR	Route Row	OXE	All	*	*	IP Group	OneBox365	None

Page 1 of 1 | 10

Selected Row #4

Name:		Destination Type:	IP Group
Routing Policy:	Default_SBCRoutingPolicy	Destination IP Group:	OneBox365
Alternative Route Options:	Route Row	Destination SIP Interface:	None
Source IP Group:	OXE	Destination Address:	
Request Type:	All	Call Setup Rules Set ID:	-1
Source Username Prefix:	*	Destination Port:	0
Destination Username Prefix:	*	Destination Transport Type:	
Source Host:	*	Group Policy:	None
Destination Host:	*	Cost Group:	None
Message Condition:	None		
ReRoute IP Group:	Any		
Call Trigger:	Any		

9.2.11 IP-to-IP Outbound Rules

▼ IP to IP Outbound Manipulation

Add + | Edit | Delete | Insert + | Up ↑ | Down ↓ | Show / Hide | ▾ All | Search in table | Search

Index	Name	Routing Policy	Additional Manipulati	Source IP Group	Destinatio IP Group	Source Username Prefix	Destinatio Username Prefix	Manipulate Item	Remove From Left	Remove From Right	Leave From Right	Prefix to Add	Suffix to Add
0	Lync vers	Default_SE	No	OneBox36	OXE	*	*	Destinatior	0	0	255		
1	Lync vers	Default_SE	No	OneBox36	OXE	*	*	Source UR	0	0	255		
2	OXE vers L	Default_SE	No	OXE	OneBox36	*	0	Destinatior	1	0	255	+33	
3	OXE vers L	Default_SE	No	OXE	OneBox36	0	*	Source UR	1	0	255	+33	

Rule 2: Remove the leading 0 and add +33 on destination URI

Selected Row #2

Rule		Action	
Name:	OXE vers Lync Dst	Manipulated Item:	Destination URI
Routing Policy:	Default_SBCRoutingPolicy	Remove From Left:	1
Additional Manipulation:	No	Remove From Right:	0
Source IP Group:	OXE	Leave From Right:	255
Destination IP Group:	OneBox365	Prefix to Add:	+33
Source Username Prefix:	*	Suffix to Add:	
Destination Username Prefix:	0	Privacy Restriction Mode:	Transparent
Source Host:	*		
Destination Host:	*		
Calling Name Prefix:	*		
Message Condition:	None		

Rule 3: Remove the leading 0 and add +33 on source URI

Selected Row #3

Rule		Action	
Name:	OXE vers Lync Src	Manipulated Item:	Source URI
Routing Policy:	Default_SBCRoutingPolicy	Remove From Left:	1
Additional Manipulation:	No	Remove From Right:	0
Source IP Group:	OXE	Leave From Right:	255
Destination IP Group:	OneBox365	Prefix to Add:	+33
Source Username Prefix:	0	Suffix to Add:	
Destination Username Prefix:	*	Privacy Restriction Mode:	Transparent
Source Host:	*		
Destination Host:	*		
Calling Name Prefix:	*		
Message Condition:	None		

9.2.12 Sip Header Manipulations

➢ **To configure Sip Headers manipulations :**

Open the 'IP to Trunk Group Routing' page (Configuration **tab >** VoIP **menu > Sip Definitions > Msg Policy & Manipulation >** Messages Manipulation**).**

▼ Message Manipulations

Add + | Edit | Delete | Insert + | Up ↑ | Down ↓ | Show / Hide | ▼ All | Search in table | Search

Index	Name	Manipulation Set ID	Message Type	Condition	Action Subject	Action Type	Action Value	Row Role
1	GW OPTIONS	1	options		header.request-u	Modify	'etesting9.etestin	Use Current
2	GW OPTIONS	2	options		header.request-u	Modify	'etesting9.etestin	Use Current
3	Lync_from	2			header.from.url.I	Modify	'10.1.2.63'	Use Current
4	Lync_ppreferred	2			header.p-preferr	Modify	header.from.url.I	Use Current
5	Lync_passerted	2			header.p-asserte	Modify	header.from.url.I	Use Current
7	Lync_remove_Priv	2	invite		header.Privacy	Remove		Use Current

Selected Row #1

Index:	1	Action Subject:	header.request-uri.url.host
Name:	GW OPTIONS	Action Type:	Modify
Manipulation Set ID:	1	Action Value:	'etesting9.etesting.lab'
Message Type:	options	Row Role:	Use Current Condition
Condition:			

Selected Row #2

Index:	2	Action Subject:	header.request-uri.url.host
Name:	GW OPTIONS	Action Type:	Modify
Manipulation Set ID:	2	Action Value:	'etesting9.etesting.lab'
Message Type:	options	Row Role:	Use Current Condition
Condition:			

Selected Row #3

Index:	3	Action Subject:	header.from.url.host
Name:	Lync_from	Action Type:	Modify
Manipulation Set ID:	2	Action Value:	'10.1.2.63'
Message Type:		Row Role:	Use Current Condition
Condition:			

Selected Row #4

Index:	4	Action Subject:	header.p-preferred-identity.url.host
Name:	Lync_ppreferred	Action Type:	Modify
Manipulation Set ID:	2	Action Value:	header.from.url.host
Message Type:		Row Role:	Use Current Condition
Condition:			

Selected Row #5

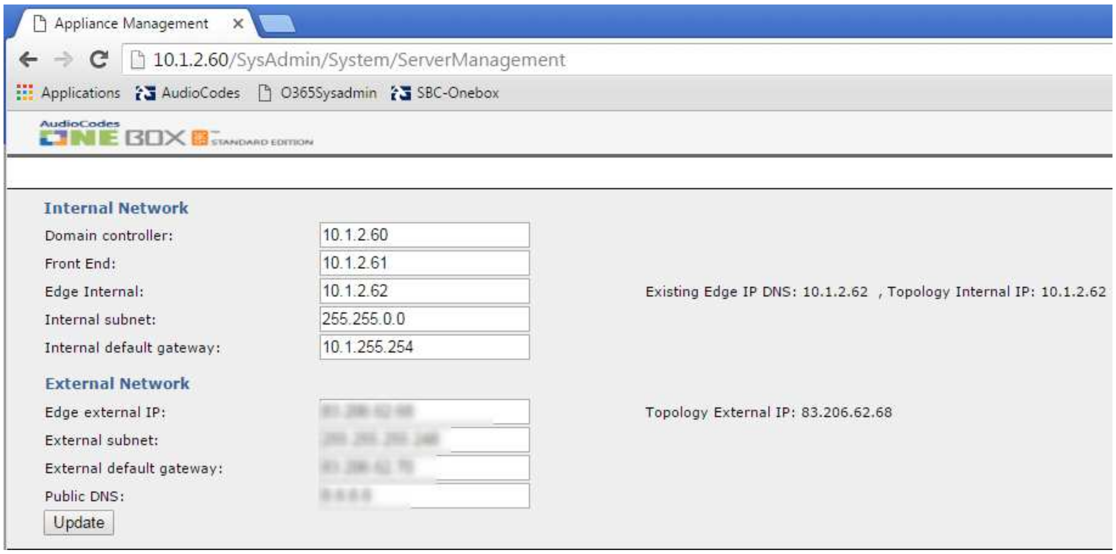
Index:	5	Action Subject:	header.p-asserted-identity.url.host
Name:	Lync_passerted	Action Type:	Modify
Manipulation Set ID:	2	Action Value:	header.from.url.host
Message Type:		Row Role:	Use Current Condition
Condition:			

Selected Row #7

Index:	7	Action Subject:	header.Privacy
Name:	Lync_remove_Priv	Action Type:	Remove
Manipulation Set ID:	2	Action Value:	
Message Type:	invite	Row Role:	Use Current Condition
Condition:			

10 Appendix B : CloudBond box configuration

Step 1: Configure the IP addresses of the servers by using the SysAdmin interface:



Step2: Put the DC as the NTP reference on all servers to be sure that they all have the same time/date.

Step 3: Create the necessary DNS entries on the enterprise DNS and the Cloudbond box DC DNS. Typically:

1. On the enterprise DNS server, a stub zone matching the CloudBond 365 resource domain Fully Qualified Domain Name (FQDN)
2. On the CloudBond 365 Controller server, a stub zone matching the corporate enterprise DNS zone.
3. On the public DNS server, a zone matching the FQDN of the SIP domain specified for CloudBond 365.

Refer to AudioCodes document:
LTRT-26323 AudioCodes CloudBond 365 Deployment Guide Ver. 7.0.pdf chapters 3.4 and B2

Step 4: Activate the PKI on the DC:

Refer to AudioCodes document:
LTRT-26443 CloudBond 365 Certificates Configuration Note Ver. 7.0.pdf, chapter B3

Step 5: Create and load the internal certificates for FE and Edge servers by using the newly created CloudBond PKI.

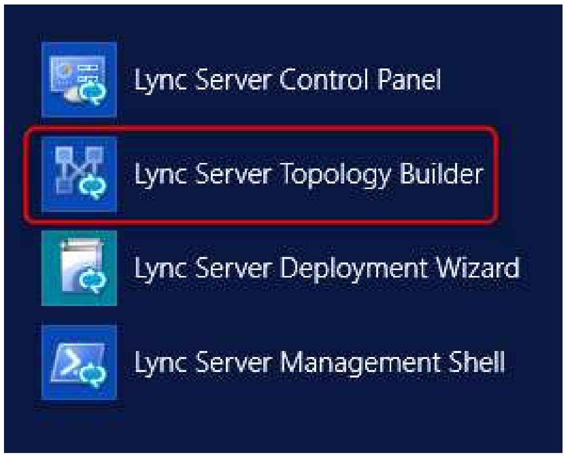
Refer to AudioCodes document:
LTRT-26443 CloudBond 365 Certificates Configuration Note Ver. 7.0.pdf, chapter 10.

Step 6: If needed, load the public certificates on Edge and Reverse Proxy server.

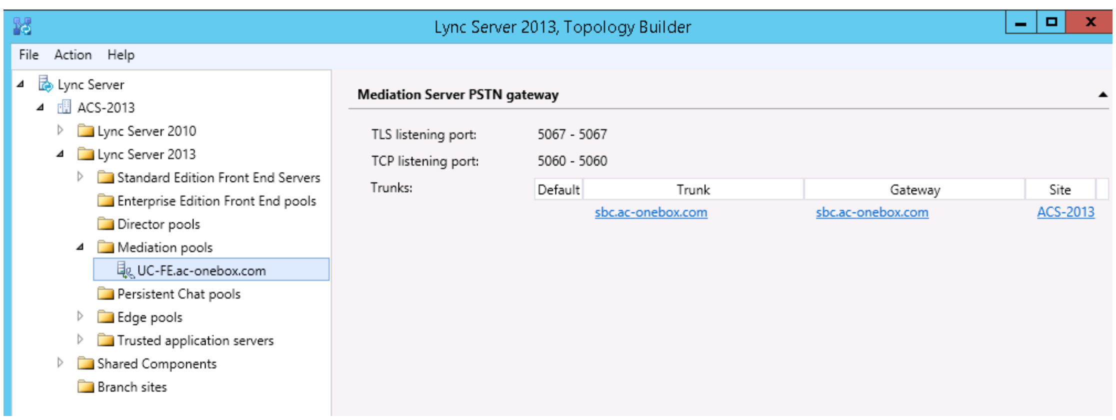
11 Appendix C : Lync 2013 Configuration

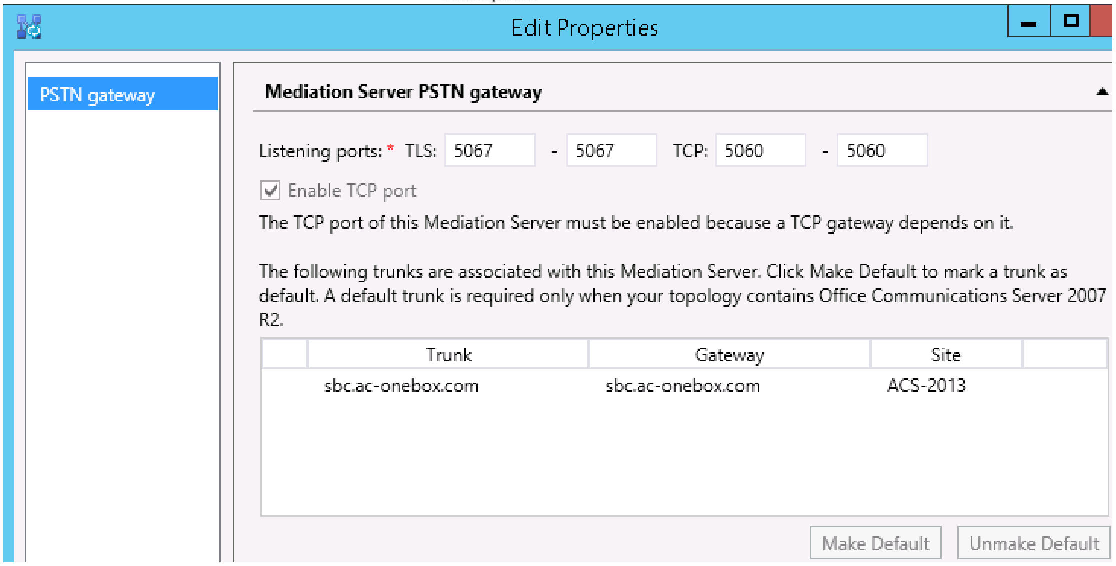
This section describes the way to configure the Mediant SBC as a PSTN Gateway and its association with the Mediation Server:

Run *Lync Server Topology Builder* program*:*

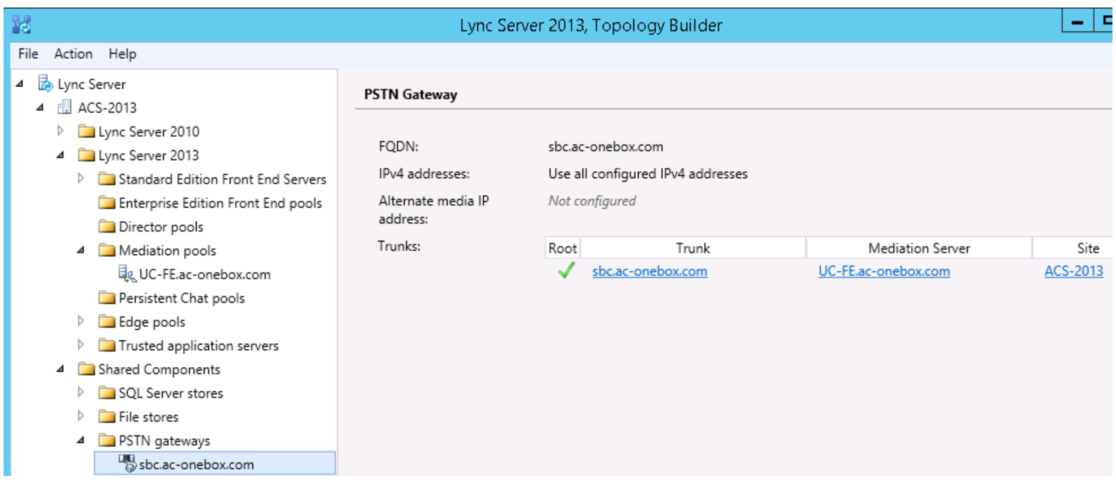


Go into *Mediation Pool* and *Edit Properties* on the server.

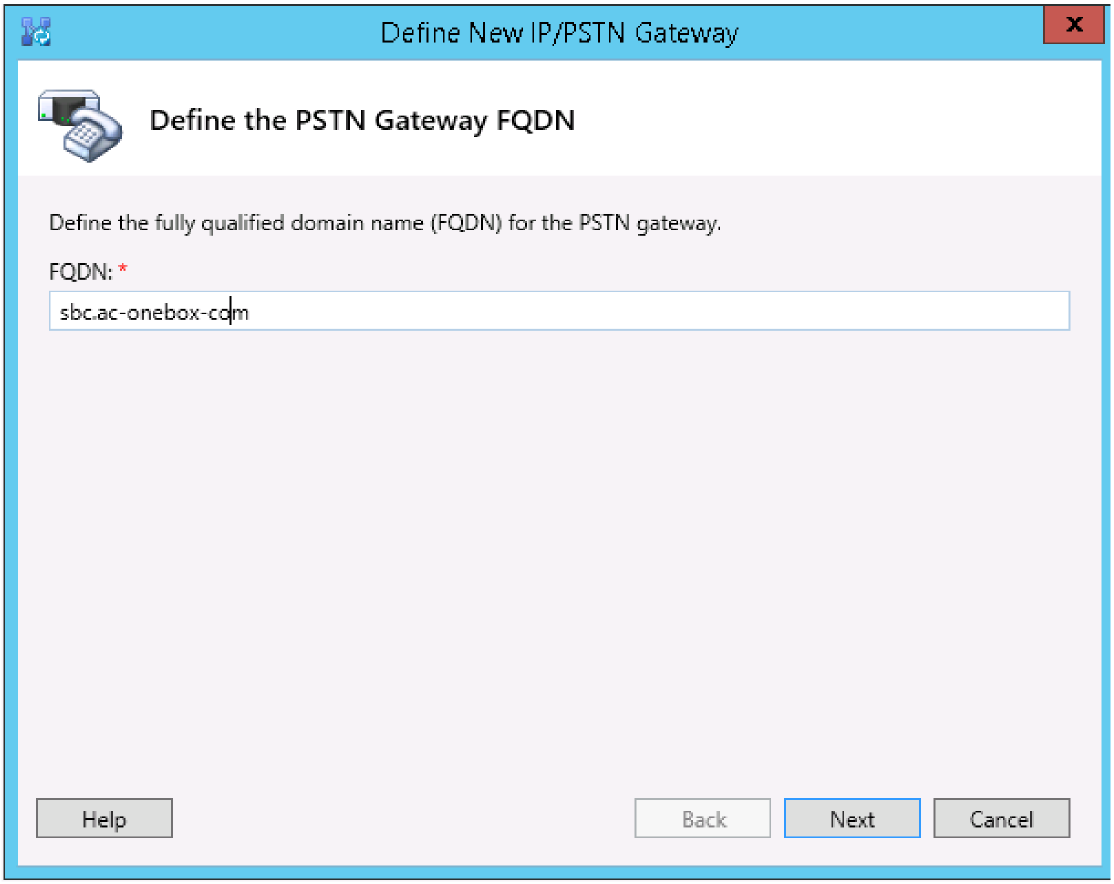




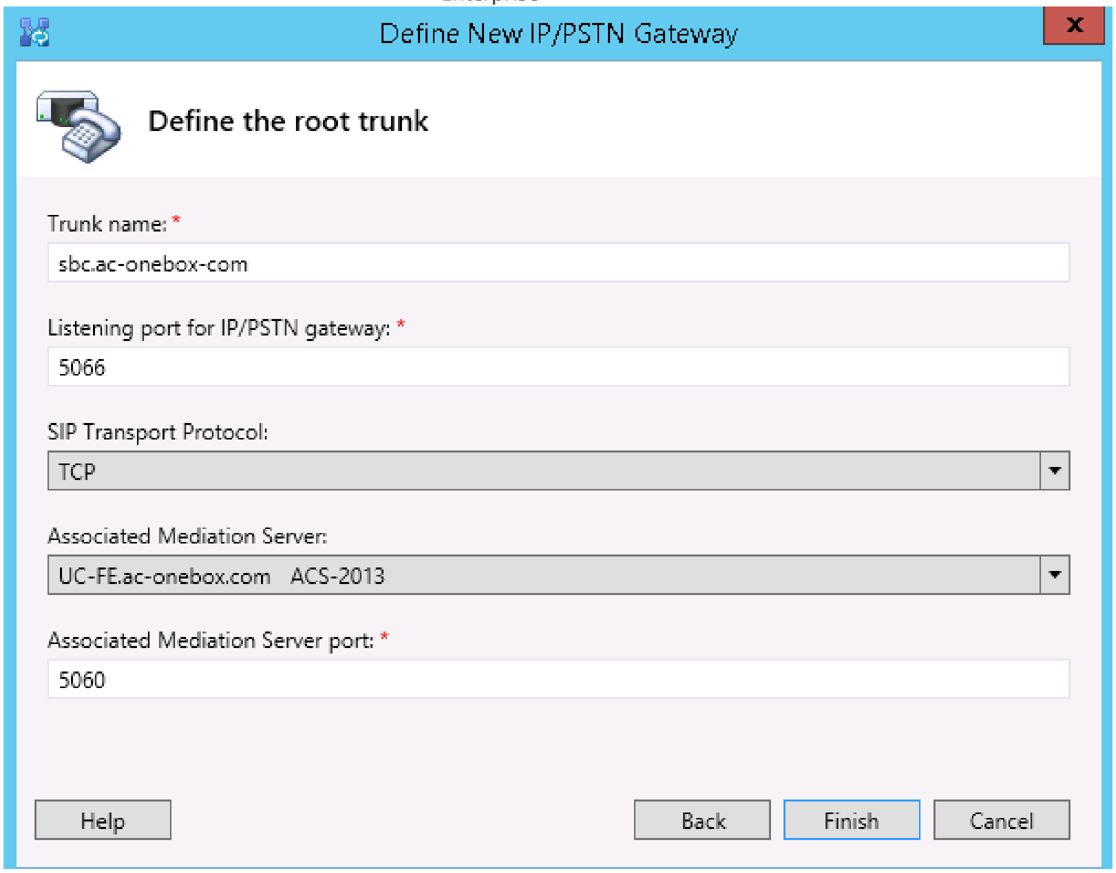
Create a new IP/PSTN gateway:



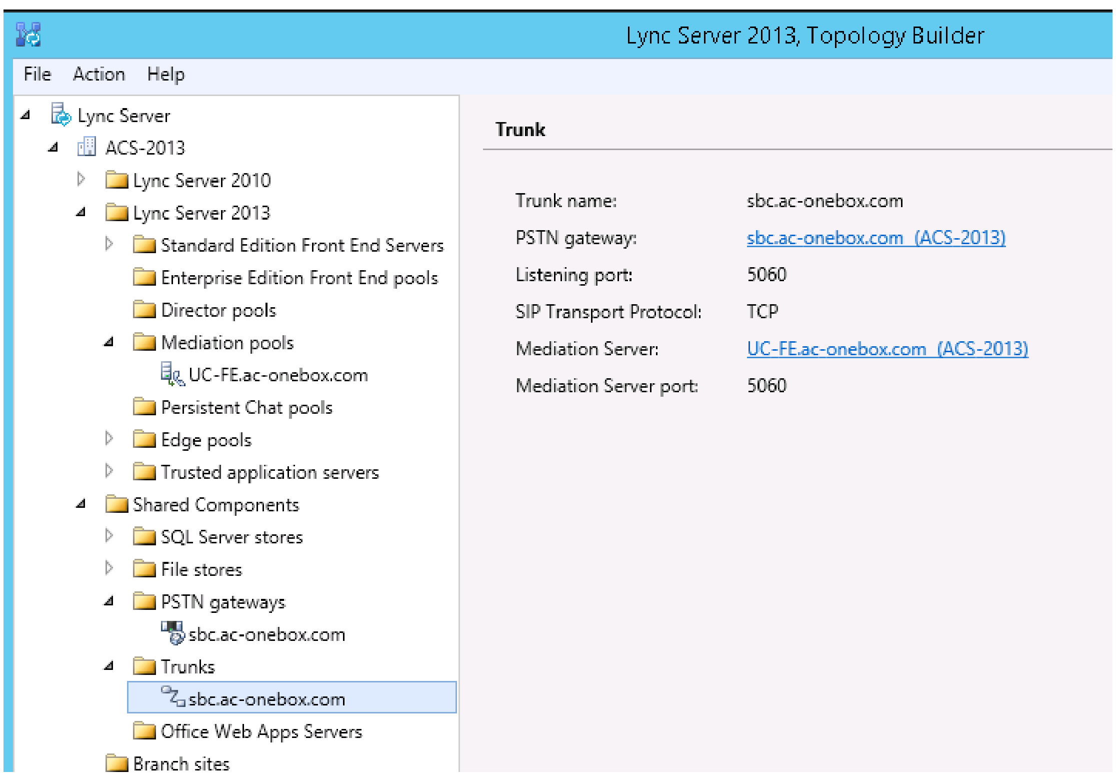
Enter the SBC FQDN or IP address:



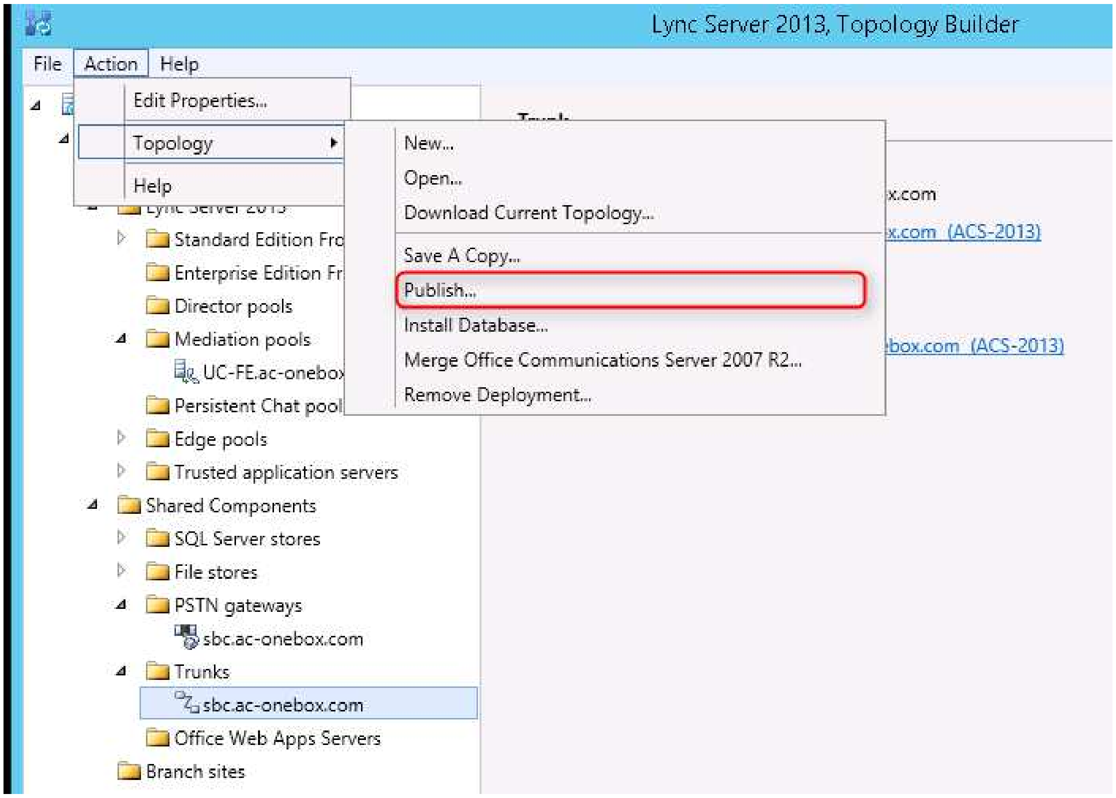
Then, choose the TCP protocol and enter the listening port of the SBC: 5060

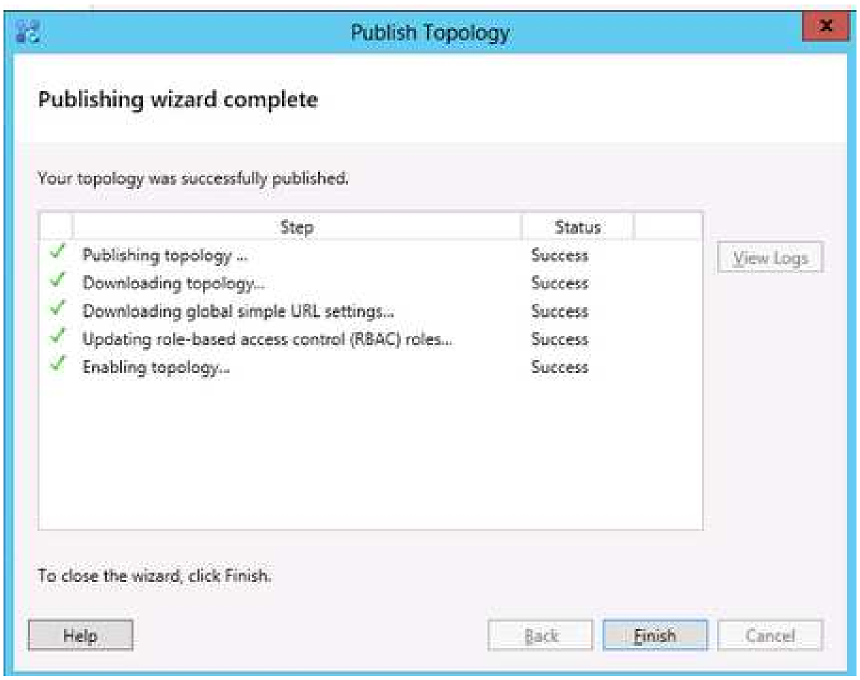


You can now see the new trunk:

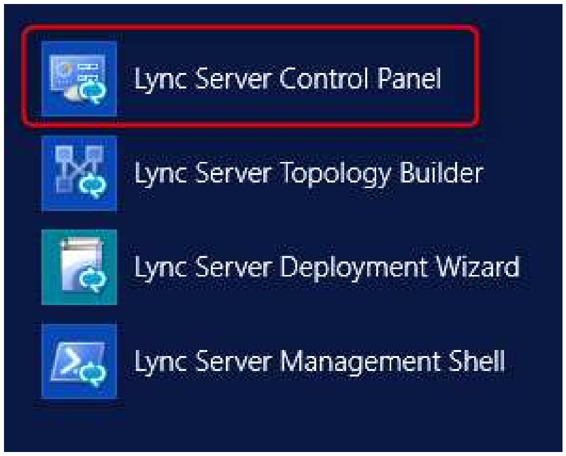


To activate all changes go to main menu and in *Action* -> *Topology* choose option *Publish*....

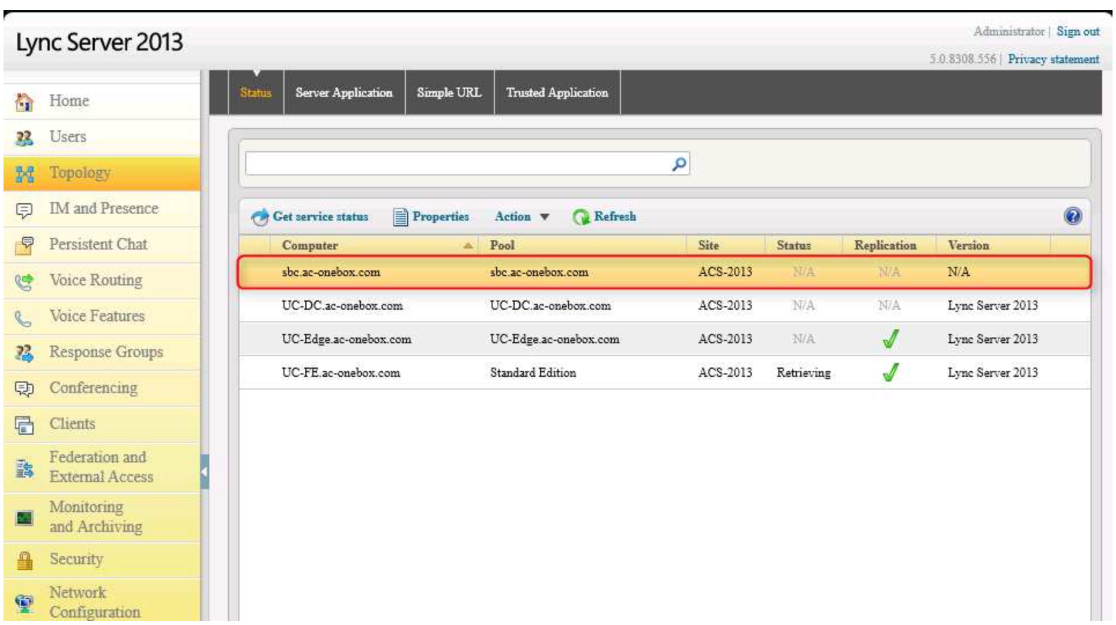




Now open *Lync Server Control Panel*

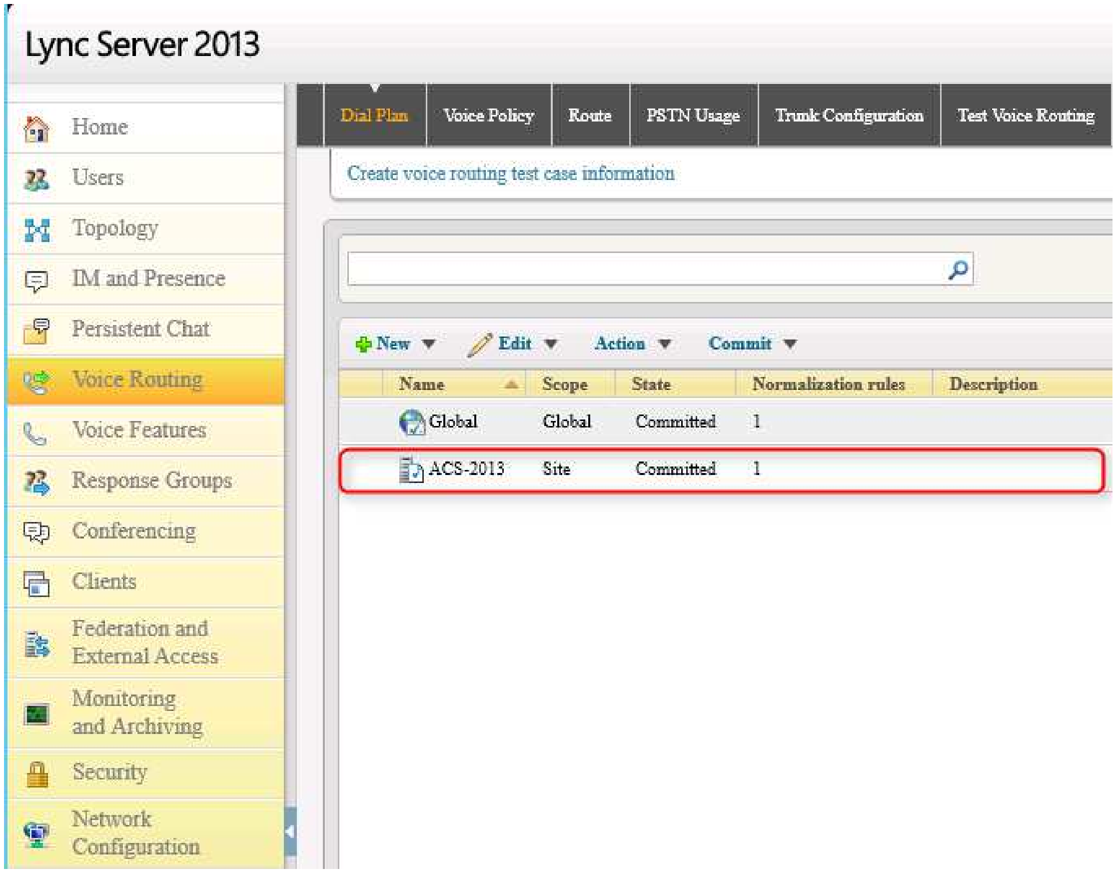


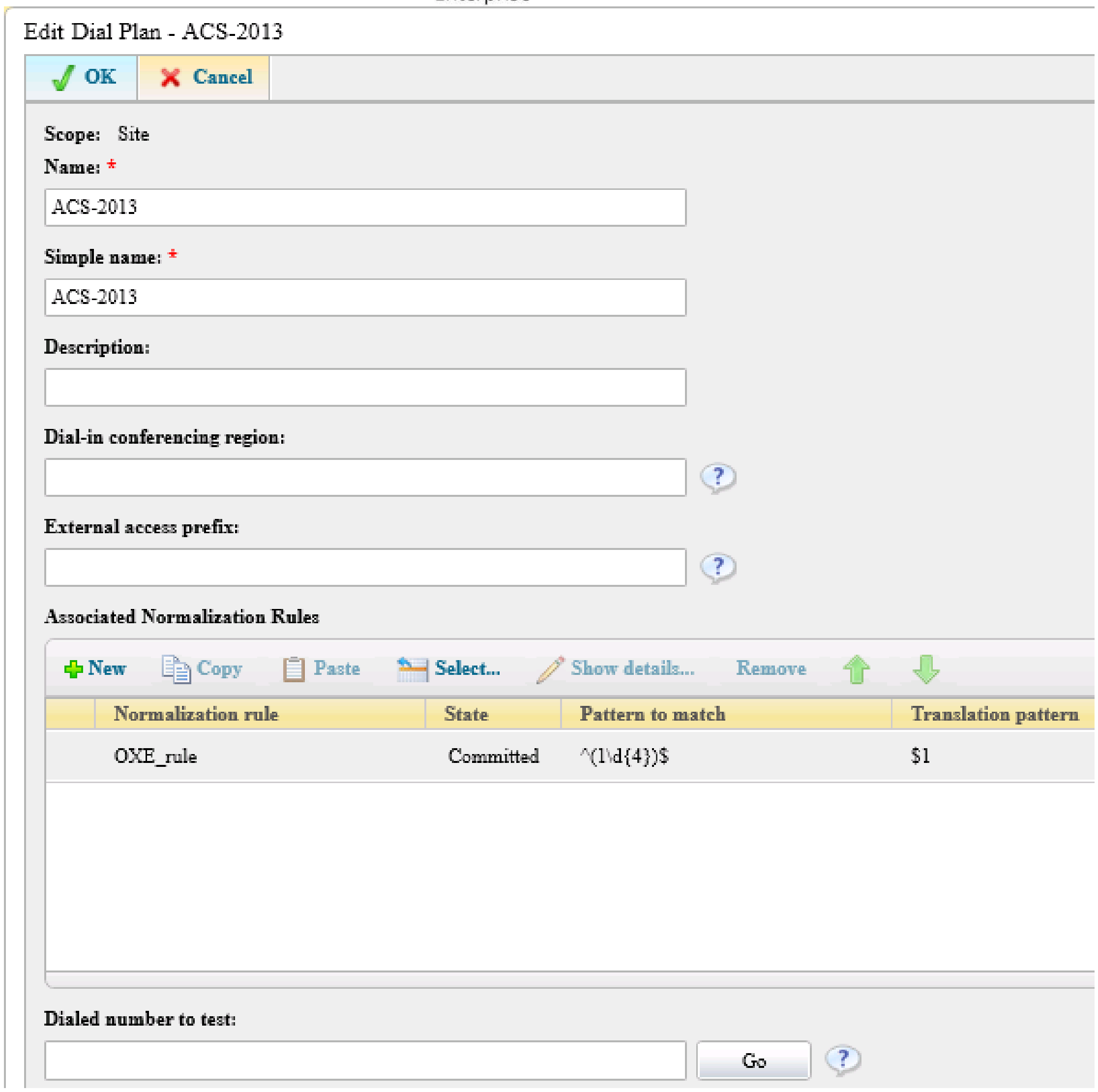
In *Topology* tab you should see the newly created gateway: sbc.ac-onebox.com



Voice Routing configuration

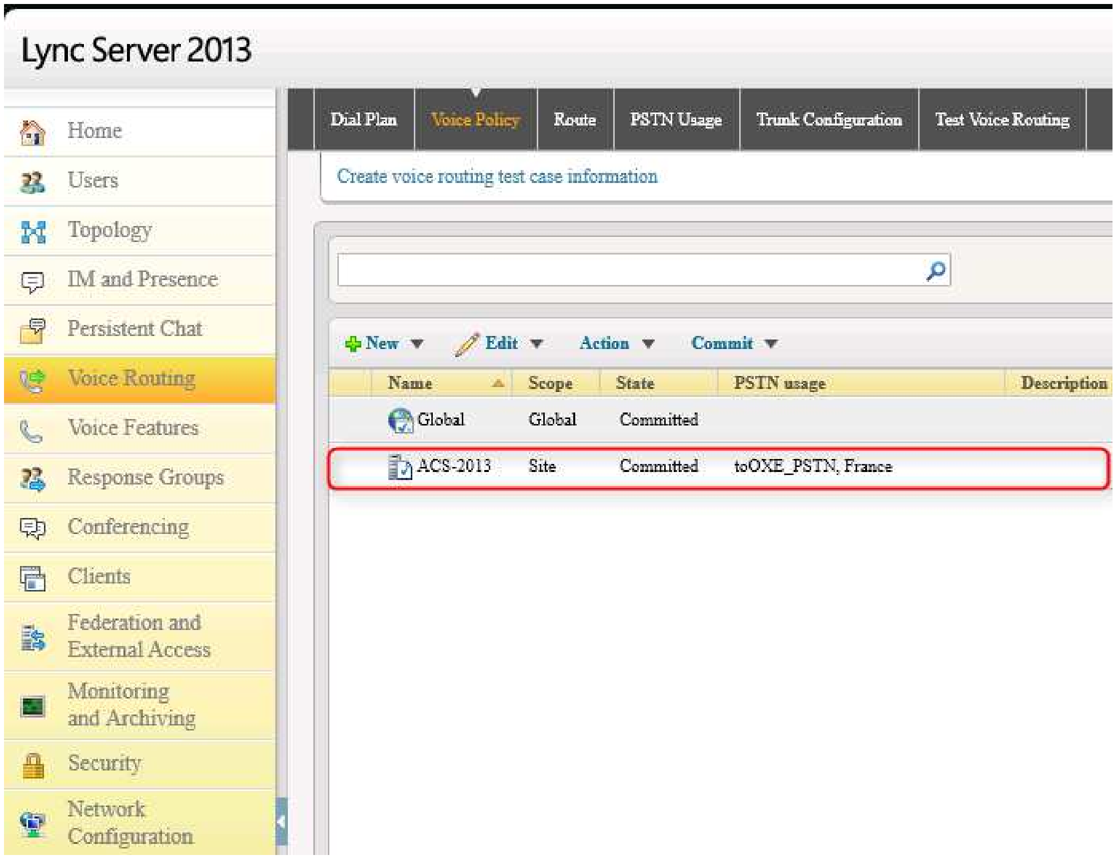
In Dial Plan section, Create a new Pool dial plan and choose the newly created PSTN gateway from the Select a service dialog box.

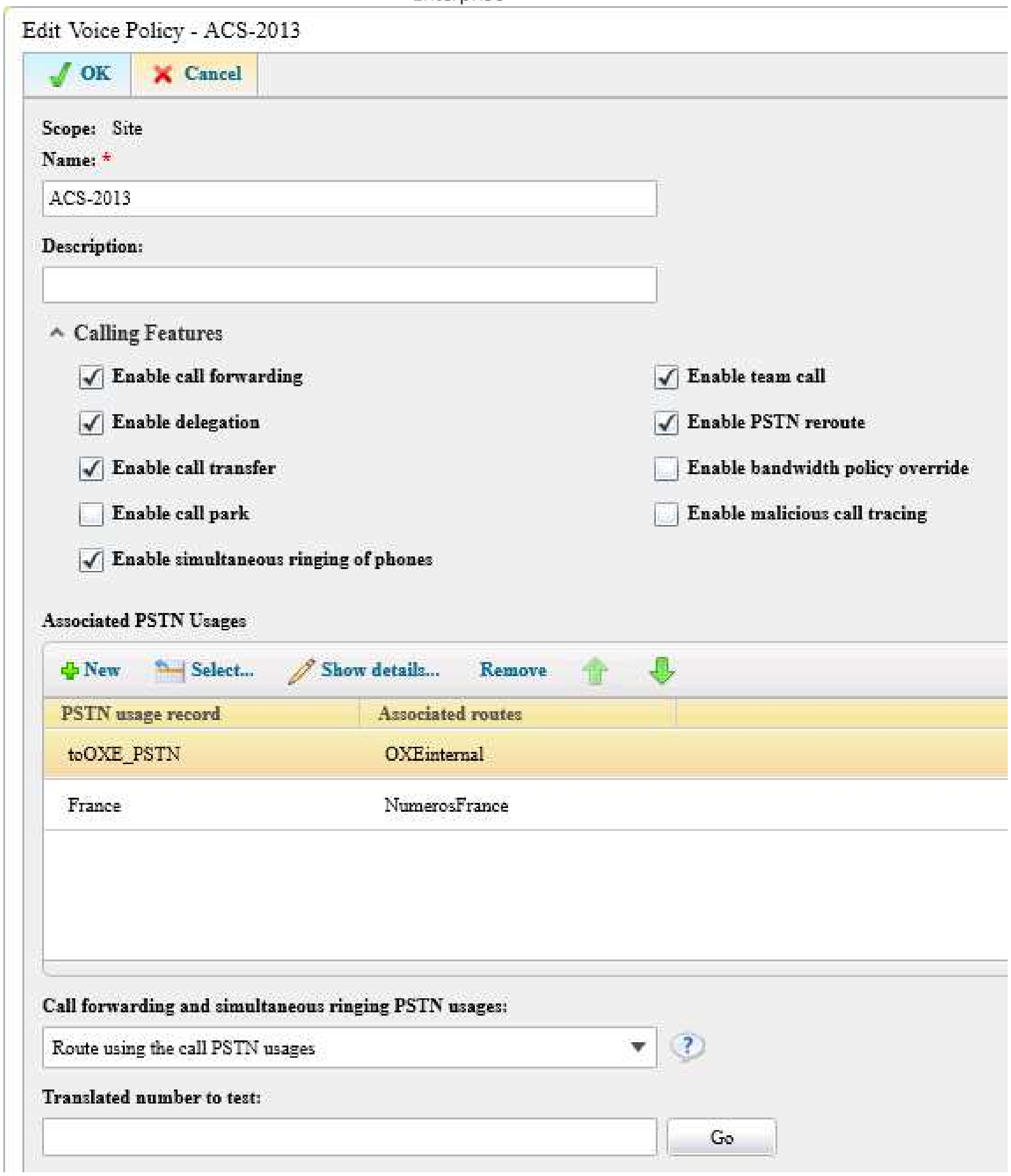




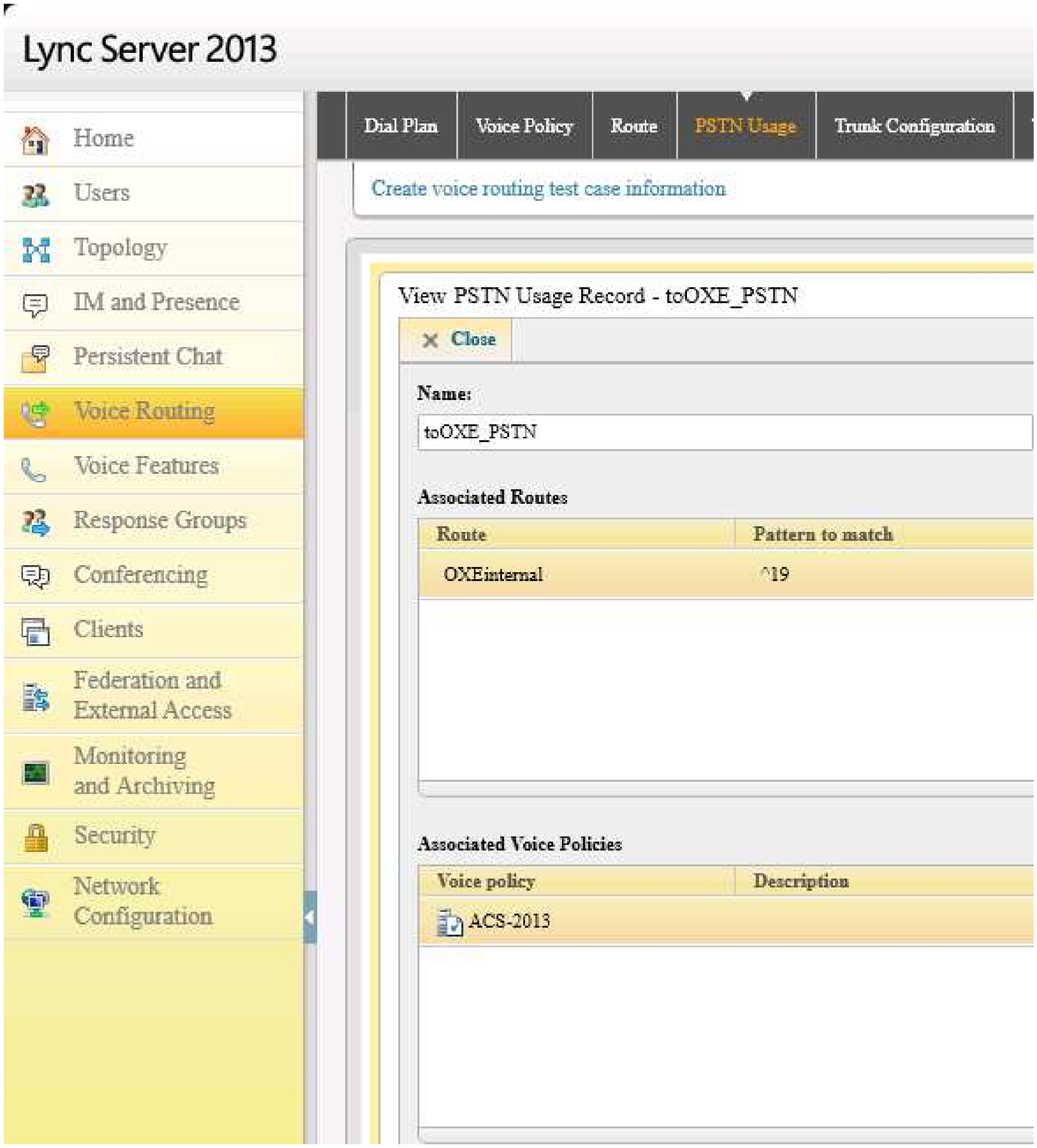
Create a normalization rule that fits your needs: OXE_rule

Create a new Voice Policy: ACS-2013

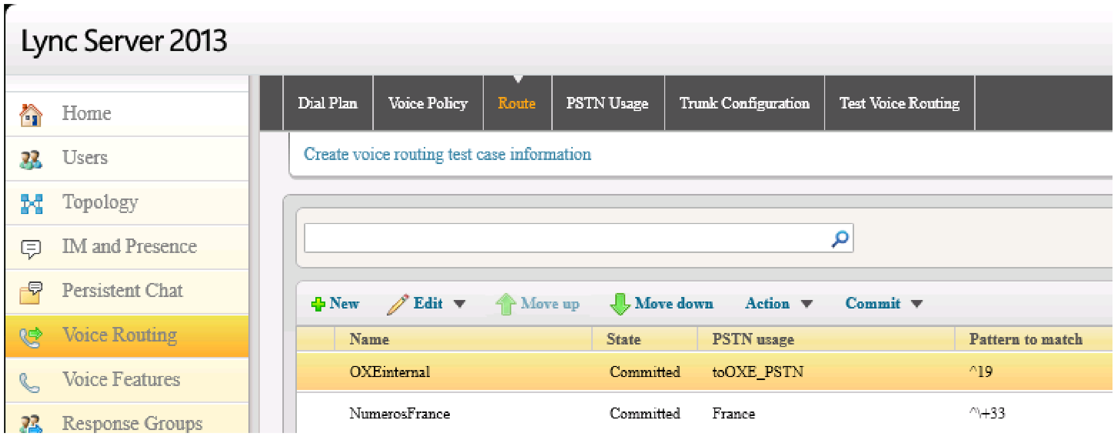


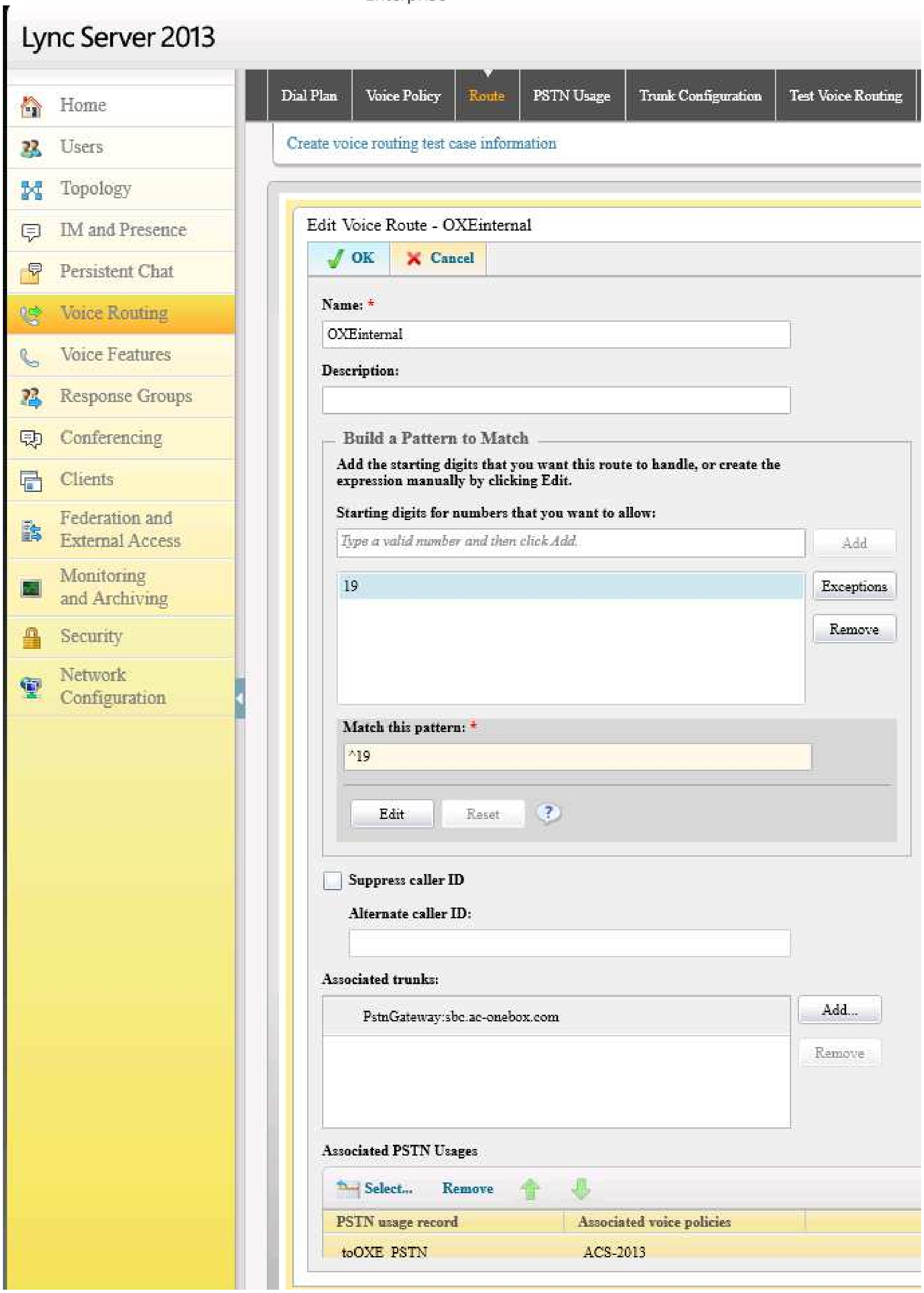

Edit Voice Policy - ACS-2013
OK
Cancel
Scope: Site
Name: *
ACS-2013
Description:
Calling Features
Enable call forwarding
Enable team call
Enable delegation
Enable PSTN reroute
Enable call transfer
Enable bandwidth policy override
Enable call park
Enable malicious call tracing
Enable simultaneous ringing of phones
Associated PSTN Usages
New
Select...
Show details...
Remove
PSTN usage record
Associated routes
toOXE_PSTN
OXEinternal
France
NumerosFrance
Call forwarding and simultaneous ringing PSTN usages:
Route using the call PSTN usages
Translated number to test:
Go

Create a new PSTN Usage ToOXE_PSTN

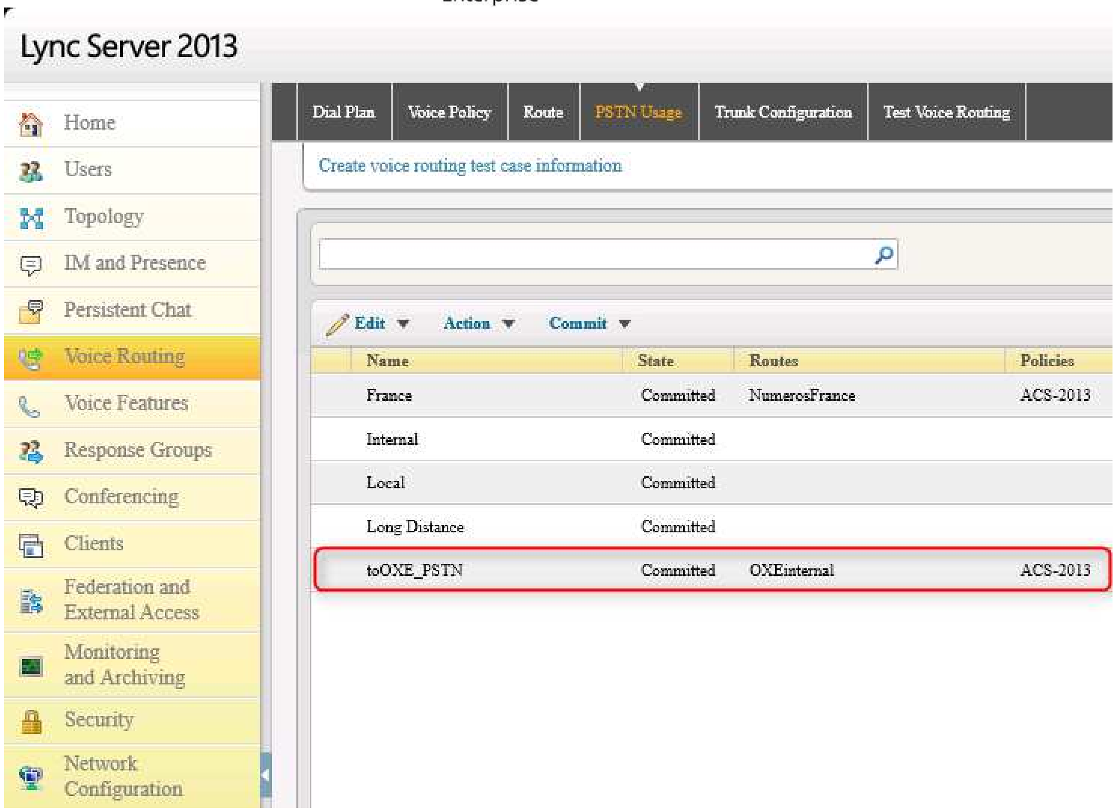


Create a new Route to the SBC: toOXE_PSTN

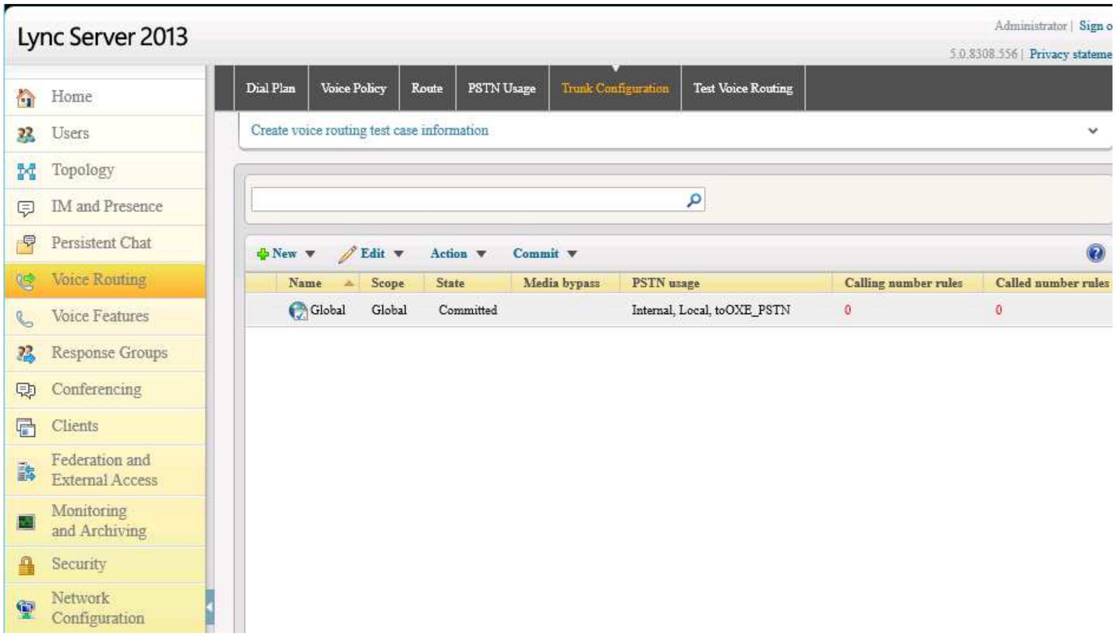


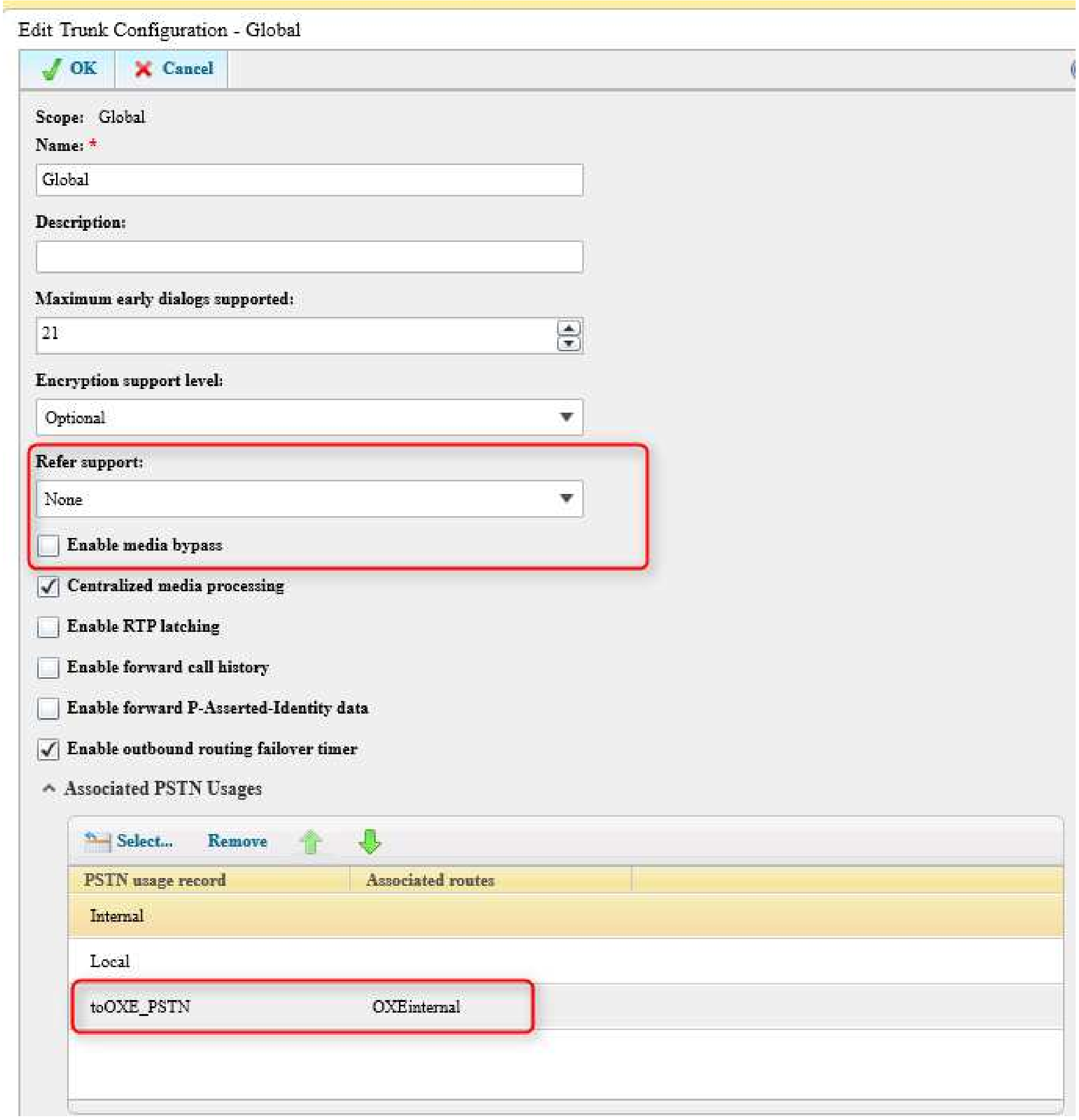


- Associated trunk = sbc.ac-onebox.com
- Associated PSTN Usage = ToOXE_PSTN


Lync Server 2013
Home
Users
Topology
IM and Presence
Persistent Chat
Voice Routing
Voice Features
Response Groups
Conferencing
Clients
Federation and External Access
Monitoring and Archiving
Security
Network Configuration
Dial Plan
Voice Policy
Route
PSTN Usage
Trunk Configuration
Test Voice Routing
Create voice routing test case information
Edit
Action
Commit
Name
State
Routes
Policies
France
Committed
NumerosFrance
ACS-2013
Internal
Committed
Local
Committed
Long Distance
Committed
toOXE_PSTN
Committed
OXEinternal
ACS-2013

Trunk configuration:



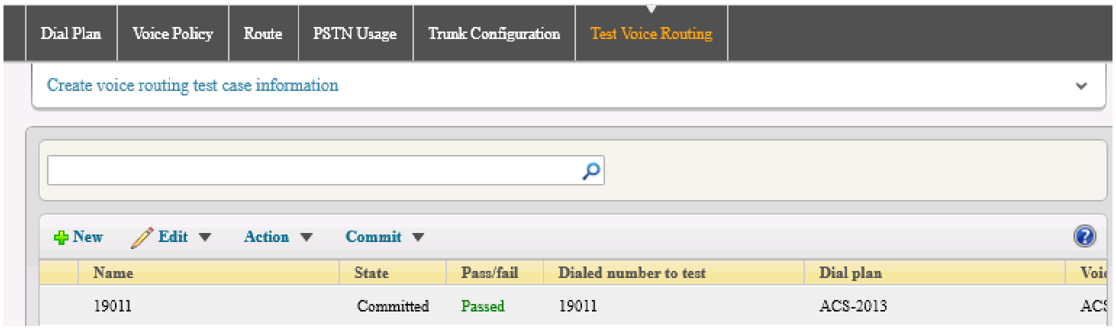


Enable media bypass: disabled

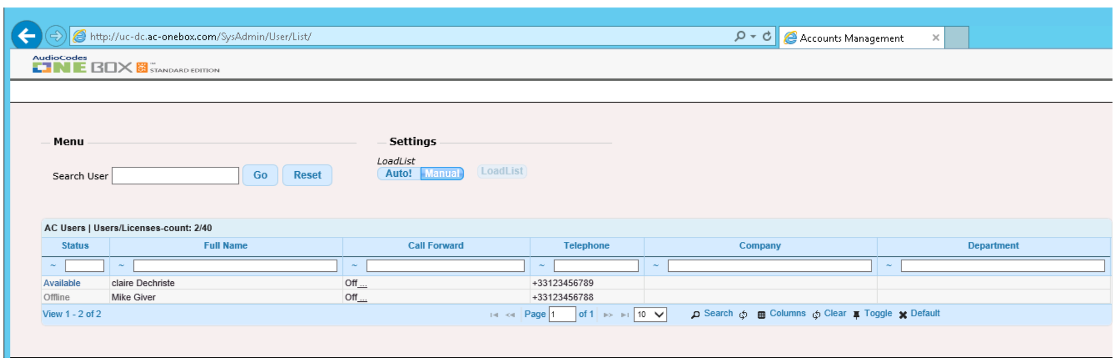
Associate a PSTN usage: add “ToOXE_PSTN”.

Commit all the previous modifications.

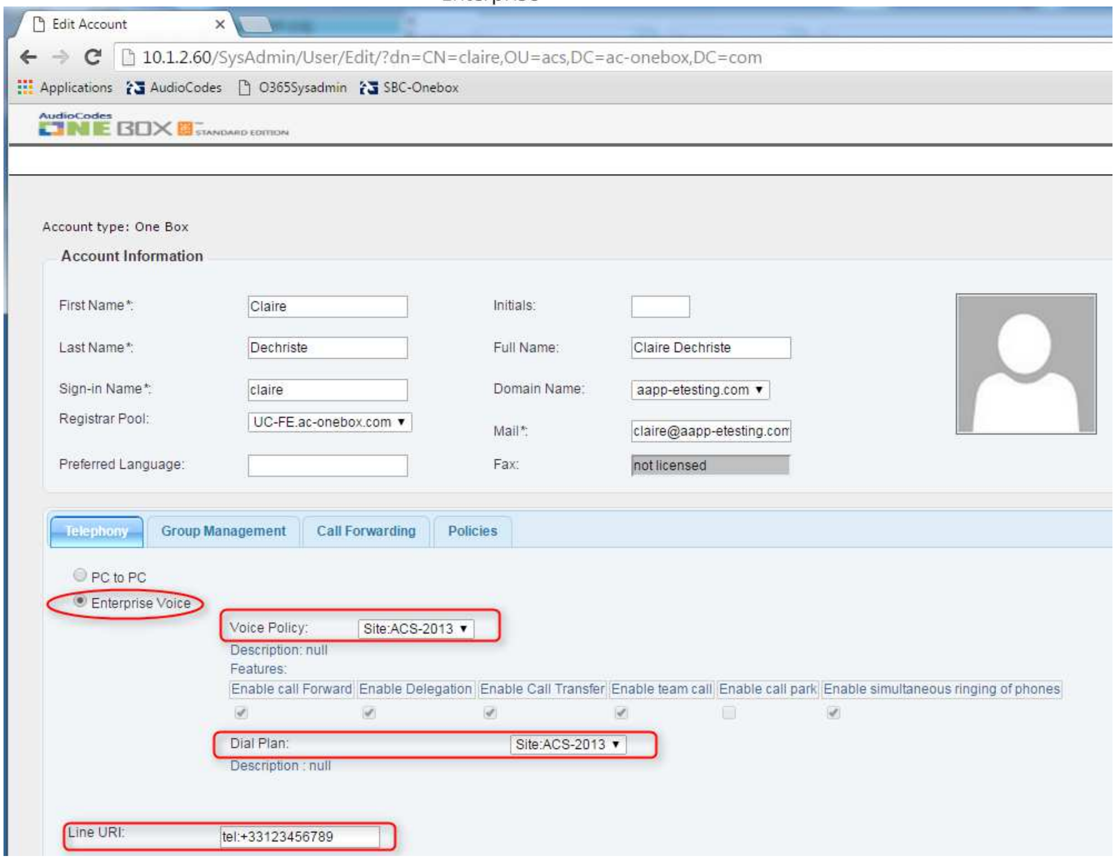
You can test this configuration by going into *Test Voice Routing* tab. Verify that the result is Passed



To create Lync users, open the Cloudband Management Suite and create your users:



Enable Enterprise Voice features:

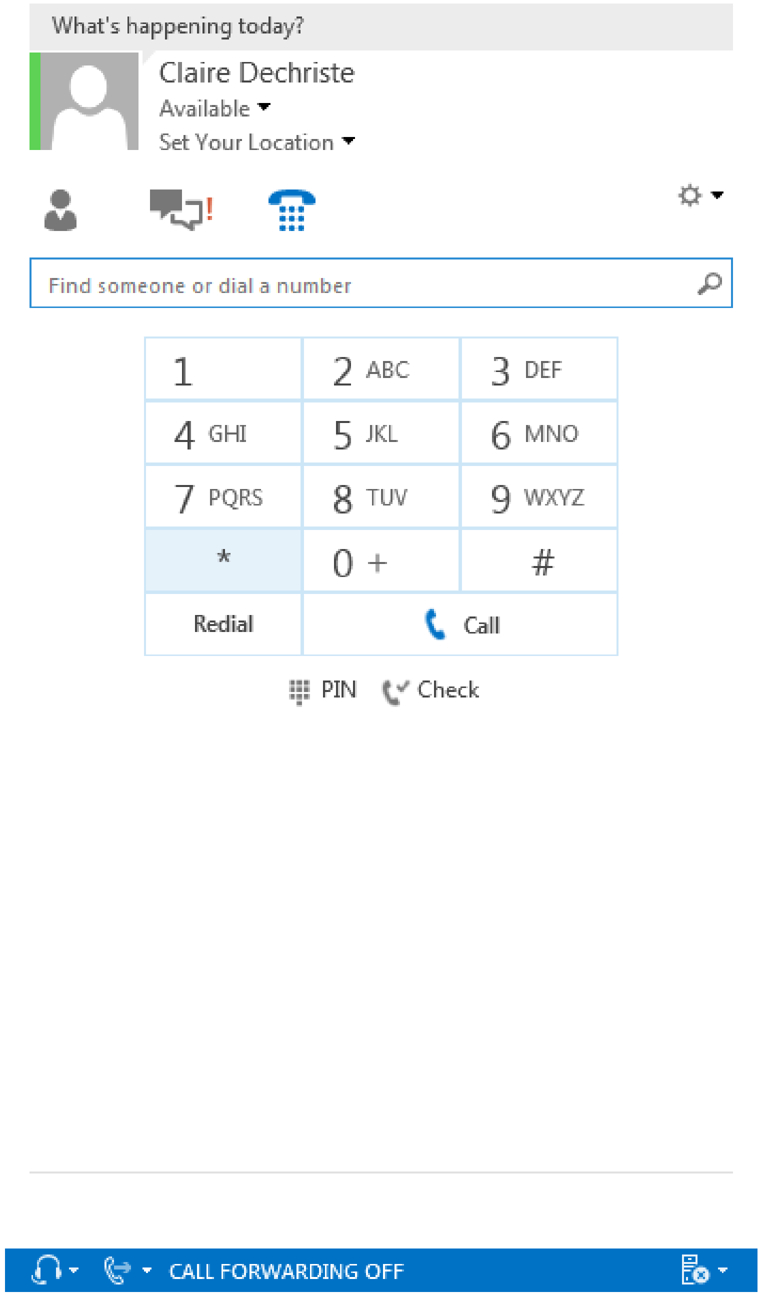


Enabled Lync Users for Enterprise Voice Features

Modify User “Telephony” from “PC-PC Only” to “Enterprise Voice” , and assign a phone number to the lync user with “Line URI”: tel:+33123456789 here.

Note: remove ext=xxxx extension if any. It causes trouble on the OXE for callback feature.

Once enabled, Lync client will display a new icon and a dial pad:


What's happening today?
Claire Dechriste
Available
Set Your Location
Find someone or dial a number
1
2 ABC
3 DEF
4 GHI
5 JKL
6 MNO
7 PQRS
8 TUV
9 WXYZ
*
0 +
#
Redial
Call
PIN
Check
CALL FORWARDING OFF

12 Appendix D: Alcatel-Lucent Enterprise Communication Platform: configuration requirements

The following mgr screenshots show the configuration of the OmniPCX Enterprise.

For more details on SIP features, refer to standard OXE technical documentation chapter "SIP configuration procedure".

12.1 SIP Trunk and ARS route management

It is necessary to create:

- An ISDN SIP trunk
- An external SIP gateway for AudioCodes Mediant E-SBC
- An ARS route, whose first route takes the SIP trunk group

12.1.1 Create a SIP Trunk

```
lqReview/Modify: Trunk Groupsqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqq
x
x        Node Number (reserved) : 109
x            Trunk Group ID : 20
x
x           Trunk Group Type + T2
x           Trunk Group Name : CloudBond
x        UTF-8 Trunk Group Name : ------------------------------------------
x        Number Compatible With : -1
x            Remote Network : 15
x          Shared Trunk Group + False
x           Special Services + Nothing
x              Node number : 9
x         Transcom Trunk Group + False
x       Auto.reserv.by Attendant + False
x       Overflow trunk group No. : -1
x            Tone on seizure + False
x          Private Trunk Group + False
x          Q931 Signal variant + ISDN all countries
x           SS7 Signal variant + No variant
x       Number Of Digits To Send : 0
x        Channel selection type + Quantified
x Auto.DTMF dialing on outgoing call + NO
x           T2 Specification + SIP
x  Homogenous network for direct RTP + NO
x          Public Network COS : 31
x            DID transcoding + False
x       Can support UUS in SETUP + True
x      Associated Ext SIP gateway : -1
x
x                Implicit Priority
x
x            Activation mode : 0
x             Priority Level : 0
x
```

x Preempter + NO
x Incoming calls Restriction COS : 10
x Outgoing calls Restriction COS : 10
x Callee number mpt1343 + NO
x Overlap dialing + NO
x Call diversion in ISDN + NO
x
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqq

Go into Trunk Group menu and specify the entity number:

lqReview/Modify: Trunk Groupqqqqqqqqqqqqqqqqqqqqqqqq
x
x Node Number (reserved) : 109
x Trunk Group ID : 20
x Instance (reserved) : 1
x
x Trunk Group Type + T2
x T2 Specification + SIP
x Public Network Ref. : ------
x VG for non-existent No. + YES
x Entity Number : 1
x Supervised by Routing + NO
x VPN Cost Limit for Incom.Calls : 0
x Immediate Trk Listening if VPNCall + YES
x VPN TS % : 50
x CSTA-Monitored + NO
x Max.% of trunks out CCD : 0
x Ratio analog.to ISDN cost : ------
x TS Distribution on Accesses + YES
x Quality profile for voice over IP + Profile #1
x Use of volume in system + YES
x Announcement for dial tone + NO
x Announcement for Ring tone + NO
x Reroute Anonymous Calls to Entity + NO
x Called Number Storage + NO
x End-to-end dialing + NO
x DTMF end-to-end signal. + NO
x Trunk group used in DISA + NO
x DISA Secret Code : ----
x Trunk COS : 31
x Sending of Progress message + YES
x No. of digits unused (ISDN) : 0
x B Channel Choice + YES
x Channels: Attendant Control (Rsvd) : 0
x Redirection For ACD (Dissuasion) + NO
x DTO joining + NO
x Consultation Call On B Channel + NO
x Automated Attendant + NO
x Calling party Rights COS : 0
x TS Overflow + YES
x Number To Be Added : --------
x Charge Calling And ADN Creation + NO
x Logical Channel + 1__15 & 17__31
x Use Split Access + NO
x Heterogeneous Remote Network + NO
x COS Restrictions - Barring mode + Not Restricted / Not barred
x ARS Class of service : 31
x External Access Server + NO
x CSTA Tracking MCDU Trk : --------
x IE External Forward + None

```
x    Max ISDN-IP and SIP connections : 0
x
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqj
```

12.1.2 Create an external SIP gateway

SIP>SIP Ext Gateway

```
lqReview/Modify: SIP Ext Gateway qqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqq
x
x           Node Number (reserved) : 109
x             Instance (reserved) : 1
x          SIP External Gateway ID : 10
x
x                    Gateway Name : CloudBond
x                SIP Remote domain : 10.1.2.63
x                  PCS IP Address : -----------------------------------------------
x                 SIP Port Number : 5060
x                  Transport type + UDP
x                 Belonging Domain : --------------------------------------------------
x                  Registration ID : ------------------------------------------------
x        Registration ID P_Asserted + False
x              Registration timer : 0
x              SIP Outbound Proxy : --------------------------------------------------
x               Supervision timer : 0
x              Trunk group number : 20
x                    Pool Number : -1
x                  Outgoing realm : --------------------------------------------------
x               Outgoing username : --------------------------------------------------
x
x               Outgoing Password : --------------------
x                        Confirm : --------------------
x
x               Incoming username : --------------------------------------------------
x
x               Incoming Password : --------------------
x                        Confirm : --------------------
x
x  RFC 3325 supported by the distant + True
x                        DNS type + DNS A
x             SIP DNS1 IP Address : 10.1.2.15
x             SIP DNS2 IP Address : -----------------------------------------------
x                      SDP in 18x + False
x     Minimal authentication method + SIP None
x   INFO method for remote extension + False
x                          To EMS + False
x                            SRTP + RTP only
x       Ignore inactive/black hole + False
x          Contact with IP address + False
x     Dynamic Payload type for DTMF : 97
x          Outbound Calls 100 REL + Supported
x          Incoming Calls 100 REL + Not Requested
x                    Gateway type + Standard type
x  Re-Trans No. for REGISTER/OPTIONS : 2
x    P-Asserted-ID in Calling Number + False
x      Trusted P-Asserted-ID header + True
x    Diversion Info to provide via  + Diversion
x Proxy identification on IP address + False
```

```
x             Outbound calls only + False
x        SDP relay on Ext. Call Fwd + Default
x         SDP Transparency Override + False
x RFC 5009 supported / Outbound call + Not Supported
x          Nonce caching activation + NO
x                FAX Procedure Type + T38 only
x  DNS SRV/Call retry on busy server : 0
x       Unattended Transfer for RSI + NO
x         Redirection functionality + NO
x                Attended Transfer + NO
x                Send BYE on REFER + YES
x       Support UTF8 characters set + NO
x       CSTA User-to-User supported + NO
x              Trusted From header + False
x     Support Re-invite without SDP + False
x         Type of codec negotiation + Default
x
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqq
```

12.1.3 Management of outgoing calls via SIP Trunk Group

ARS routing is used to route outgoing calls to external SIP GW through Public SIP TG.

Create a new Dialing Command Table:
Select Translator > Automatic Route Selection > Numbering Command Table

```
lqCreate: Numbering Command Tableqqqqqqqqqqqqqqqqqqqqqqqqqqqq
x
x          Node Number (reserved) : 109
x             Instance (reserved) : 1
x             Instance (reserved) : 1
x                      Table ID : 20
x
x              Carrier Reference : 0
x                       Command : I
x       Associated Ext SIP gateway : 10
x
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqq
```

Create an ARS route list with a route using the SIP trunk group and the previously created dialing command table:

```
lqReview/Modify: ARS Route listqqqqqqqqqqqqqqqqqqqqqqqqqqk
x
x          Node Number (reserved) : 109
x             Instance (reserved) : 1
x             Instance (reserved) : 1
x                 ARS Route list : 20
x
x                          Name : oneboxARS
x                      PIN Code + False
x
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqq
```

Select Translator > Automatic Route Selection > ARS Route list > ARS Route

```
lqReview/Modify: ARS Routeqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqk
x
x          Node Number (reserved) : 109
```

```
x            Instance (reserved) : 1
x            Instance (reserved) : 1
x               ARS Route list : 20
x                    Route : 1
x
x                      Name : oneboxsip
x           Trunk Group Source + Route
x                Trunk Group : 20
x        No.Digits To Be Removed : 0
x              Digits To Add : ------------------------------
x      Numbering Command Tabl. ID : 20
x             VPN Cost Limit : 0
x              Protocol Type + Dependant on Trunk Group Type
x             NPD identifier : 255
x                Route Type + Public
x             ATM Address ID : -1
x                 Preempter + False
x
x                         Quality
x
x        [  Add   ]    [ Remove ]    [  Next  ]     [Previous]
x
x                    Quality + Speech
x
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqj
```

Create the time-based route list.
Select Translator > Automatic Route Selection > ARS Route list > Time-based Route List

```
lqCreate: Time-based Route Listqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqk
x
x         Node Number (reserved) : 1
x            Instance (reserved) : 1
x            Instance (reserved) : 1
x               ARS Route list : 20
x       Time-based Route List ID : 1
x
x                    Time-based Route
x
x        [  Add   ]    [ Remove ]    [  Next  ]     [Previous]
x
x                    Time-based Route
x
x                 Route Number : 1
x           Waiting Cost Limit : -1
x          Stopping Cost Limit : -1
x
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqj
```

Select Translator > External Numbering Plan > Numbering Discriminator

```
lqCreate: Numbering Discriminatorqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqk
x
x         Node Number (reserved) : 1
x            Instance (reserved) : 1
x            Instance (reserved) : 1
x              Discriminator No. : 20
x
x                      Name : Onebox
x
```

```
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqq
```

Select Translator > External Numbering Plan > Numbering Discriminator > Dsc Hierarchy > Discriminator Rule

```
lqCreate: Discriminator Rule qqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqx
x
x          Node Number (reserved) : 1
x            Instance (reserved) : 1
x            Instance (reserved) : 1
x             Discriminator No. : 20
x                 Call Number : 0
x
x                 Area Number : 1
x           ARS Route List Number : 20
x               Schedule Number : -1
x              Number of Digits : 10
x
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqq
```

Select Translator > External Numbering Plan > Ext. Callback Translation Tables
Create an external callback translation table:

```
lqCreate: Ext.Callback Translation Tablesqqqqqqqqqqqqqqqqqqqqqk
x
x          Node Number (reserved) : 1
x            Instance (reserved) : 1
x            Instance (reserved) : 1
x          External Callback Table : 1
x
x                Country Codes : 0
x                 Country Name : Default
x
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqj
```

Create a callback translation Rule for this table:

```
lqReview/Modify: Ext.Callback Translation Rulesqqqqqqk
x
x          Node Number (reserved) : 109
x            Instance (reserved) : 1
x            Instance (reserved) : 1
x          External Callback Table : 1
x                 Basic Number : DEF
x
x          No.Digits To Be Removed : 0
x                Digits To Add : 200
x
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqj
```

Create an Entity for Lync users:

```
lqReview/Modify: Entitiesqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqq
x
x          Node Number (reserved) : 109
x                Entity Number : 1
x
x                       Name : ENTITY_1
x                 UTF-8 Name : ------------------------------------------------------------
```

```
x          Attendant Group Manager : -1
x                     Priority + NO
x           Emergency call to attd + NO
x                Traffic Overflow + Disallowed
x          Installation No. (ISDN) : ------------------------------
x    Supplement.Install.No. (ISDN) : ------------------------------
x                Caller ID Secret + No
x      AdvOfCharg2 requests (AOC2) + NO
x      AdvOfCharg3 requests (A0C3) + NO
x                  Auto. Locking : 0
x    Voice Mail Box No.for attendt : --------
x                  Trunk Group ID : 0
x          External Callback Table : 1
x
x                        Call Distribution
x
x             Overflow Routing No. : --------
x            Forwarding on routing + YES
```

Associate the system discriminator with the entity discriminator:
Select Entity > Descend Hierarchy> Discriminator Selector

```
lqReview/Modify: Discriminator Selectorqqqqqqqqqqqqqqk
x
x           Node Number (reserved) : 109
x                  Entity Number : 1
x              Instance (reserved) : 1
x
x                Discriminator 00 : 0
x                Discriminator 01 : 1
x                Discriminator 02 : 20
x                Discriminator 03 : 12
x                Discriminator 04 : 0
x                Discriminator 05 : 0
x                Discriminator 06 : 0
x                Discriminator 07 : 0
x
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqj
```

Activate the transfer feature in Class of services > Phone feature COS

```
lqReview/Modify: Phone Features COSqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqk
x
x           Node Number (reserved) : 109
x              Instance (reserved) : 1
x               Phone Features COS : 0
x
x                   Phone COS Name : --------------------
x
x                          Rights
x
x     Prot.against dir.call pickup : 0
x    Protected against all barge-in : 1
x    Protected against set barge-in : 1
x              Outgoing calls only : 0
x          Forward to external No. : 1
x     Prot.against multi-l ringing : 1
x     Protected against forwarding : 0
x Protected (against barge-in, etc.) : 1
x       Prot.against call announc. : 1
x       Remote wake-up/appointment : 1
```

x Auto call back on busy trk-grp : 0
x Transfer on no answer : 1
x ISDN remote charge service : 0
x Bypass on forwarding : 0
x Prot.against bypass onforward : 1
x Interphony : 1
x Secret Code, Repertory Key : 1
x Night Serv.Answ.Pick up : 0
x Night Serv.Direct call pick-up : 0
x Attendant Call Privil.on PAI : 0
x Busy priv.to public overfl. : 0
x Server-Minitel PC : 0
x Prot.against Priv.Call : 0
x Prot.against.Rem.Forward. : 0
x Beep On Ext.Call : 0
x O/S private to public overflow : 0
x Transfer outgoing - incoming : 1
x Transfer Outgoing-Outgoing : 1

To get the display name of Lync user on OXE user when the OXE user calls the Lync user or when Lync user calls the OXE user, you must configure Calling Name Presentation to True.

System -> Other System Parameter ->Descend Hierarchy -> External signaling Parameter -> review Modify -> calling name presentation -> put it in True.

lqReview/Modify: External Signaling Parametersqqqqqqqqqqqqqqqqqqqqqqqqqqqk
x
x Node Number (reserved) : 109
x Instance (reserved) : 1
x Instance (reserved) : 1
x System Option + Calling Name Presentation
x
x Calling Name Presentation + True
x
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqj

To get the display name of Lync user on OXE user during a call, check the OXE user phone feature Cos number (Tel facility Category Id), then Go to:

Classes of Service -> Phone Feature Cos -> Calling name display (CNIP/I CNAM) put it in 1

x Calling name display (CNIP/I-CNAM) : 1

Set G729 in System > other system param> compression parameters>compression Type

lqReview/Modify: Compression Parametersqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqk
x
x Node Number (reserved) : 109
x Instance (reserved) : 1
x Instance (reserved) : 1
x System Option + Compression Type
x
x Compression Type + G 729
x
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqj

System > other system parameters> compression parameters>Multi algorithms for compression

```
lqReview/Modify: Compression Parametersqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqk
x
x          Node Number (reserved) : 109
x            Instance (reserved) : 1
x            Instance (reserved) : 1
x                System Option + Multi. Algorithms for Compression
x
x  Multi. Algorithms for Compression + False
x
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqj
```

List of prefixes and suffixes defined on etesting9 OXE system:

dir	mean	info	digit
0	Professional_trunk_seize	2	-1
484	Cancel_Remote_forward	-1	-1
485	Overfl_busy_to_assoc_set	-1	-1
486	Overf_busy/no_repl_assoc_set	-1	-1
487	Recording_Conversation	-1	-1
490	Ubiquity_Mobile_Programming	-1	-1
491:493	Ubiquity_Services_Pfx	-1	-1
495	Ubiquity_Assistant	-1	-1
496		-1	-1
497		-1	-1
500	Last_Caller_Call_back	-1	-1
501	Remote_forward	-1	-1
502	Overflow_on_associated_set	-1	-1
503	Cancel_Overfl_on_assoc_set	-1	-1
504	Protection_against_beeps	-1	-1
505	Substitution	-1	-1
506	Wake_up/appointment_remind	-1	-1

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dir	mean	info	digit
507	Cancel_Wake_up	-1	-1
508	Forward_cancel_by_destinat	-1	-1
509	Meet_me_Conference	-1	-1
51	Immediate_forward	-1	-1
52	Immediate_forward_on_busy	-1	-1
53	Forward_on_no_reply	-1	-1
54	Forward_on_busy_or_no_reply	-1	-1
55	Direct_call_pick_up	-1	-1
56	Group_call_pick_up	-1	-1
570	Voice_Mail_Deposit	-1	-1
580	Tone_test	-1	-1
581	Personal_directory_Progr	-1	-1
582	Personal_Directory_Use	-1	-1
583	Force_type_identification_pfx	-1	-1
584	Suite_Wakeup	-1	-1
585	Suite_Wakeup_Cancel	-1	-1
586	Suite_Dont_Disturb	-1	-1

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dir	mean	info	digit
587	Room_status_management	-1	-1
588	Mini_bar	-1	-1
589	Direct_Paging_Call	-1	-1
599	Professional_trunk_seize	0	-1
666	Pabx_address_in_DPNSS	-1	-1
67	ARS_Prof_Trg_Grp_Without_Subad	-1	-1
9	Attendant_Group_Call	-1	-1
*	DTMF_End_to_End_Dialling	-1	-1
#	Speed_call_to_associated_set	-1	-1

13 Appendix E: AAPP member's escalation process

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

14 Appendix F: AAPP program

14.1 Alcatel-Lucent Application Partner Program (AAPP)

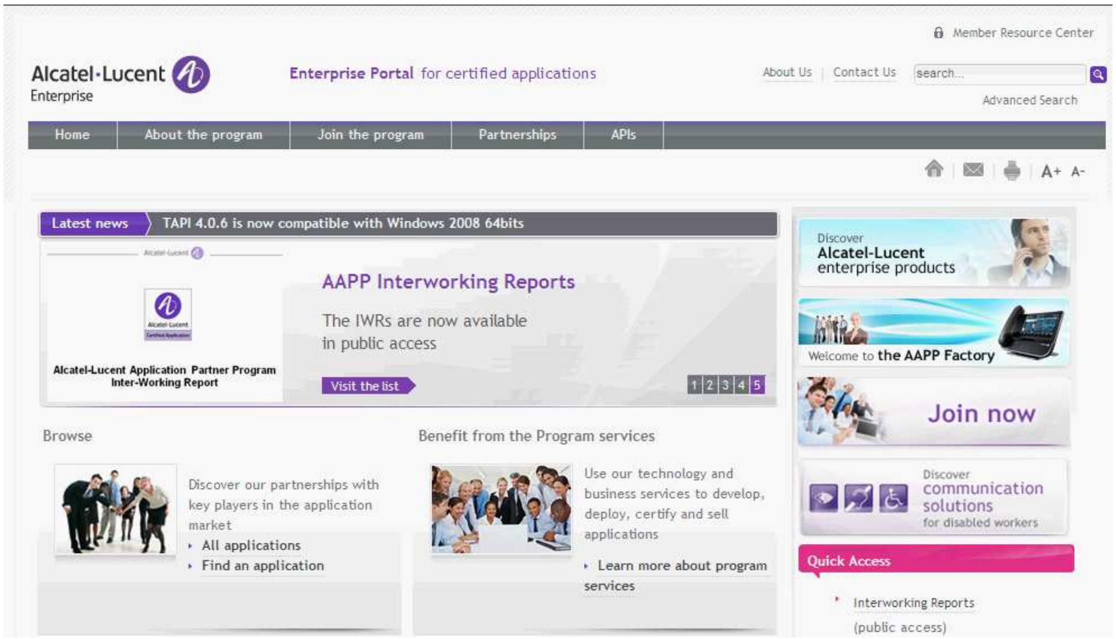
The Application Partner Program is designed to support companies that develop communication applications for the enterprise market, based on Alcatel-Lucent Enterprise's product family. The program provides tools and support for developing, verifying and promoting compliant third-party applications that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

The Alcatel-Lucent Application Partner Program (AAPP) has two main objectives:
Provide easy interfacing for Alcatel-Lucent Enterprise communication products:
Alcatel-Lucent Enterprise's communication products for the enterprise market include infrastructure elements, platforms and software suites. To ensure easy integration, the AAPP provides a full array of standards-based application programming interfaces and fully-documented proprietary interfaces. Together, these enable third-party applications to benefit fully from the potential of Alcatel-Lucent Enterprise products.
Test and verify a comprehensive range of third-party applications:
to ensure proper inter-working, ALE International tests and verifies selected third-party applications that complement its portfolio. Successful candidates, which are labelled Alcatel-Lucent Enterprise Compliant Application, come from every area of voice and data communications.

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

Web site

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



14.2 Enterprise.Alcatel-Lucent.com

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

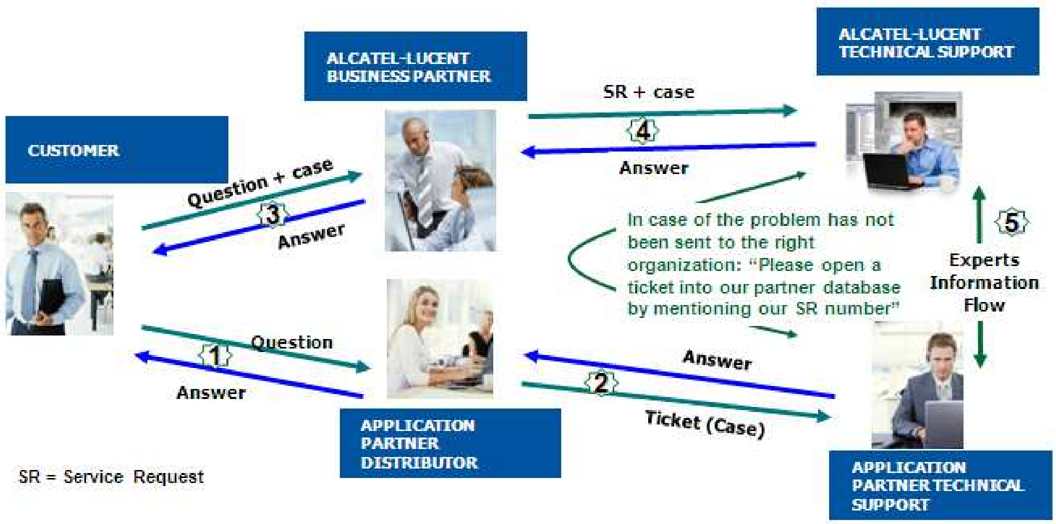
15 Appendix G: AAPP Escalation process

15.1 Introduction

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

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

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



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

15.2 Escalation in case of a valid Inter-Working Report

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

This defines the scope of what has been certified.

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

Case 1: the responsibility can be established 100% on ALE International side.

In that case, the problem must be escalated by the ALE Business Partner to the ALE International Support Center using the standard process: open a ticket (eService Request –eSR)

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

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

The Application Partner shall be contacted first by the Business Partner (responsible for the application, see figure in previous page) for an analysis of the problem.

The ALE International Business Partner will escalate the problem to the ALE International Support Center only if the Application Partner has demonstrated with traces a problem on the ALE International side or if the Application Partner (not the Business Partner) needs the involvement of ALE International

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

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

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

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

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

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

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

15.3 Escalation in all other cases

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

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

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

15.4 Technical support access

The ALE International **Support Center** is open 24 hours a day; 7 days a week:
e-Support from the Application Partner Web site (if registered Alcatel-Lucent Application Partner): http://applicationpartner.alcatel-lucent.com
e-Support from the ALE International Business Partners Web site (if registered Alcatel-Lucent Enterprise Business Partners): https://businessportal2.alcatel-lucent.com click under "Contact us" the *eService Request* link
e-mail: Ebg_Global_Supportcenter@al-enterprise.com
Fax number: +33(0)3 69 20 85 85
Telephone numbers:

ALE International Business Partners Support Center for countries:

Country	Supported language	Toll free number
France		
Belgium	French	
Luxembourg		
Germany		
Austria	German	
Switzerland		
United Kingdom		
Italy		
Australia		
Denmark		
Ireland		
Netherlands		+800-00200100
South Africa		
Norway	English	
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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