



# Microsoft Lync 2013 [v5.0.8308.0] to Verizon Business SIP Trunk via the Cisco Unified Border Element 10.5 [IOS 15.4(3)M]

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## Table of Contents

Introduction .....	2
Network Topology.....	4
System Components .....	5
Hardware Components .....	5
Software Requirements .....	5
Features .....	6
Features Supported.....	6
Features Not Supported .....	6
Caveats .....	7
Configuration.....	8
Configuring Cisco Unified Border Element (CUBE) .....	8
Configuring the Lync 2013 Server .....	14
Lync Topology builder: Adding PSTN Gateway .....	14
Lync Topology builder: Associating Gateway with Mediation pool .....	15
Lync Server: Control Panel: Adding users .....	16
Lync Control panel: Configuring Dial Plan.....	17
Lync Control panel: Configuring Voice Policy .....	18
Lync Control panel: Configuring Route .....	19
Lync Control panel: Trunk Configuration .....	21
Acronyms .....	22
<b>Important Information</b> .....	23



## Introduction

Service Providers today, such as Verizon Business, are offering alternative methods to connect to the PSTN via their IP network. Most of these services utilize SIP as the primary signaling method and a centralized IP to TDM gateway to provide on-net and off-net services. Verizon Business SIP trunk is a SP offering that allows connection to the PSTN and may offer the end customer a viable alternative to traditional PSTN connectivity via either Analog or T1 lines. A demarcation device between these services and customer owned services is recommended. The Cisco Unified Border Element provides demarcation, security, interworking and session management services.

- This application note describes how to configure a Lync 2014 with a Cisco Unified Border Element (CUBE) for connectivity to Verizon Business SIP trunk service. The deployment model covered in this application note is Lync 2013/CUBE to Verizon Business SIP trunk. This document does not address 911 emergency outbound calls. For 911 feature service details contact Verizon Business, directly.
- Testing was performed in accordance to Cisco's SIP Trunk Test Plan and all features were verified. Key features verified are:
  - o CPE outbound to SP Offnet gateway(PSTN)
  - o SP Offnet gateway(PSTN) inbound to CPE (G.729 offered first)
  - o CPE Telephone Number Support – digit translations
  - o CPE Offnet Call Conference
  - o CPE Intra-Site Call Conference
  - o CPE Intra-Site Attended Call Transfer
  - o CPE Intra-Site Unattended Call Transfer
  - o CPE Call Hold and Resume (call hold is always done on the IP PBX side)
  - o CPE Voice Mail
  - o SP Voice Mail
  - o CPE Find Me (CFU)
  - o Simultaneous Calls
  - o CPE Auto Attendant
  - o CPE to PSTN offnet gateway international call
  - o CPE Find Me (Call Forward Don't Answer)
  - o Codec mid-call re-negotiation (to be tested without transcoder)
  - o Dial plans
  - o PRACK with SDP



- The Cisco Unified Border Element configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between Verizon Business SIP network and Cisco Unified Communications. The configuration described in this document details the important commands to have enabled for interoperability to be successful and care must be taken, by the network administrator deploying CUBE, to ensure these commands are set per each dial-peer requiring to interoperate to Verizon Business SIP network.

### Network Topology

The network topology includes the Microsoft Lync 2013 Enterprise Edition and 2 Lync clients. Cisco UBE published as a PSTN gateway in the Lync topology. Verizon was used as the service provider with a SIP trunk to the Cisco UBE.

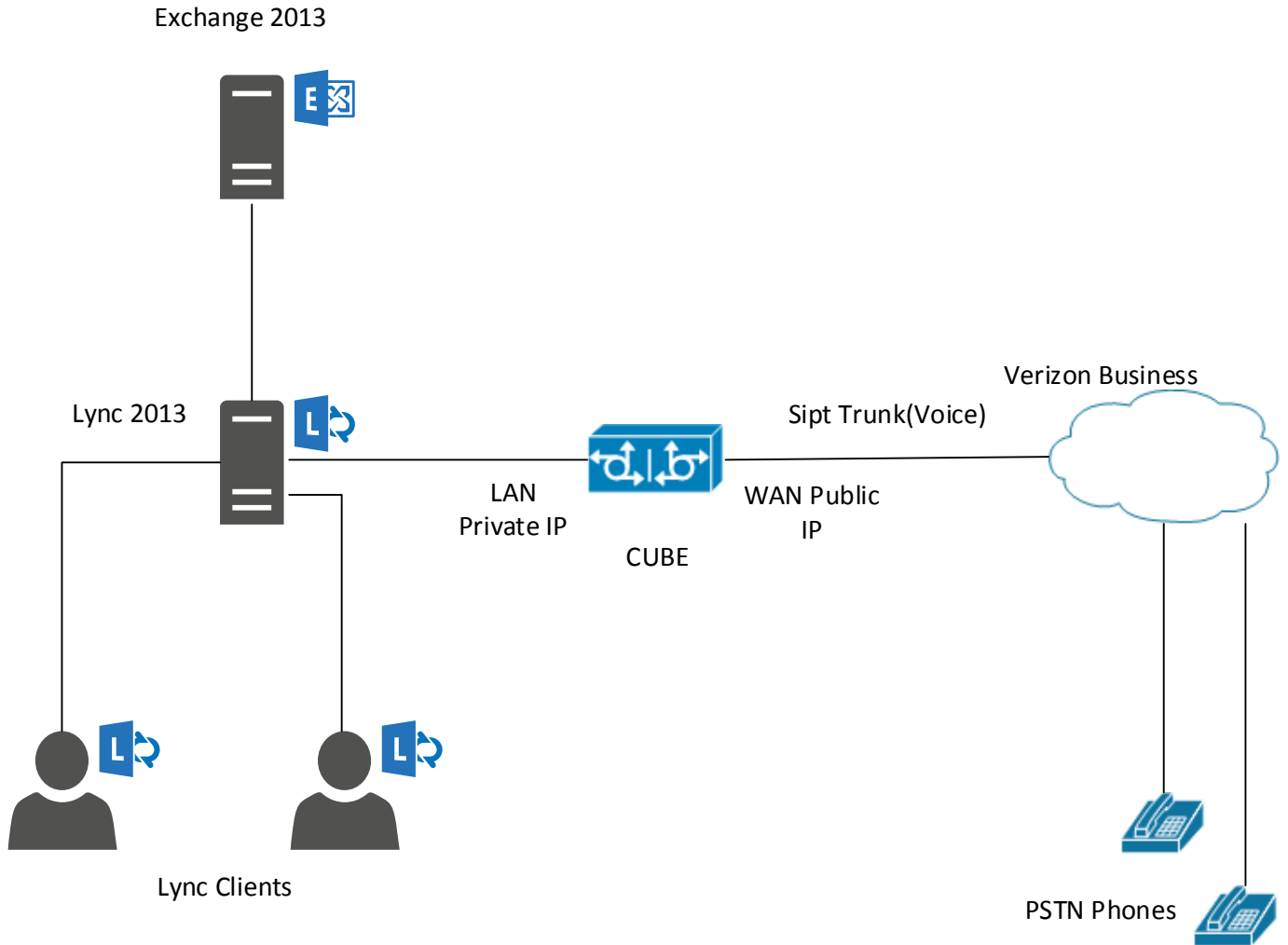


Figure 1. Basic Call Setup



## **System Components**

### **Hardware Components**

- Cisco UBE 3945

### **Software Requirements**

- Cisco UBE IOS 15.4(3)M
- Microsoft Lync 2013 V5.0.8308.0
- Microsoft Exchange 2013 CU7



## Features

### Features Supported

- Lync only supports attended and semi-attended. Call can be completed. However, caller ID on the terminating side does not get updated after transfer is completed
- Call from/to PSTN to/from CPE – Basic and International calls , digit translations
- Hold/Resume
- DTMF
- Call Forwarding CFA and CFNA
- Support for early media

### Features Not Supported

- Lync does not support codec G.729
- Lync need a third party vendor to support Fax.
- Lync does not support blind transfers.
- Lync does not have a call forward busy feature.



### **Caveats**

- Caller ID updates are not observed on call transfer scenarios.
- Lync does not support Call Forwarding on Busy .
- Lync doesn't support G729 calls.Hence all calls were tested with G711ulaw.Codec negotiation tests were done with G711u and Alaw.
- Lync cannot send a call with anonymous caller id
- There is a work around needed for CPE outbound calls. If the workaround is not implemented calls are disconnected after 30 minutes.
- The Diversion header needs to be manipulated for call forward to work.



## Configuration

### Configuring Cisco Unified Border Element (CUBE)

Cisco IOS Software, C3900e Software (C3900e-UNIVERSALK9-M), Version 15.4(3)M, RELEASE SOFTWARE (fc1)  
Technical Support: <http://www.cisco.com/techsupport>  
Copyright (c) 1986-2014 by Cisco Systems, Inc.  
Compiled Mon 21-Jul-14 12:23 by prod\_rel\_team

ROM: System Bootstrap, Version 15.1(1r)T5, RELEASE SOFTWARE (fc1)

centurylink1 uptime is 4 days, 20 hours, 14 minutes  
System returned to ROM by power-on  
System image file is "flash0:c3900e-universalk9-mz.SPA.154-3.M.bin"  
Last reload type: Normal Reload  
Last reload reason: power-on

This product contains cryptographic features and is subject to United States and local country laws governing import, export, transfer and use. Delivery of Cisco cryptographic products does not imply third-party authority to import, export, distribute or use encryption. Importers, exporters, distributors and users are responsible for compliance with U.S. and local country laws. By using this product you agree to comply with applicable laws and regulations. If you are unable to comply with U.S. and local laws, return this product immediately.

A summary of U.S. laws governing Cisco cryptographic products may be found at:  
<http://www.cisco.com/www/export/crypto/tool/stqrg.html>

If you require further assistance please contact us by sending email to [export@cisco.com](mailto:export@cisco.com).

Cisco CISCO3925-CHASSIS (revision 1.0) with C3900-SPE200/K9 with 1797120K/300032K bytes of memory.  
Processor board ID FTX1744AMAD  
4 Gigabit Ethernet interfaces  
4 Channelized (E1 or T1)/PRI ports  
1 Virtual Private Network (VPN) Module  
DRAM configuration is 72 bits wide with parity enabled.  
256K bytes of non-volatile configuration memory.  
4001760K bytes of ATA System CompactFlash 0 (Read/Write)

License Info:

License UDI:

```

-----
Device#  PID          SN
-----
*1      C3900-SPE200/K9    FOC174142TF

```

Technology Package License Information for Module:'c3900e'







```
!
multilink bundle-name authenticated
!
!
!
!
!
cts logging verbose
!
!
voice-card 0
!
!
!
voice service voip
ip address trusted list
! The IP address below are the Service Provider and Lync 2013
  ipv4 XX.XX.XX.XX
  ipv4 YY.YY.YY.YY
no ip address trusted authenticate
address-hiding
allow-connections sip to sip
no supplementary-service sip moved-temporarily
no supplementary-service sip refer
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
sip
rel1xx supported "rel100"
header-passing
referto-passing
early-offer forced
midcall-signaling passthru
sip-profiles inbound
!
!
voice class uri LYNC sip
! ** Lync IP address **
host ipv4:XX.XX.XX.XX
!
voice class uri PSTN sip
! ** SIP trunk provider IP Address **
host ipv4:YY.YY.YY.YY

voice class codec 1
codec preference 1 g711ulaw
!
!
! ** This allows call forwards to work ***
voice class sip-profiles 1
request INVITE peer-header sip Referred-By copy "sip:(.*)@" u01
request INVITE sip-header Diversion add "Diversion: <sip:sip-uri@97.79.185.189>"
request INVITE sip-header Diversion modify "sip-uri" "\u01"
request INVITE sip-header Referred-By remove
!
! ** This allows calls to last longer than 30 minutes **
voice class sip-profiles 2
request INVITE sip-header Session-Expires modify "(.*)" "\1;refresher=uas"
!
!
```



```
voice class sip-copylist 1
sip-header Referred-By
!
!
!
!
hw-module pvdn 0/0
!
hw-module pvdn 0/1
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
interface GigabitEthernet0/0
! ** WAN IP address **
ip address XX.XX.XX.XX 255.255.255.128
ip virtual-reassembly in
ip virtual-reassembly out
duplex auto
speed auto
!
interface GigabitEthernet0/1
! ** LAN IP address **
ip address YY.YY.YY.YY 255.255.255.0
ip virtual-reassembly in
ip virtual-reassembly out
duplex auto
speed auto
!
interface GigabitEthernet0/2
no ip address
shutdown
duplex auto
speed auto
!
interface GigabitEthernet0/3
no ip address
shutdown
duplex auto
speed auto
!
ip forward-protocol nd
!
no ip http server
no ip http secure-server
!
! ** Default route **
ip route 0.0.0.0 0.0.0.0 XX.XX.XX.XX
!
!
```



```
nls resp-timeout 1
cpd cr-id 1
!
!
control-plane
!
!
!
!
!
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
!
!
!
dial-peer voice 400 voip
description ** LAN side to Lync **
destination-pattern 9725511[01]
session protocol sipv2
session target ipv4:XX.XX.XX.XX
session transport tcp
voice-class codec 1
dtmf-relay rtp-nte
!
dial-peer voice 500 voip
description ** WAN side to PSTN **
destination-pattern .T
session protocol sipv2
session target ipv4:YY.YY.YY.YY
session transport udp
voice-class codec 1
voice-class sip profiles 1
no voice-class sip copy-list
no voice-class sip refer-to-passing
dtmf-relay rtp-nte
!
dial-peer voice 401 voip
description ** LAN side from Lync **
session protocol sipv2
session transport tcp
incoming uri via LYNC
voice-class codec 1
voice-class sip profiles 2 inbound
voice-class sip copy-list 1
dtmf-relay rtp-nte
!
dial-peer voice 501 voip
description ** WAN side from PSTN **
session protocol sipv2
session transport tcp
incoming uri via PSTN
voice-class codec 1
dtmf-relay rtp-nte
```



```
!  
!  
!  
!  
gatekeeper  
  shutdown  
!  
!  
!  
line con 0  
line aux 0  
line vty 0 4  
  login local  
  transport input telnet  
!  
scheduler allocate 20000 1000  
!  
end
```



## Configuring the Lync 2013 Server

The validation of Microsoft Lync with Cisco UBE includes the following integration steps from Lync perspective:

- Adding the Cisco UBE as a PSTN Gateway on the Lync Topology Builder
- Associating Gateway with a Mediation pool
- Adding Lync users with DIDs provided by the Service provider.
- Configuring a Dial Plan
- Configuring a Voice Policy
- Configuring a Route
- Trunk configuration to enable all relevant features required for the test.

## Lync Topology builder: Adding PSTN Gateway

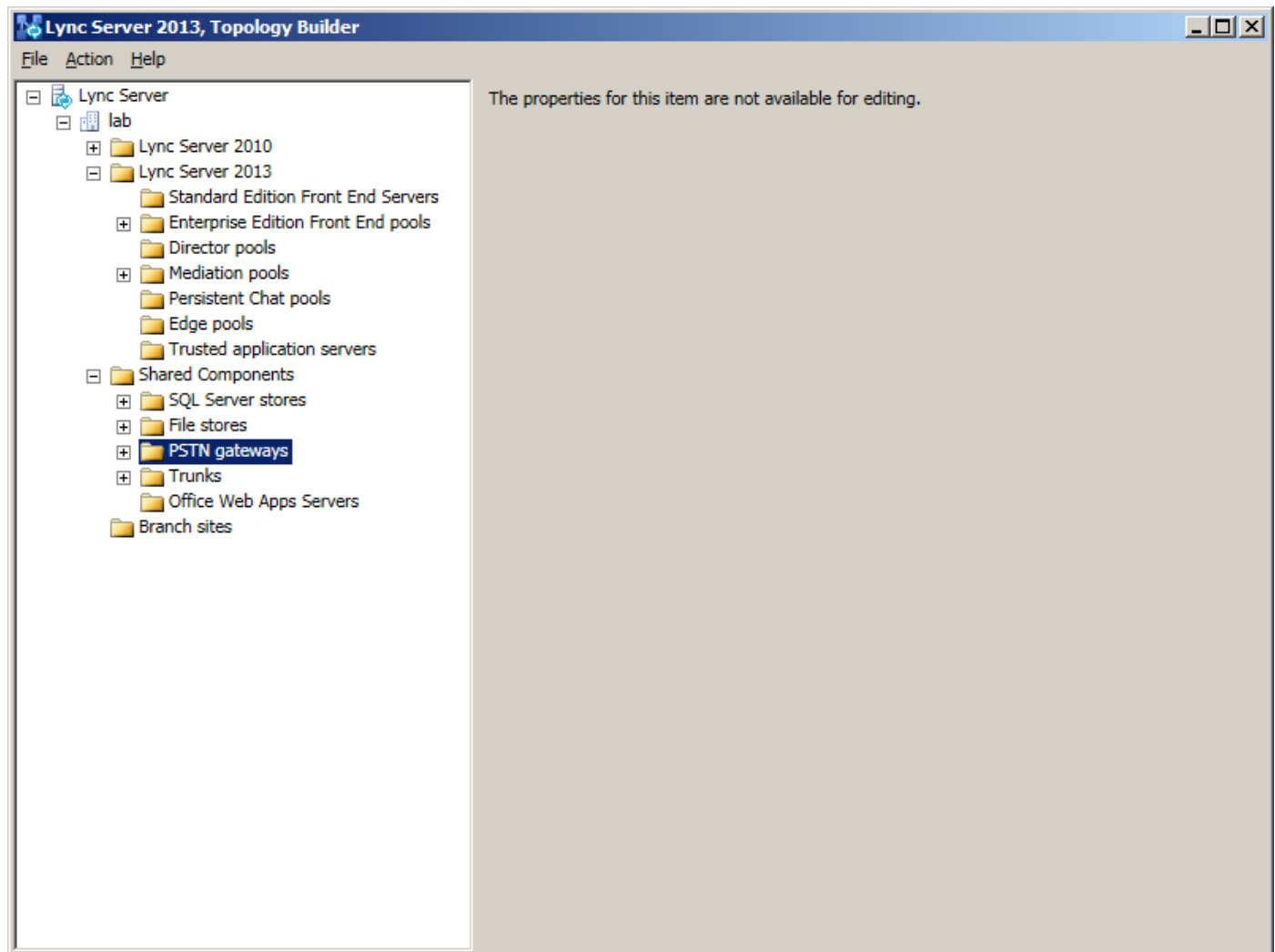


Figure 2. Lync Topology Builder- PSTN Gateway

### Lync Topology builder: Associating Gateway with Mediation pool

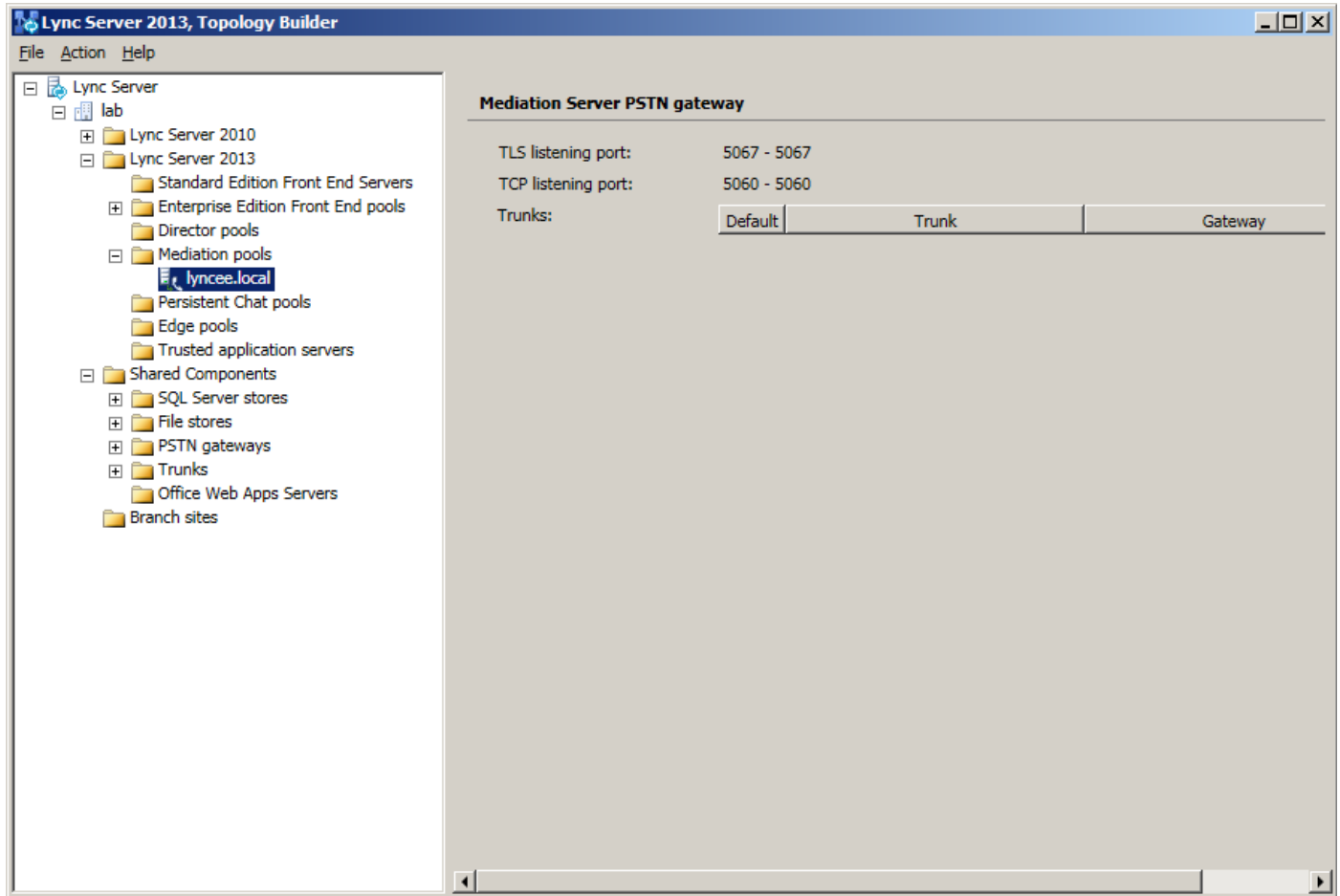


Figure 3. Lync Topology Builder- Mediation pools



## Lync Server: Control Panel: Adding users

The screenshot shows the Microsoft Lync Server 2013 Control Panel. The main area is titled "User Search" and contains a search interface. The search results show two users:

Display name	Enabled	SIP address	Registrar pool	Telephony
test	✓	sip:test@lyncee.local	lyncee.local	Enterprise Voice
test1	✓	sip:test1@lyncee.local	lyncee.local	Enterprise Voice

Figure 4. Lync Server: Control Panel- Users



Lync Control panel: Configuring Dial Plan

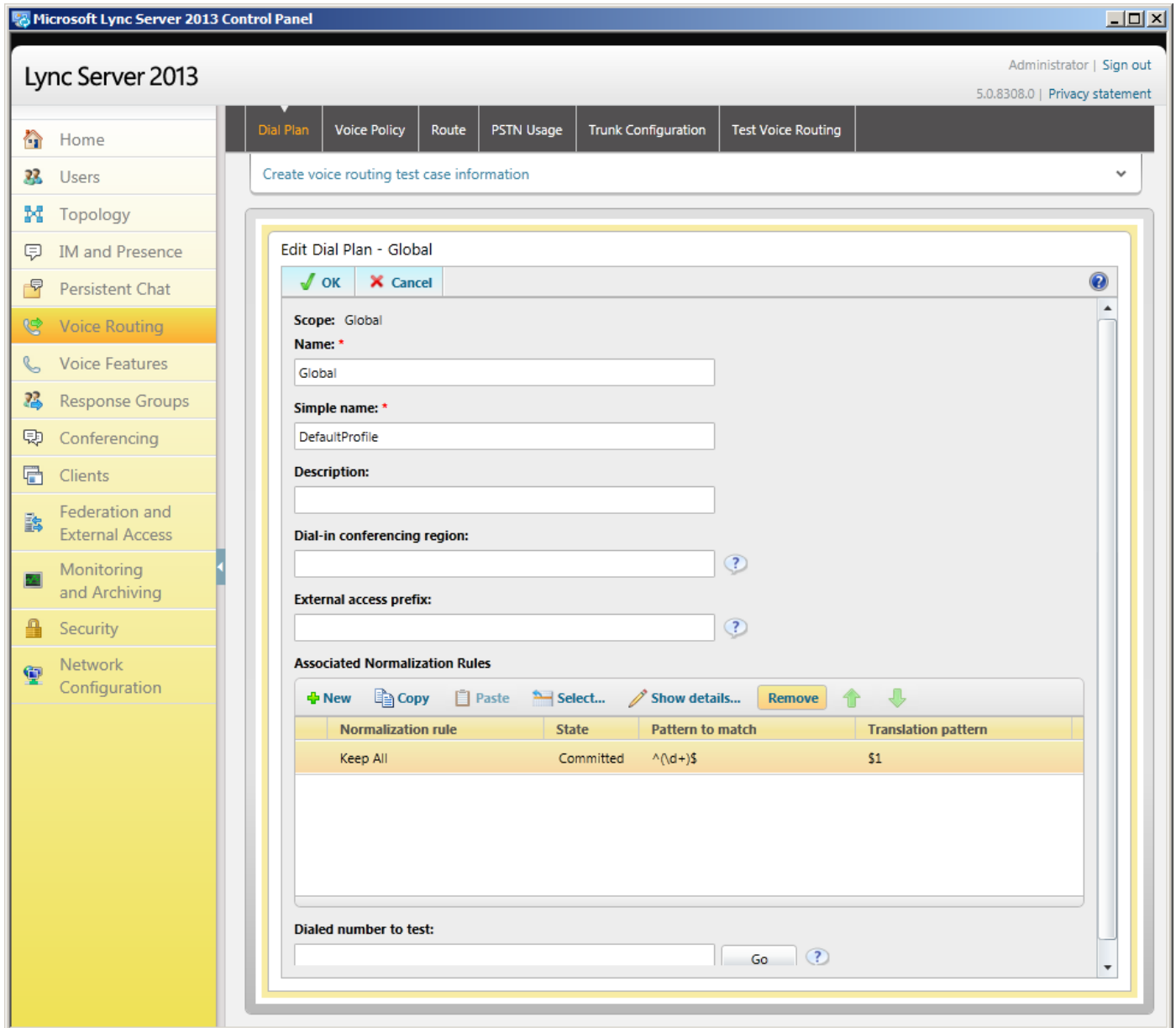
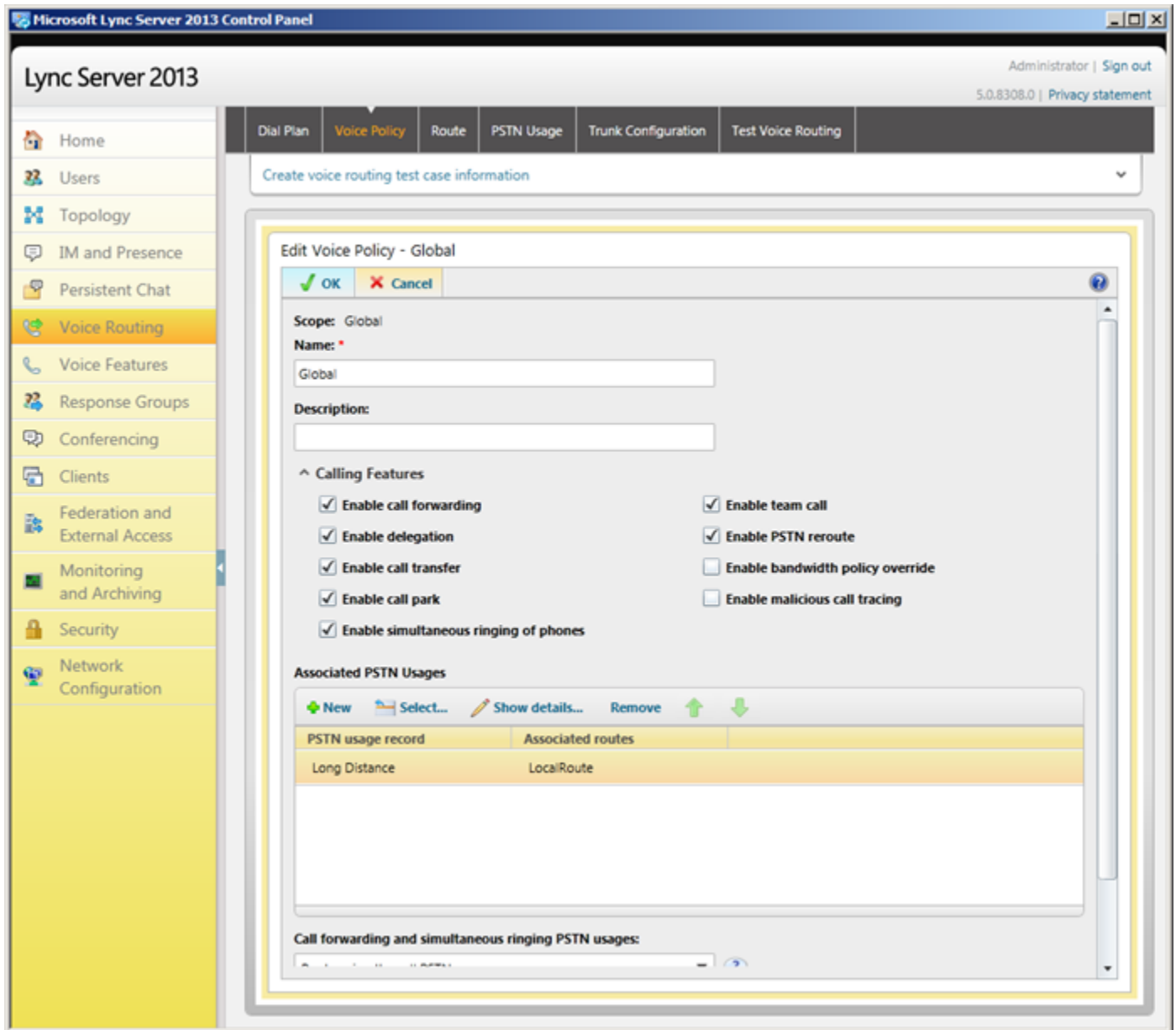


Figure 5. Lync Control panel - Dial Plan

## Lync Control panel: Configuring Voice Policy



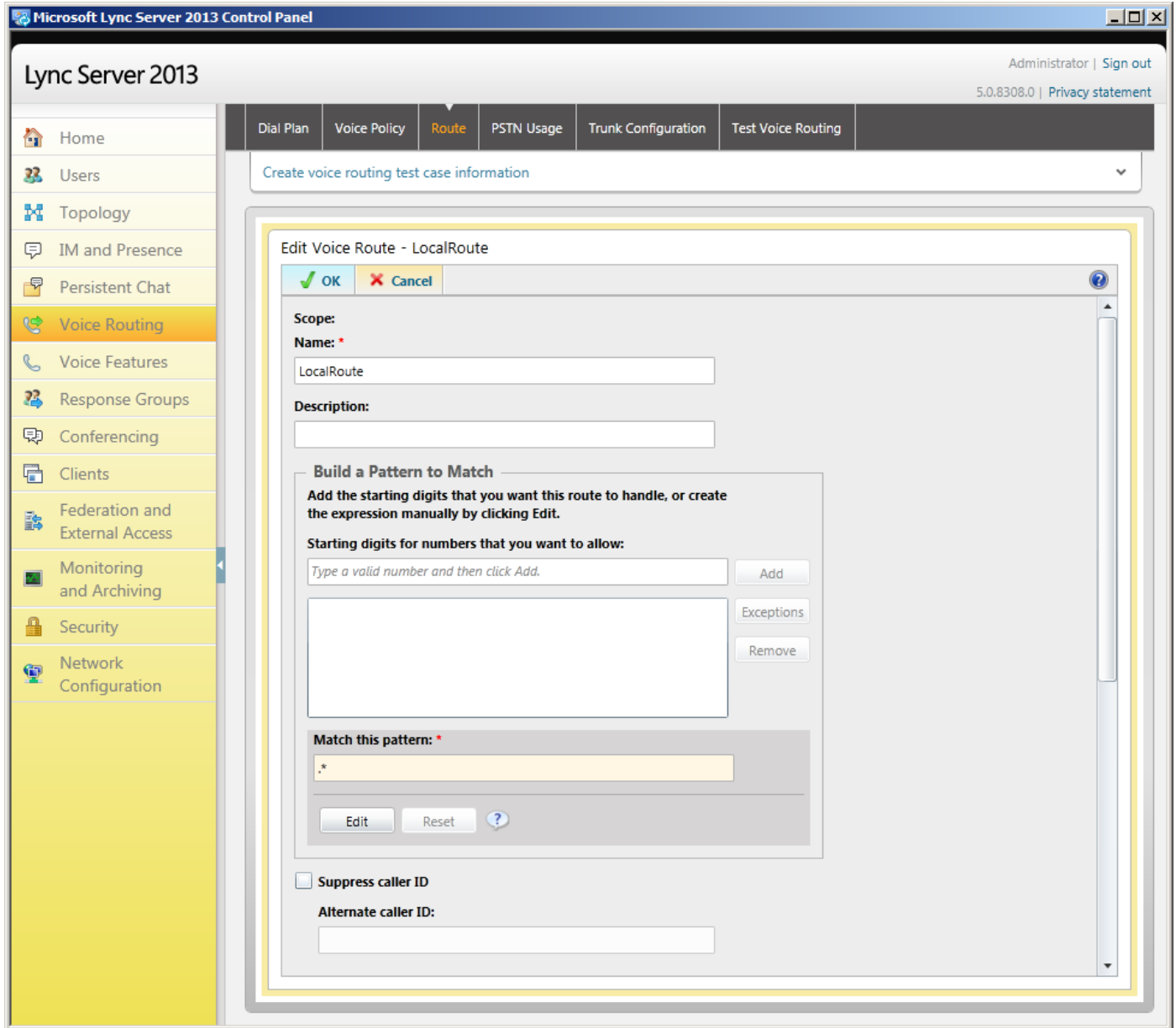
The screenshot displays the Microsoft Lync Server 2013 Control Panel interface. The left-hand navigation pane includes options such as Home, Users, Topology, IM and Presence, Persistent Chat, Voice Routing (highlighted), Voice Features, Response Groups, Conferencing, Clients, Federation and External Access, Monitoring and Archiving, Security, and Network Configuration. The main content area is titled 'Lync Server 2013' and shows a top navigation bar with tabs for Dial Plan, Voice Policy (selected), Route, PSTN Usage, Trunk Configuration, and Test Voice Routing. Below the tabs, there is a search bar and a dropdown menu for 'Create voice routing test case information'. The central focus is the 'Edit Voice Policy - Global' dialog box, which contains the following configuration options:

- Scope:** Global
- Name:** Global
- Description:** (empty text field)
- Calling Features:**
  - Enable call forwarding
  - Enable delegation
  - Enable call transfer
  - Enable call park
  - Enable simultaneous ringing of phones
  - Enable team call
  - Enable PSTN reroute
  - Enable bandwidth policy override
  - Enable malicious call tracing
- Associated PSTN Usages:**
  - Buttons: New, Select..., Show details..., Remove, Up arrow, Down arrow
  - Table:

PSTN usage record	Associated routes
Long Distance	LocalRoute
- Call forwarding and simultaneous ringing PSTN usages:** (dropdown menu)

Figure 6. Lync Control Panel- Voice Policy

## Lync Control panel: Configuring Route



The screenshot displays the Microsoft Lync Server 2013 Control Panel interface. The top navigation bar includes 'Dial Plan', 'Voice Policy', 'Route' (selected), 'PSTN Usage', 'Trunk Configuration', and 'Test Voice Routing'. The left sidebar lists various management areas, with 'Voice Routing' highlighted. The main content area shows the 'Edit Voice Route - LocalRoute' dialog box. This dialog has 'OK' and 'Cancel' buttons at the top. It contains the following fields and sections:

- Scope:**
  - Name:** LocalRoute
  - Description:** (empty text box)
- Build a Pattern to Match** (collapsible section):
  - Instruction: "Add the starting digits that you want this route to handle, or create the expression manually by clicking Edit."
  - Starting digits for numbers that you want to allow:** (empty text box with placeholder "Type a valid number and then click Add.")
  - Buttons: Add, Exceptions, Remove
  - Match this pattern:** .\*
  - Buttons: Edit, Reset, ?
- Suppress caller ID**
- Alternate caller ID:** (empty text box)

Figure 7. Lync Control Panel - Route (1/2)

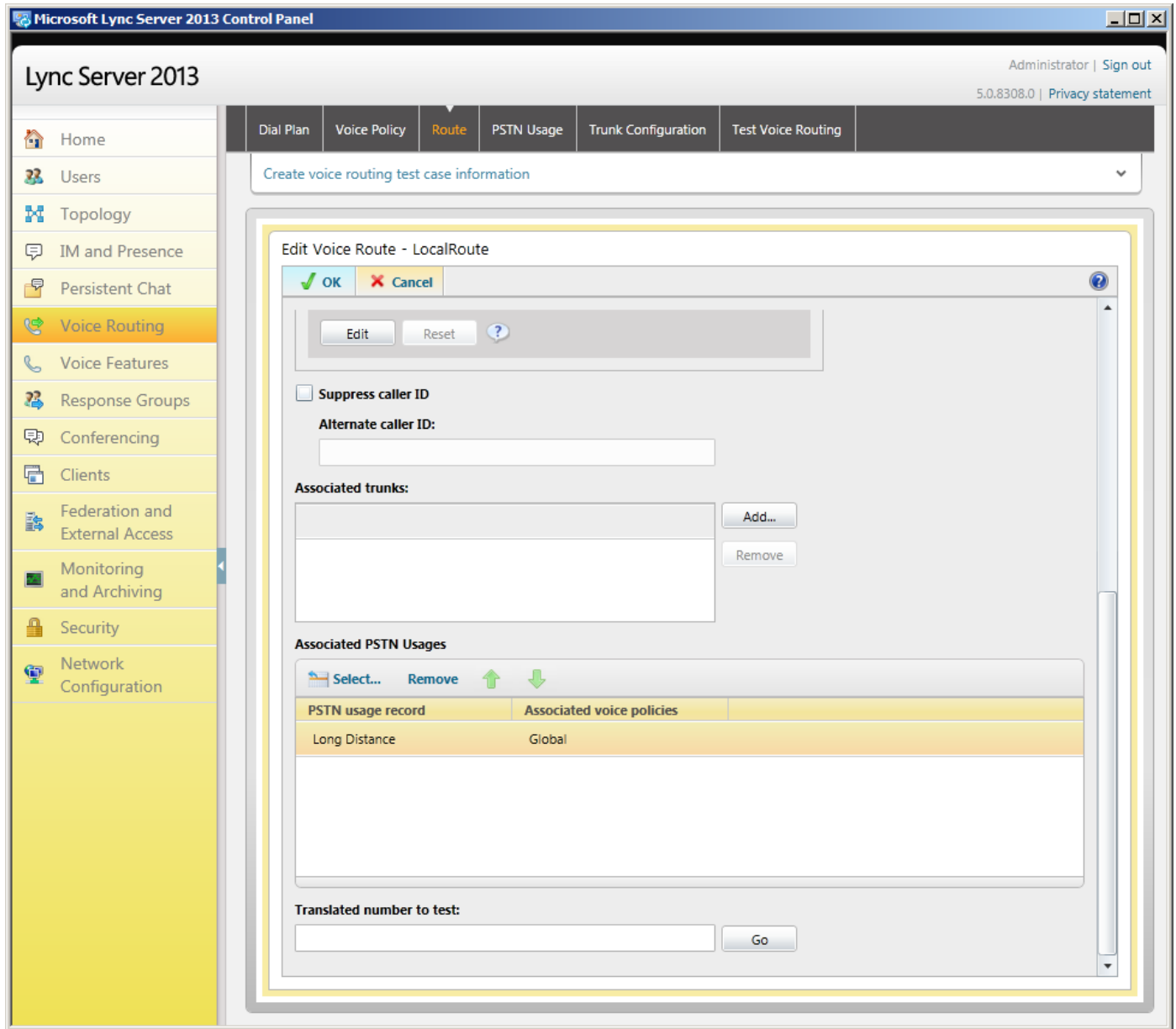
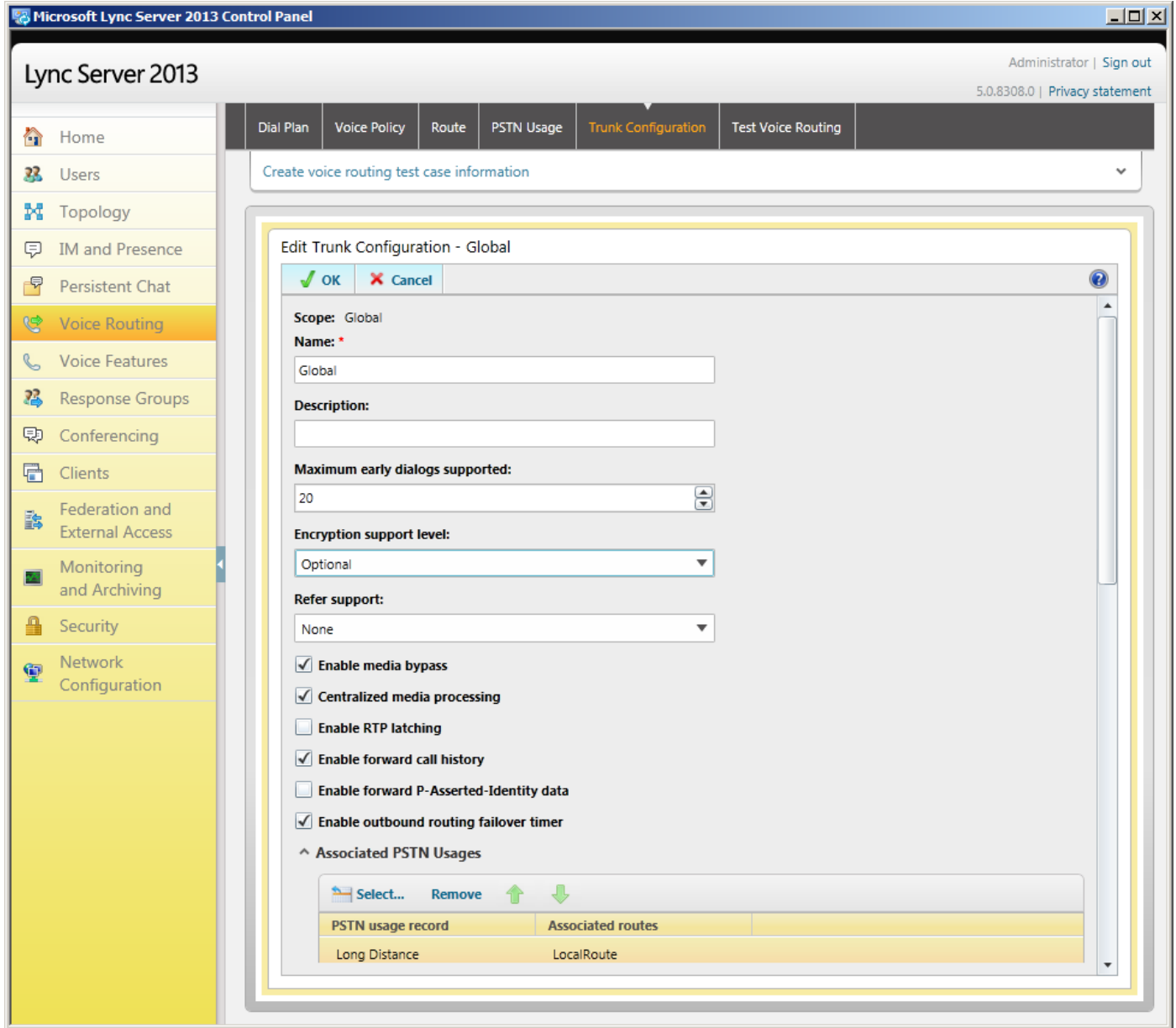


Figure 8. Lync Control Panel - Route (2/2)

## Lync Control panel: Trunk Configuration



The screenshot shows the Microsoft Lync Server 2013 Control Panel interface. The left sidebar contains navigation options: Home, Users, Topology, IM and Presence, Persistent Chat, Voice Routing (highlighted), Voice Features, Response Groups, Conferencing, Clients, Federation and External Access, Monitoring and Archiving, Security, and Network Configuration. The top navigation bar includes Dial Plan, Voice Policy, Route, PSTN Usage, Trunk Configuration (selected), and Test Voice Routing. A search bar contains the text "Create voice routing test case information".

The main content area displays the "Edit Trunk Configuration - Global" dialog box. The dialog has "OK" and "Cancel" buttons at the top left. The configuration details are as follows:

- Scope:** Global
- Name:** Global
- Description:** (empty text box)
- Maximum early dialogs supported:** 20
- Encryption support level:** Optional
- Refer support:** None
- Checkboxes:**
  - Enable media bypass
  - Centralized media processing
  - Enable RTP latching
  - Enable forward call history
  - Enable forward P-Asserted-Identity data
  - Enable outbound routing failover timer
- Associated PSTN Usages:**
  - Select... Remove ↑ ↓
  - Table with 2 columns: PSTN usage record, Associated routes
  - Row 1: Long Distance, LocalRoute

Figure 9. Lync Control Panel - Trunk Configuration



**Acronyms**

<b>Acronym</b>	<b>Definitions</b>
SIP	Session Initiation Protocol
MGCP	Media Gateway Control Protocol
SCCP	Skinny Client Control Protocol
CUCM	Cisco Unified Communications Manager
CUBE	Cisco Unified Border Element



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**Note:** Testing was conducted in Tekvizion labs





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