

Integrating Microsoft Lync Server 2013 and Cisco ISR 3845

Document Revision History

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# Topology

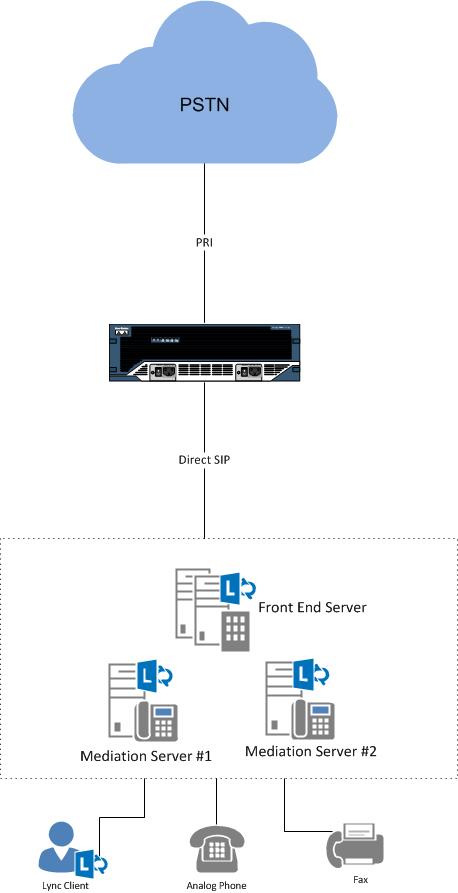


Figure : Deployment Topology

# Components Information

**Lync Server 2013 Version**

|  |  |
| --- | --- |
| **Vendor** | Microsoft |
| **Models** | Lync Server 2013 |
| **Software Version** | RTM: Release 2013 5.0.8308.0 |
| **VoIP Protocol** | SIP |
| **Additional Notes** | None |

# **Cisco ISR 3845 Gateway**

|  |  |
| --- | --- |
| **Vendor** | Cisco |
| **Models** | ISR Gateway 3845 |
| **Software Version** | 15.1-4.M6 |
| **VoIP Protocol** | SIP |
| **Additional Notes** | None |

# Configuration Overview

Call transfer on Lync Client with REFER fail, so REFER is disabled. As REFER is disabled and Media Bypass is enabled on the trunk, Lync Server makes it mandatory to disable RTCP and enabled Session Timer on the trunk.

## Lync Server 2013

|  |  |
| --- | --- |
| Feature | Configuration |
| REFER | Disabled |
| Media Bypass | Enabled |
| Session Timer | Enabled |
| RTCP | Disabled |
| EncryptionLevel | Support Encryption |

## Cisco ISR Gateway

|  |  |
| --- | --- |
| Feature | Configuration |
| PRACK/reliable early media | Disabled |

# ISR 3845 Configuration

Current configuration : 10627 bytes

!

! Last configuration change at 19:13:12 UTC Wed Oct 16 2013 by cisco

version 15.1

service timestamps debug datetime msec

service timestamps log datetime msec

no service password-encryption

!

hostname ciscogd2[[1]](#footnote-1)

!

boot-start-marker

boot system flash c3845-adventerprisek9-mz.151-4.M6.bin

boot-end-marker

!

card type t1 2 1

logging buffered 51200 warnings

enable password tekV1z10n

!

no aaa new-model

!

no network-clock-participate slot 2

!

dot11 syslog

ip source-route

!

ip cef

!

ip domain name lab.tekvizion.com[[2]](#footnote-2)

ip name-server 10.64.1.3[[3]](#footnote-3)

no ipv6 cef

!

multilink bundle-name authenticated

!

isdn switch-type primary-ni

!

voice-card 0

dsp services dspfarm

!

voice-card 2

!

voice service voip

ip address trusted list

ipv4 0.0.0.0 0.0.0.0

ipv4 10.85.0.252

rtcp keepalive

allow-connections sip to sip

no supplementary-service sip refer

supplementary-service media-renegotiate

sip

min-se 600 session-expires 600

asserted-id pai

midcall-signaling passthru

privacy-policy passthru

privacy-policy send-always

sip-profiles 1

block 181

!

voice class codec 1[[4]](#footnote-4)

codec preference 1 g711ulaw

codec preference 2 g711alaw

!

voice class sip-profiles 1

request INVITE sip-header Expires remove

request INVITE sip-header Session-Expires add "Session-expires: 600"

!

voice translation-rule 1

rule 1 /\(..........\)/ /+1\1/

!

voice translation-rule 2

rule 1 /\+1\(..........\)/ /\1/

!

voice translation-rule 3

rule 1 /\(..........\)/ /+\1/

!

!

voice translation-profile toAnalog

translate called 3

!

voice translation-profile toLync

translate called 1

!

voice translation-profile toPRI

translate called 2

!

!

!

application[[5]](#footnote-5)

service dsapp

param dialpeer 7

param callHold TRUE

!

!!

controller T1 2/0

pri-group timeslots 1-24

!

interface GigabitEthernet0/1

ip address 10.64.3.36 255.255.0.0

duplex full

speed 100

media-type rj45

!

interface Serial2/0:23[[6]](#footnote-6)

no ip address

encapsulation hdlc

isdn switch-type primary-ni

isdn timer T310 300000

isdn not-end-to-end 64

isdn incoming-voice voice

isdn map address .\* plan isdn type national

no cdp enable

!

ip default-gateway 10.64.1.1

ip forward-protocol nd

ip http server

ip http secure-server

!

ip route 0.0.0.0 0.0.0.0 10.64.1.1

ip route 10.0.0.0 255.0.0.0 GigabitEthernet0/1

!

control-plane

!

!

voice-port 2/0:23

!

mgcp profile default

!

!

dial-peer voice 1 voip[[7]](#footnote-7)

description outgoing to Lync

translation-profile outgoing toLync

huntstop

rtp payload-type comfort-noise 13

session protocol sipv2

session target dns:medpool.lynclabkm2013.local

session transport tcp

incoming called-number .

voice-class codec 1

voice-class sip localhost dns:ciscogd2.lab.tekvizion.com

no voice-class sip early-offer forced

no voice-class sip block 183

no voice-class sip block 181

voice-class sip options-keepalive

dtmf-relay rtp-nte

!

dial-peer voice 1852263 pots[[8]](#footnote-8)

translation-profile outgoing toPRI

destination-pattern +1[2-9]..[2-9]......$

no digit-strip

direct-inward-dial

port 2/0:23

!

dial-peer voice 2 pots[[9]](#footnote-9)

destination-pattern 8522617

clid network-number 9728522617

port 0/0/0

!

dial-peer voice 3 pots8

destination-pattern 8522618

clid network-number 9728522618

port 0/0/1

!

dial-peer voice 4 voip[[10]](#footnote-10)

description outgoint toAnalog

huntstop

destination-pattern 9728522617

rtp payload-type comfort-noise 13

session protocol sipv2

session target dns:medpool.lynclabkm2013.local

session transport tcp

incoming called-number .

voice-class codec 1

voice-class sip localhost dns:ciscogd2.lab.tekvizion.com

no voice-class sip early-offer forced

no voice-class sip block 183

no voice-class sip block 181

voice-class sip options-keepalive

dtmf-relay rtp-nte sip-notify sip-kpml

!

dial-peer voice 6 voip[[11]](#footnote-11)

description outgoint toAnalog

huntstop

destination-pattern 9728522618

rtp payload-type comfort-noise 13

session protocol sipv2

session target dns:medpool.lynclabkm2013.local

session transport tcp

incoming called-number .

voice-class codec 1

voice-class sip localhost dns:ciscogd2.lab.tekvizion.com

no voice-class sip early-offer forced

no voice-class sip block 183

no voice-class sip block 181

voice-class sip options-keepalive

dtmf-relay rtp-nte sip-notify sip-kpml

!

dial-peer voice 7 voip[[12]](#footnote-12)

translation-profile outgoing toLync

huntstop

destination-pattern 972852263.

rtp payload-type comfort-noise 13

session protocol sipv2

session target dns:medpool.lynclabkm2013.local

session transport tcp

voice-class codec 1

voice-class sip localhost dns:ciscogd2.lab.tekvizion.com

no voice-class sip early-offer forced

voice-class sip block 183 sdp absent[[13]](#footnote-13)

no voice-class sip block 181

voice-class sip options-keepalive

dtmf-relay rtp-nte

!

dial-peer voice 8 voip[[14]](#footnote-14)

description toPSTN

translation-profile outgoing fromAnalog

destination-pattern 1..........

rtp payload-type comfort-noise 13

session protocol sipv2

session target dns:medpool.lynclabkm2013.local

session transport tcp

incoming called-number .

voice-class codec 1

voice-class sip localhost dns:ciscogd2.lab.tekvizion.com

no voice-class sip early-offer forced

no voice-class sip block 183

no voice-class sip block 181

voice-class sip options-keepalive

dtmf-relay rtp-nte sip-notify sip-kpml

!

sip-ua

set pstn-cause 31 sip-status 480

timers expires 1800000

!

# Lync Server Configuration

## Add Cisco ISR Gateway to Lync Topology

Lync recognizes Cisco ISR as a PSTN gateway connected by SIP. So we need to add Cisco ISR to the Lync topology by adding it as a PSTN gateway.

1. To add a PSTN gateway to the Lync topology, run Lync Server Topology Builder as a user in the CSAdministrator group. Then add the Cisco ISR Gateway to the PSTN gateway topology

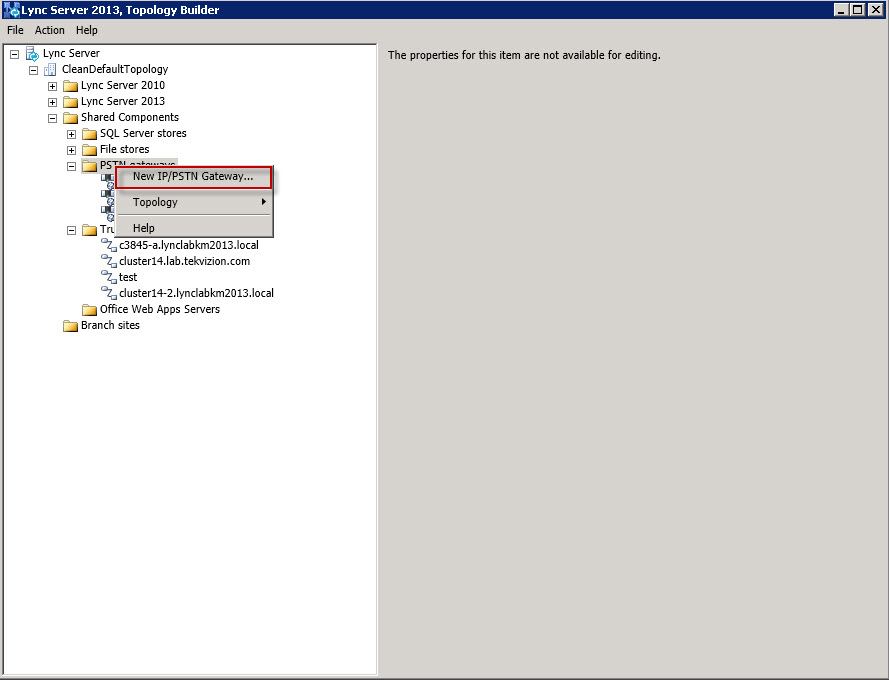


Figure : Configure PSTN Gateway -1

1. Set **FQDN**: This is the IP Address or FQDN of the Cisco ISR Gateway.

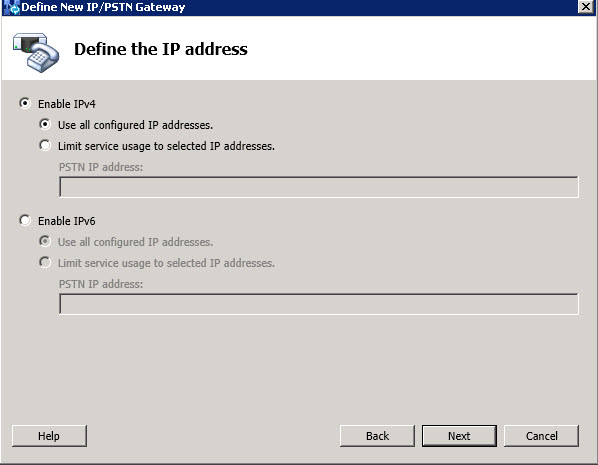


Figure : Configure PSTN Gateway -2

1. Set **Trunk Name**: This is the FQDN of the ISR
2. Set **Listening port for IP/PSTN gateway**: 5060 for TCP
3. Set **SIP Transport Protocol**: TCP
4. Set **Associate Mediation Server**: Assign this PSTN gateway to the Mediation Server. Medpool.lynclabkm2013.local is used here for example.

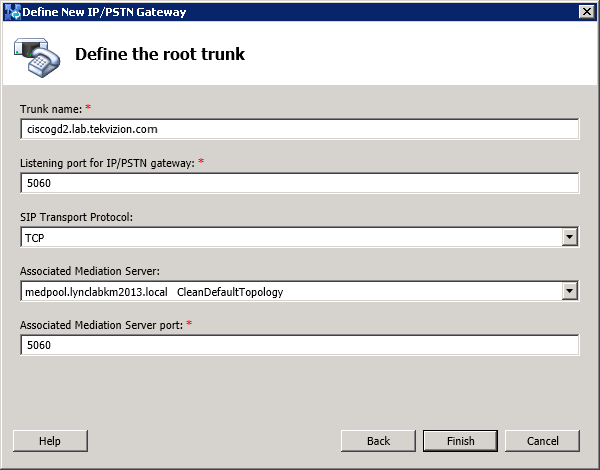


Figure : Configure PSTN Gateway -3

1. Publish topology to make the changes effective, refer to below screen capture for the process.

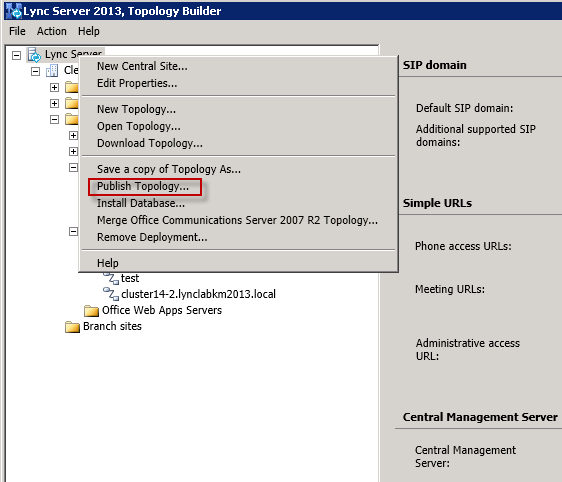


Figure : Publish Topology

## Trunk Configuration

*Navigation: Voice Routing -> Trunk Configuration*

1. Create a **Pool Trunk** by selecting New
2. Select **Service**: Select the trunk to ISR you created in topology builder
3. Set **Maximum early dialogs supported**: 20
4. Set **Encryption support level**: Optional
5. Set **Refer Support**: None
6. Confirm **Enable media bypass**: is checked
7. Confirm **Centralized media processing**: is checked
8. Confirm **Enable RTP latching**: is unchecked
9. Confirm **Enable forward call history**: is unchecked
10. Confirm **Enable forward P-Asserted-Identity data**: is unchecked
11. Confirm **Enable outbound routing failover timer**: is checked

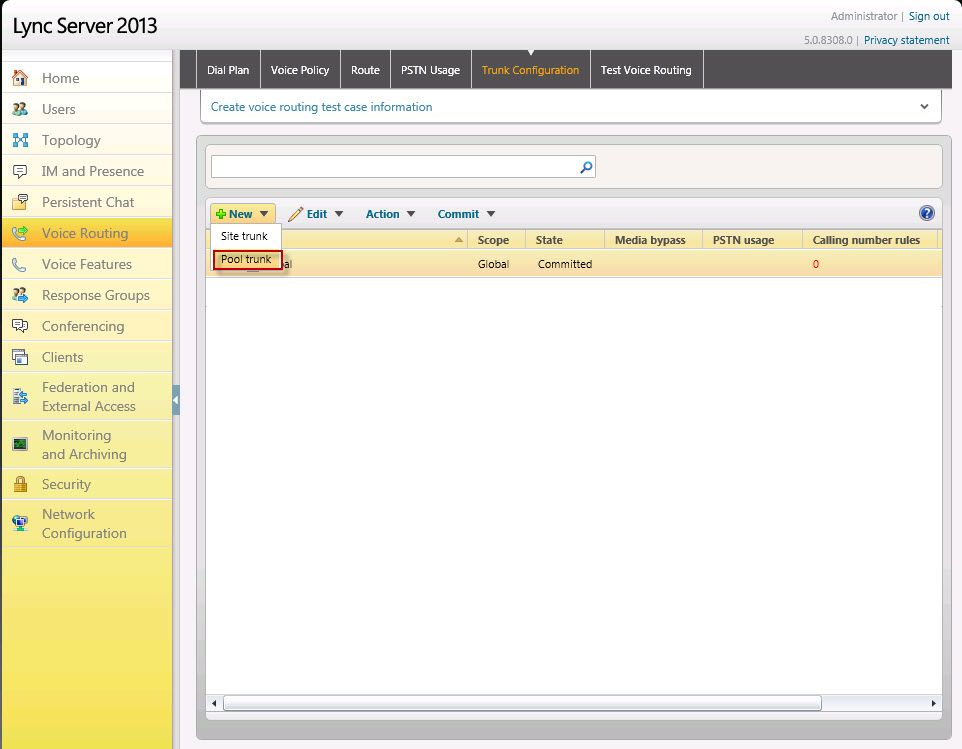


Figure : Trunk Configuration -1

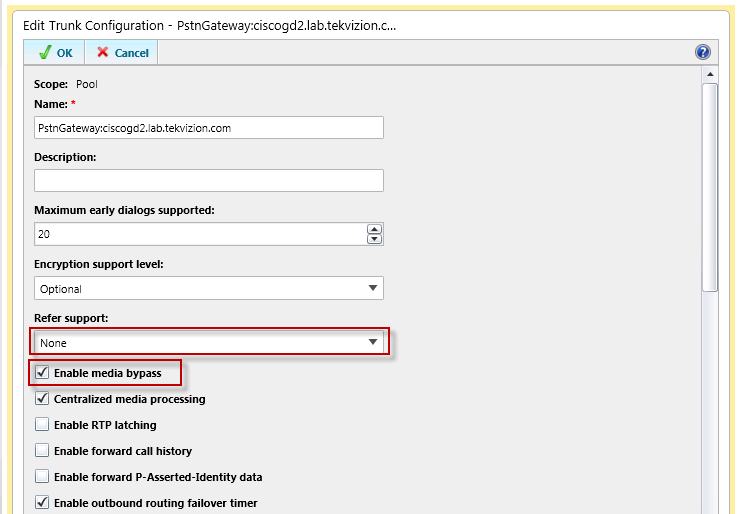


Figure : Trunk Configuration -2

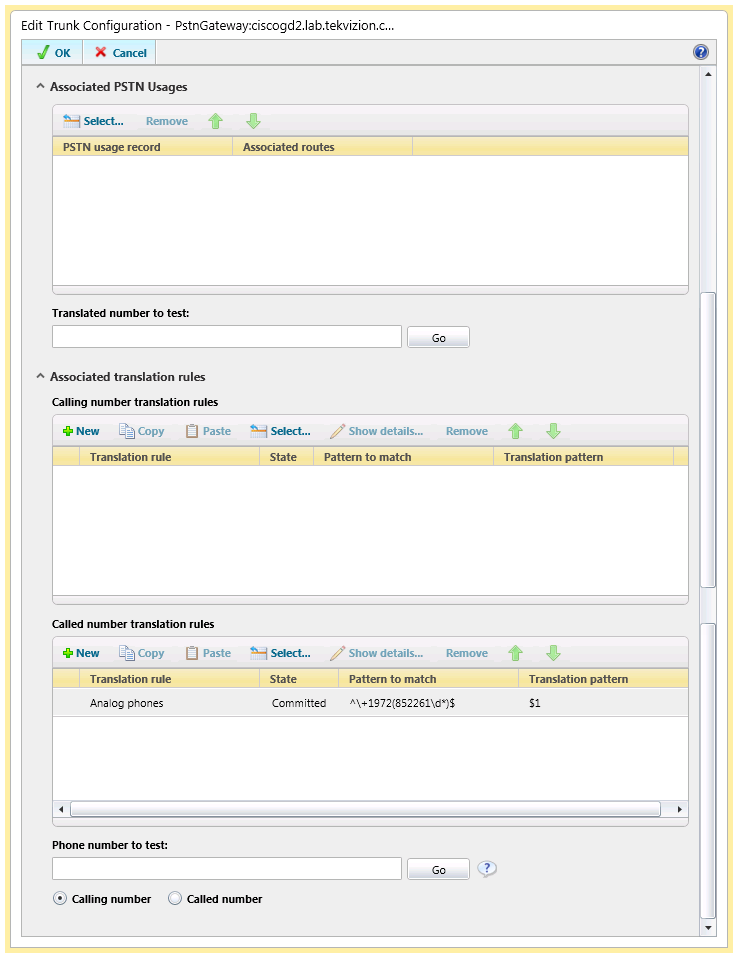


Figure : Trunk Configuration -3

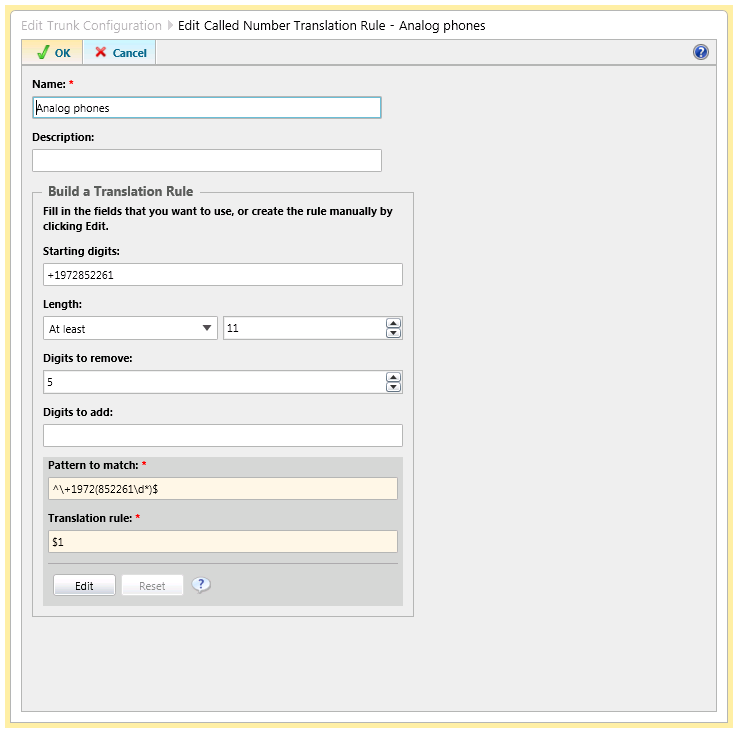


Figure : Called Number Translation Rule-Analog Phones

## Route

*Navigation: Voice Routing -> Route*

1. Set **Name**: Enter a name for this route
2. Add **Associated gateways**: Add the gateway (ISR here) to which this route should send all the calls.

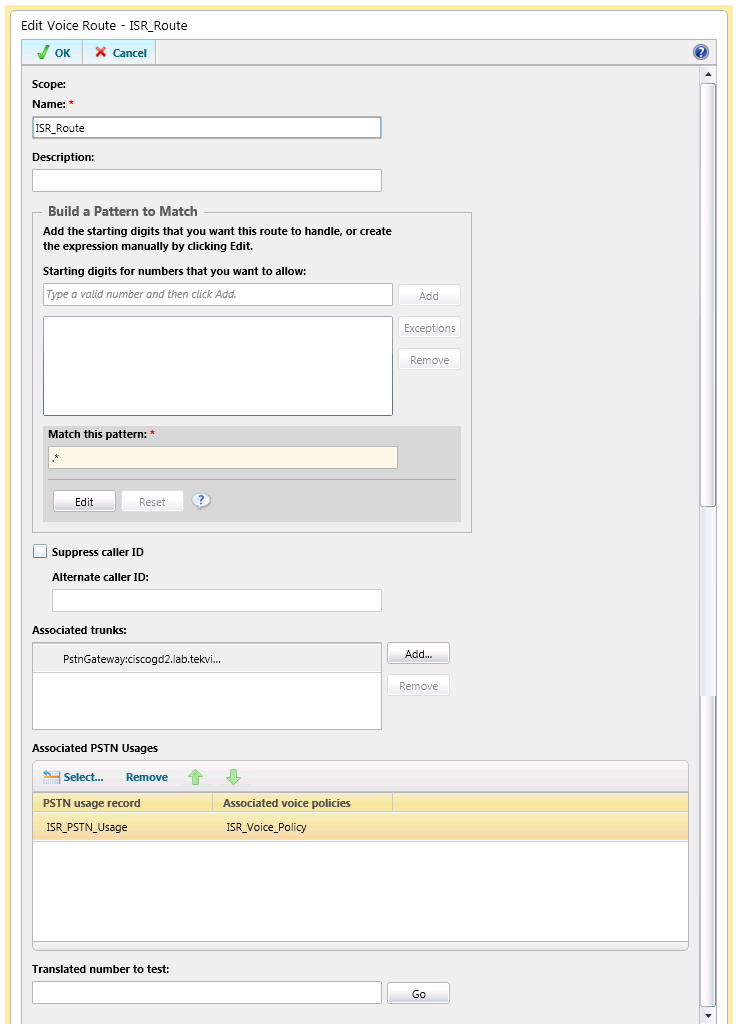


Figure : Route Configuration

## Voice Policy and PSTN Usage

*Navigation: Voice Routing -> Voice Policy*

1. Create a **User policy** by selecting New
2. Set **Name**: Enter a name for this Voice Policy
3. Set **Calling Features**:
   1. Enable call forwarding : Checked
   2. Enable delegation : Checked
   3. Enable call transfer : Checked
   4. Enable call park : Checked
   5. Enable simultaneous ringing of phones : Checked
   6. Enable team call : Checked
   7. Enable PSTN reroute : Checked
   8. Enable bandwidth policy override : Unchecked
   9. Enable malicious call tracing : Unchecked
4. Set **Associated PSTN Usages**:
   1. Select New to create a new PSTN Usage
   2. Set **Name**: Enter a name for this PSTN Usage
   3. Set **Associated Routes**: Select the route you created under Voice Routing -> Route

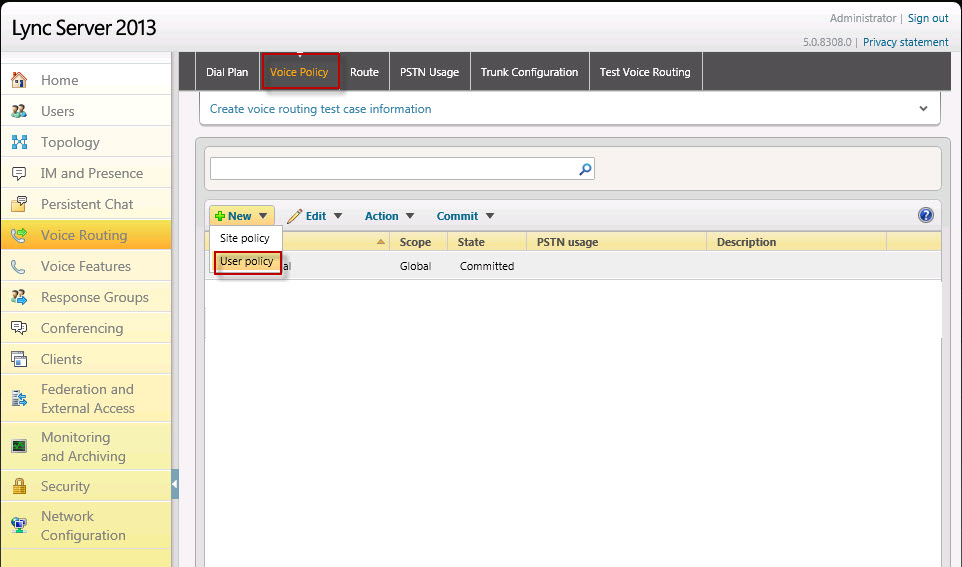


Figure : Voice Policy -1

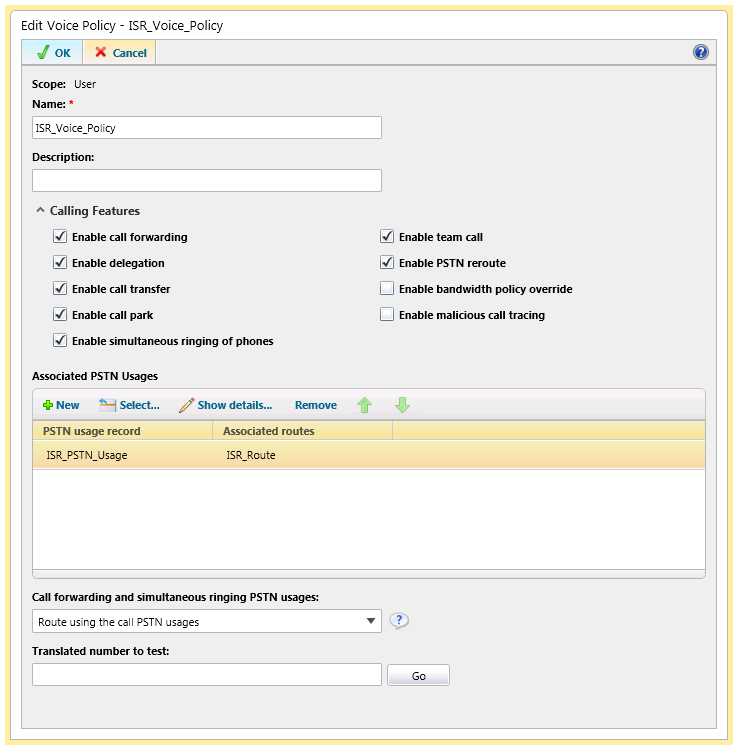


Figure : Voice Policy -2

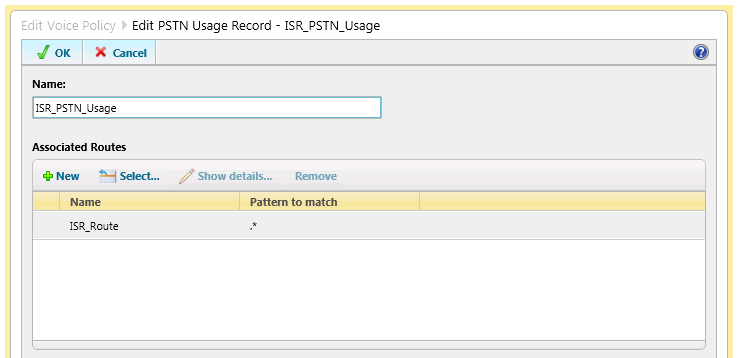
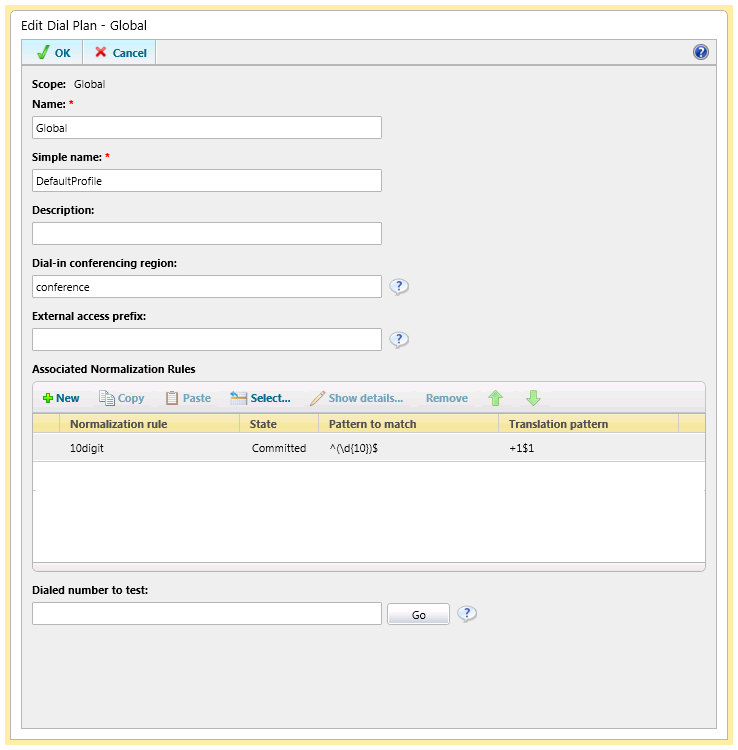


Figure : PSTN Usage

## Dial Plan

Create a dial plan with normalization rules for all the enterprise and local voice calls.



## Configure Media Bypass

*Navigation: Network Configuration -> Global*

1. Check ‘**Enable media bypass**’ in Global setting.
2. Confirm you have also disabled media bypass in the trunk configuration

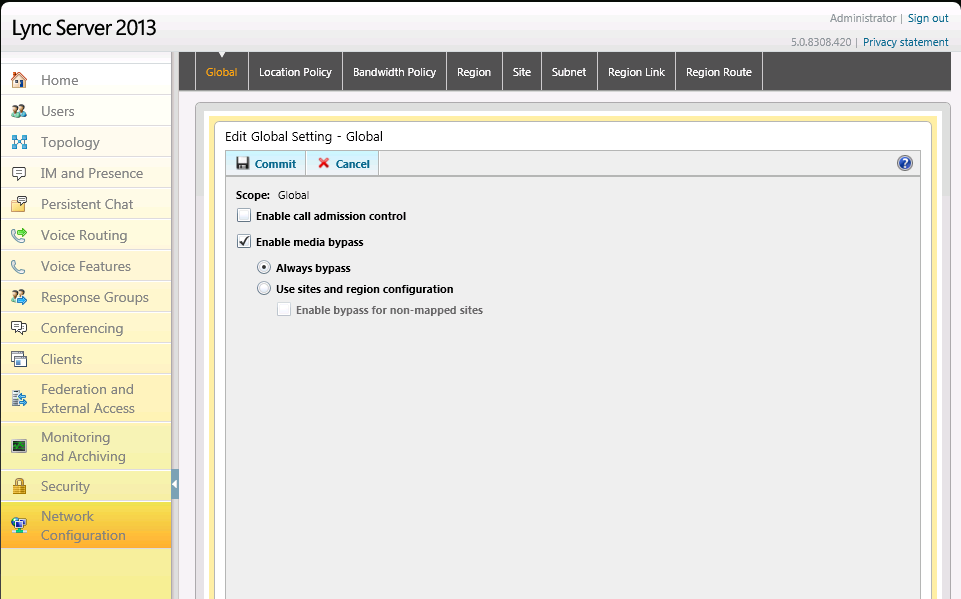


Figure : Media Bypass

Configure RTCP, Session Timer through the Windows PowerShell® command line interface because they are not configurable on Lync Server Control Panel

1. RTCPCallsonHold and RTCPActiveCalls must be turned off. RTCP is a control channel that is opened and is used to monitor the network specific conditions of the RTP channel. As REFER is disabled and Media Bypass is enabled on ISR Trunk, RTCP should be disabled on Lync Server.

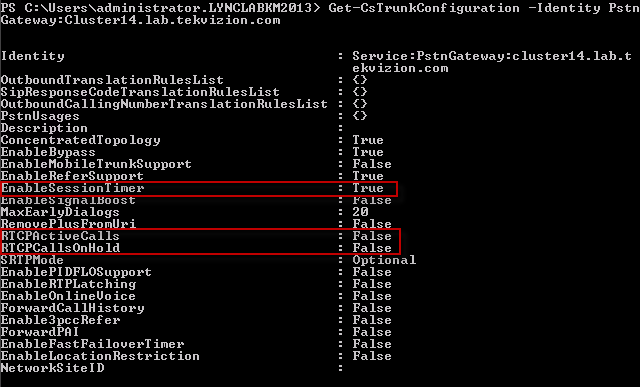
**Set-CsTrunkConfiguration –identity <Trunk name> –RTCPActiveCalls $false –RTCPCallsonHold $false**

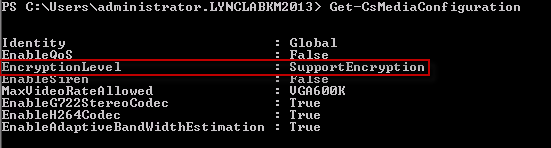
2. SessionTimer must be enabled. Because the RTCP channel is disabled, session timers must be enabled so that calls don’t stay up indefinitely in case we don’t get proper teardown of the call.

**Set-CsTrunkConfiguration –identity <Trunk name> –EnableSessionTimer $true**

3. Media EncryptionLevel must be set to SupportEncryption. Since we do not support SRTP to Cisco through Direct SIP, we need to set the media configuration’s EncryptionLevel to SupportEncryption so that SRTP will only be used if it can be negotiated. By default, this parameter is set to RequireEncryption, meaning SRTP must be used.

**Set-CsMediaConfiguration –identity Global -EncryptionLevel SupportEncryption**





1. Host Name [↑](#footnote-ref-1)
2. Domain Name [↑](#footnote-ref-2)
3. IP Address of DNS Server [↑](#footnote-ref-3)
4. List of supported codecs in the order of preference [↑](#footnote-ref-4)
5. This is an example of the configuration required to enable call hold on Analog Endpoints. To make this active you must enter “service dsapp” in the voice dialpeer. [↑](#footnote-ref-5)
6. PRI interface on ISR Gateway [↑](#footnote-ref-6)
7. Dial peer for all calls going out to Lync [↑](#footnote-ref-7)
8. Dial peer for calls starting with +1 going out to PRI [↑](#footnote-ref-8)
9. Dial peer for analog port [↑](#footnote-ref-9)
10. Dial peer for calls to an Analog extension with 9728522617 [↑](#footnote-ref-10)
11. Dial peer for calls to an Analog extension with 9728522618 [↑](#footnote-ref-11)
12. Dial peer for calls going out to Lync with destination pattern 972852263. [↑](#footnote-ref-12)
13. Block 183 without sdp [↑](#footnote-ref-13)
14. Dial peer for calls going out from Analog extensions [↑](#footnote-ref-14)