Lync Server 2013 using SIP trunk to Cisco Unified Communications Manager Release 10.0

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Introduction

This document describes the steps and configurations necessary for Cisco Unified Communications Manager (Cisco UCM) release 10.0 to interoperate with the Lync Server 2013 using SIP. End points are configured on both Cisco UCM and Lync Server with connectivity to PSTN. A SIP Trunk is configured between Cisco UCM and Cisco Unity for Voicemail connectivity, Lync Server users access Cisco Unity Voicemail via Cisco UCM.

Key Points:

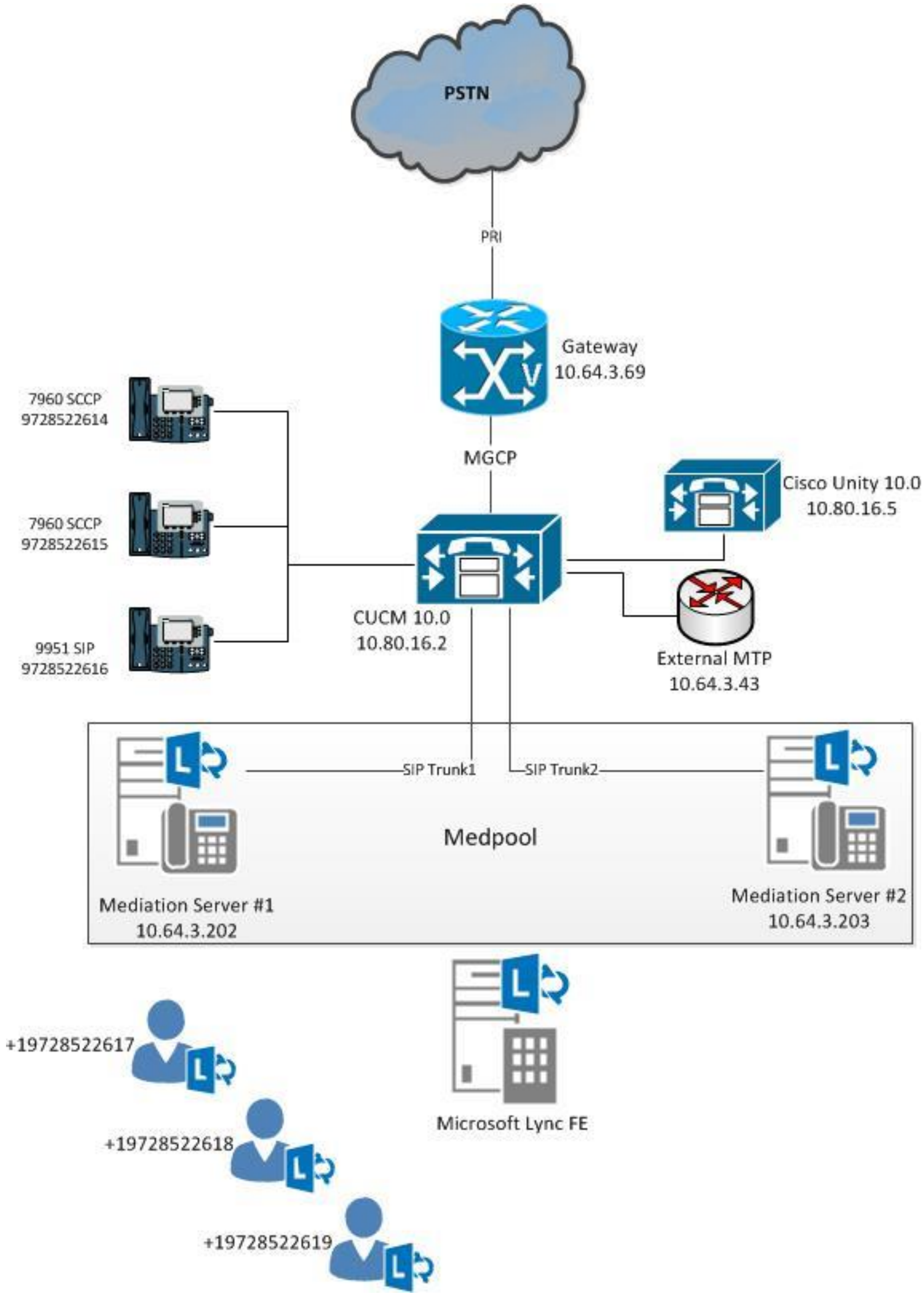
- The testing has been performed with only IPv4 using TCP for signaling.
- CISCO UCM is connected to the PSTN network via MGCP Gateway, as seen in the topology diagram.
- Basic call, call transfer with and without refer, call forwarding, conference call, call hold and resume, call park, RTCP, PRACK, Voice Mail work successfully.
- Testing was performed with Cisco UCM 10.0(1), but a later release fixing the defects CSCum00523 and CSCun13435 is required to resolve REFER and Call Hold issues.
- A Lua script is used to modify the bandwidth line during call hold, to manipulate the user=phone parameter in SIP URIs, to change History-Info headers in inbound INVITEs to Diversion headers, to change Referred-By headers to Diversion headers and to provide ring back at the call originator when PRACK is enabled on the SIP trunk.
- An external MTP is used on CISCO UCM to enable RTCP from CISCO UCM.
- “IP RTCP Interval threshold” on MGCP gateway is changed to 5000 to prevent Lync from dropping the call while the call is on hold with MOH enabled.
- Configuration of multiple SIP trunks and associated routing in Cisco UCM is necessary to support redundant Lync Mediation servers.

The following items were tested:

- Basic outbound and inbound calls between Lync and PSTN through Unified Communications Manager and verification of voice path.
- SIP Headers: E.164 and non-E.164, phone-context, long Request-URI
- Anonymous caller representation
- Codecs: G.711ulaw, G.711alaw, DTMF, Comfort Noise
- Early Media: PRACK, IVR
- RTP and RTCP
- Call transfer: attended, early unattended (only for Cisco endpoints) and blind (only for Lync endpoints).

- Call Park and retrieve
- Shared Lines on Cisco endpoints.
- Call forwarding: Call Forward Unconditional (CFU), simultaneous ring, Call Forward No Answer, Call Forward Busy (only for Cisco endpoints)
- Hold and resume with music on hold and without music on hold.
- Three-way conferencing
- Voice Mail

Network Topology



System Components

Hardware Requirements

The following hardware was tested

- Cisco UCS-C240-M3S VMWare Host running ESXi 5.5
- Microsoft Windows Server 2012 running Hyper-V
- Cisco 3925 Chassis as MGCP Gateway

Software Requirements

The following software was tested:

- Lync Server release 2013 and version 5.0.8308.556
- Cisco Unified Communications Manager version 10.0.1.10000-24
- Cisco Gateway version 15.4(1)T
- Cisco Unity version 10.0.1.10000-24
- A later release of Cisco UCM fixing the defects CSCum00523 and CSCun13435 is required to resolve REFER and Call Hold issues observed in testing.

Features

This section lists supported and unsupported features. Deviance from the configuration presented in this guide is not supported by Cisco. Please see the Limitations section below for more information.

Features Supported:

- Attended call transfer.
- Blind call transfer(only for Lync Endpoints)
- Early unattended transfer (only for Cisco Endpoints)
- Call forwarding unconditional
- Call Forward No Answer
- Call Forward Busy
- Hold and resume with and without refer
- Conference call
- Audio Codecs G711ualaw and G711alaw
- RTCP
- Call Park
- Failover
- Early Media

- MWI on Cisco Phones
- Shared lines on Cisco Endpoints
- Voice Mail Deposit and Retrieval
- Message Waiting Indicator (only for Cisco Endpoints)

Features Not Supported or Not Tested:

- Message Waiting Indicator on Lync Endpoints

Limitations

These are the known limitations, caveats, or integration issues:

- When simultaneous ring is set on Lync client to an IVR and PSTN user makes an inbound call to Lync, the call originator does not hear the early media from IVR.
- No message waiting indicator on Lync for voice mail. Lync rejects the NOTIFY from Unity as it does not have 'Notify' as either Supported or Allowed on the call leg to Unity
- Lync users do not receive Comfort Noise. Cisco provides local Comfort Noise via Cisco IP Phones and Gateways
- The external MTP configured on Unified Communications Manager does not pass-through the RTCP packets coming from Lync or the MGCP Gateway when it receives a=inactive in the SDP on call hold from Lync.
- When Unified Communications Manager and external MTP are configured for G.729 only and if it receives a call with G.711, Unified Communications Manager sends back a "503 Service Unavailable"

Cisco Configuration

The goal of this guide is to provide an overview of the integration between Cisco Unified Communication Manager, Cisco Unity and Lync Server. The deployment will interconnect the UC systems using SIP and using MGCP Gateway for PSTN connectivity to Unified Communications Manager. The following sections provide the required configurations for a successful integration.

Cisco Unified Communications Manager Configuration

SIP Trunk Security Profile Configuration

Navigation: System -> Security -> SIP Trunk Security Profile

1. Set **Name**: Enter a name for the security profile.
   When you save the new profile, the name displays in the SIP Trunk Security Profile drop-down list box in the Trunk Configuration window.
2. Set **Description**: Enter a description relevant to your security profile
3. Confirm **Accept unsolicited notification**: is checked
   If you want Cisco Unified Communications Manager to accept incoming non-INVITE, unsolicited notification messages that come via the SIP trunk, check this check box.
4. Confirm **Accept replaces header**: is checked
   If you want Cisco Unified Communications Manager to accept new SIP dialogs, which have replaced existing SIP dialogs, check this check box

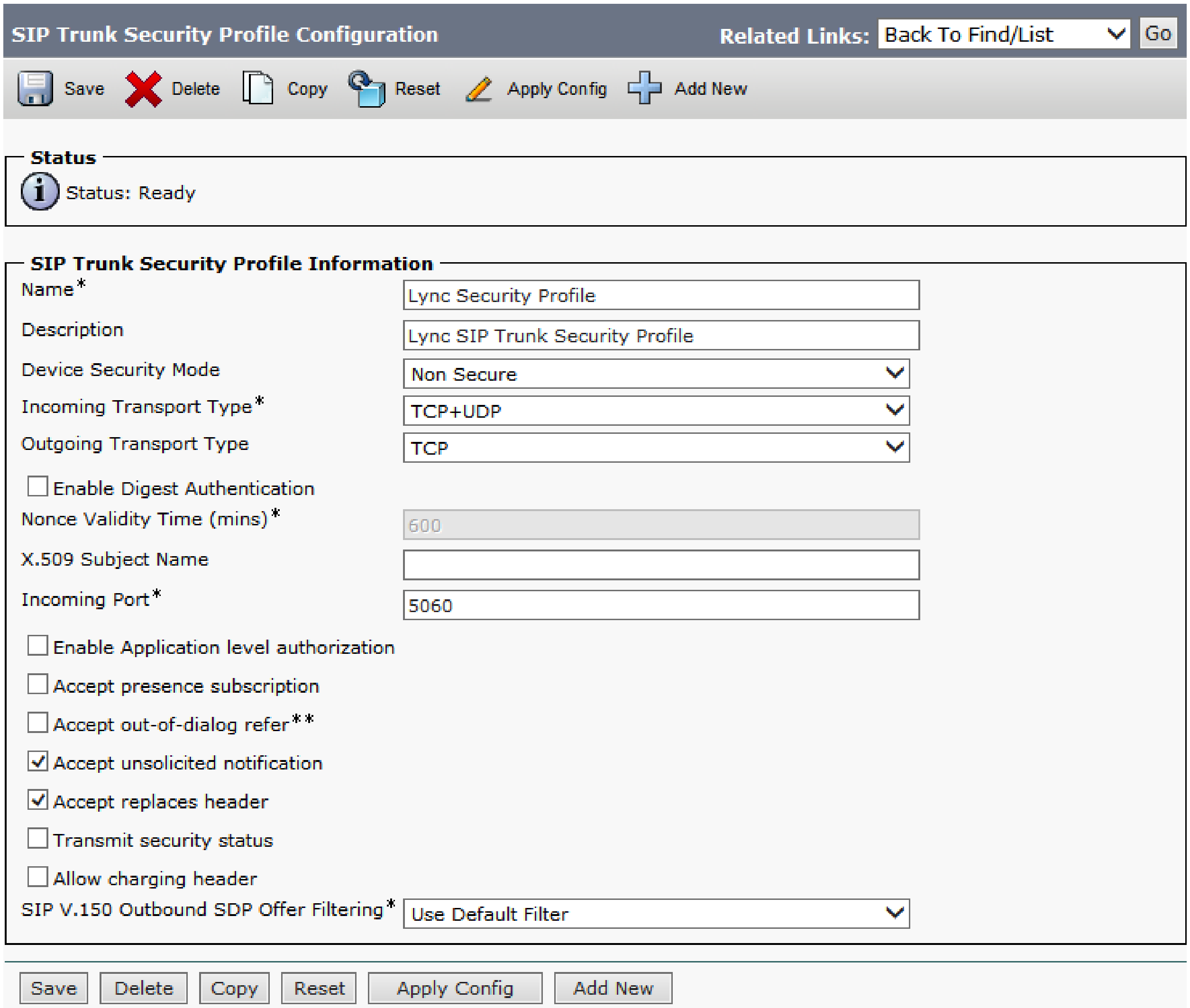


Figure 1: SIP Trunk Security Profile

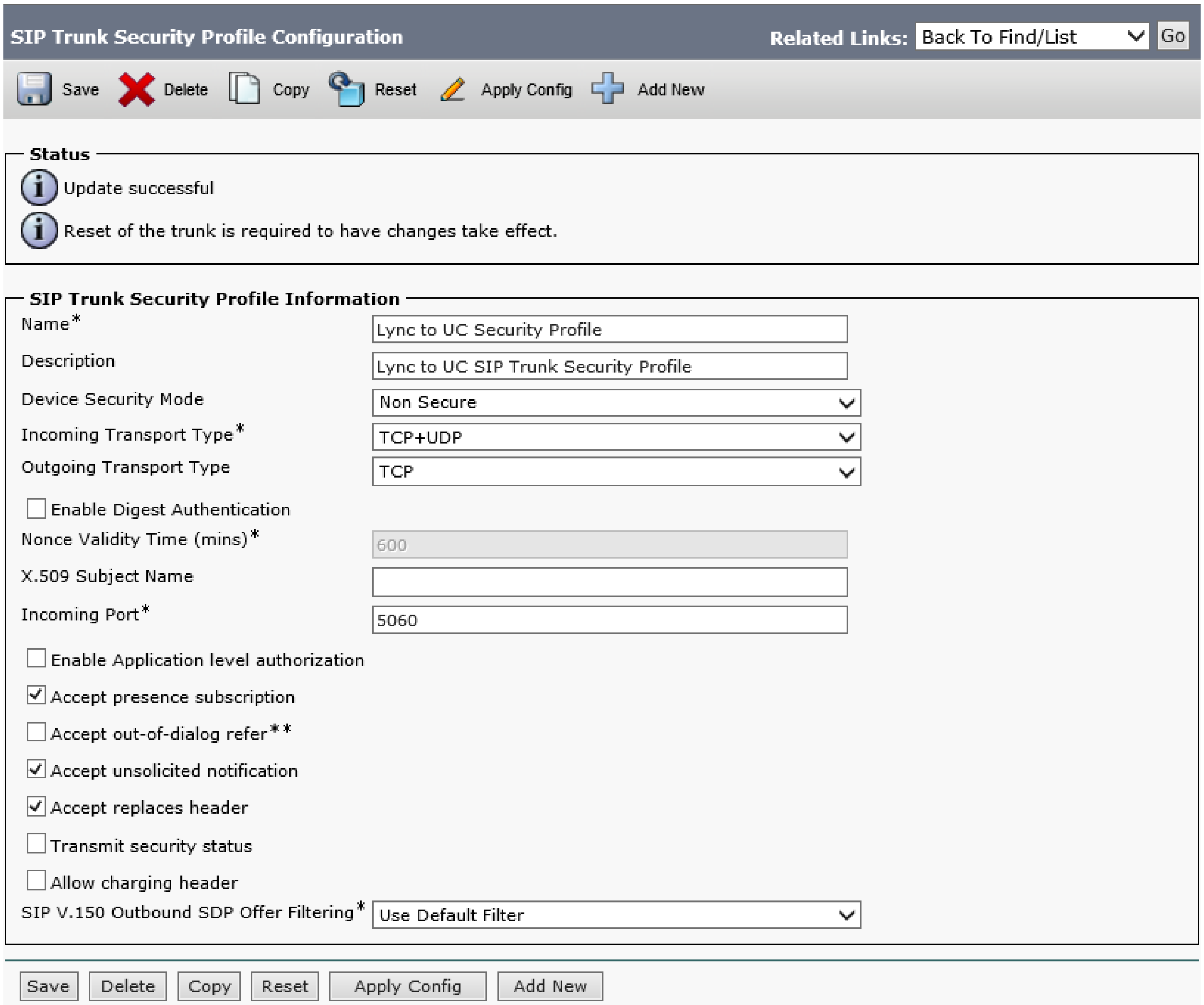


Figure 2: Unity Connection Security Profile

SIP Profile Configuration

Navigation: Device -> Device Settings -> SIP Profile

SIP Profile Information

1. Set **Name**: Enter a name for the SIP Profile
   When you save the new profile, the name displays in the SIP Profile drop-down list box in the Trunk Configuration window
2. Set **Description**: Enter a description relevant to your profile
3. Set **SDP Session-level Bandwidth Modifier for Early Offer and Re-invites**: TIAS and AS
   The Session Level Bandwidth Modifier specifies the maximum amount of bandwidth needed when all the media streams are used.
4. Confirm **Fall back to local RSVP:** is checked
5. Set **SIP Rel1XX Options:** Send PRACK if 1xx contains SDP
6. Confirm **Early Offer support for voice and video calls**: is unchecked
   Check this check box to make this trunk support early offer.
7. Confirm **Send send-receive SDP in mid-call INVITE**: is unchecked
8. Confirm **SIP OPTIONS Ping**: is enabled

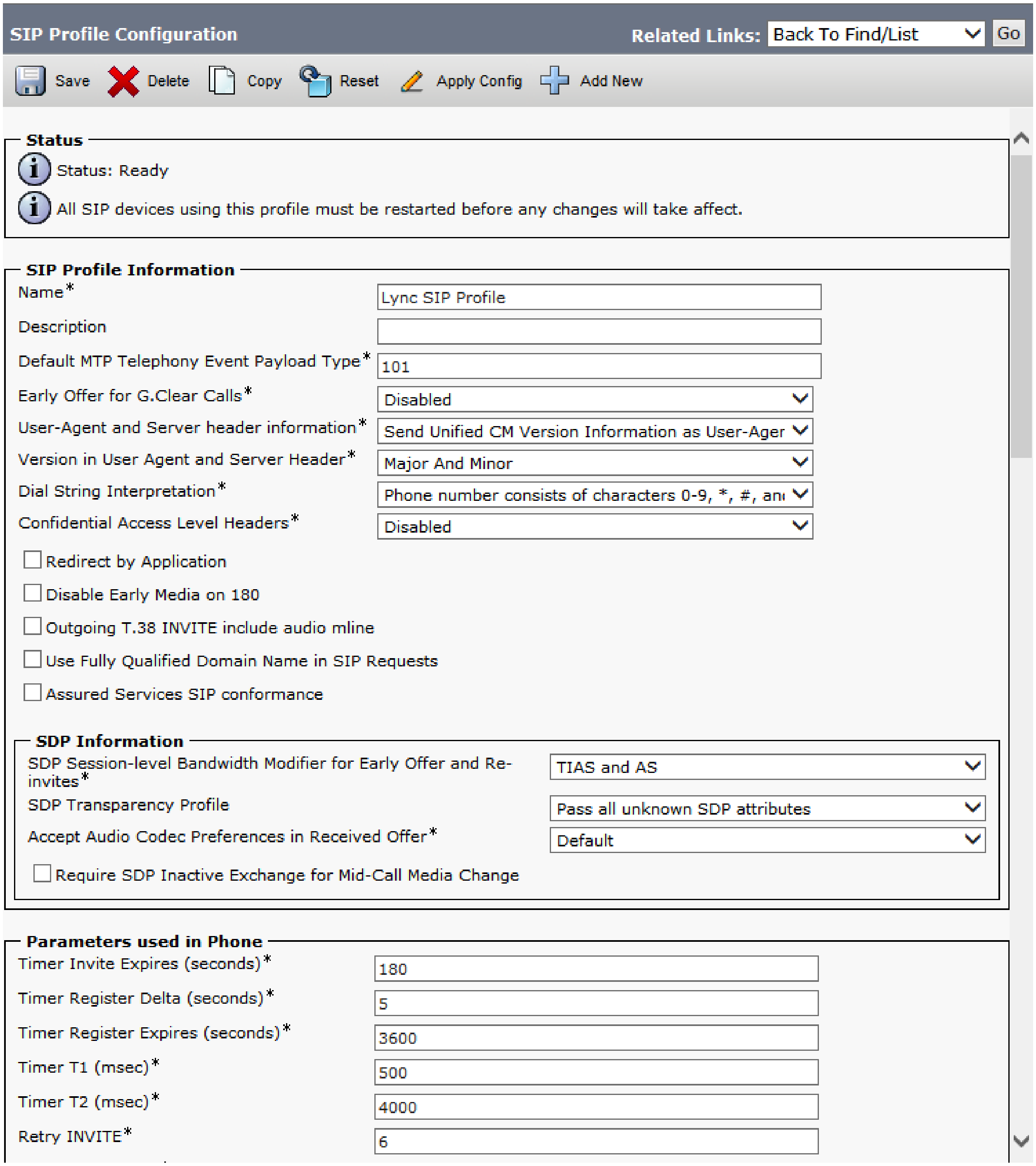
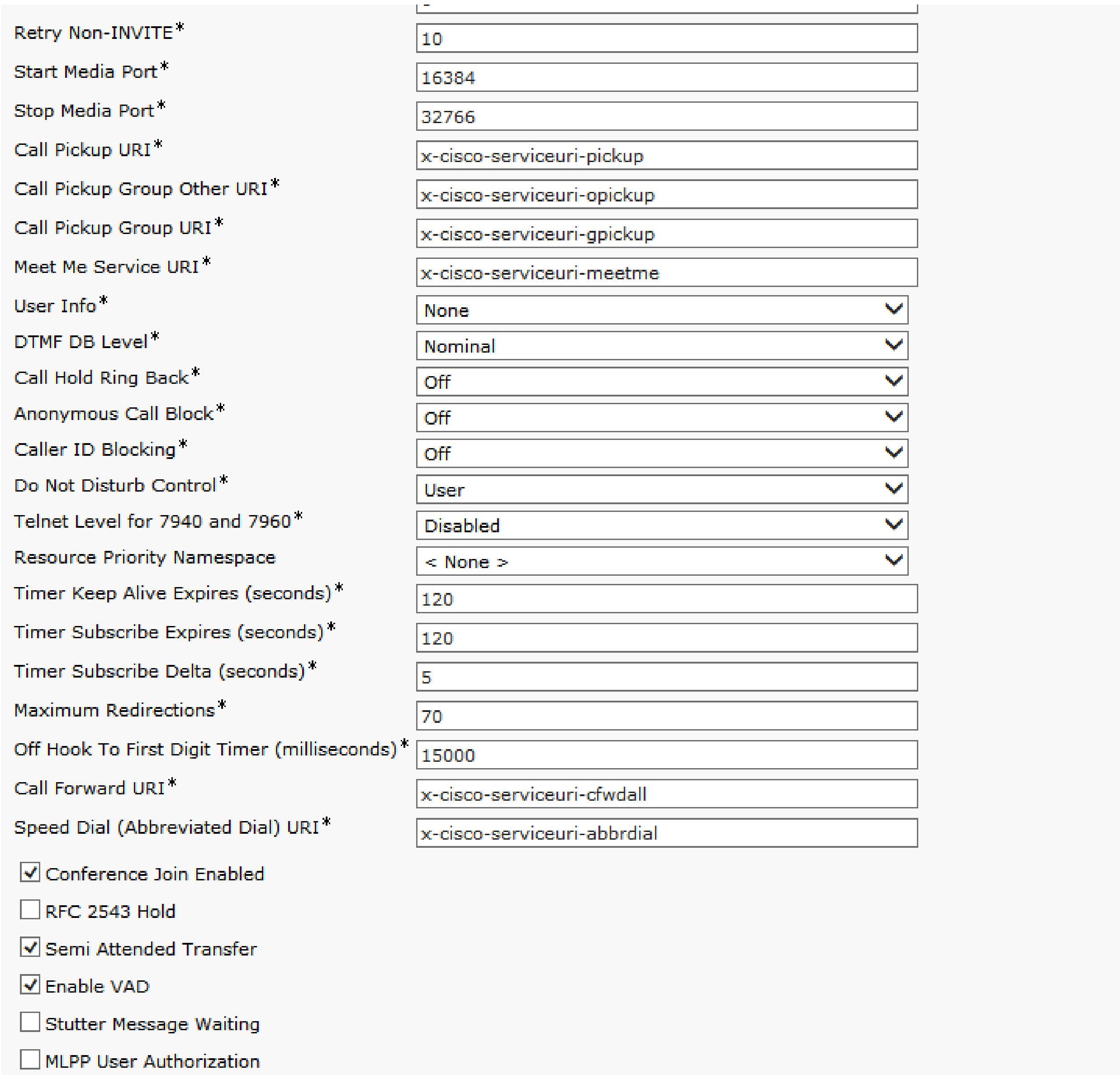


Figure 3: SIP Profile Configuration -1



Retry Non-INVITE*	10
Start Media Port*	16384
Stop Media Port*	32766
Call Pickup URI*	x-cisco-serviceuri-pickup
Call Pickup Group Other URI*	x-cisco-serviceuri-opickup
Call Pickup Group URI*	x-cisco-serviceuri-gpickup
Meet Me Service URI*	x-cisco-serviceuri-meetme
User Info*	None
DTMF DB Level*	Nominal
Call Hold Ring Back*	Off
Anonymous Call Block*	Off
Caller ID Blocking*	Off
Do Not Disturb Control*	User
Telnet Level for 7940 and 7960*	Disabled
Resource Priority Namespace	< None >
Timer Keep Alive Expires (seconds)*	120
Timer Subscribe Expires (seconds)*	120
Timer Subscribe Delta (seconds)*	5
Maximum Redirections*	70
Off Hook To First Digit Timer (milliseconds)*	15000
Call Forward URI*	x-cisco-serviceuri-cfwdall
Speed Dial (Abbreviated Dial) URI*	x-cisco-serviceuri-abbrdial

- [x] Conference Join Enabled
- [ ] RFC 2543 Hold
- [x] Semi Attended Transfer
- [x] Enable VAD
- [ ] Stutter Message Waiting
- [ ] MLPP User Authorization

Figure 4: SIP Profile Configuration-2

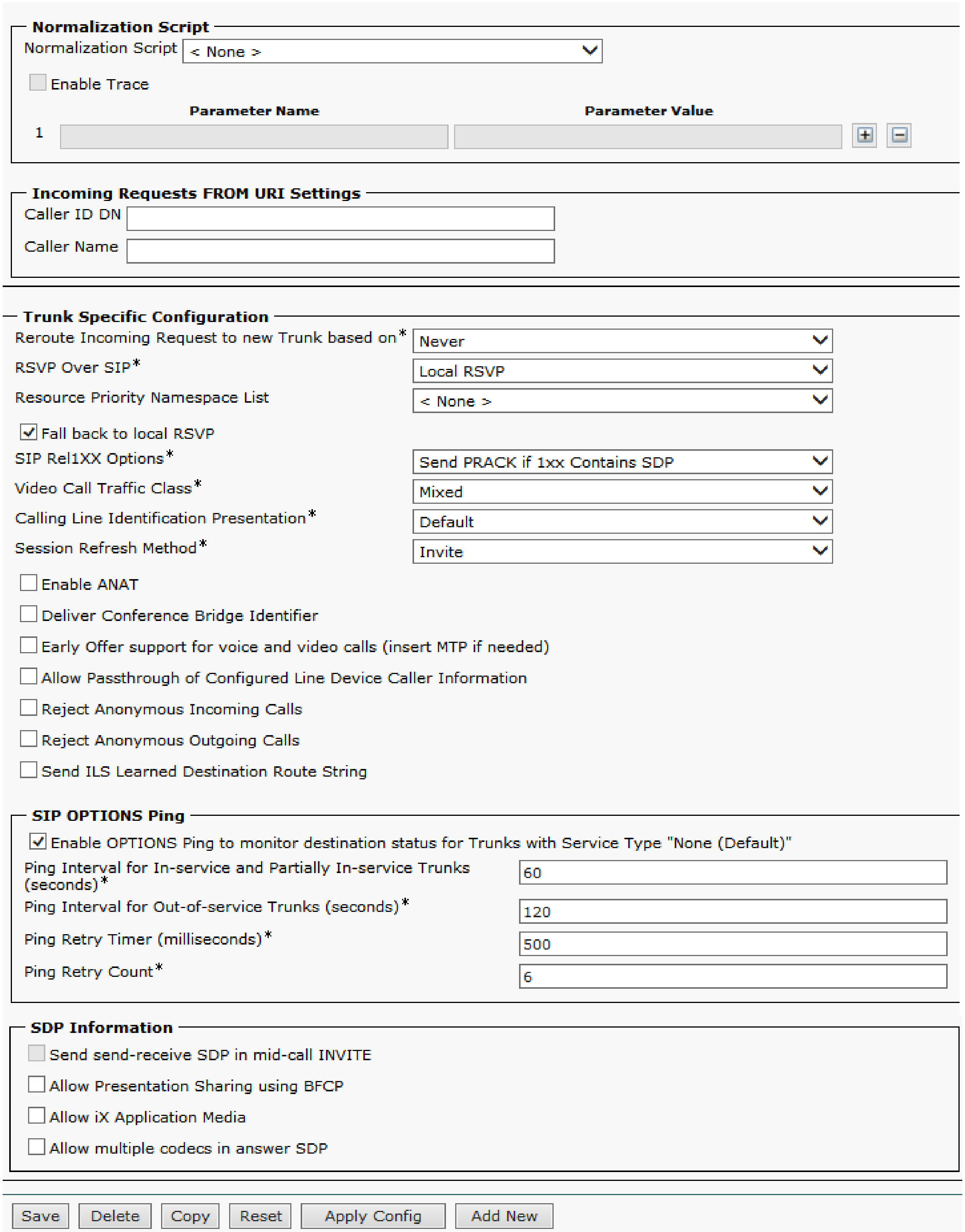


Figure 5: SIP Profile Configuration-3

Media Termination Point

Navigation: Media Resources->Media Termination Point

1. Set **Media Termination Point Name:** Enter the name of the external media termination point
2. Set **Device Pool:** Select the device pool, default device pool is used in this configuration.

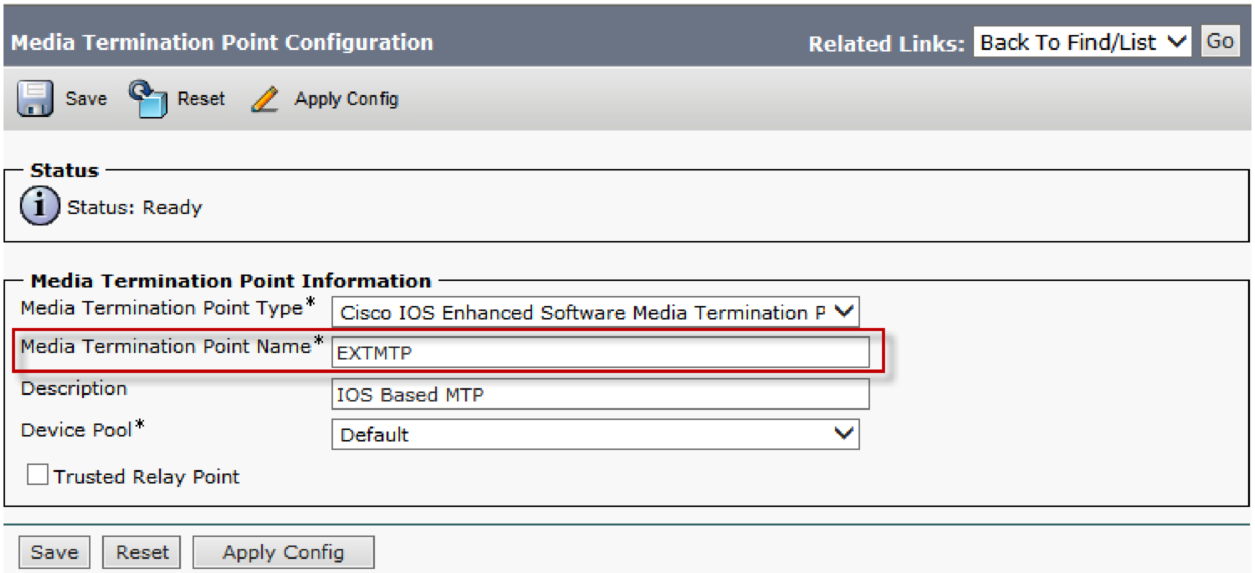


Figure 6: Media Termination Point-1

3. Confirm, the configured MTP is registered with the Unified Communications Manager

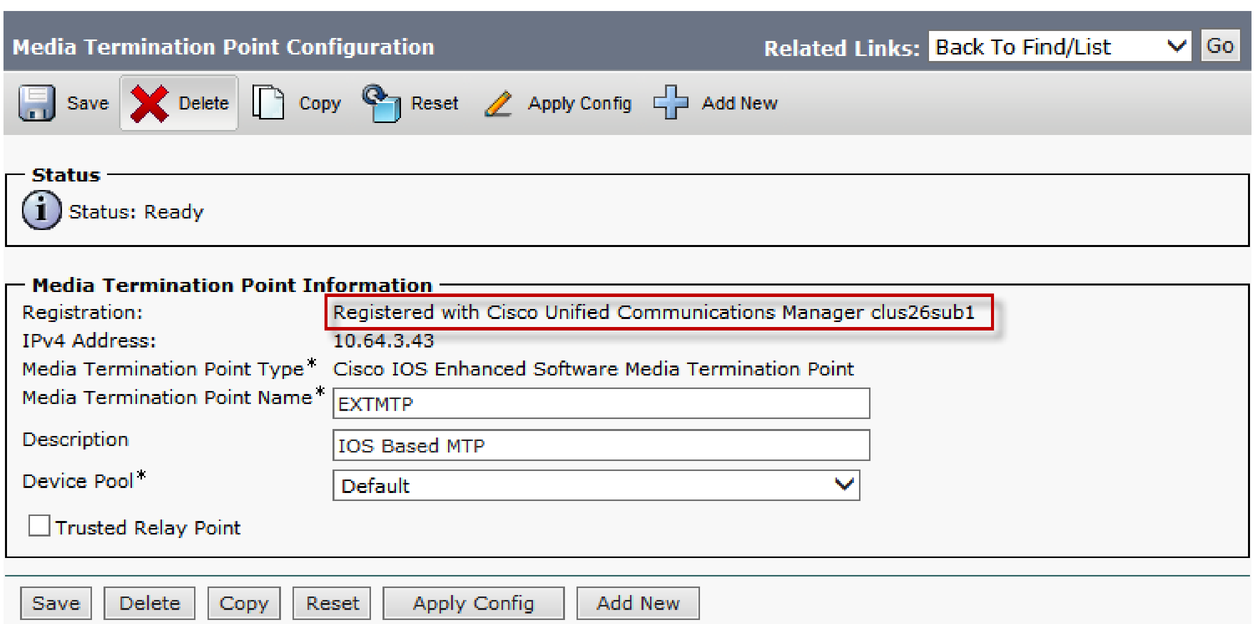


Figure 7: Media Termination Point-2

Media Resource Group

Navigation: Media Resources -> Media Resource Group

1. Add New Media Resource Group
2. Set Name: Enter a name for this group
3. Add Resources: Select the available resources as shown in the screen capture below.

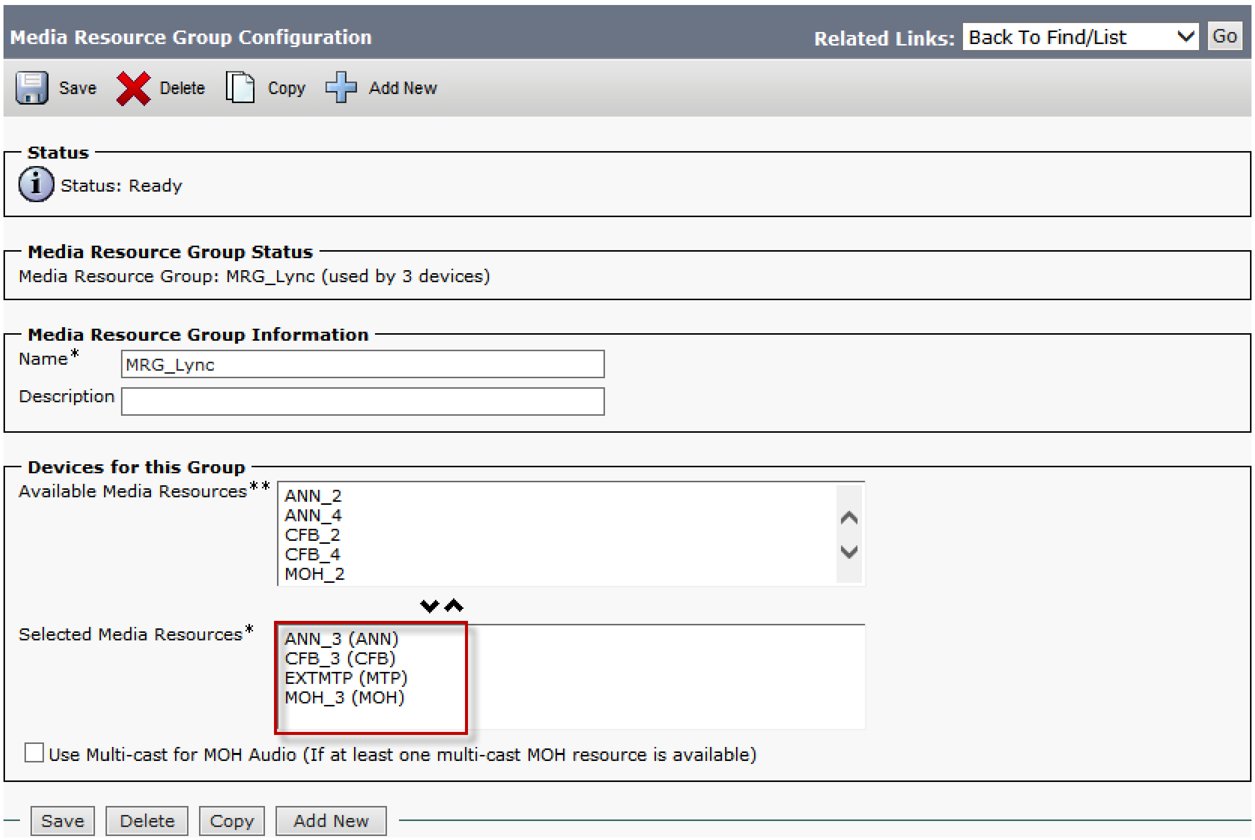


Figure 8: Media Resource Group Configuration

Media Resource Group List Configuration

Navigation: Media Resources -> Media Resource Group List

1. Add New Media Resource Group List
2. Set **Name**: Enter name for this list
3. Select the media resource group you created under Media Resources -> Media Resource Group from the available groups

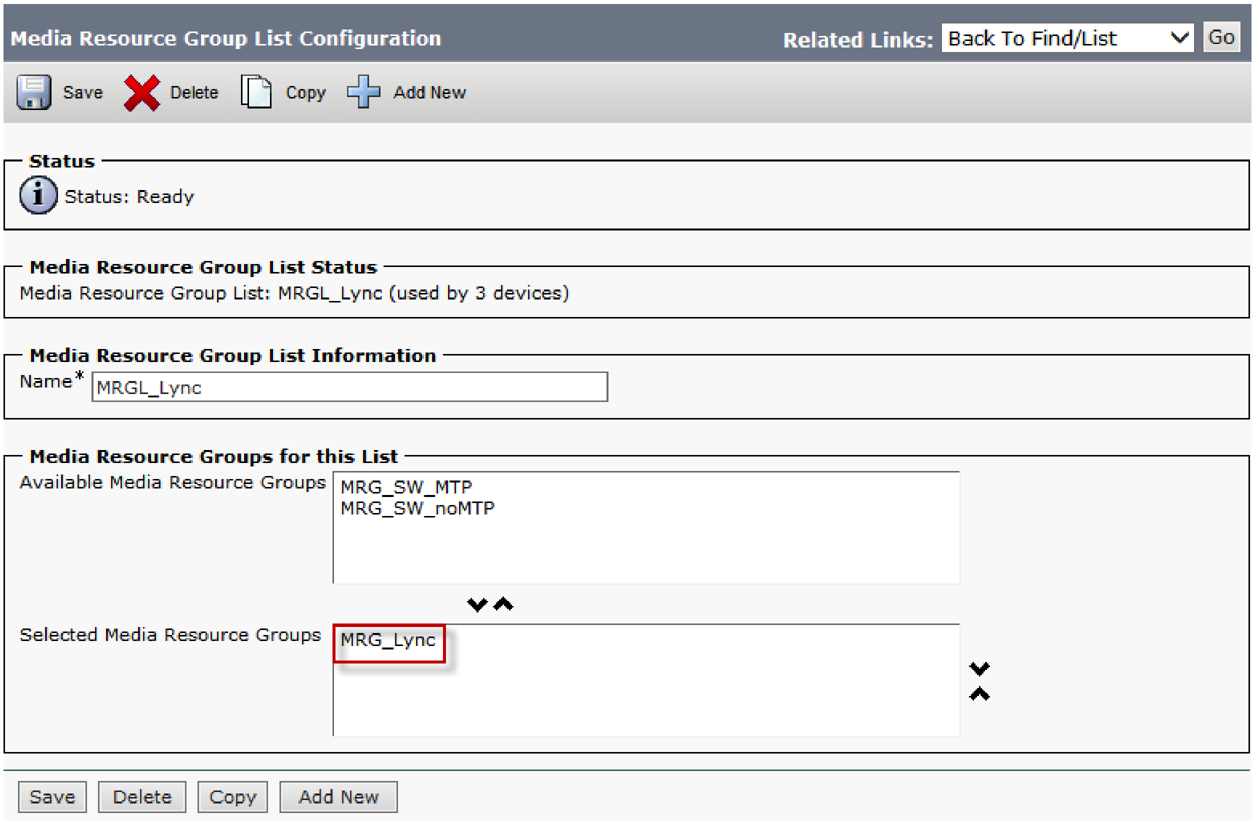


Figure 9: Media Resource Group List Configuration

4. Repeat the same to create Media Resource Group Lists with MTP and without MTP. Select the relevant Media Resource Group for each list as shown in the screen captures below.

SIP Trunk Configuration to Lync

Navigation: Device -> Trunk

Trunks are created from Unified Communications Manager to each Lync Mediation Server for trunk failover and also to enable communications between Cisco UCM and Lync Mediation Servers. The FQDNs are used for configuring the trunks to the Lync Mediation Servers. However, due to the current limitation on Cisco UCM, if a SIP trunk is associated to a SIP-Route Pattern, the same trunks is not available to be included in a Route-List. This creates a need for a duplicate set of trunks to each Lync Mediation Server using IPv4 address. This makes the total number of trunks required to be four (two trunks using FQDN and two trunks using IP) to enable the provisioning of Route List and SIP Route Patterns to the Lync Mediation Servers.

Device Information

1. Set **Trunk Type**: SIP Trunk
2. Set **Device Protocol**: SIP
3. Set **Trunk Service Type**: None
4. Set **Device Name**: Enter a name for the trunk
5. Set **Description:** Enter a description relevant to your trunk
6. Set **Device Pool**: Default
   For trunks, device pools specify a list of Cisco Unified Communications Managers that the trunk uses to distribute the call load dynamically
7. Set **Media Resource Group List**: MRGL_Lync, this is the list you created under
   Media Resources -> Media Resource Group List.
   This list provides a prioritized grouping of media resource groups. An application chooses the required media resource, such as a Music on Hold server, from among the available media resources according to the priority order that a Media Resource Group List defines.
8. Confirm **Media Termination Point Required**: is checked
   This check box is used to indicate whether a media termination point (MTP) is used to implement features that H.323 does not support (such as hold and transfer).
9. Confirm **Retry Video Calls as Audio**: is checked
10. Confirm **Run On All Active Unified CM Nodes**: is checked

SIP Information

11. Set the **Destination Address**: Enter the FQDN of the Mediation Server to which you are establishing a trunk.
12. Set **SIP trunk Security Profile**: Select the security profile you created under System -> Security -> SIP Security Profile
13. Set **SIP Profile**: Select the SIP Profile you created under Device -> Device Settings -> SIP Profile
14. Set **Normalization Script:** Select the normalization script to modify the bandwidth line *b=CT:64* during call hold, ring back issue with PRACK enabled.

Trunk to Mediation Server 1 using FQDN

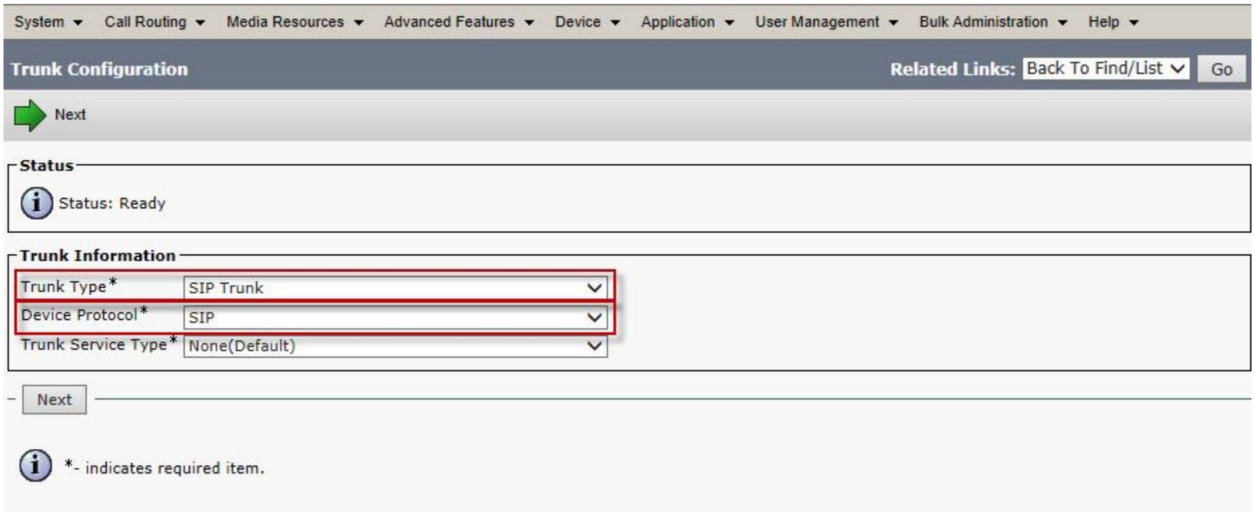


Figure 10: SIP Trunk to Lync Configuration -1_1

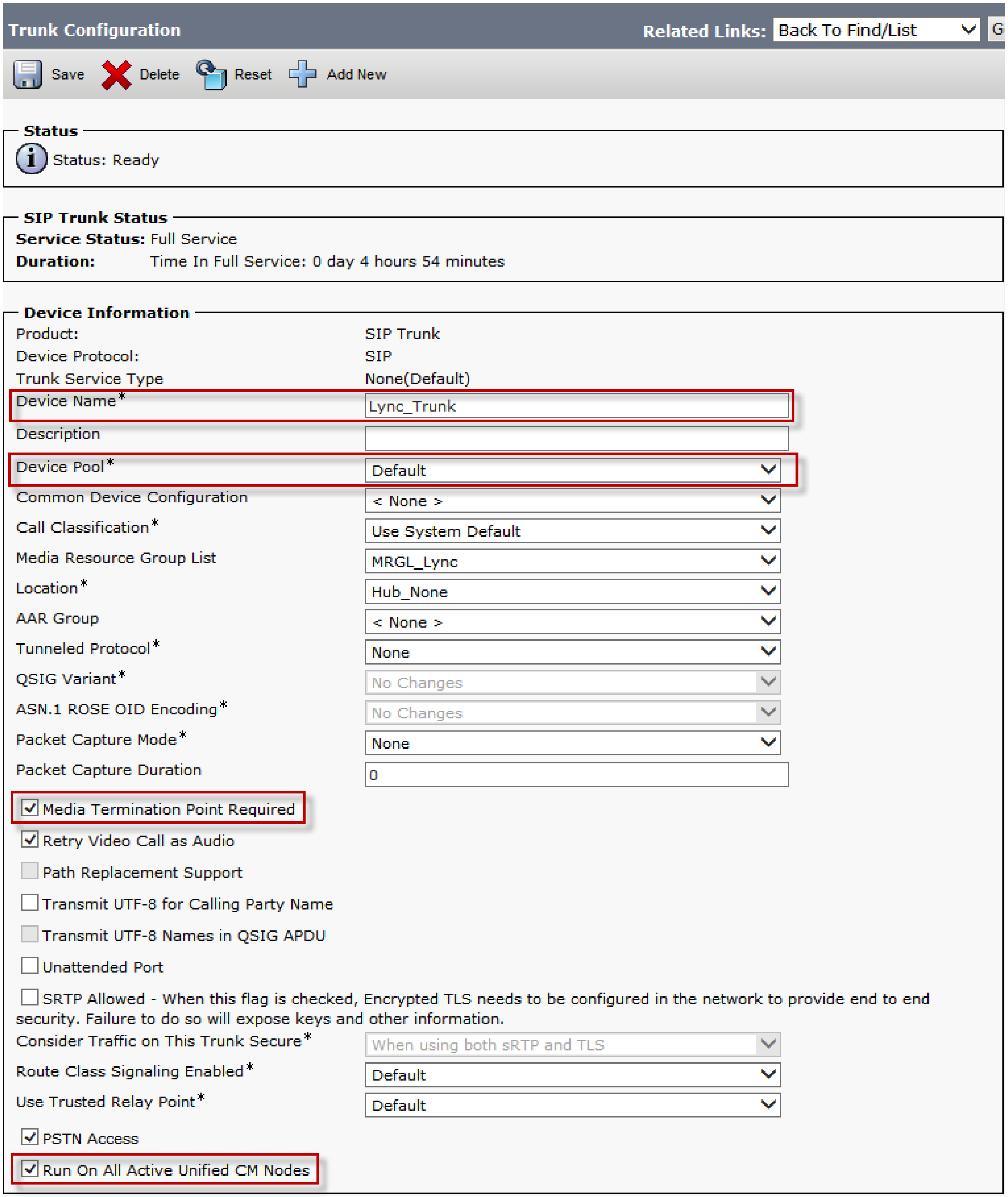


Figure 11: SIP Trunk to Lync Configuration -1_2

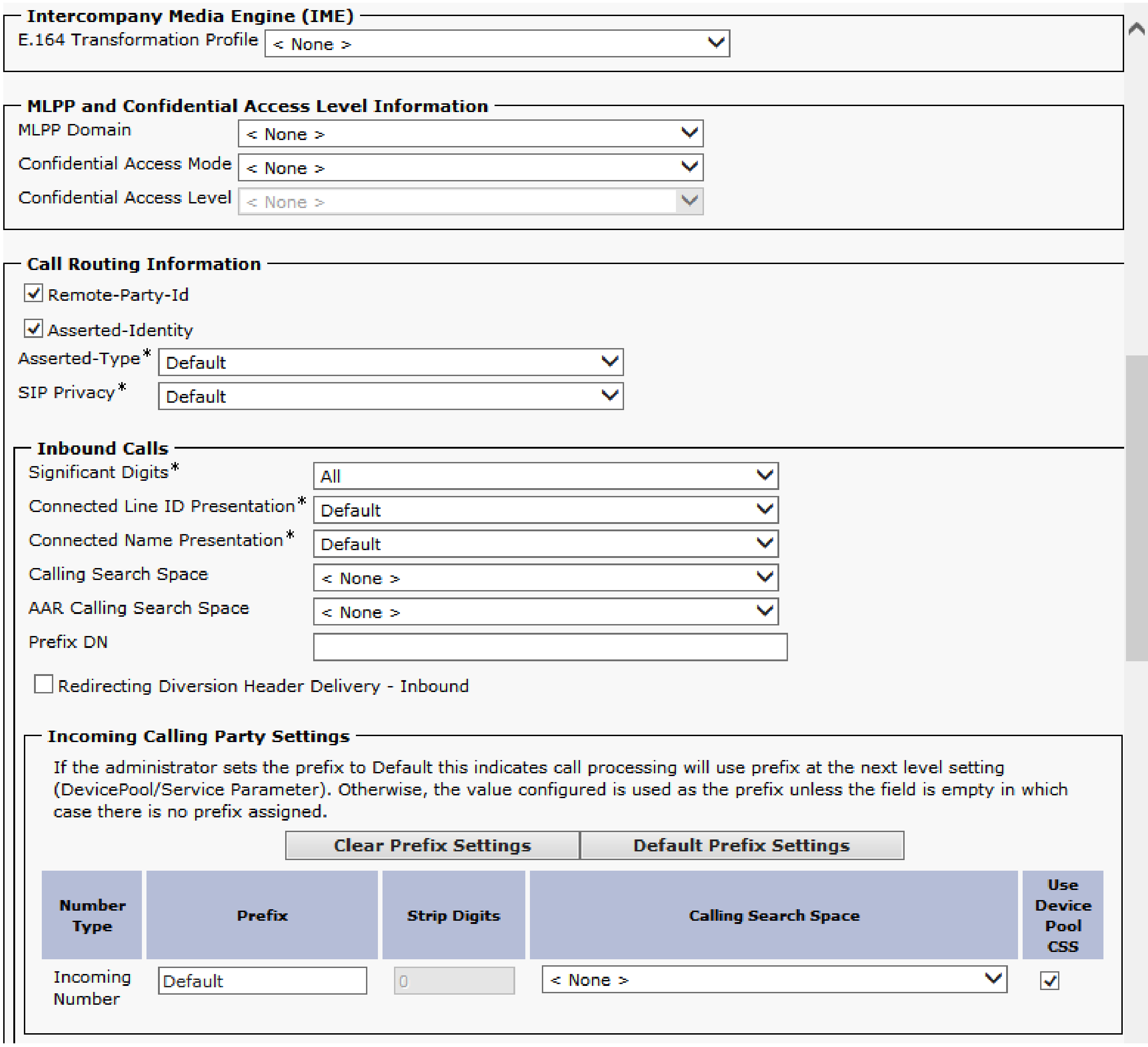


Figure 12: SIP Trunk to Lync Configuration -1_3

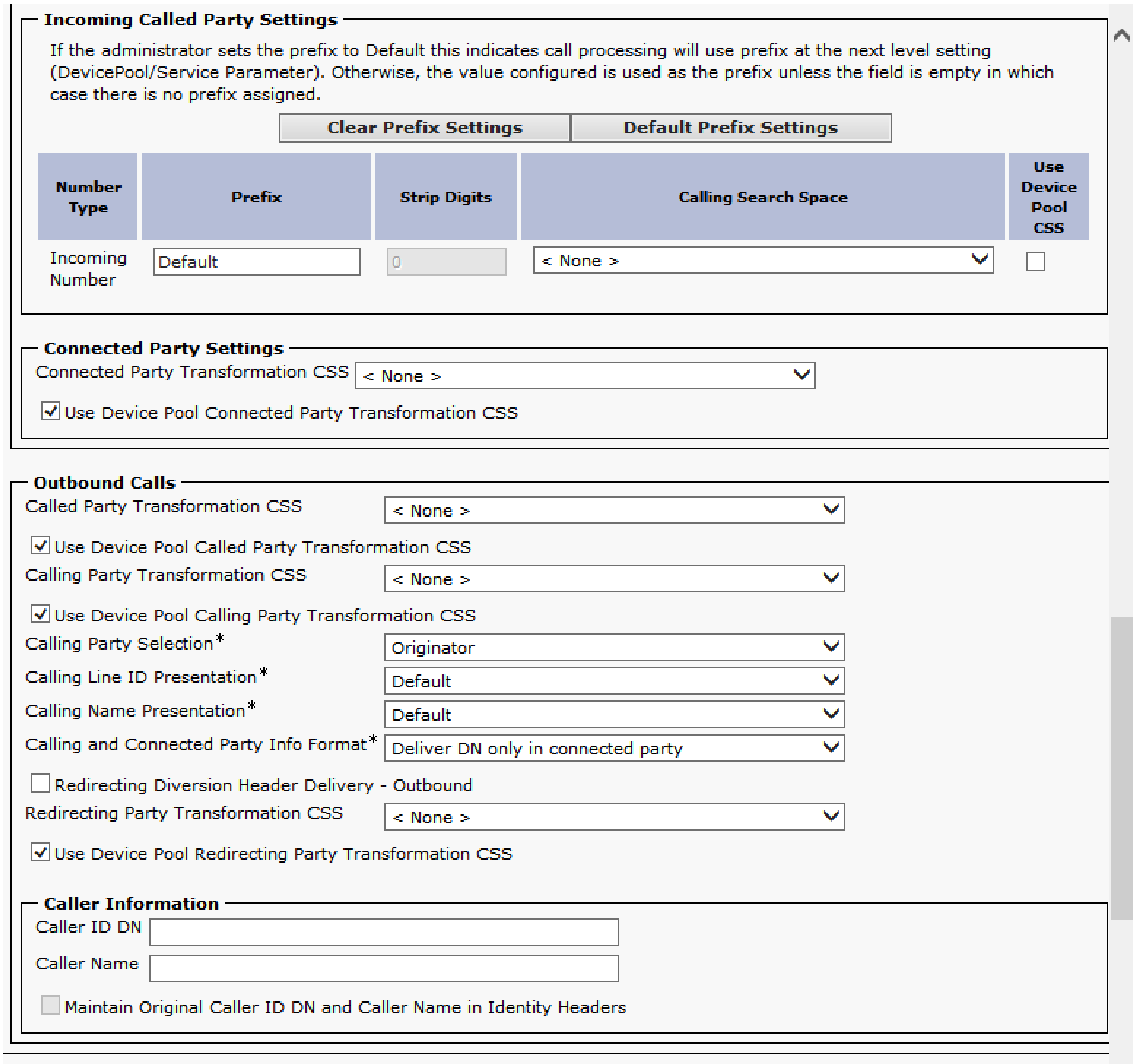


Figure 13: SIP Trunk to Lync Configuration -1_4

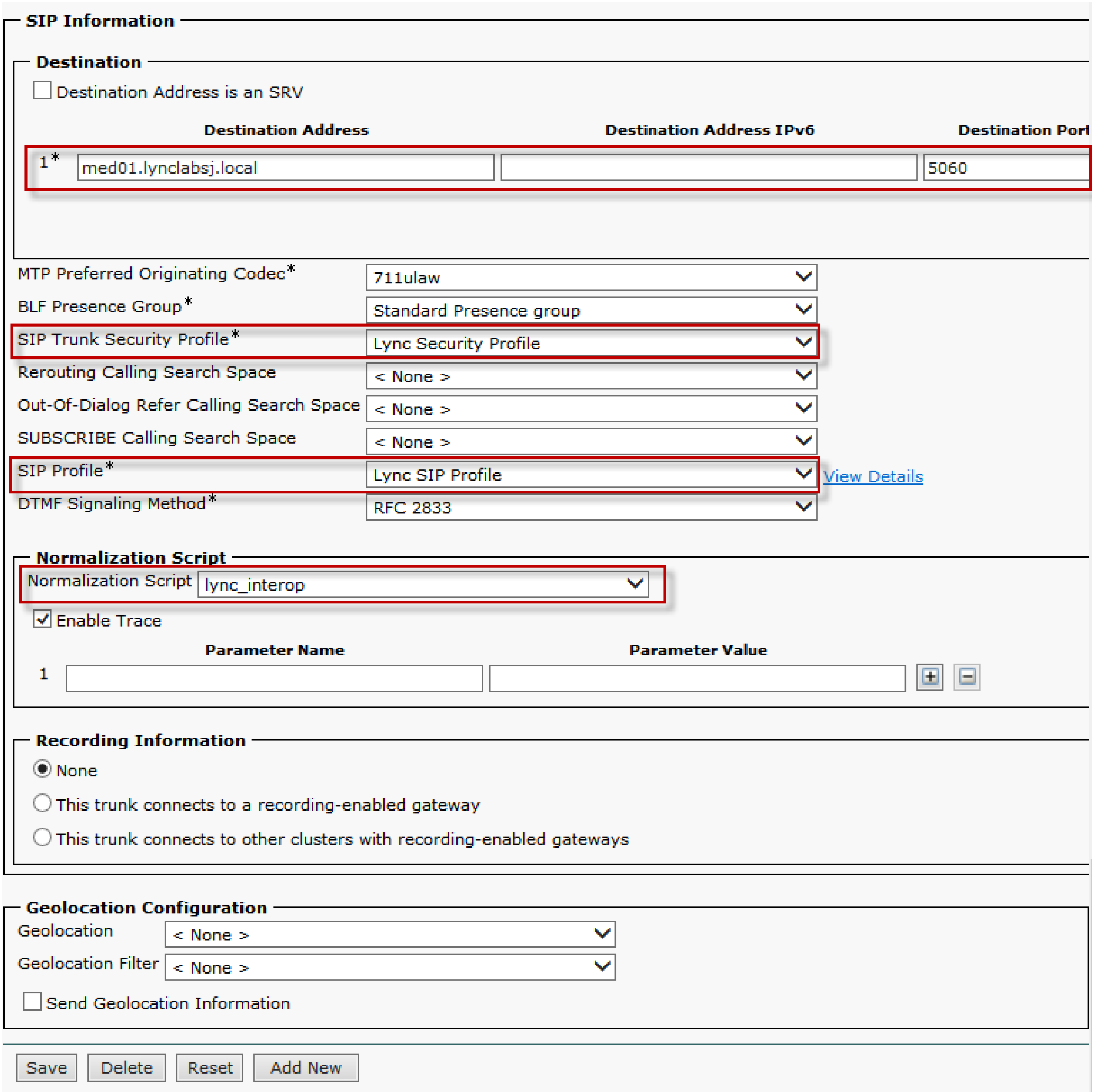
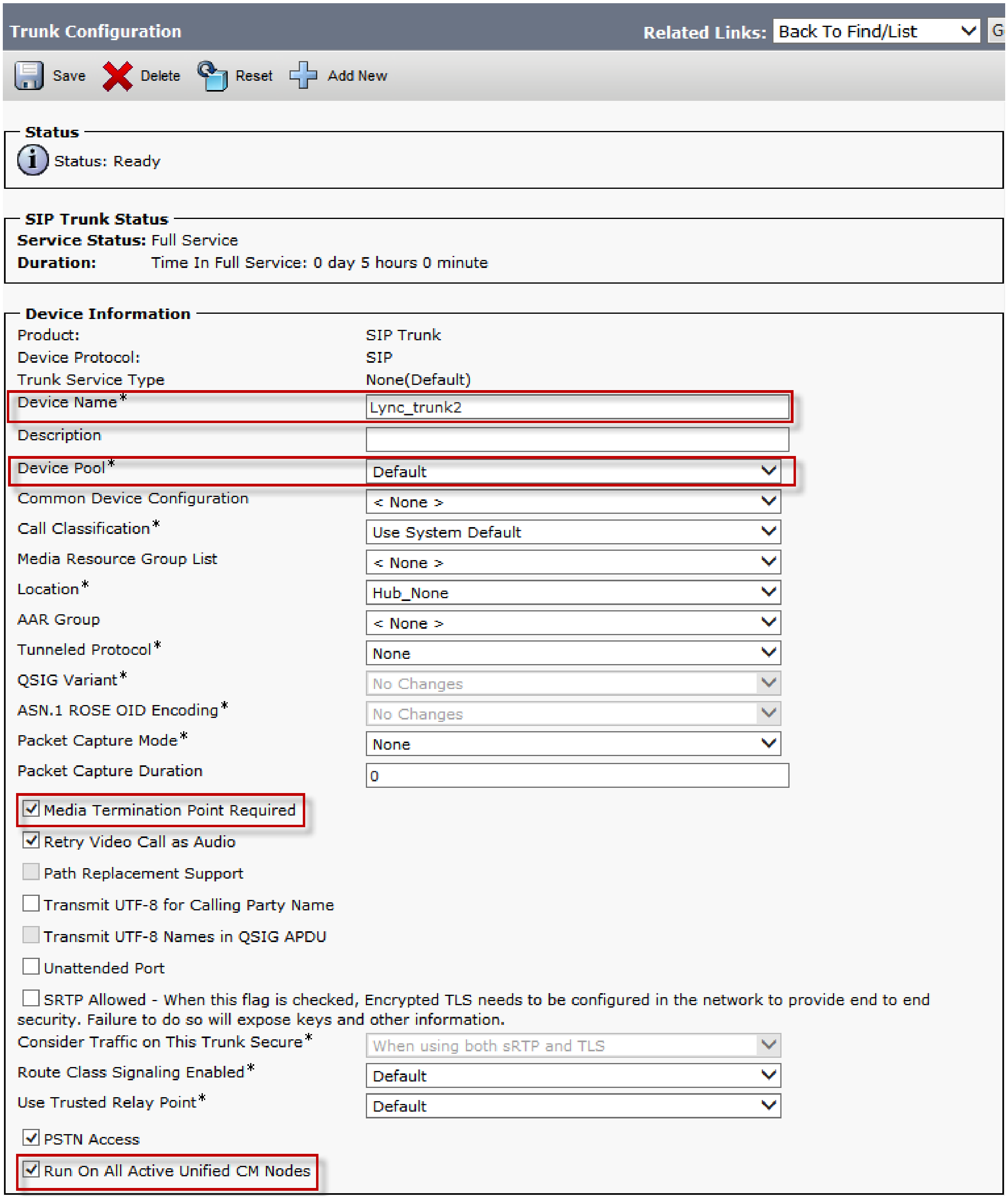


Figure 14: SIP Trunk to Lync Configuration -1_5

Trunk to Mediation Server 2 using FQDN



Trunk Configuration Related Links: Back To Find/List

Save Delete Reset Add New

Status

Status: Ready

SIP Trunk Status

Service Status: Full Service

Duration: Time In Full Service: 0 day 5 hours 0 minute

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	Lync_trunk2
Description	
Device Pool*	Default
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	< None >
Location*	Hub_None
AAR Group	< None >
Tunneled Protocol*	None
QSIG Variant*	No Changes
ASN.1 ROSE OID Encoding*	No Changes
Packet Capture Mode*	None
Packet Capture Duration	0

- [x] Media Termination Point Required
- [x] Retry Video Call as Audio
- [ ] Path Replacement Support
- [ ] Transmit UTF-8 for Calling Party Name
- [ ] Transmit UTF-8 Names in QSIG APDU
- [ ] Unattended Port
- [ ] SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.

Consider Traffic on This Trunk Secure*	When using both sRTP and TLS
Route Class Signaling Enabled*	Default
Use Trusted Relay Point*	Default

- [x] PSTN Access
- [x] Run On All Active Unified CM Nodes

Figure 15: SIP Trunk to Lync Configuration -2_1

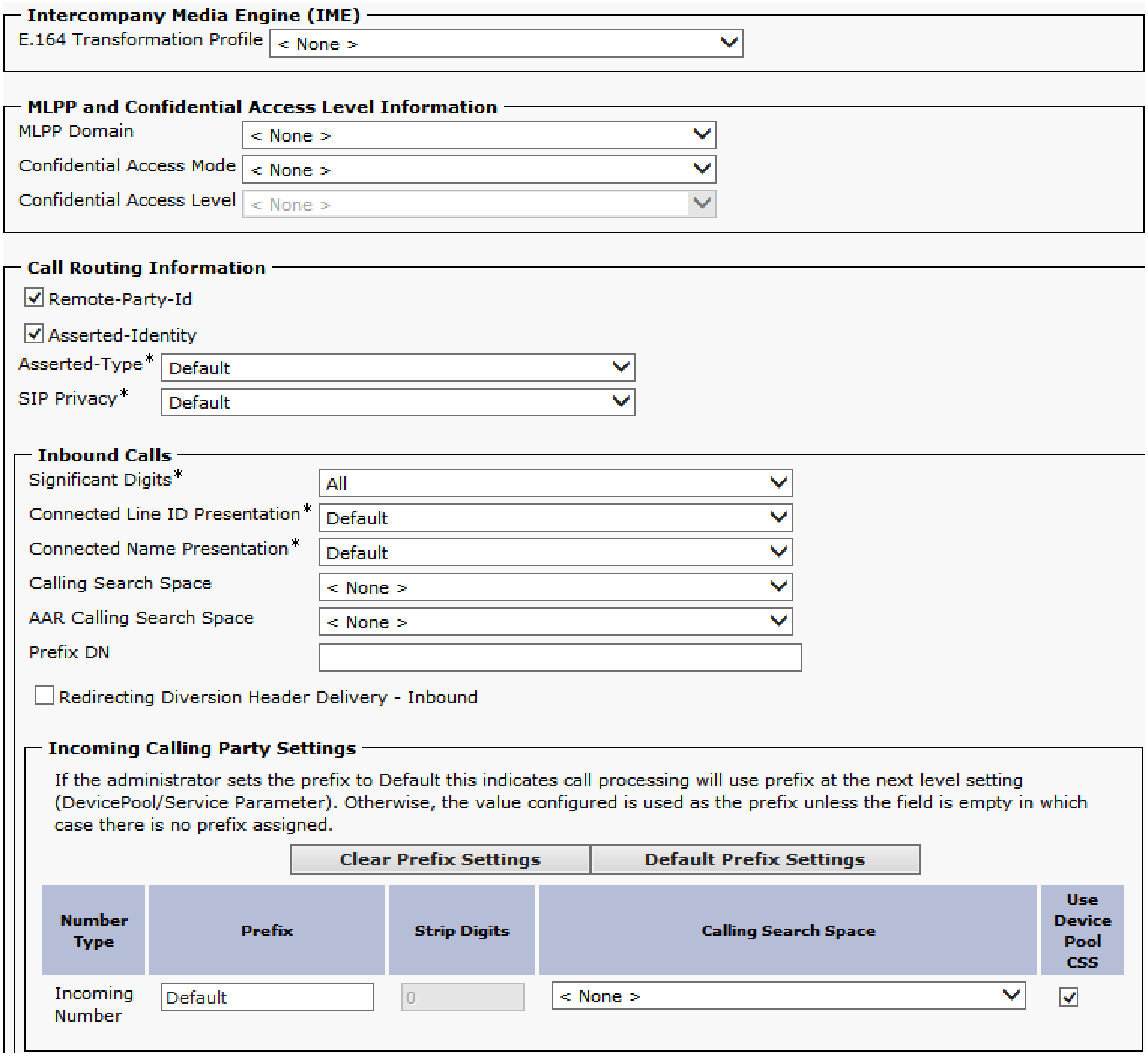


Figure 16: SIP Trunk to Lync Configuration -2_2

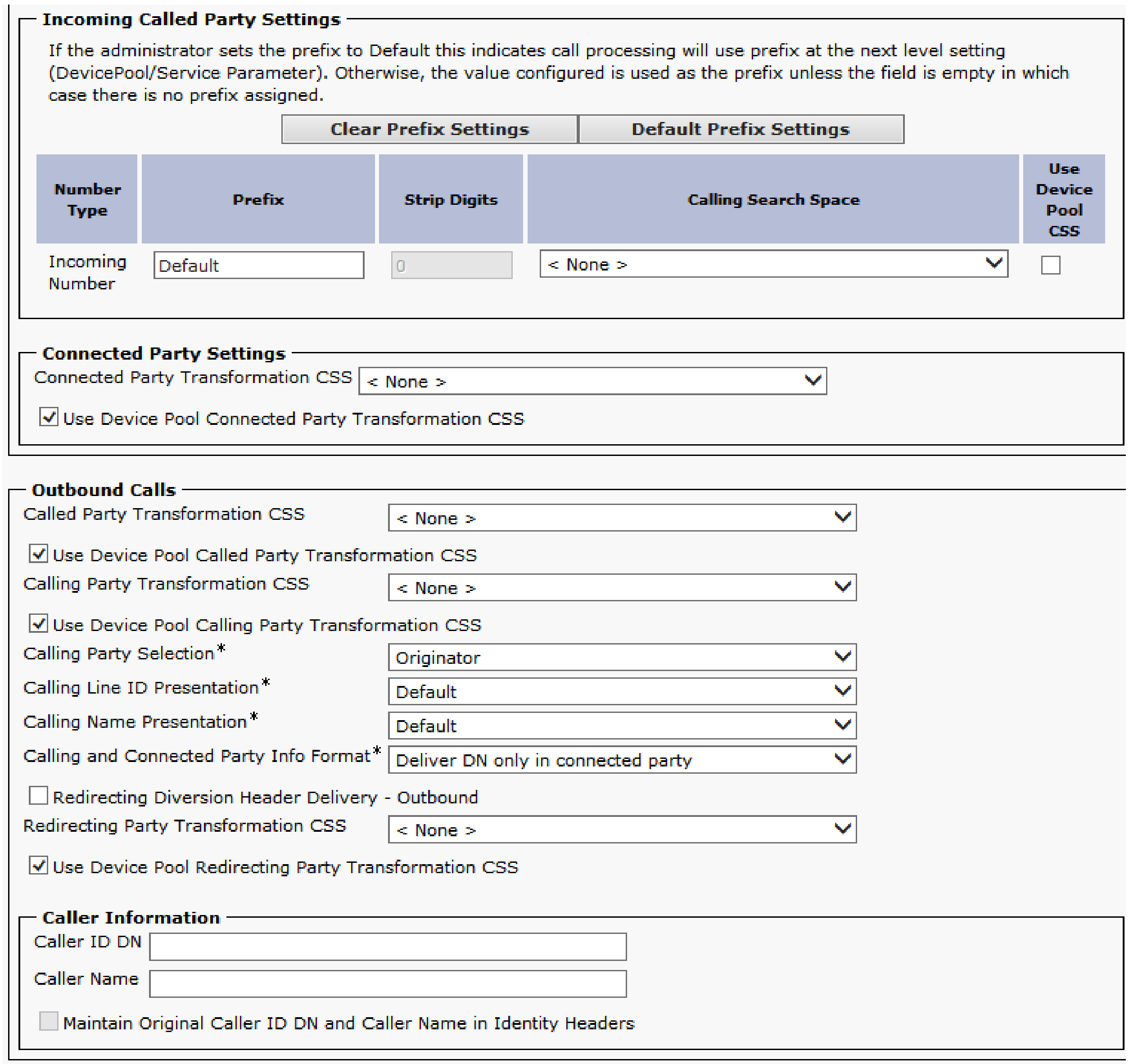


Figure 17: SIP Trunk to Lync Configuration -2_3

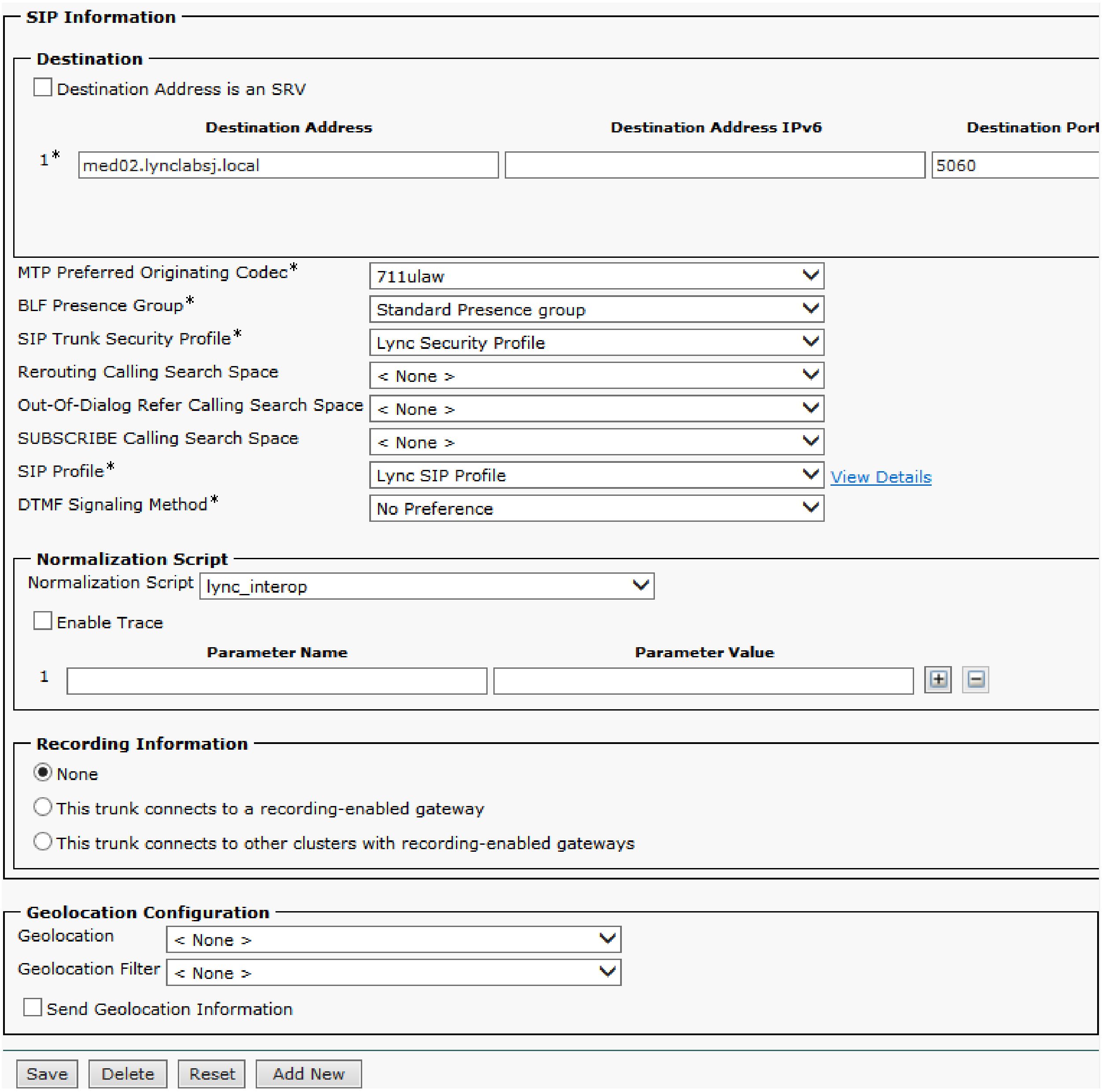


Figure 18: SIP Trunk to Lync Configuration -2_4

Trunk to Mediation Server 1 using IP Address

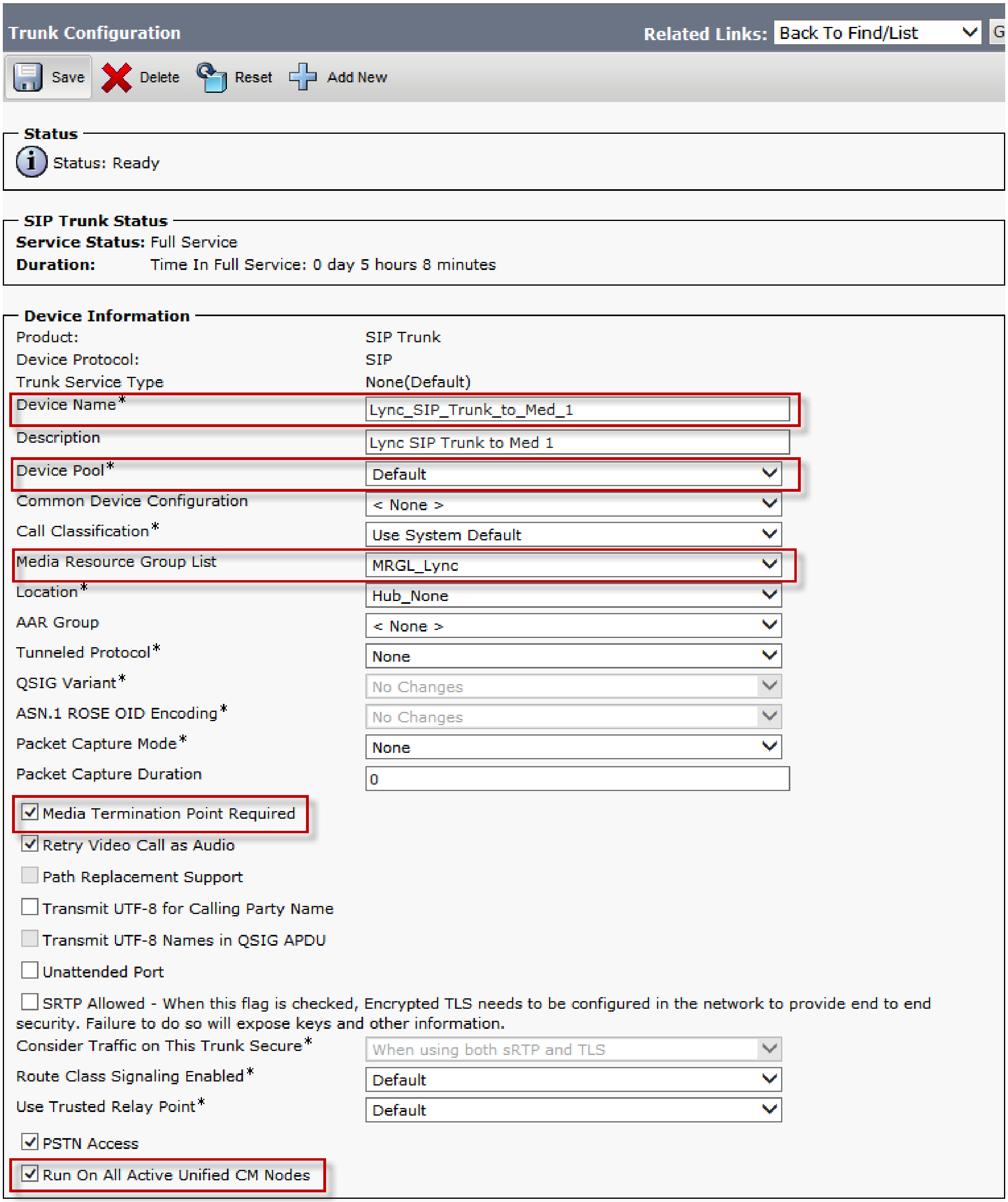


Figure 19: SIP Trunk to Lync Configuration -3_1

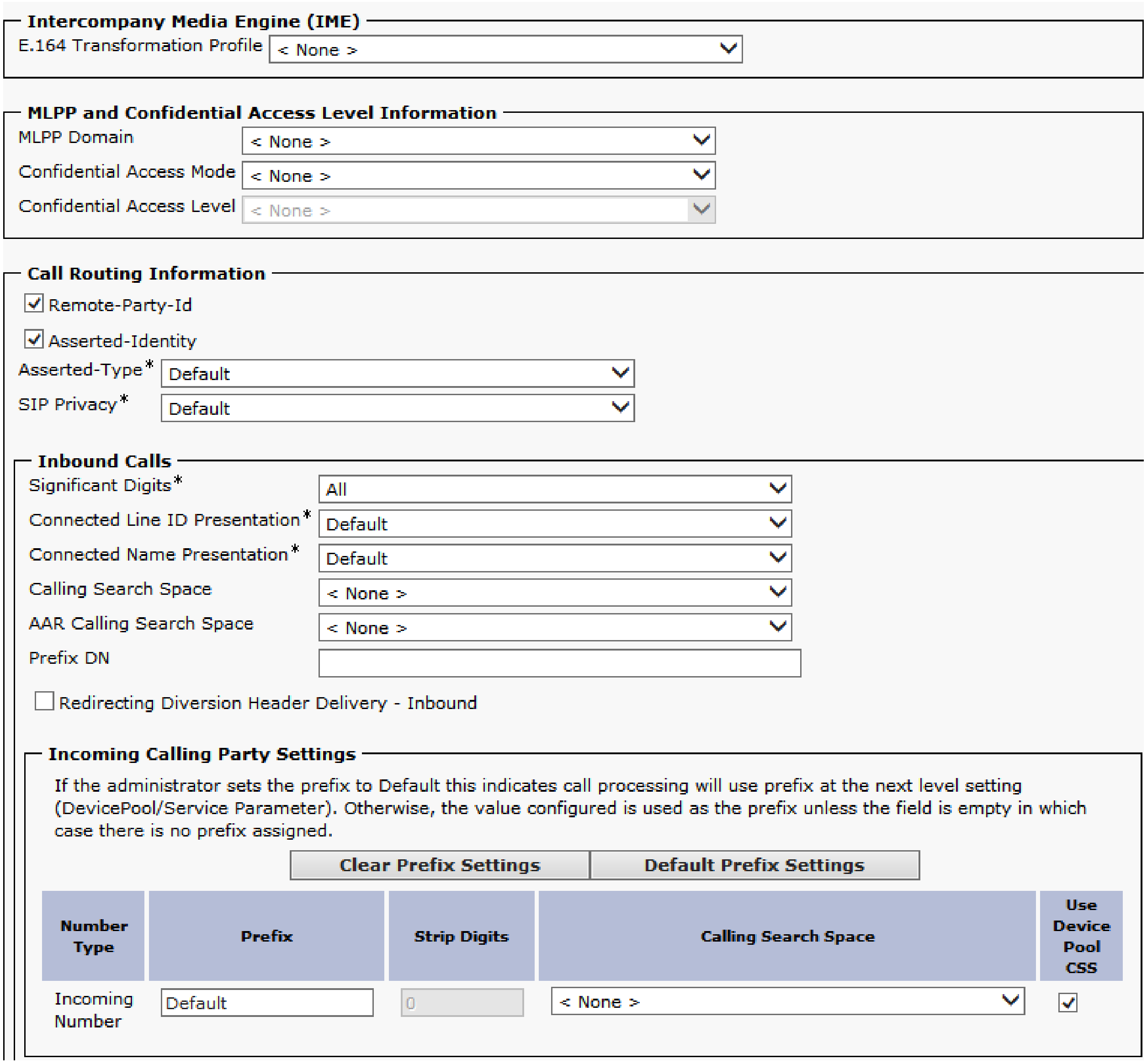


Figure 20: SIP Trunk to Lync Configuration -3_2

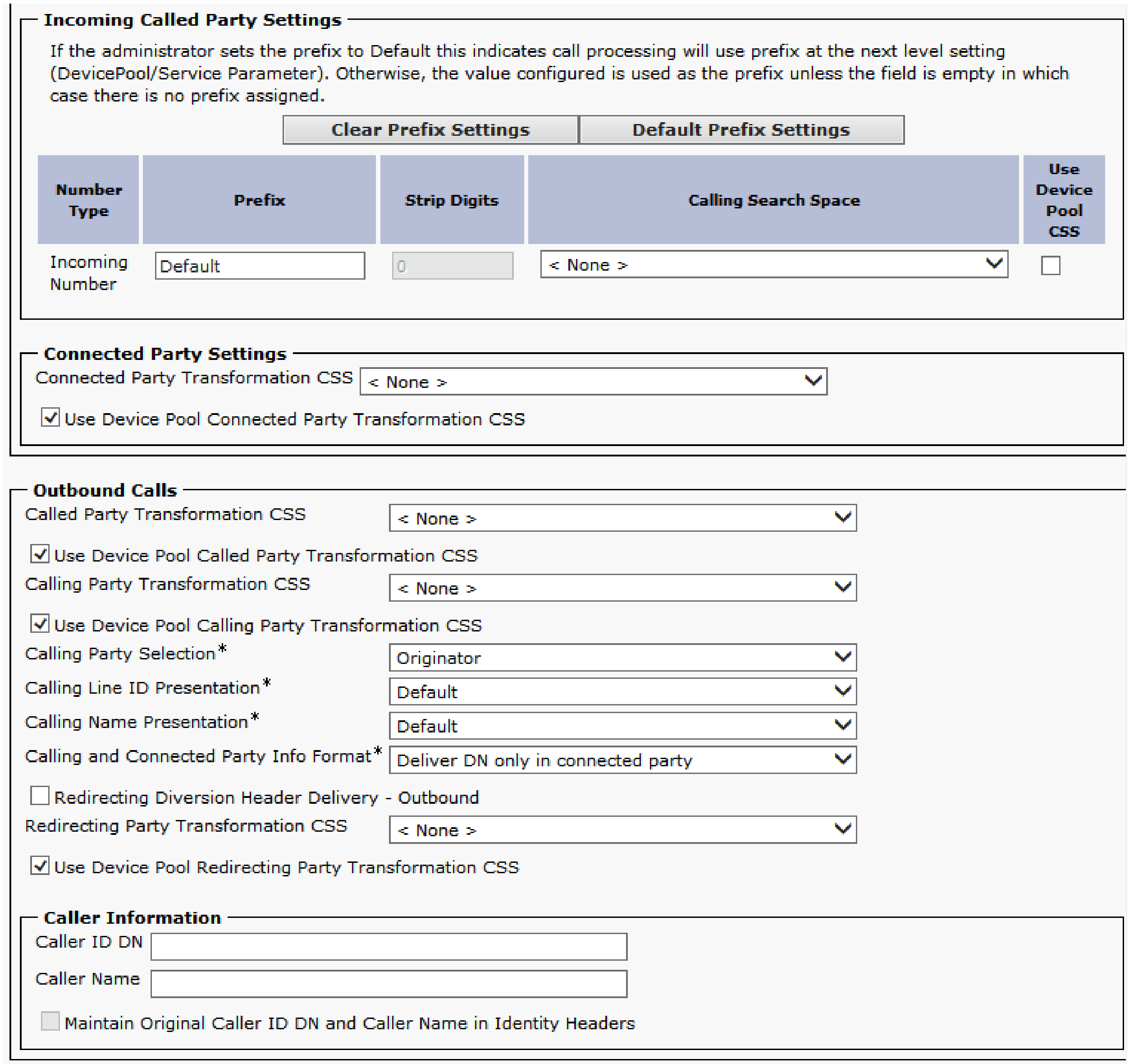


Figure 21: SIP Trunk to Lync Configuration -3_3

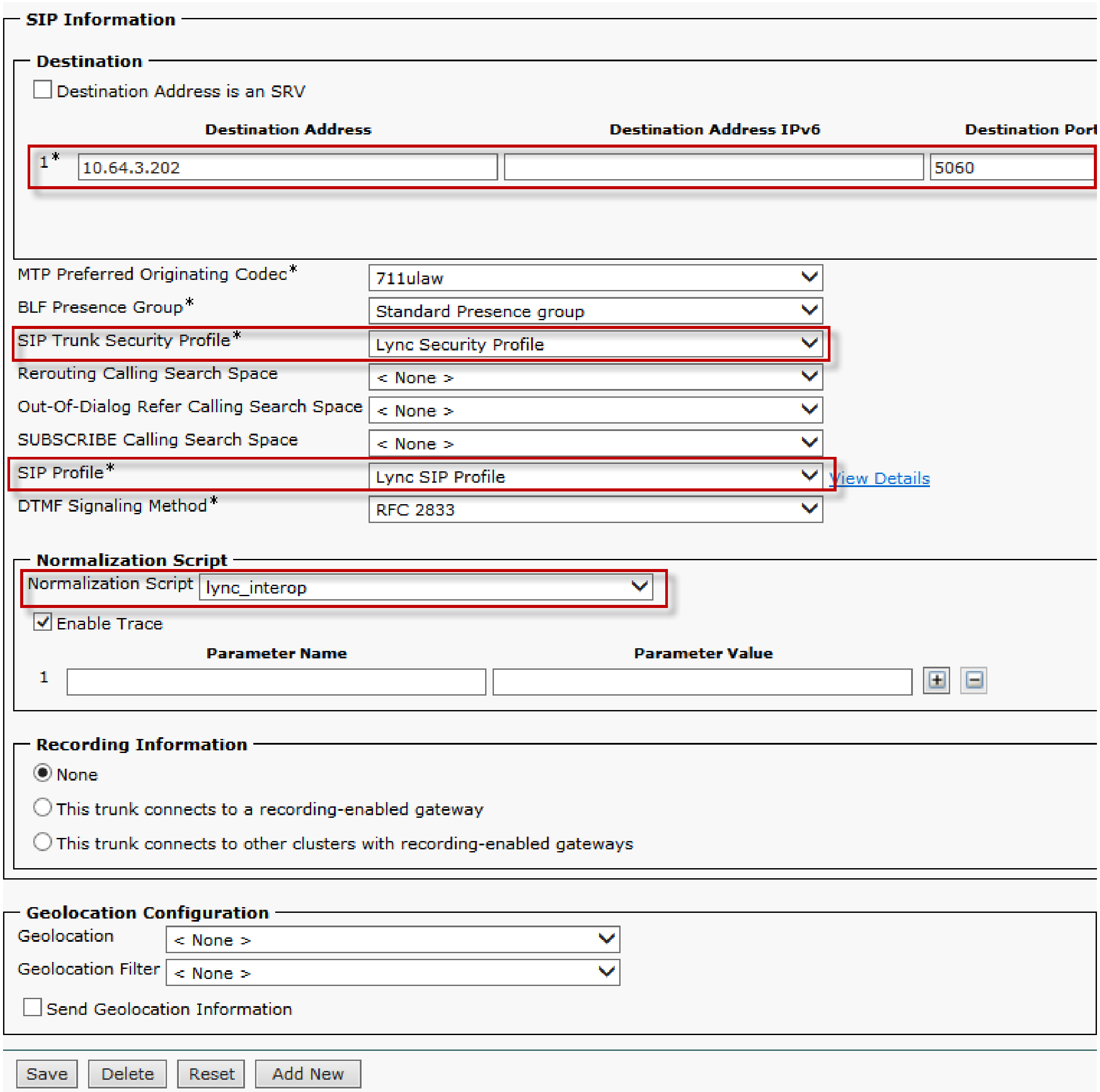


Figure 22: SIP Trunk to Lync Configuration -3_4

Trunk to Mediation Server 2 using IP Address

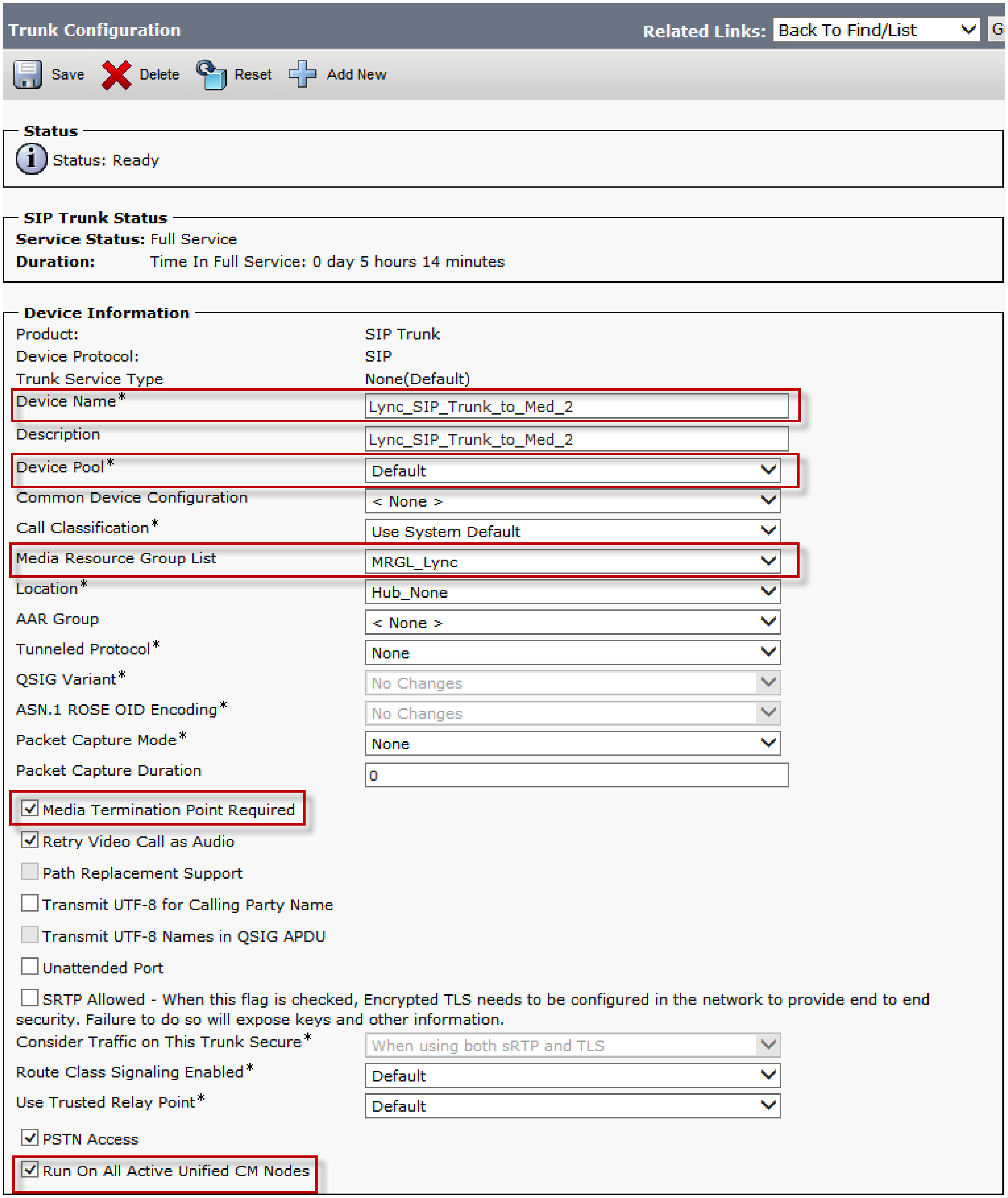


Figure 23: SIP Trunk to Lync Configuration -4_1

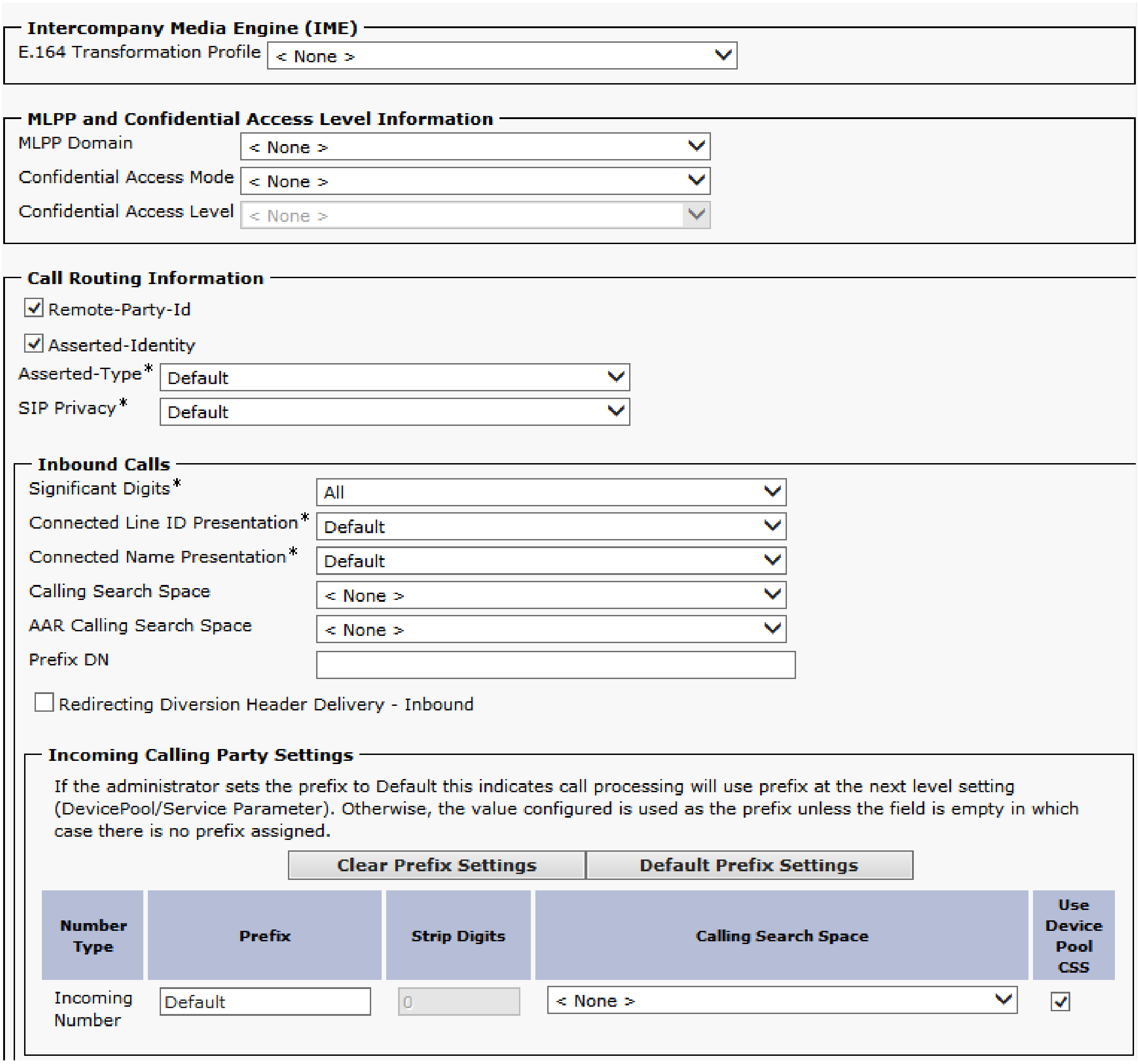


Figure 24: SIP Trunk to Lync Configuration -4_2

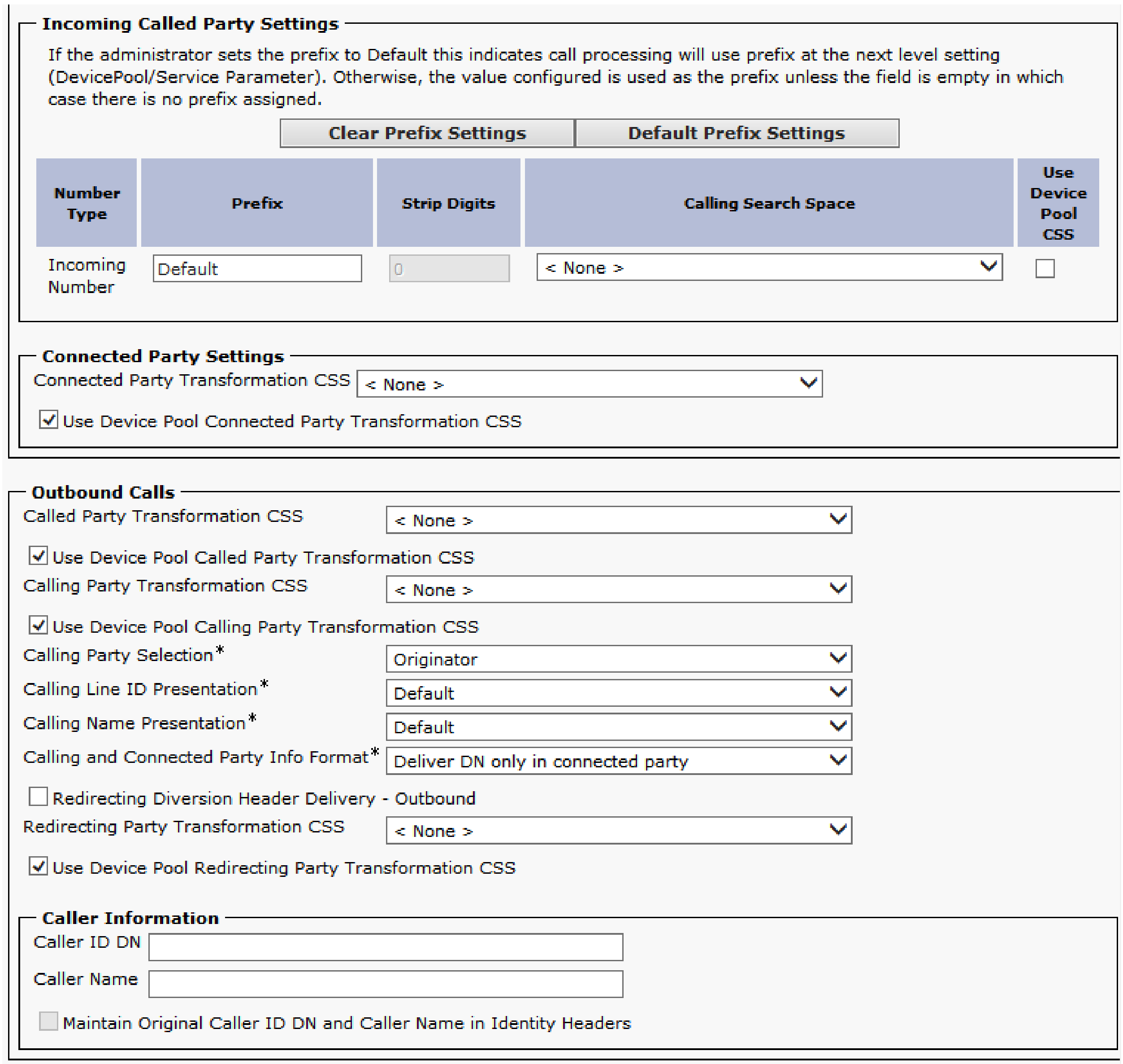


Figure 25: SIP Trunk to Lync Configuration -4_3

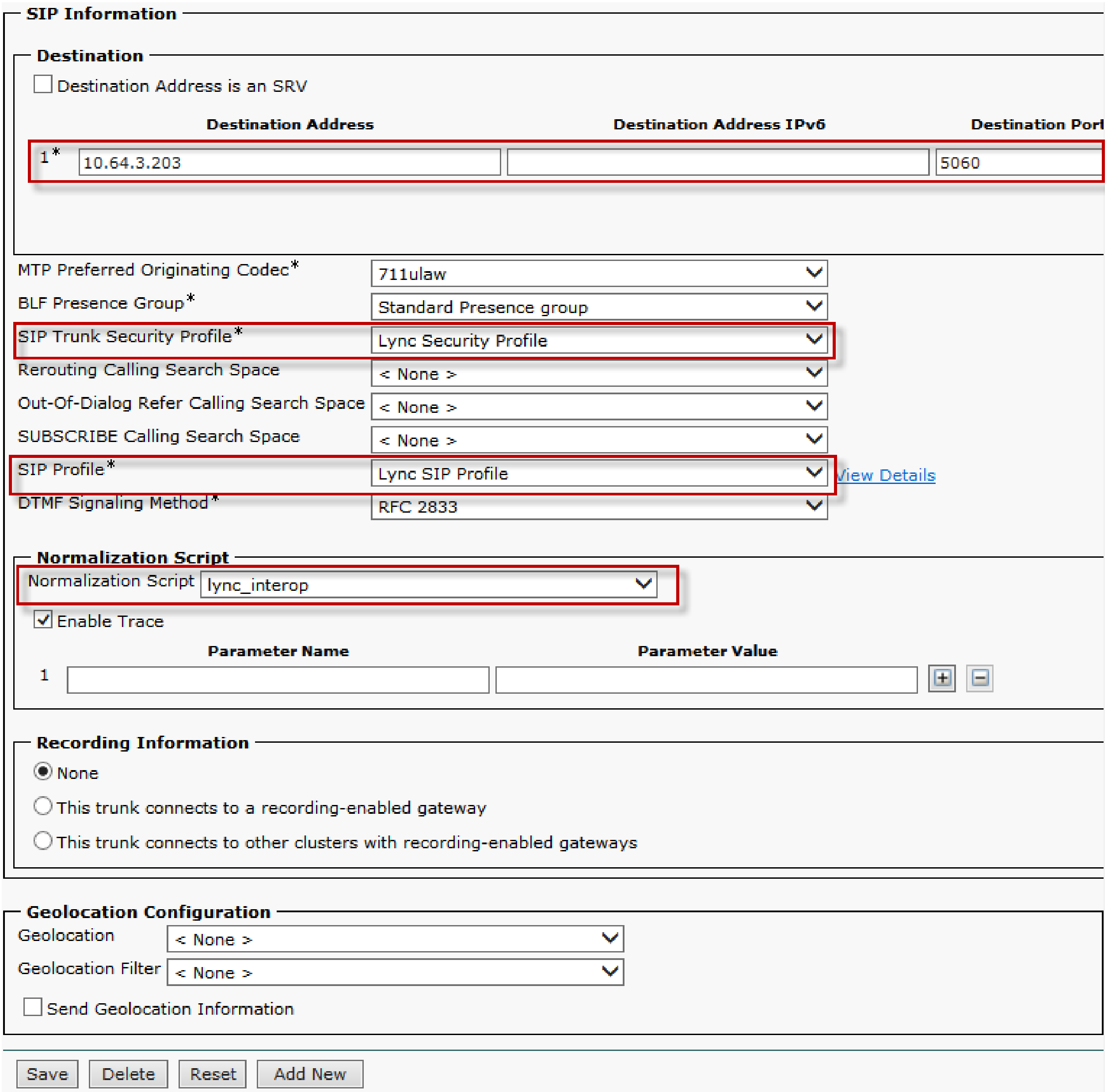


Figure 26: SIP Trunk to Lync Configuration -4_4

Trunk to Cisco Unity Voice Mail Server

Navigation: Device -> Trunk

The following procedure describes the trunk configuration from Unified Communications Manager to Cisco Unity Voice Mail Server

Device Information

1. Set **Trunk Type**: SIP Trunk
2. Set **Device Protocol**: SIP
3. Set **Trunk Service Type**: None
4. Set **Device Name**: Enter a name for the trunk
5. Set **Description:** Enter a description relevant to your trunk
6. Set **Device Pool**: Default
   For trunks, device pools specify a list of Cisco Unified Communications Managers that the trunk uses to distribute the call load dynamically
7. Confirm **Media Termination Point Required**: is checked
   This check box is used to indicate whether a media termination point (MTP) is used to implement features that H.323 does not support (such as hold and transfer).
8. Confirm **Retry Video Calls as Audio**: is checked

SIP Information

9. Set the **Destination Address**: Enter the FQDN/IP Address of the Unity Server to which you are establishing a trunk.
10. Set **SIP trunk Security Profile**: Select the security profile you created under System -> Security -> SIP Security Profile

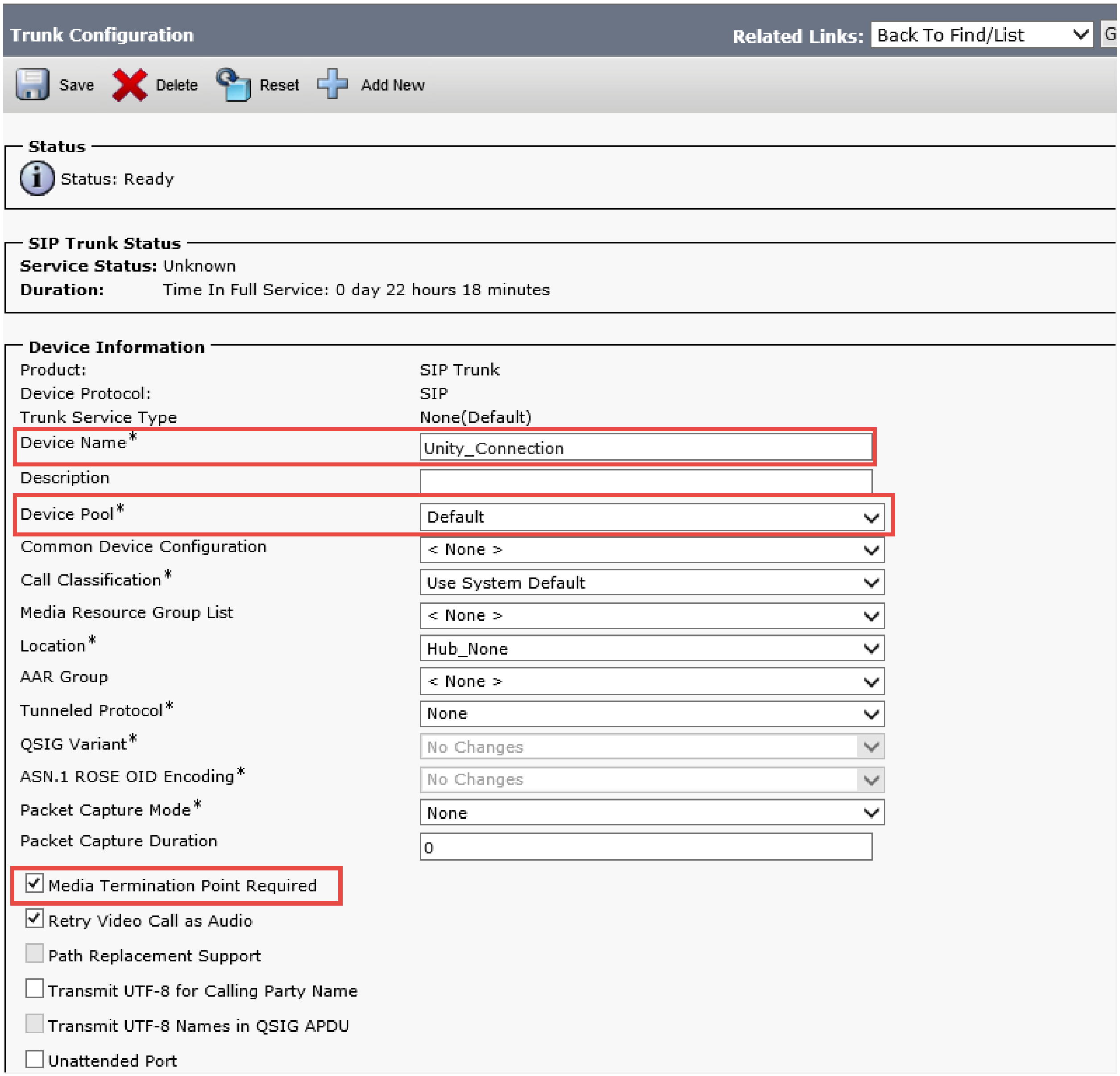


Figure 27: SIP Trunk to Cisco Unity-1

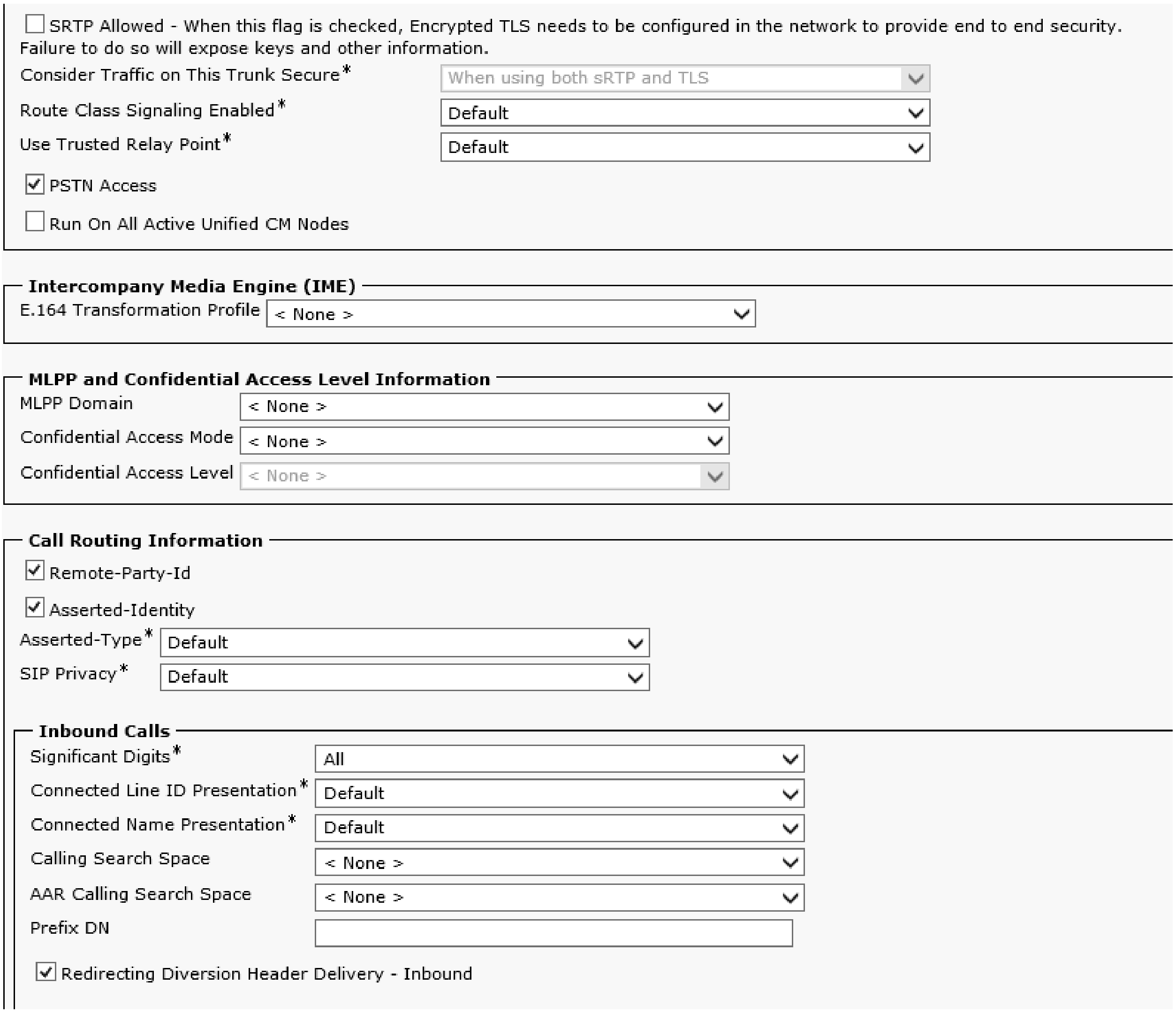


Figure 28: SIP Trunk to Cisco Unity-2

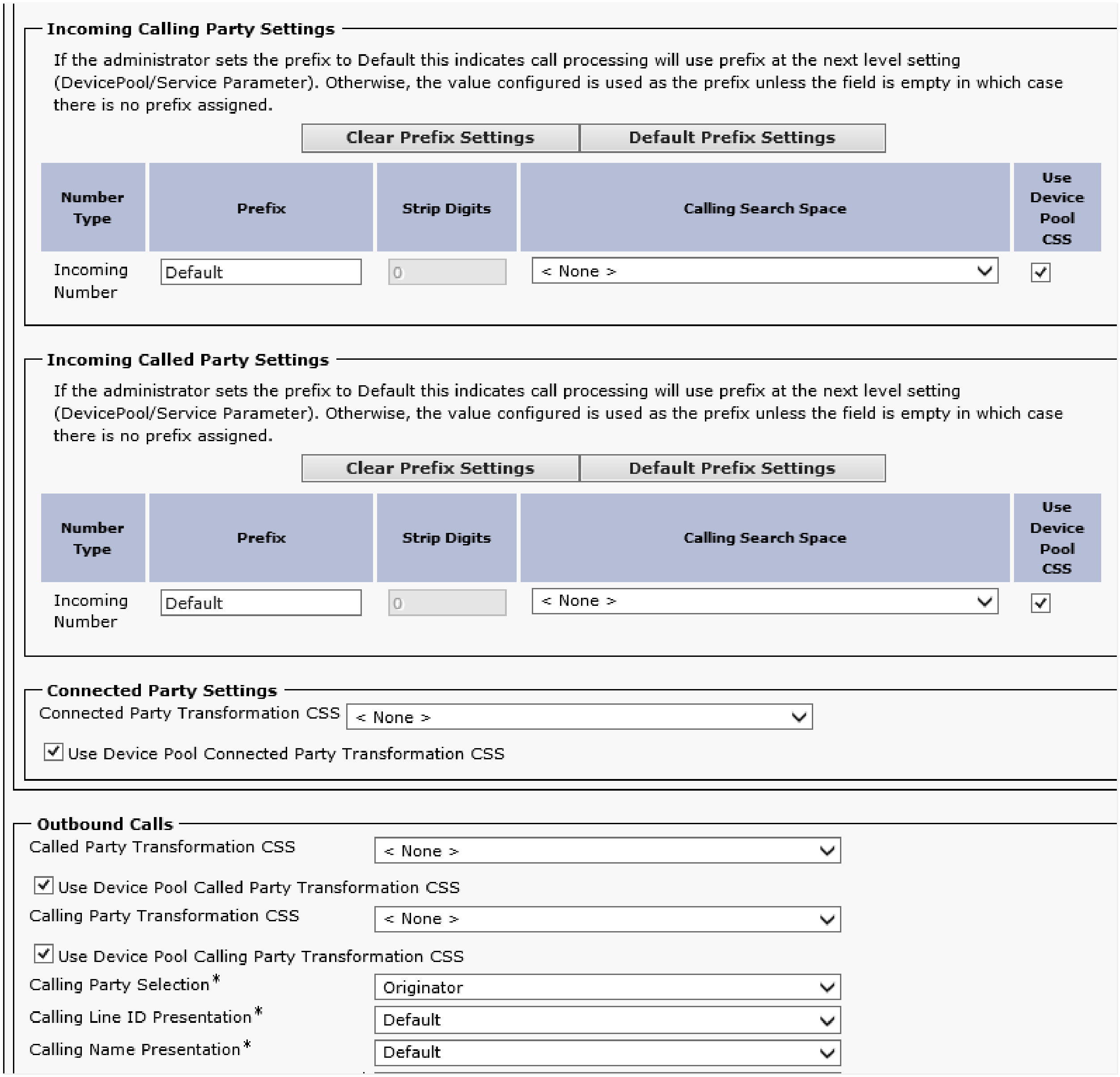


Figure 29: SIP Trunk to Cisco Unity-3

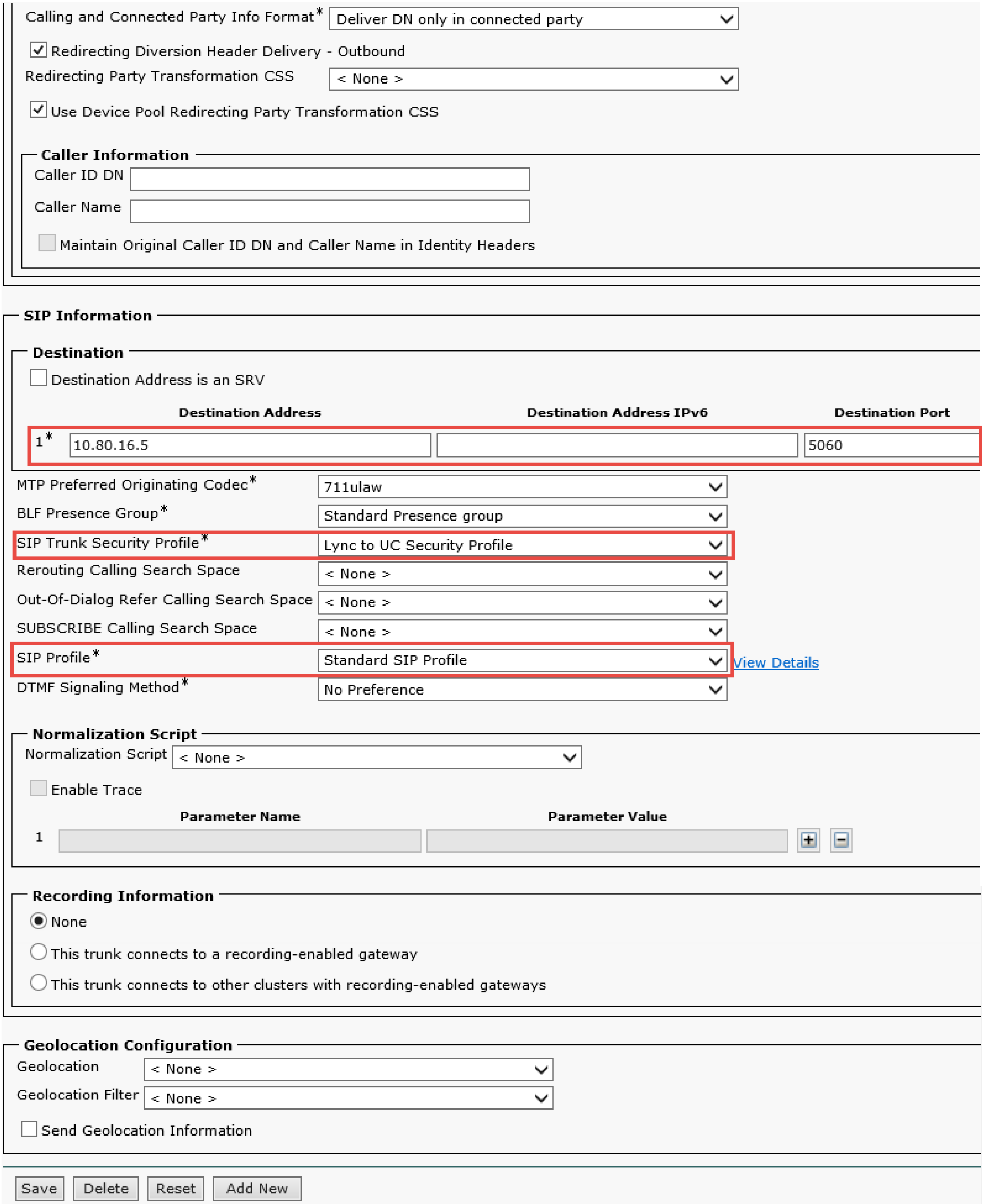


Figure 30: SIP Trunk to Cisco Unity-4

Create Normalization Script

Navigation: Device->Device Settings->Normalization Script

This normalization script is used to modify the bandwidth line during call hold, to manipulate the user=phone parameter in SIP URIs , to modify History-Info on inbound INVITEs to Diversion header, and Referred-by to Diversion header and to provide ring back at the call originator when PRACK is enabled on trunk. Below is snap shot of Script configuration.

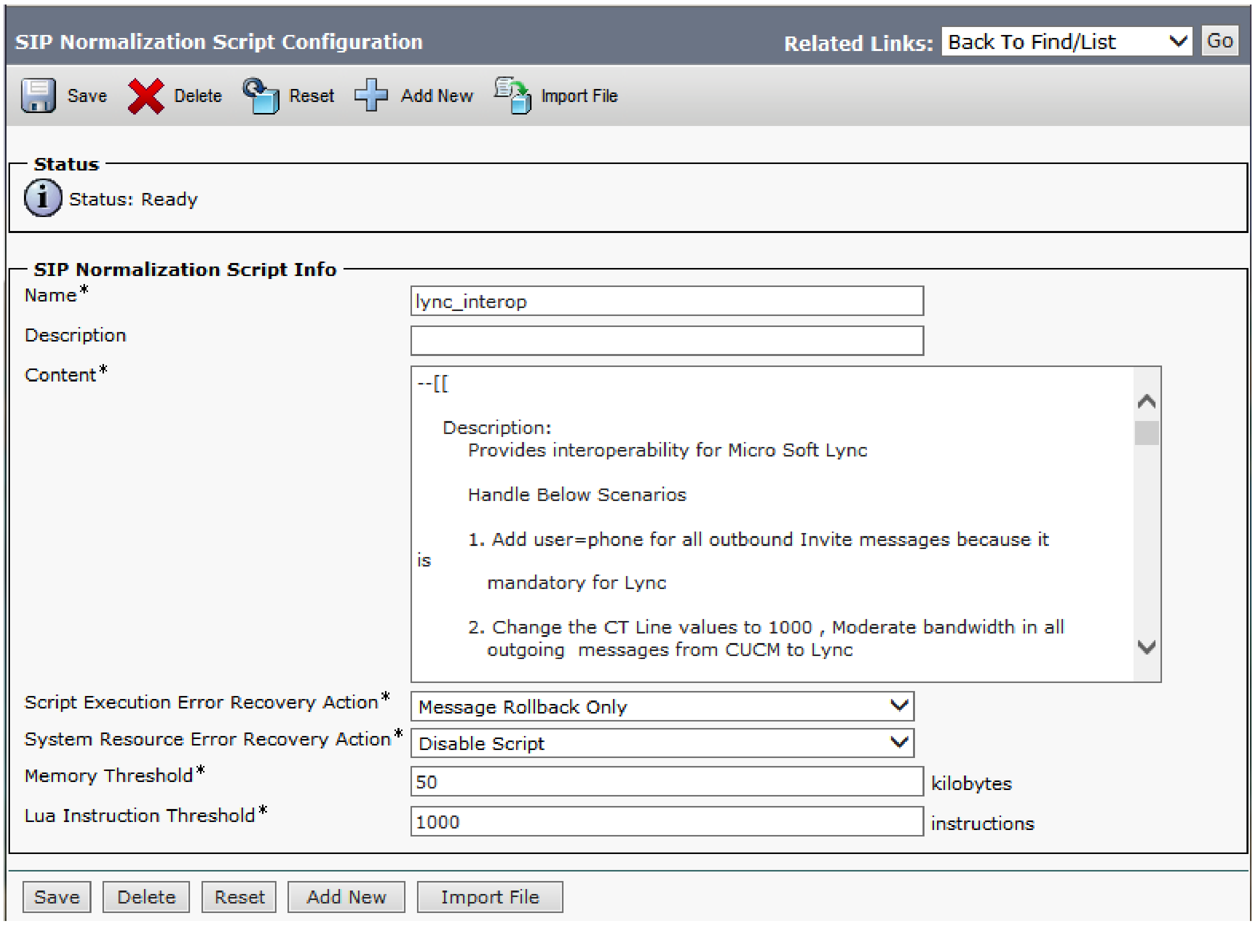


Figure 31: Normalization Script

CISCO UCM Normalization Script

Download the script “SIP normalization script (version 1.2) for audio interoperability between Microsoft Lync 2013 and Cisco Unified Communications Manager” (lync_interop_lua.zip) at Downloads Home > Products > Unified Communications > Call Control > Cisco Unified Communications Manager (CallManager) > Cisco Unified Communications Manager Version 10.0 > SIP Normalization and Transparency Scripts-Scripts:

http://software.cisco.com/download/release.html?i=!y&mdfid=284603137&softwareid=284695022&release=Scripts&os=

Translation Pattern

Navigation: Call Routing-> Translation Pattern Configuration

1. Set **Translation Pattern**: Enter the ten digit number pattern to be translated
2. Set **Called Party Transform Mask**: Enter the four digit number pattern to be translated to, these will be the Cisco Phone extension pattern.

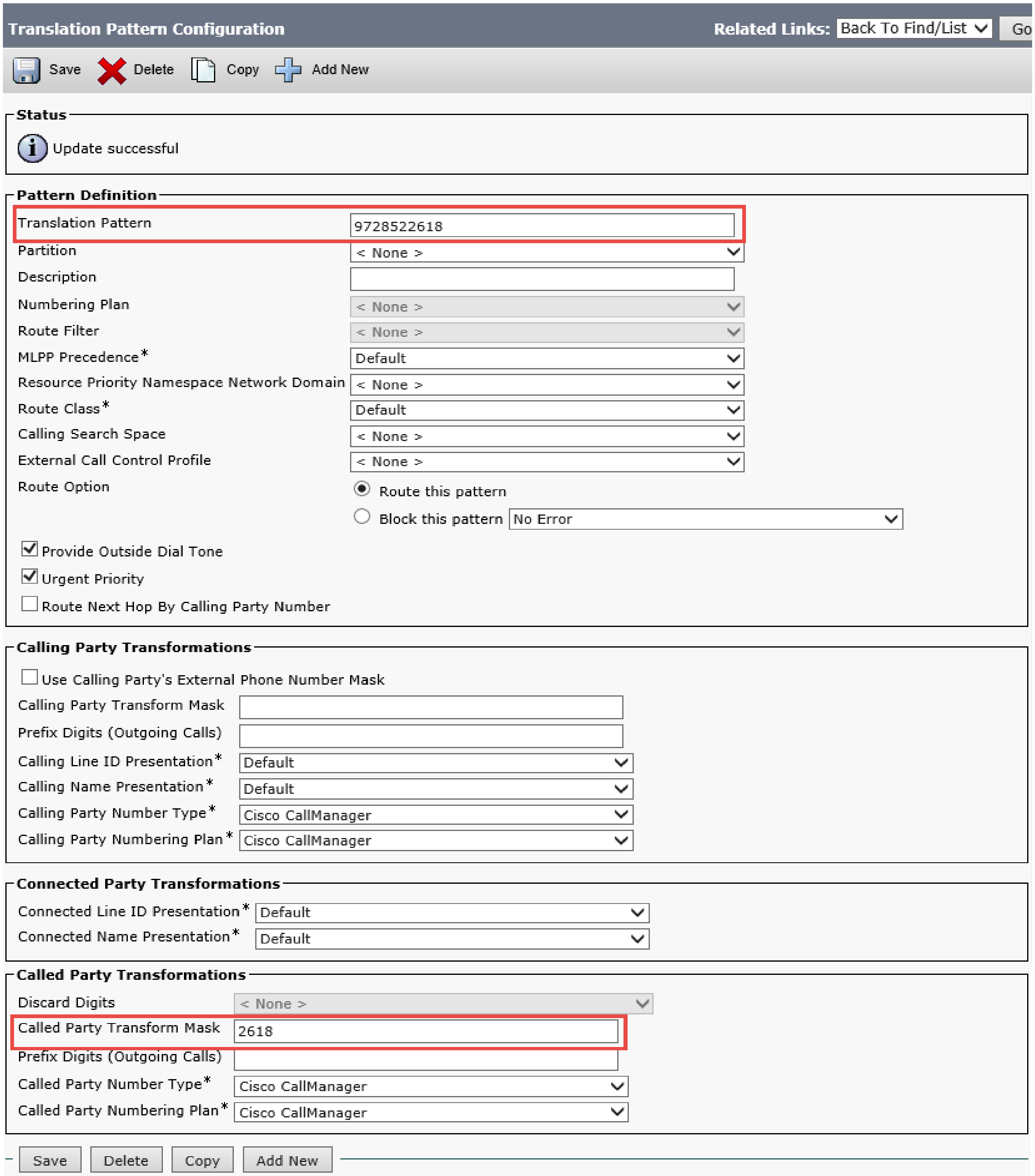


Figure 32: Translation Pattern

Route Group

Navigation: Call Routing -> Route/Hunt -> Route Group

1. Set **Route Group Name:** Enter a name for the route group
2. Set **Distribution Algorithm:** Select the preferred distribution algorithm from the available list

3. Select the available devices and them to the route group, once selected they should be available in the **Current Route Group Members**

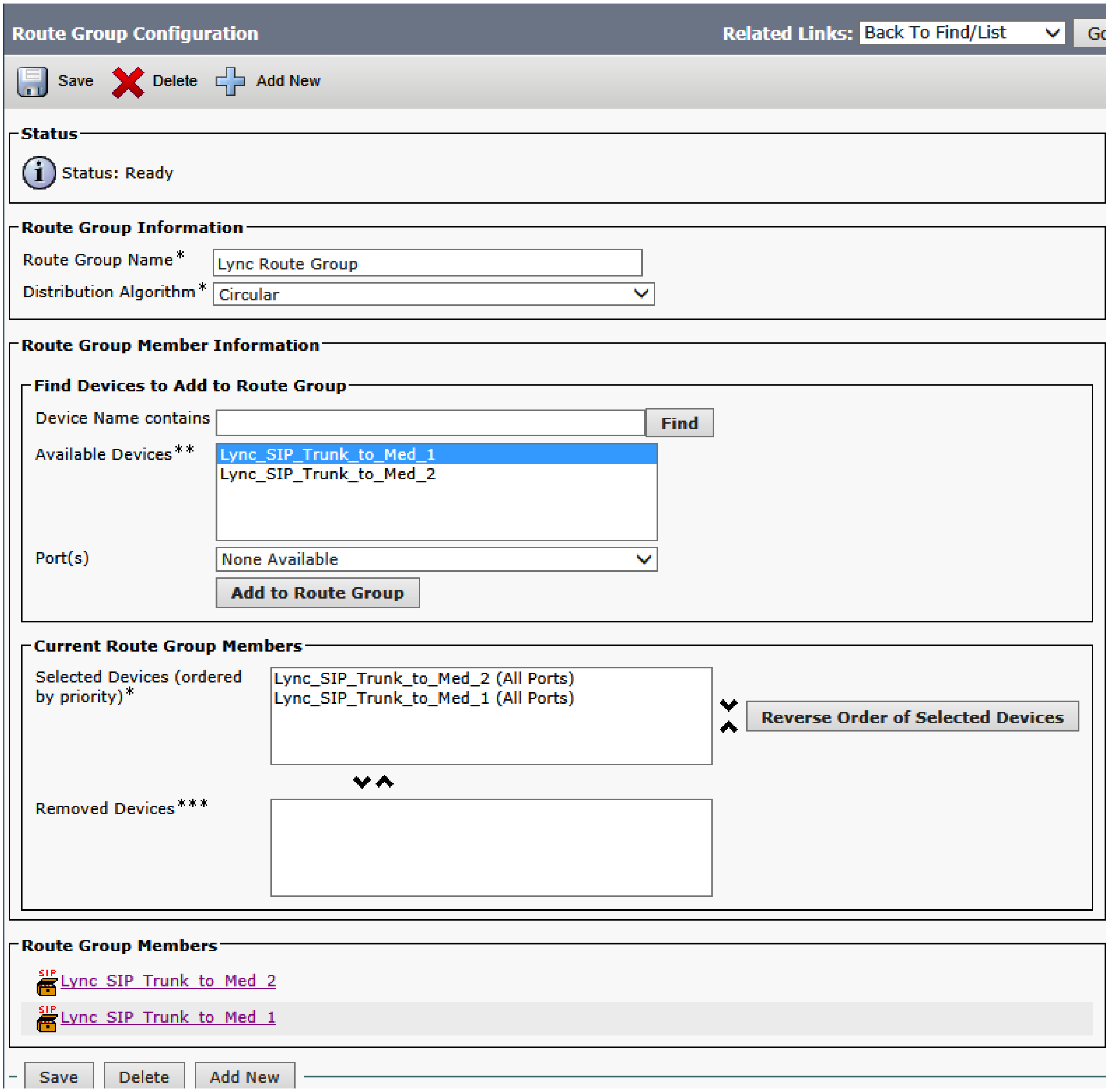


Figure 33: Route Group

Route List

Navigation: Call Routing -> Route/Hunt -> Route List

1. Set **Name:** Enter a name for the route list

2. Set **Cisco Unified Communications Manager Group:** Default
3. Add Route Group: Under Route List Member Information, click the **Add Route Group** and add the route group configured in the previous section.

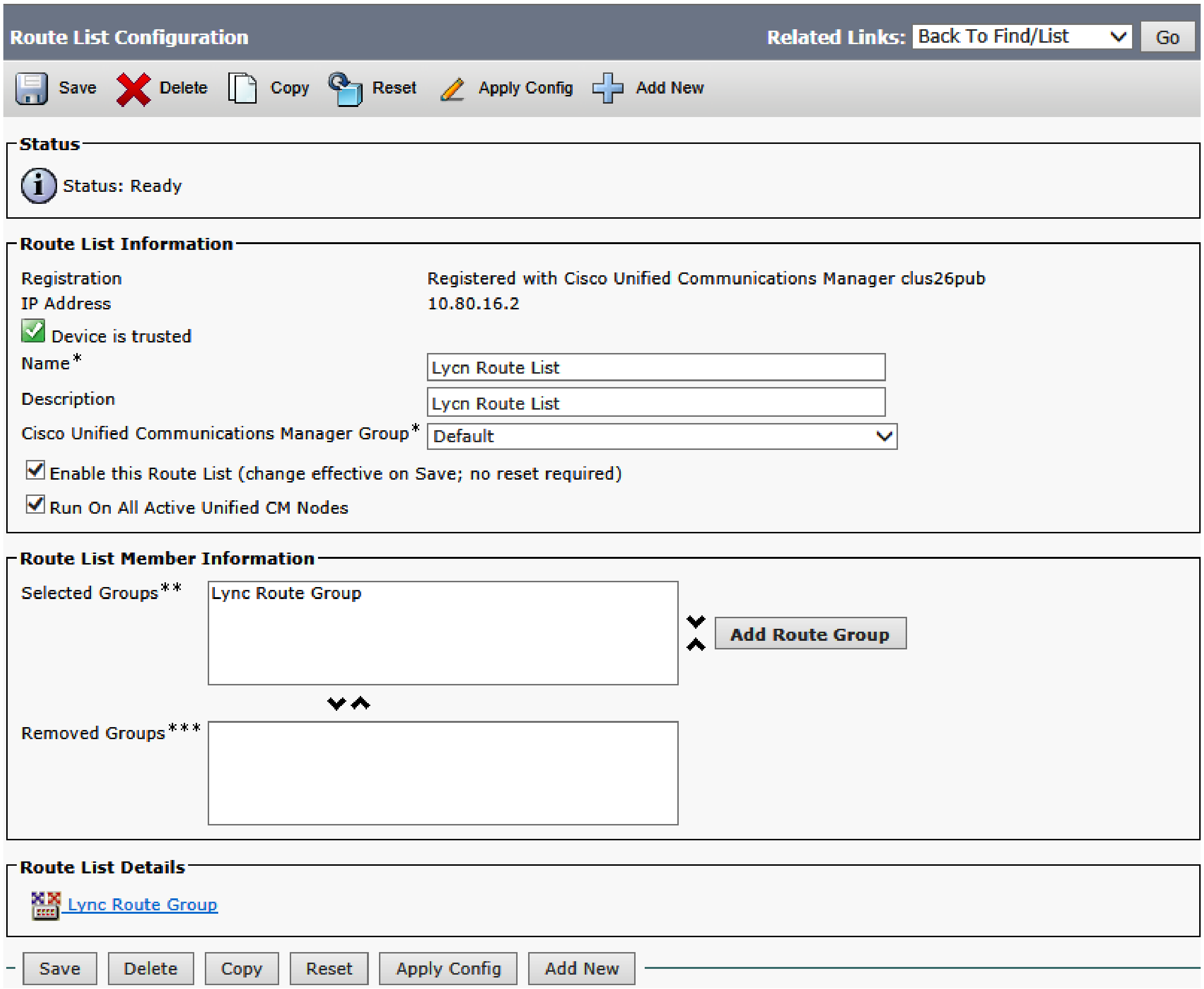


Figure 34: Route List

Route Pattern Configuration

Route Pattern to Lync

Navigation: Call Routing -> Route/Hunt -> Route Pattern

Pattern Definition

1. Set **Route Pattern**: Enter the routing pattern
2. Set **Gateway/Route List**: Select the Route List you have created under Call Routing -> Route/Hunt -> Route List

Calling Party Transformations

3. Set **Prefix Digits (Outgoing Calls)**: +1

Called Party Transformations

4. Set **Prefix Digits(Outgoing Calls)**: +1

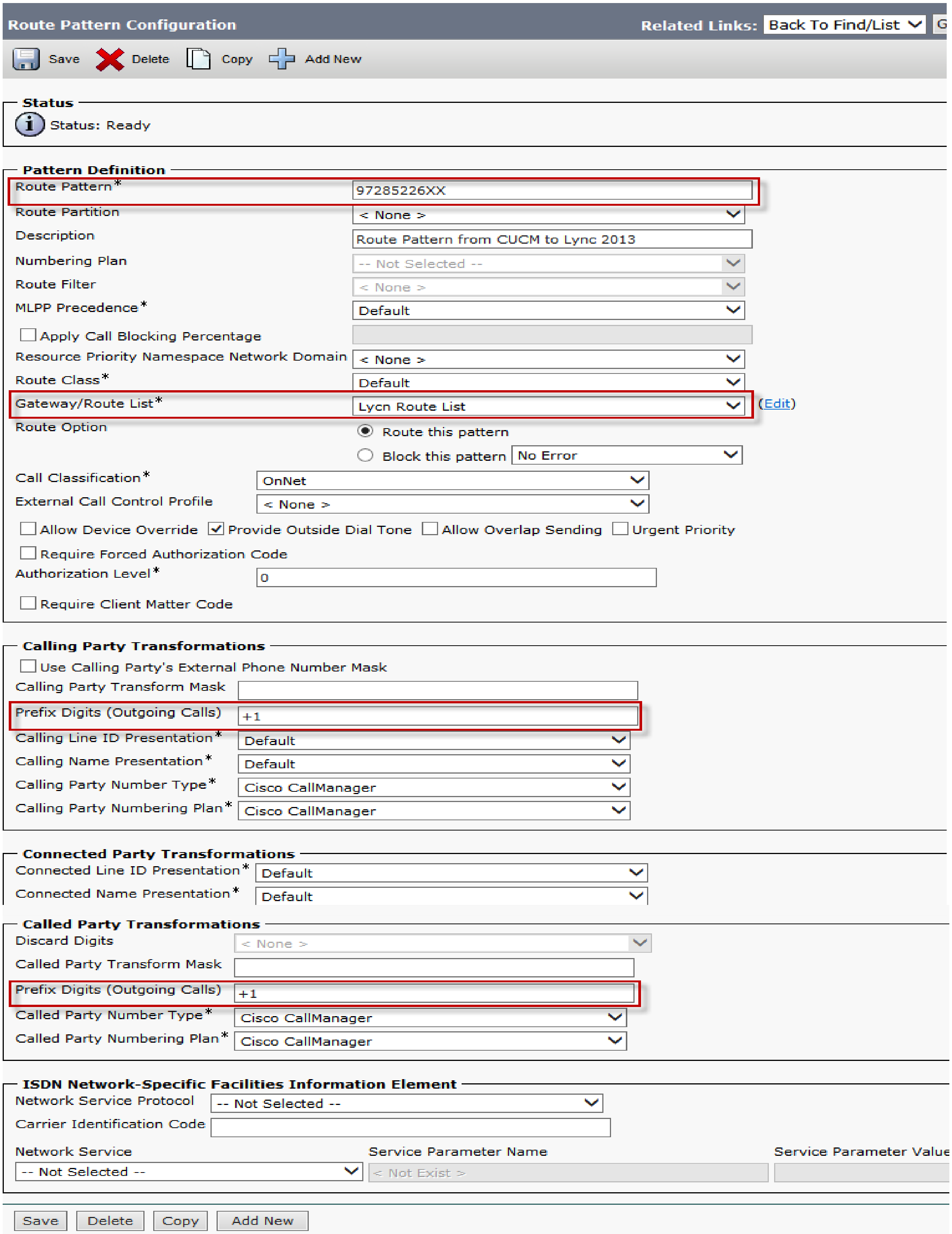
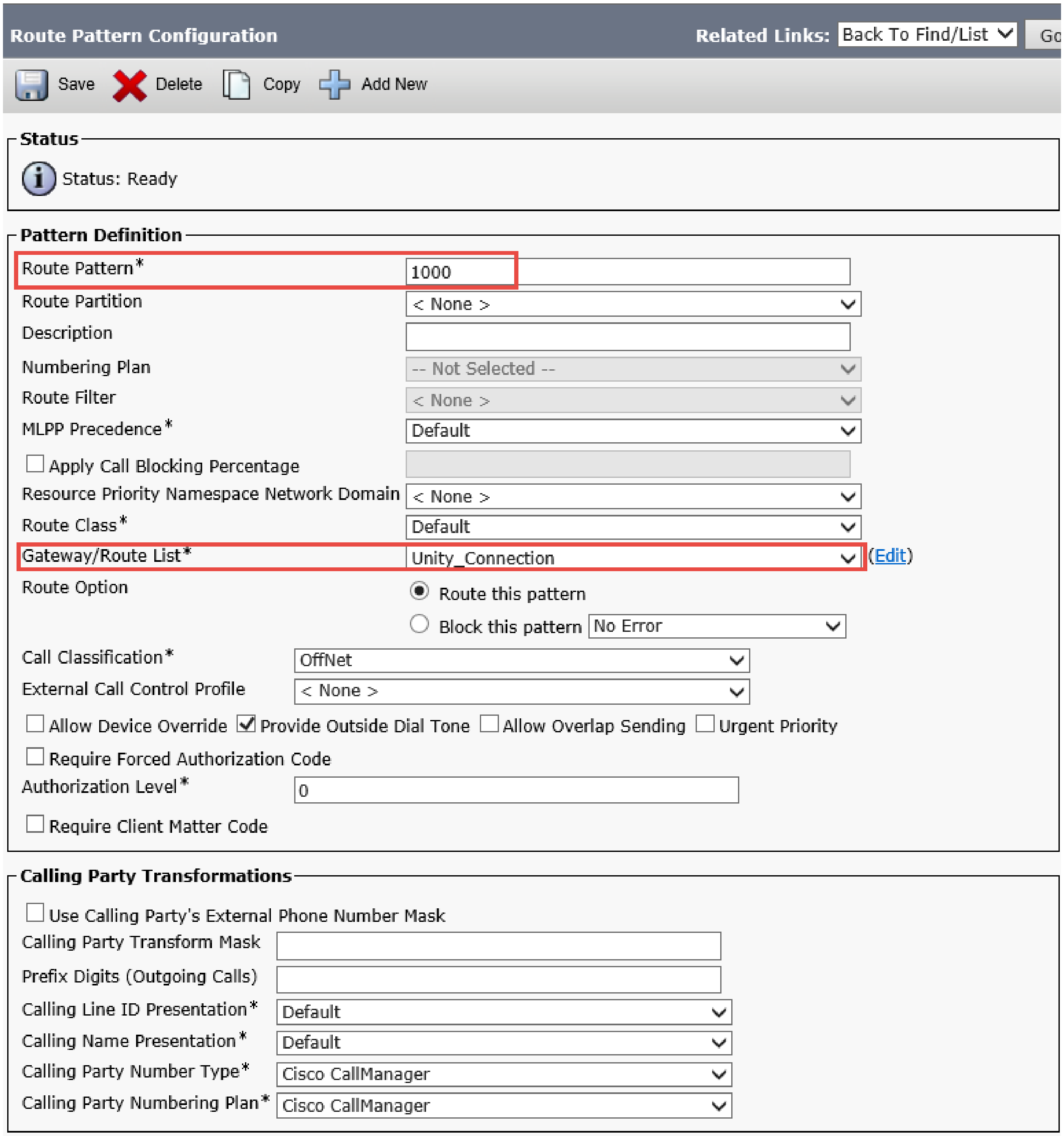


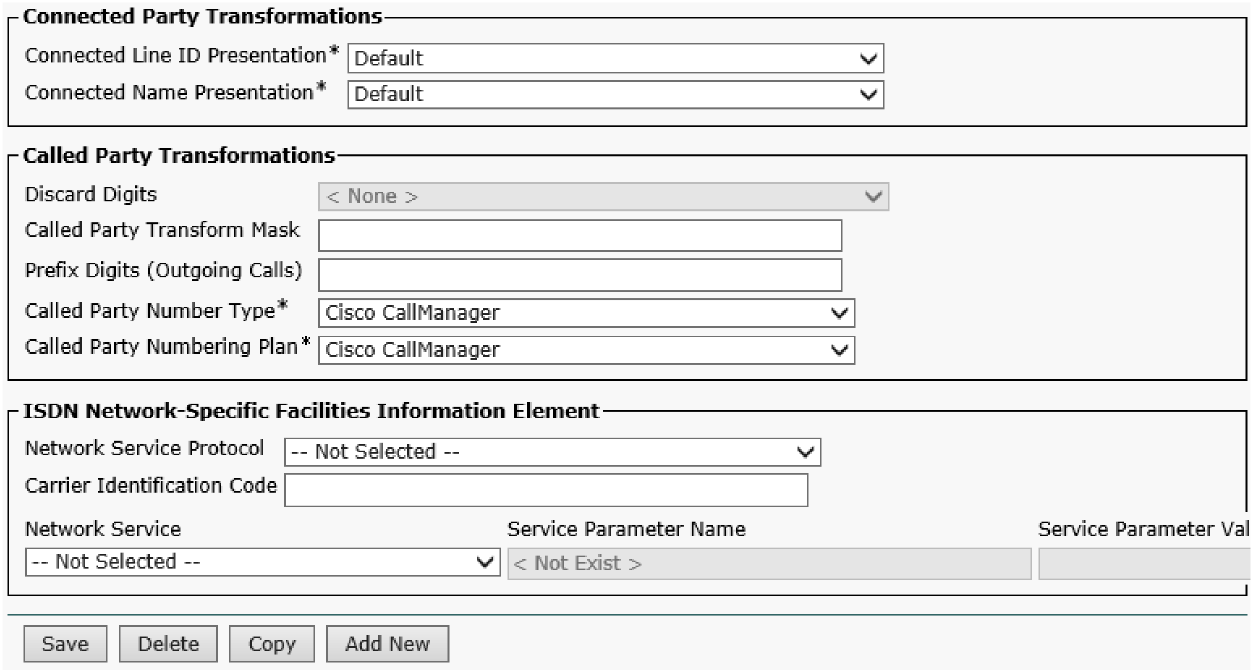
Figure 35: Route Pattern -1

Route Pattern to Cisco Unity

Pattern Definition

1. Set **Route Pattern**: Enter the routing pattern
2. Set **Gateway/Route List**: Select the Trunk you have created under Device ->Trunk





Connected Party Transformations

Connected Line ID Presentation* Default

Connected Name Presentation* Default

Called Party Transformations

Discard Digits < None >

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type* Cisco CallManager

Called Party Numbering Plan* Cisco CallManager

ISDN Network-Specific Facilities Information Element

Network Service Protocol -- Not Selected --

Carrier Identification Code

Network Service	Service Parameter Name	Service Parameter Val
-- Not Selected --	< Not Exist >	

Save Delete Copy Add New

Route Pattern to Gateway

Navigation: Call Routing -> Route/Hunt -> Route Pattern

1. Set **Route Pattern**: \+1XXXXXXXXXX
2. Set **Gateway/Route List**: Select the End-Point you have created in the gateway configuration under Device -> Gateway
3. Set Called Party Transform Mask: XXXXXXXXXX

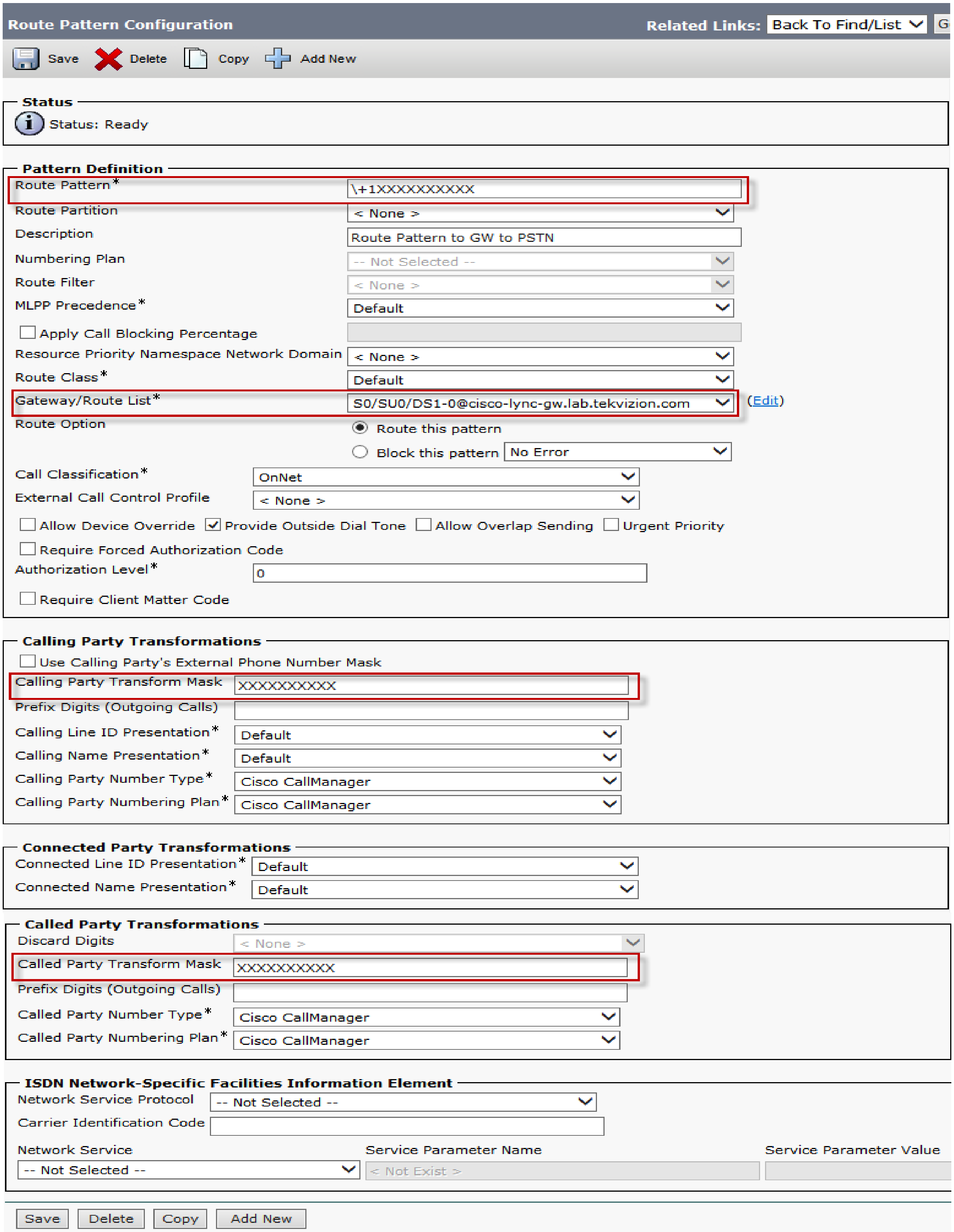


Figure 36: Route Pattern to Gateway

SIP Route Pattern

Navigation: Call Routing -> SIP Route Pattern

1. Set **IPv4 Pattern**: Enter the FQDN of the mediation server
2. Set **Description**: Enter the description of the SIP Route Pattern
3. Set **SIP Trunk**: From the drop-down list select your trunk to Lync Server [this will be the trunk, other than the one added in route group].

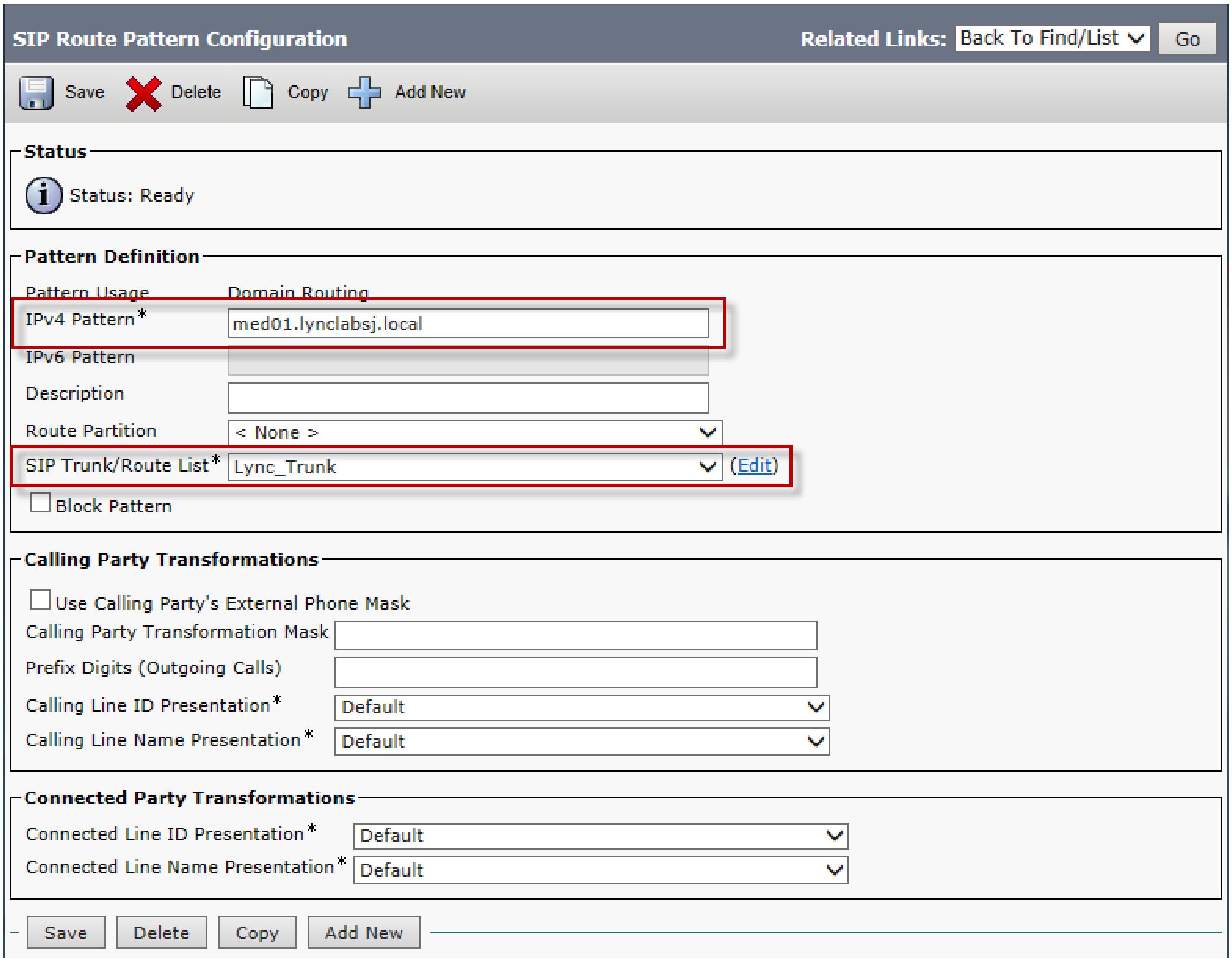


Figure 37: SIP Route Pattern-1

Create SIP Route Patterns similar to the above configuration to all mediation servers in the mediation pool as shown below

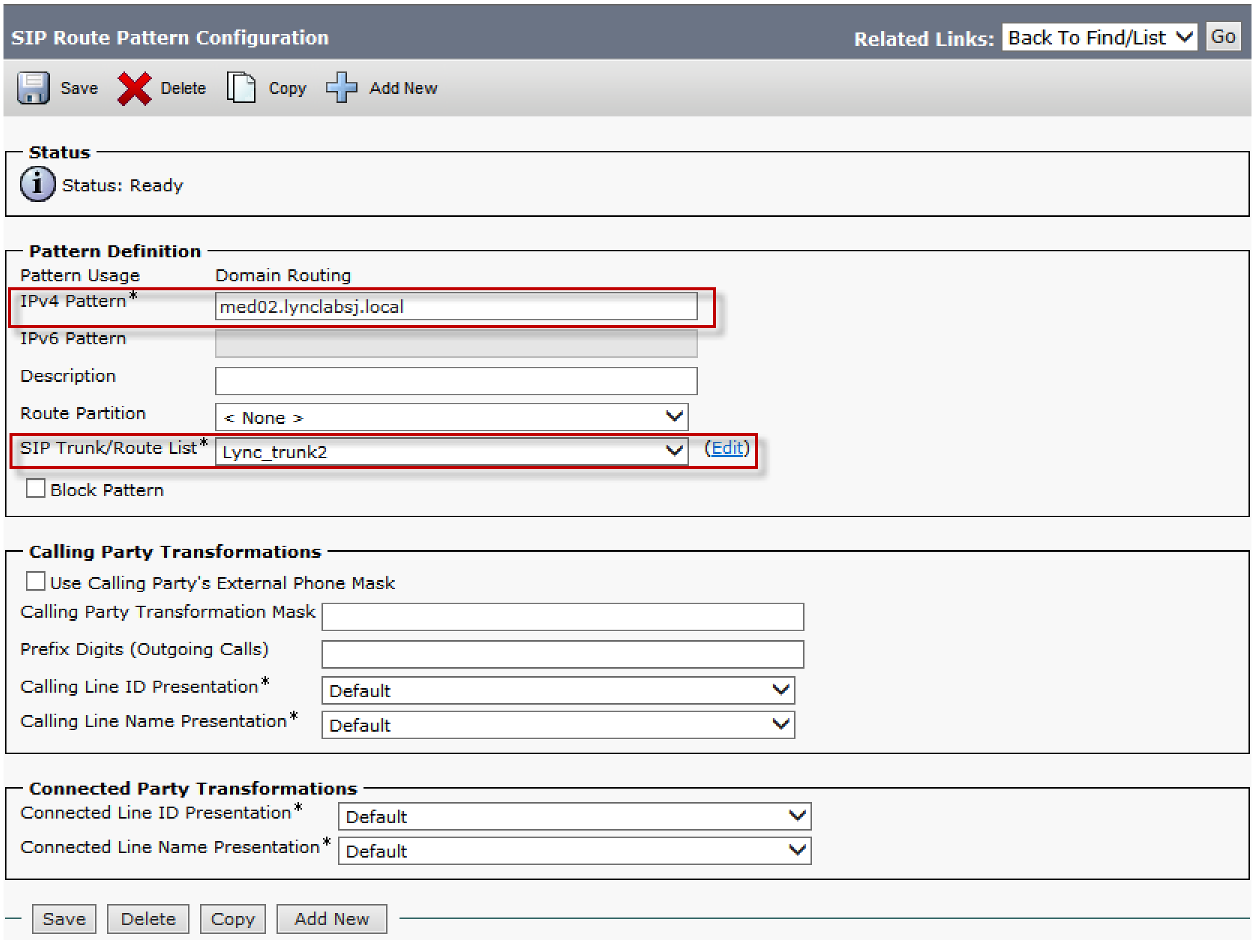


Figure 38: SIP Route Pattern-2

Cisco End Point Configuration

Navigation: Device->Phone

SIP Phone Configuration

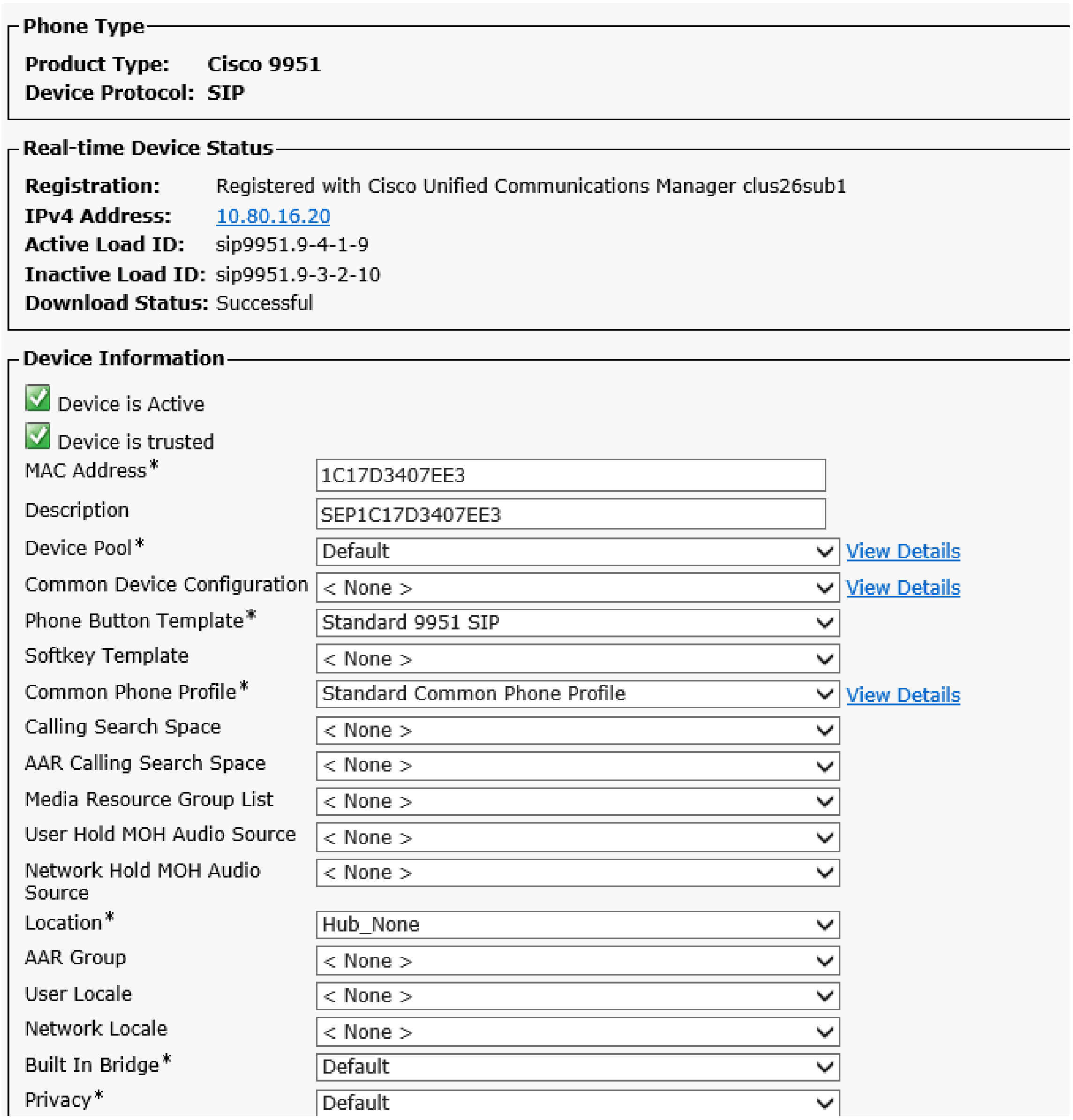


Figure 39: SIP Phone Configuration-1

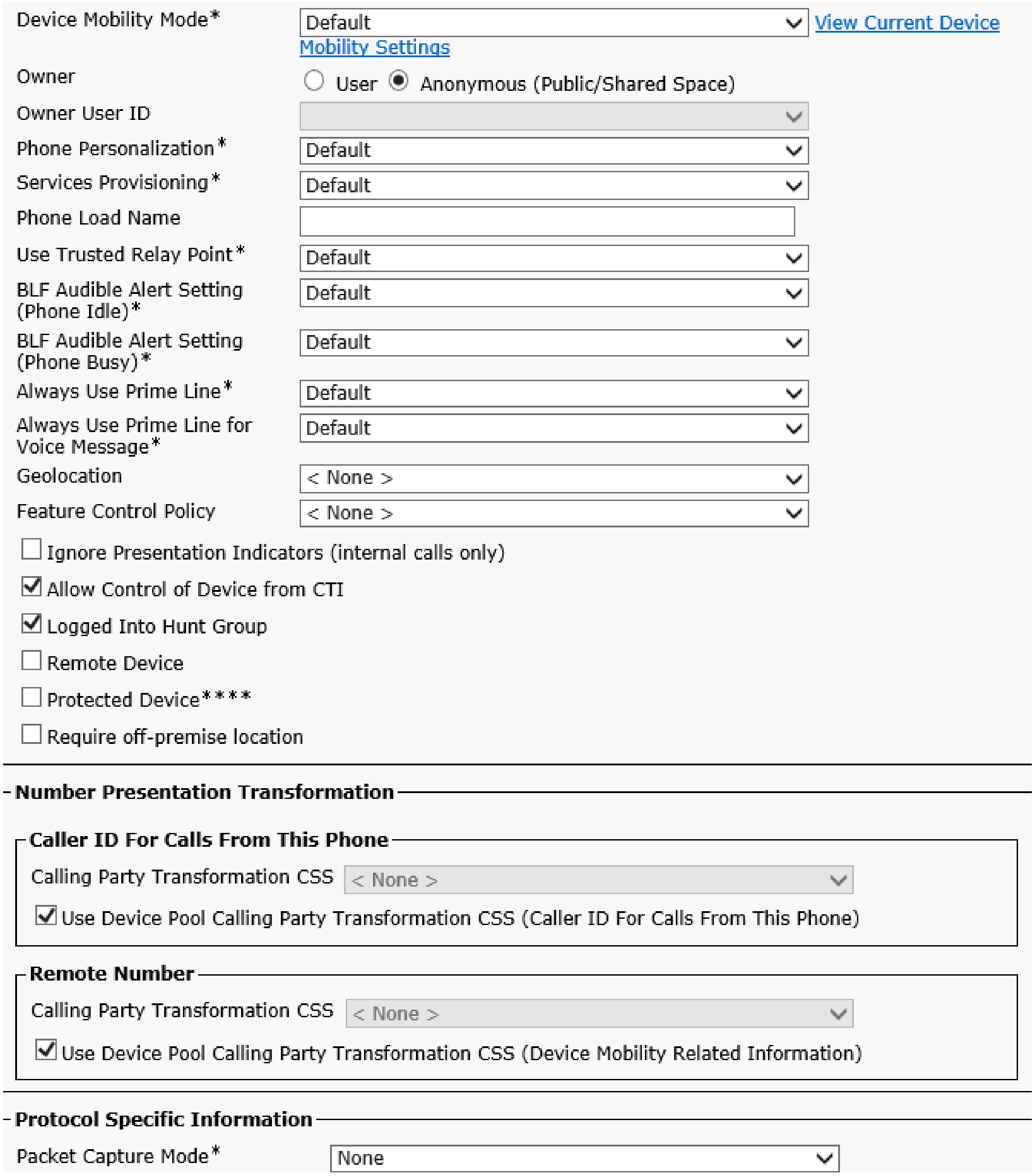


Figure 40: SIP Phone Configuration-2

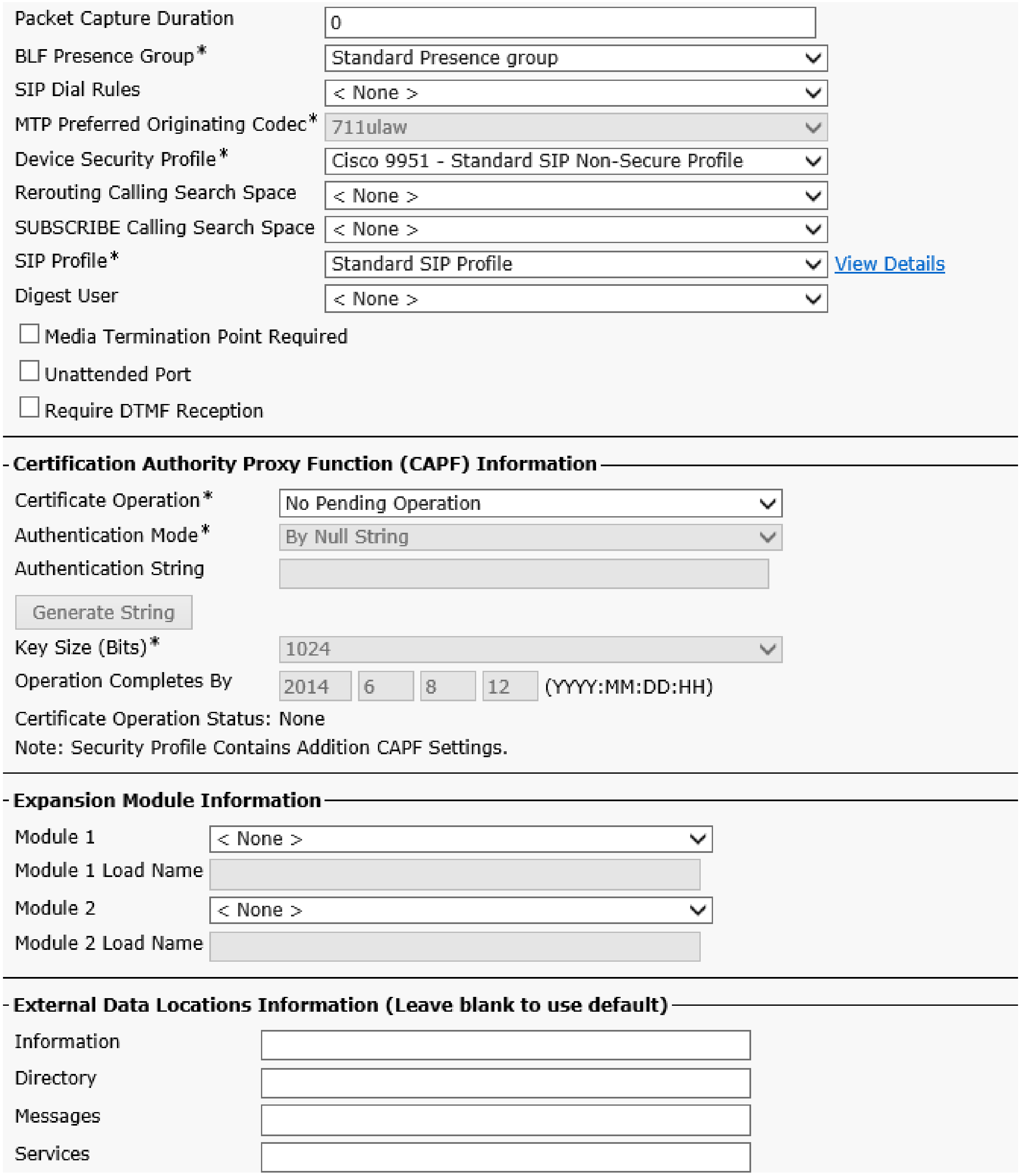


Figure 41: SIP Phone Configuration-3

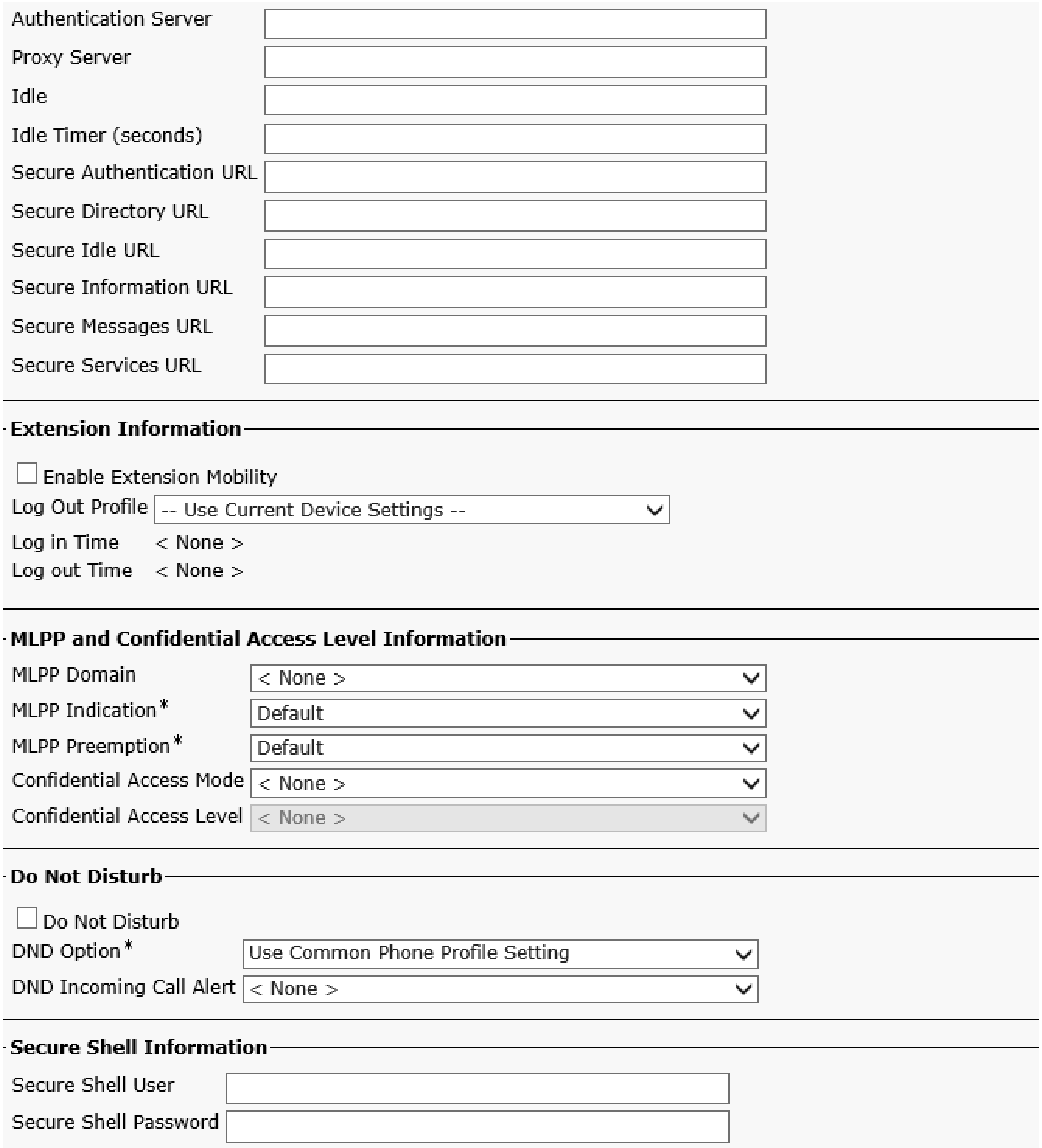


Figure 42: SIP Phone Configuration-4

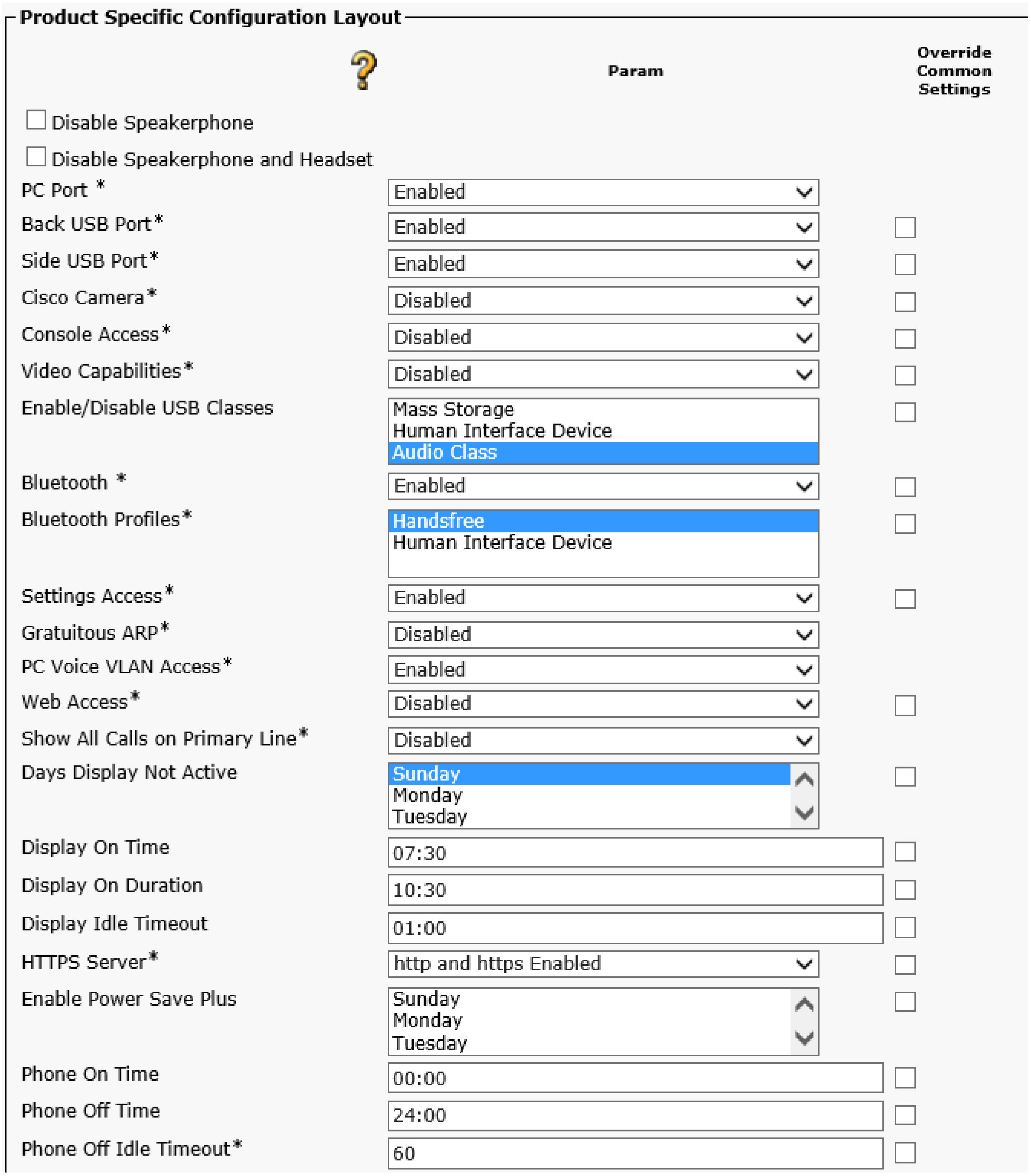


Figure 43: SIP Phone Configuration-5

☐ Enable Audible Alert		☐
EnergyWise Domain		☐
EnergyWise Endpoint Security Secret		☐
☐ Allow EnergyWise Overrides		☐
Span to PC Port*	Disabled	
Logging Display*	Disabled	
Load Server		☐
IPv6 Load Server		☐
Recording Tone*	Disabled	
Recording Tone Local Volume*	100	
Recording Tone Remote Volume*	50	
Recording Tone Duration		
Display On When Incoming Call*	Enabled	☐
RTCP*	Enabled	☑
Log Server		☐
IPv6 Log Server		☐
Remote Log*	Disabled	☐
Log Profile	Default Preset Telephony	☐
Advertise G.722 and iSAC Codecs*	Use System Default	
Wideband Headset UI Control*	Enabled	
Wideband Headset*	Enabled	
Peer Firmware Sharing*	Enabled	☐
Cisco Discovery Protocol (CDP): Switch Port*	Enabled	☐
Cisco Discovery Protocol (CDP): PC Port*	Enabled	☐
Link Layer Discovery Protocol - Media Endpoint Discover (LLDP-MED): Switch Port*	Enabled	☐
Link Layer Discovery Protocol (LLDP): PC Port*	Enabled	☐
LLDP Asset ID		
LLDP Power Priority*	Unknown	
802.1x Authentication*	User Controlled	☐

Figure 44: SIP Phone Configuration-6

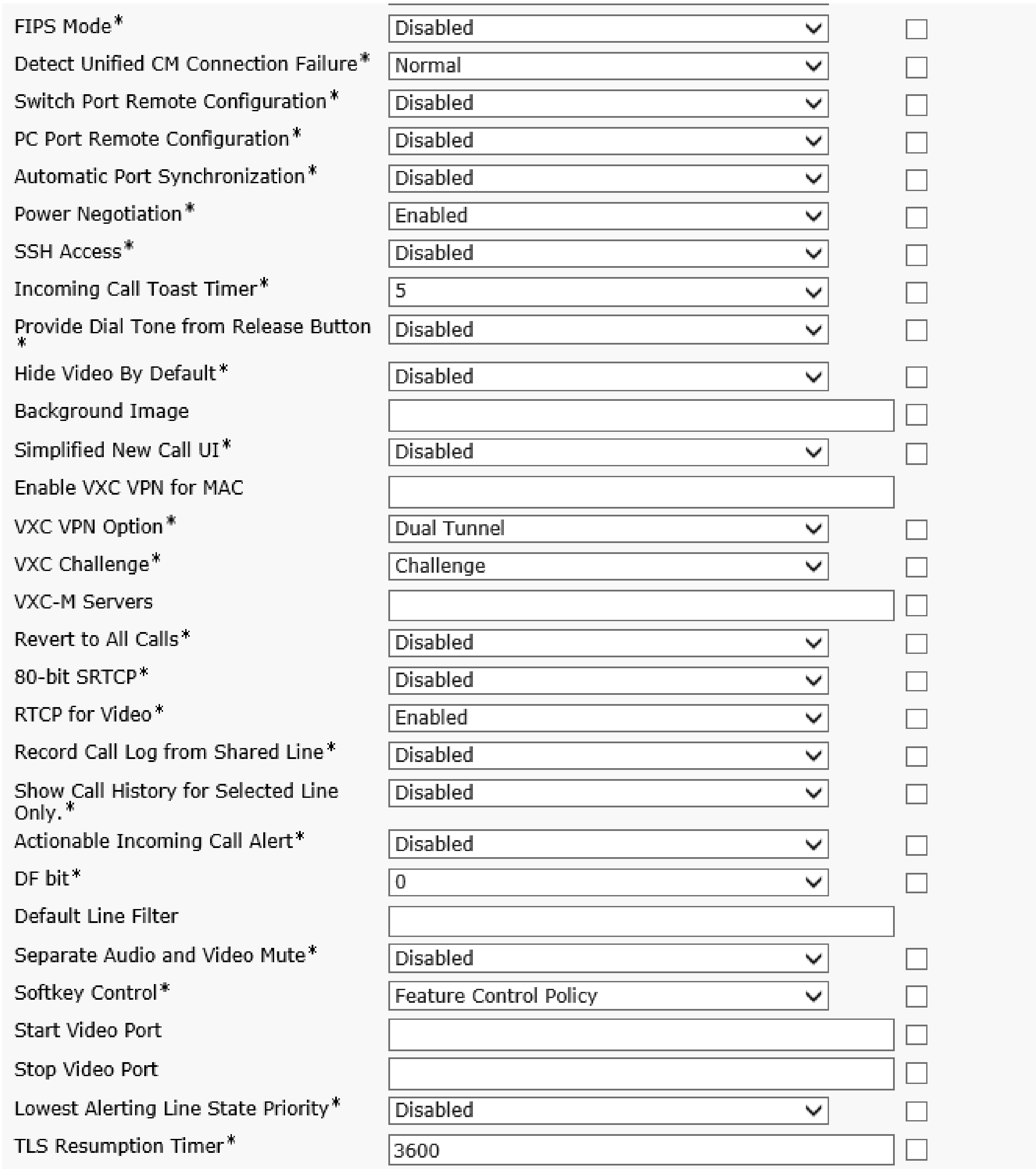


Figure 45: SIP Phone Configuration-7

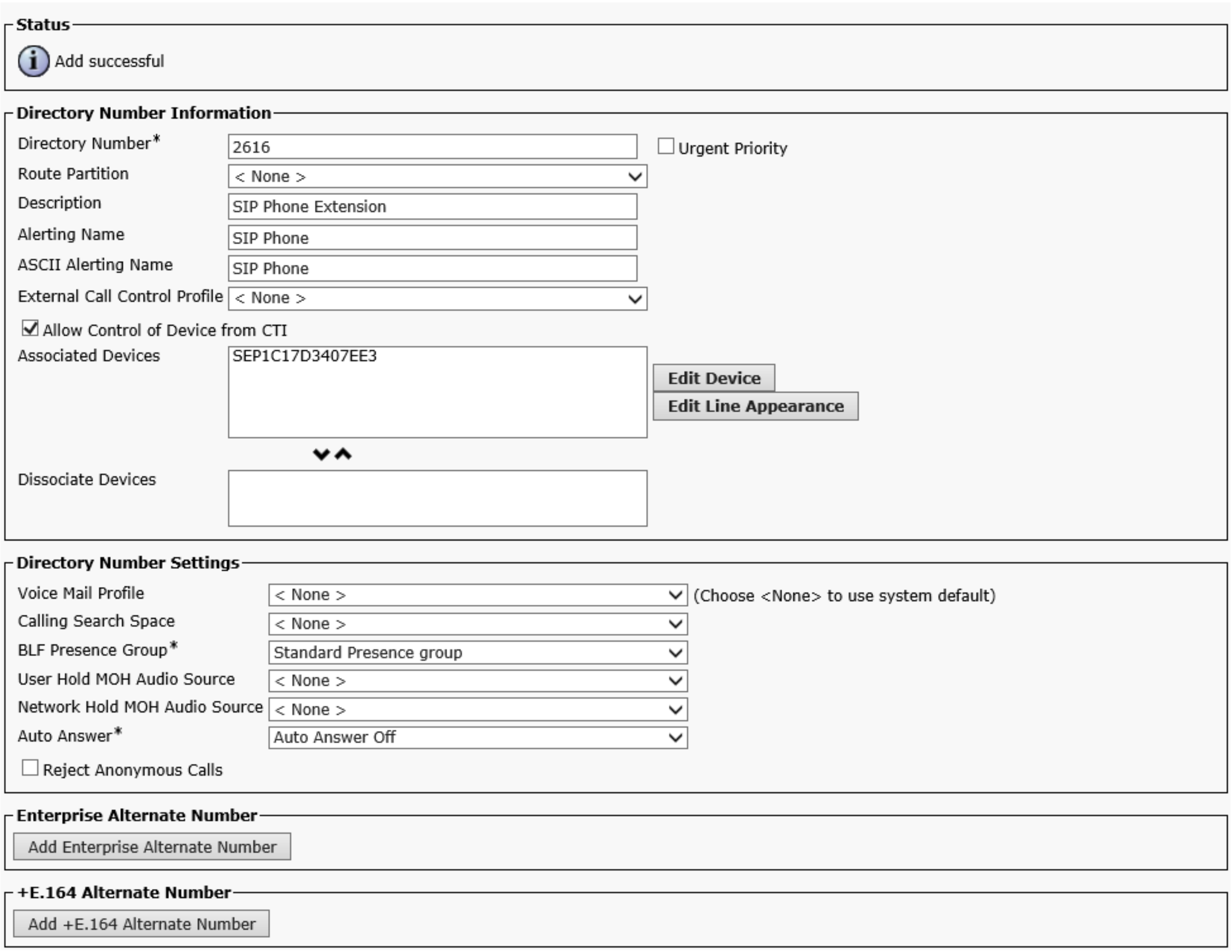


Figure 46: SIP Phone Configuration-8

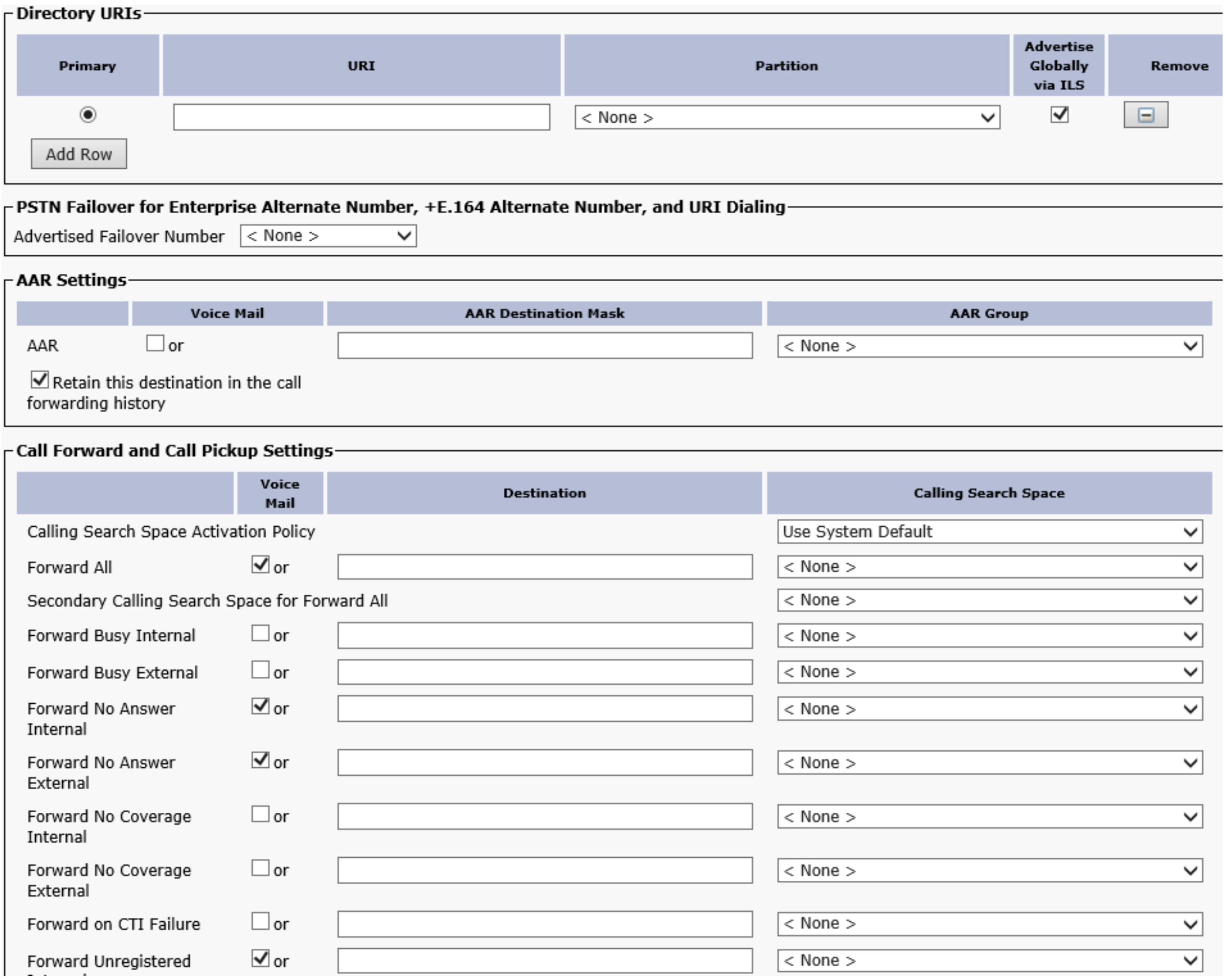


Figure 47: SIP Phone Configuration-9

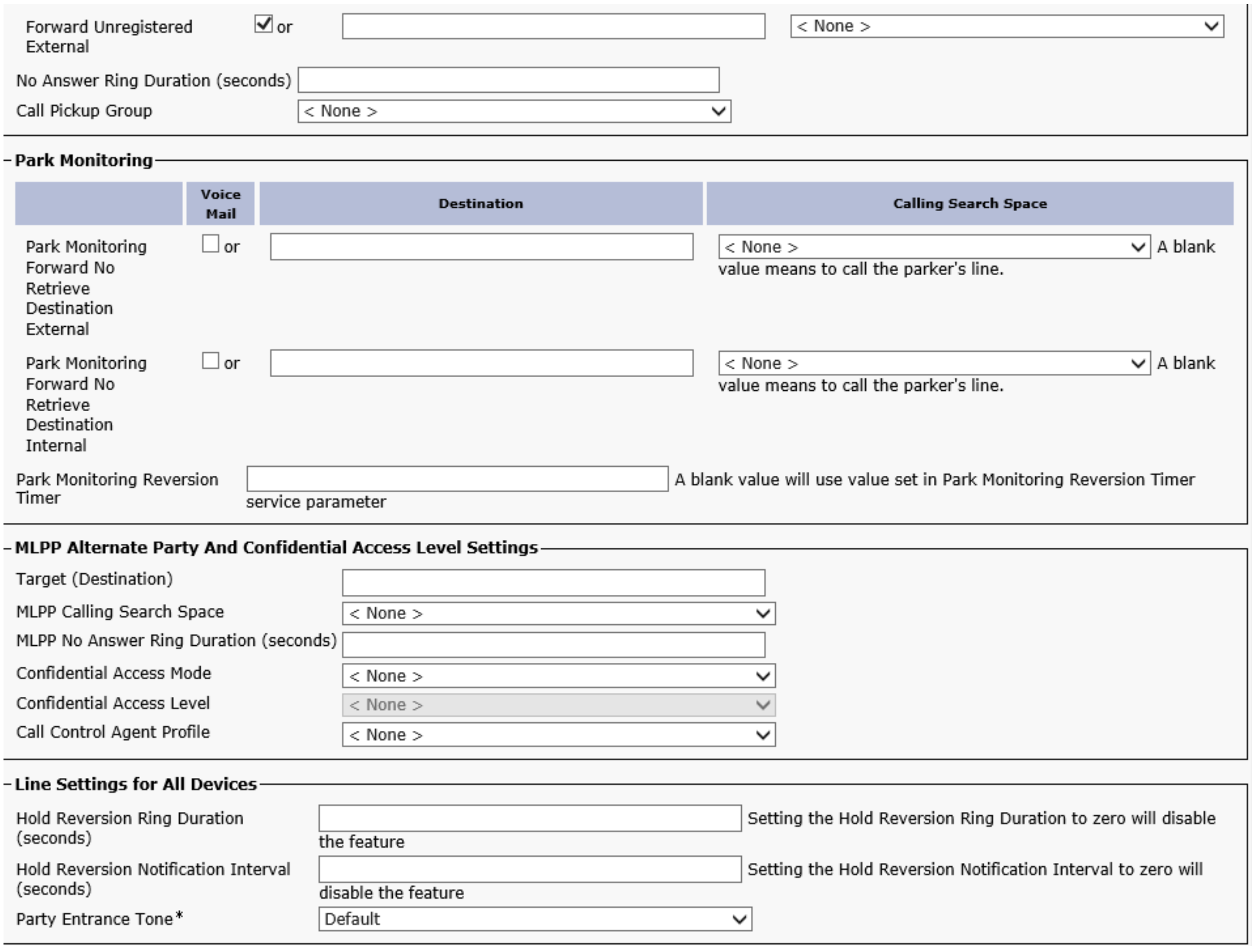
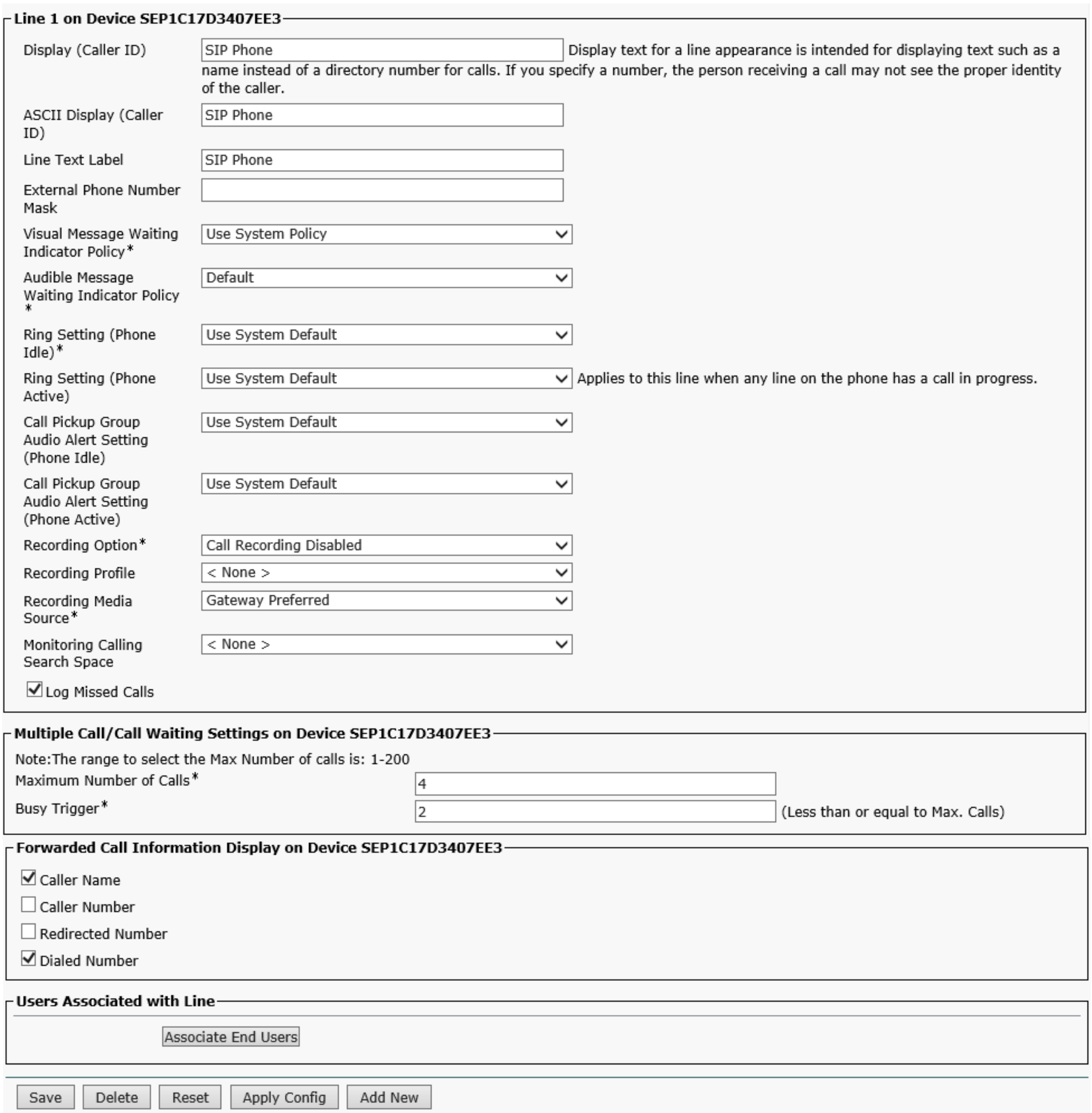


Figure 48: SIP Phone Configuration-10



Line 1 on Device SEP1C17D3407EE3

Display (Caller ID)	SIP Phone — Display text for a line appearance is intended for displaying text such as a name instead of a directory number for calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.
ASCII Display (Caller ID)	SIP Phone
Line Text Label	SIP Phone
External Phone Number Mask	
Visual Message Waiting Indicator Policy*	Use System Policy
Audible Message Waiting Indicator Policy*	Default
Ring Setting (Phone Idle)*	Use System Default
Ring Setting (Phone Active)	Use System Default — Applies to this line when any line on the phone has a call in progress.
Call Pickup Group Audio Alert Setting (Phone Idle)	Use System Default
Call Pickup Group Audio Alert Setting (Phone Active)	Use System Default
Recording Option*	Call Recording Disabled
Recording Profile	< None >
Recording Media Source*	Gateway Preferred
Monitoring Calling Search Space	< None >

☑ Log Missed Calls

Multiple Call/Call Waiting Settings on Device SEP1C17D3407EE3

Note:The range to select the Max Number of calls is: 1-200

Maximum Number of Calls* 4

Busy Trigger* 2 (Less than or equal to Max. Calls)

Forwarded Call Information Display on Device SEP1C17D3407EE3

☑ Caller Name

☐ Caller Number

☐ Redirected Number

☑ Dialed Number

Users Associated with Line

Associate End Users

Save | Delete | Reset | Apply Config | Add New

Figure 49: SIP Phone Configuration-11

SCCP Phone Configuration

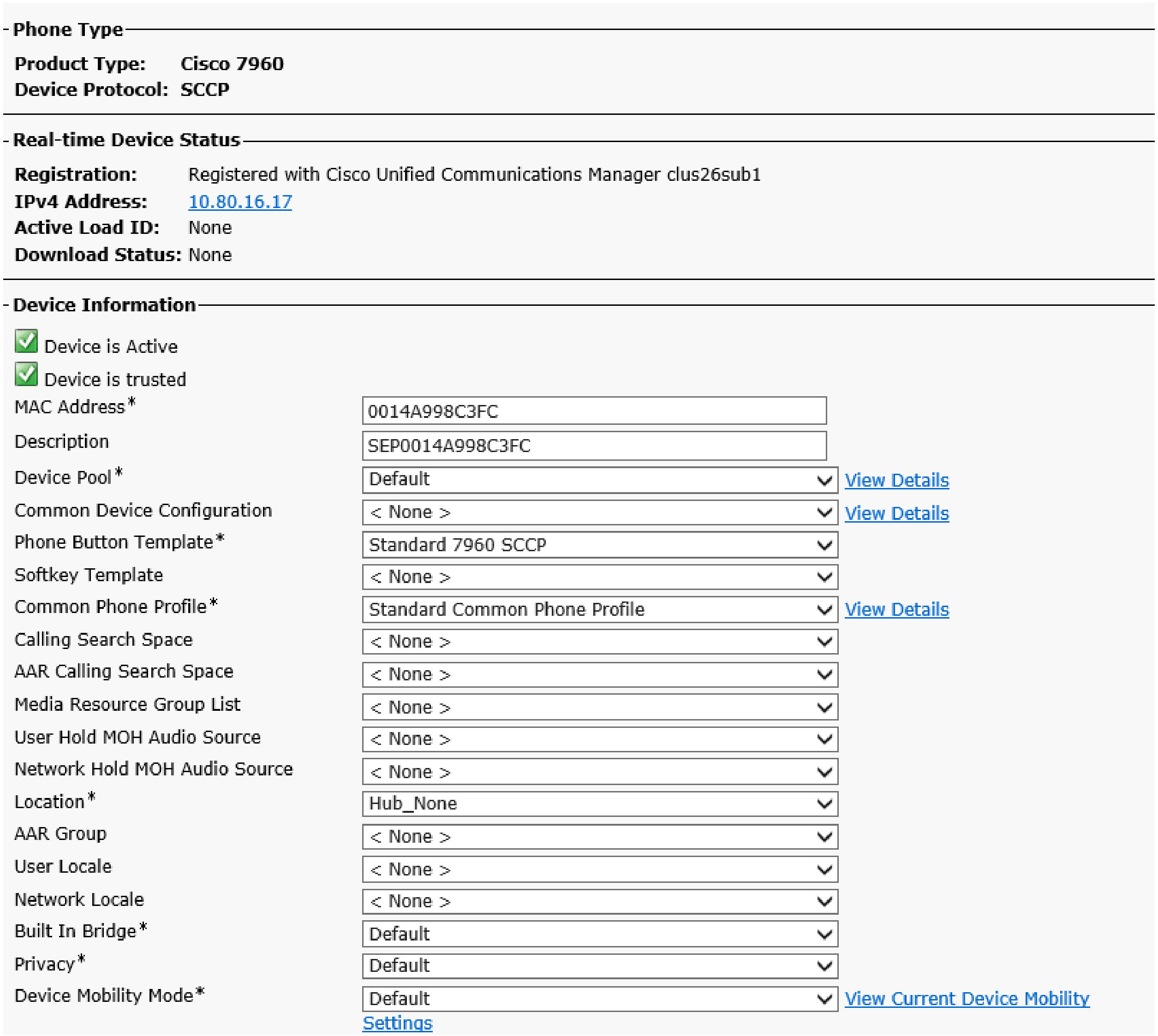


Figure 50: SCCP Phone Configuration-1

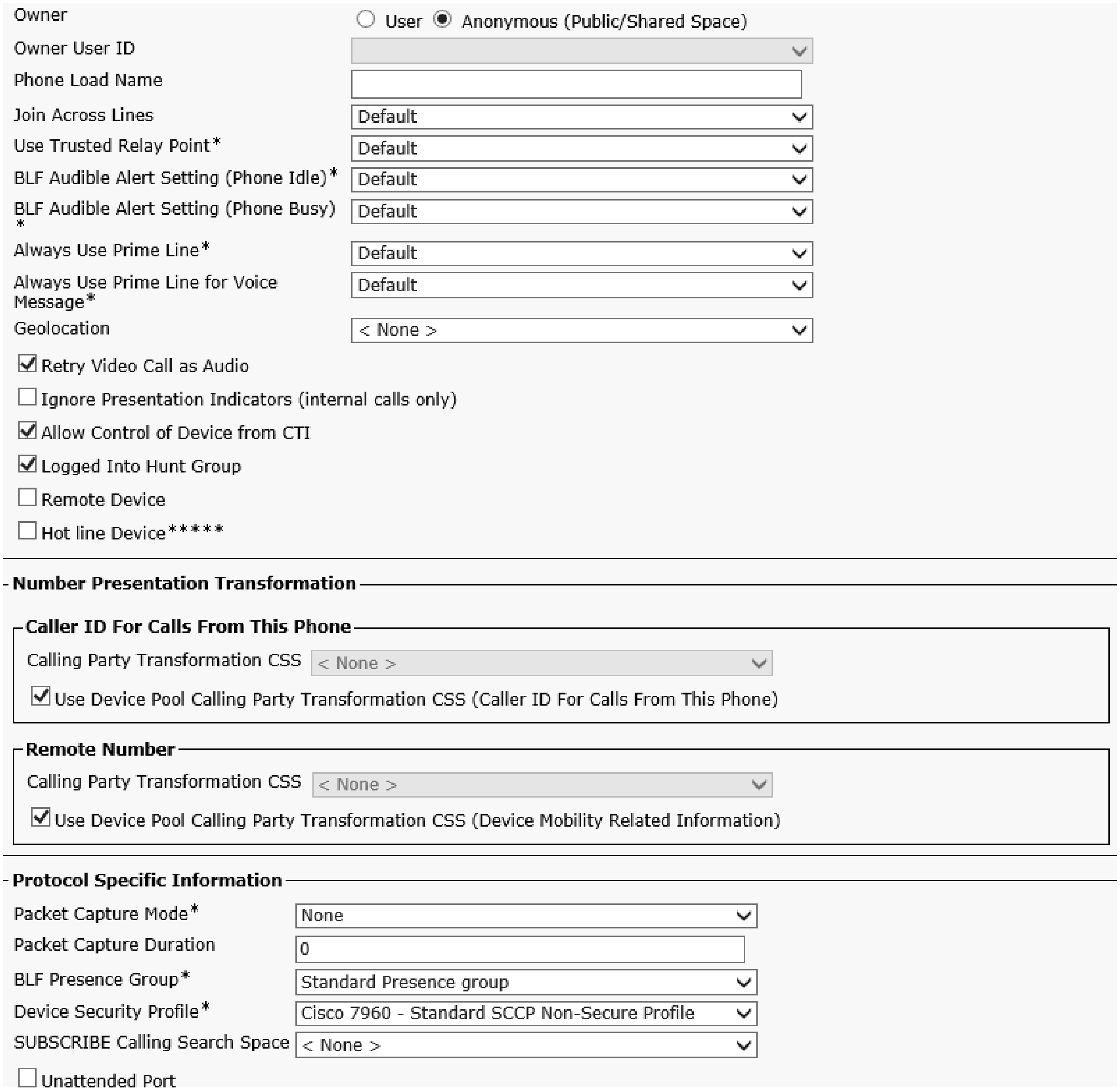


Figure 51: SCCP Phone Configuration-2

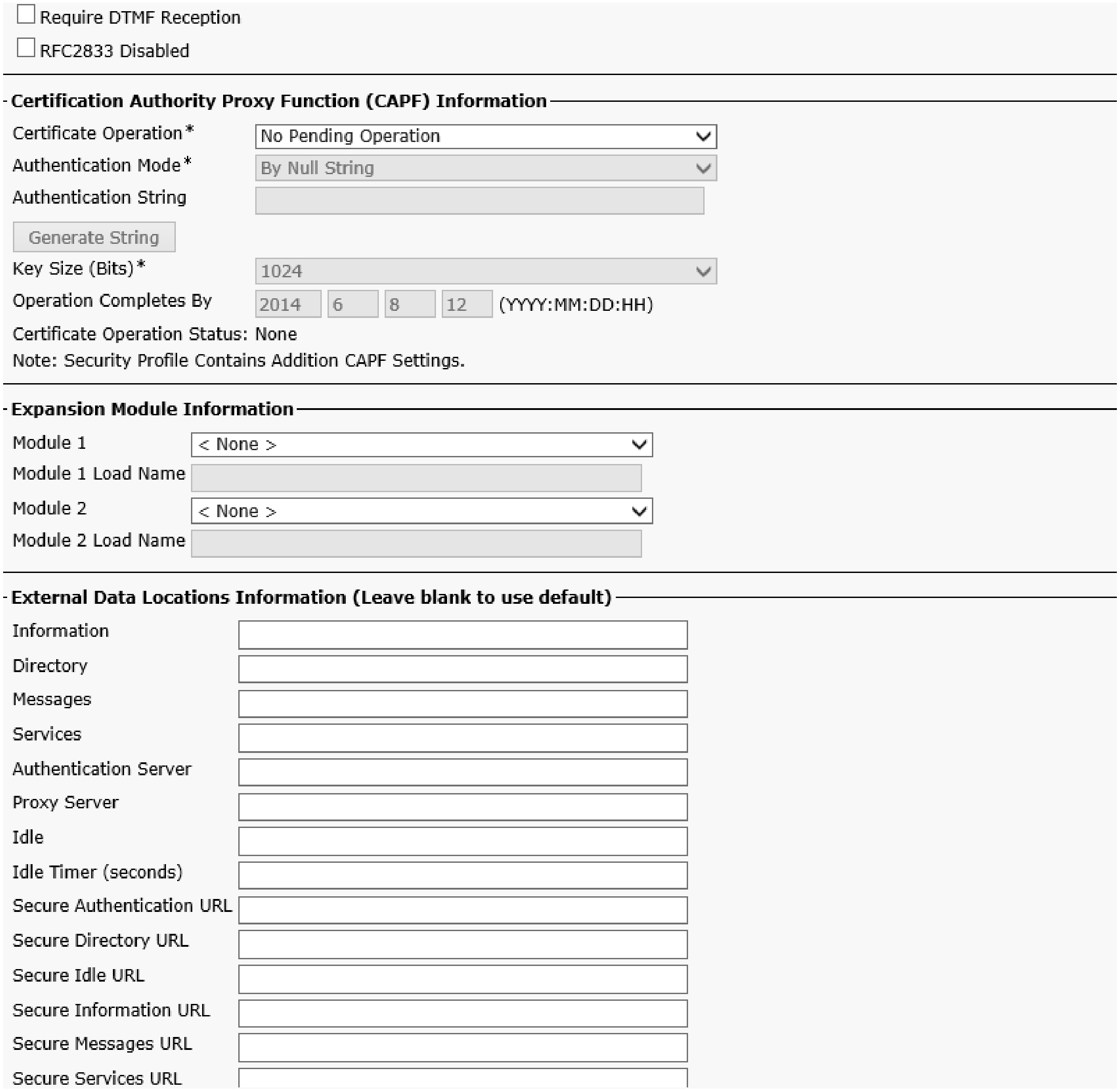


Figure 52: SCCP Phone Configuration-3

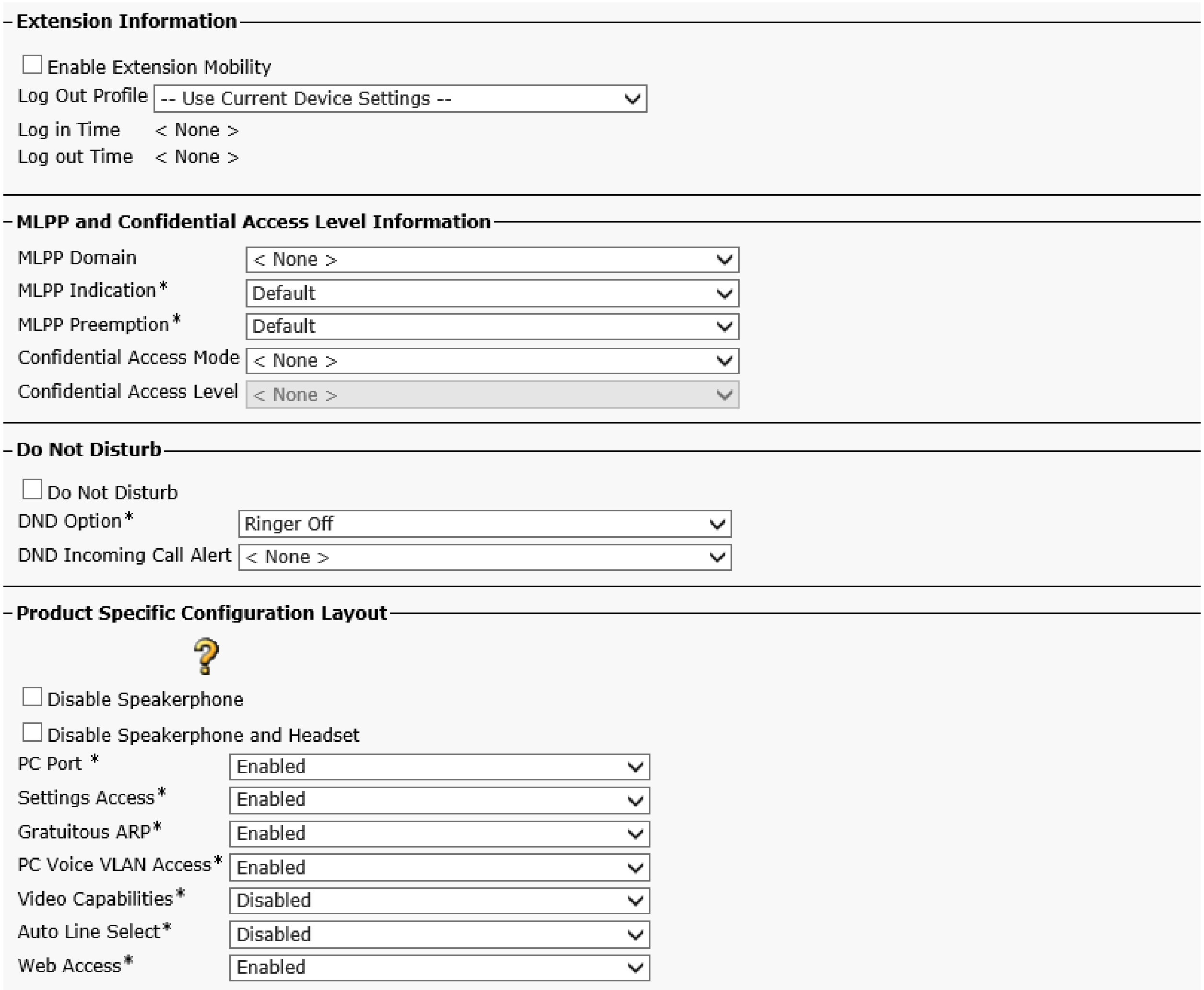


Figure 53: SCCP Phone Configuration-4

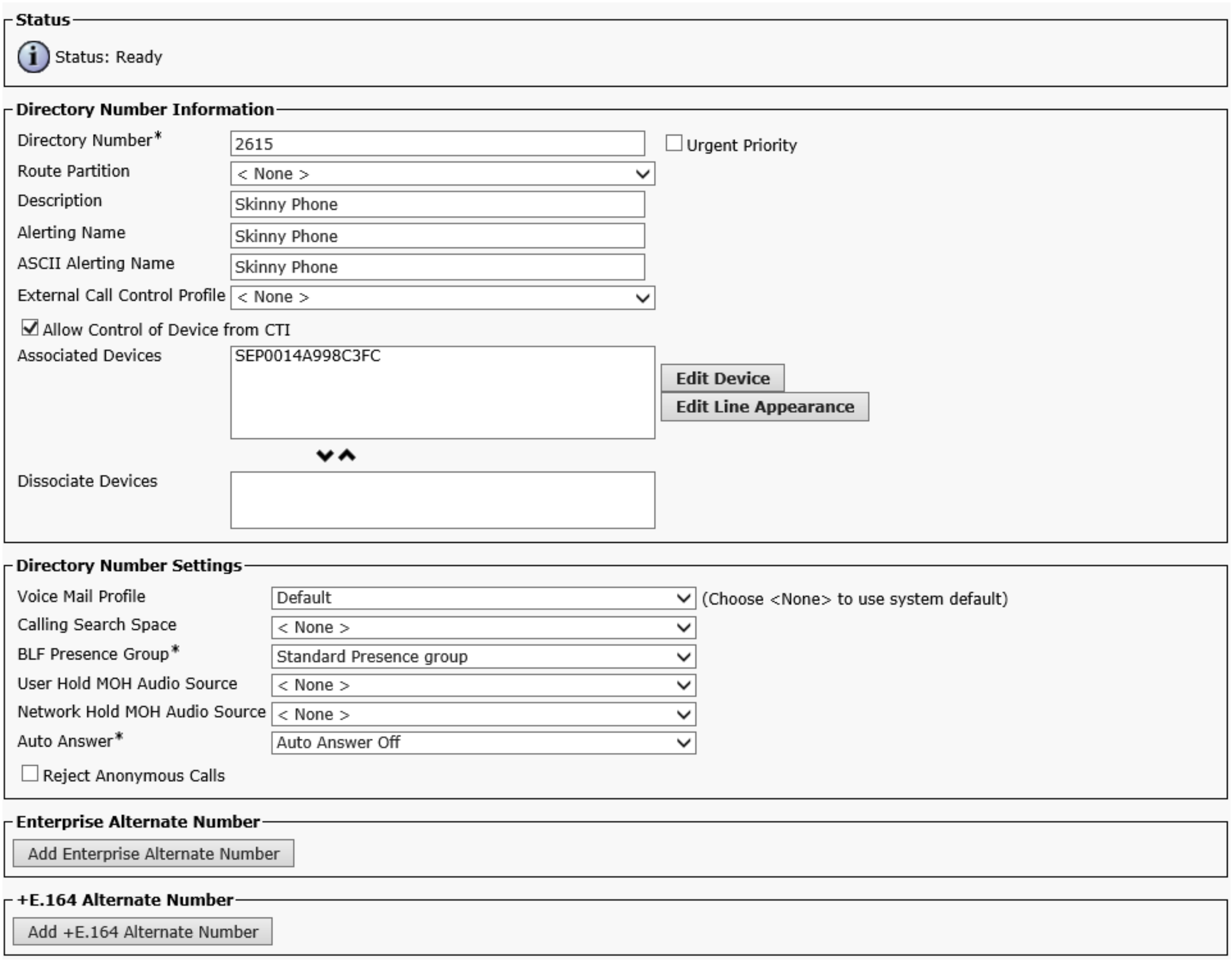


Figure 54: SCCP Phone Configuration-7

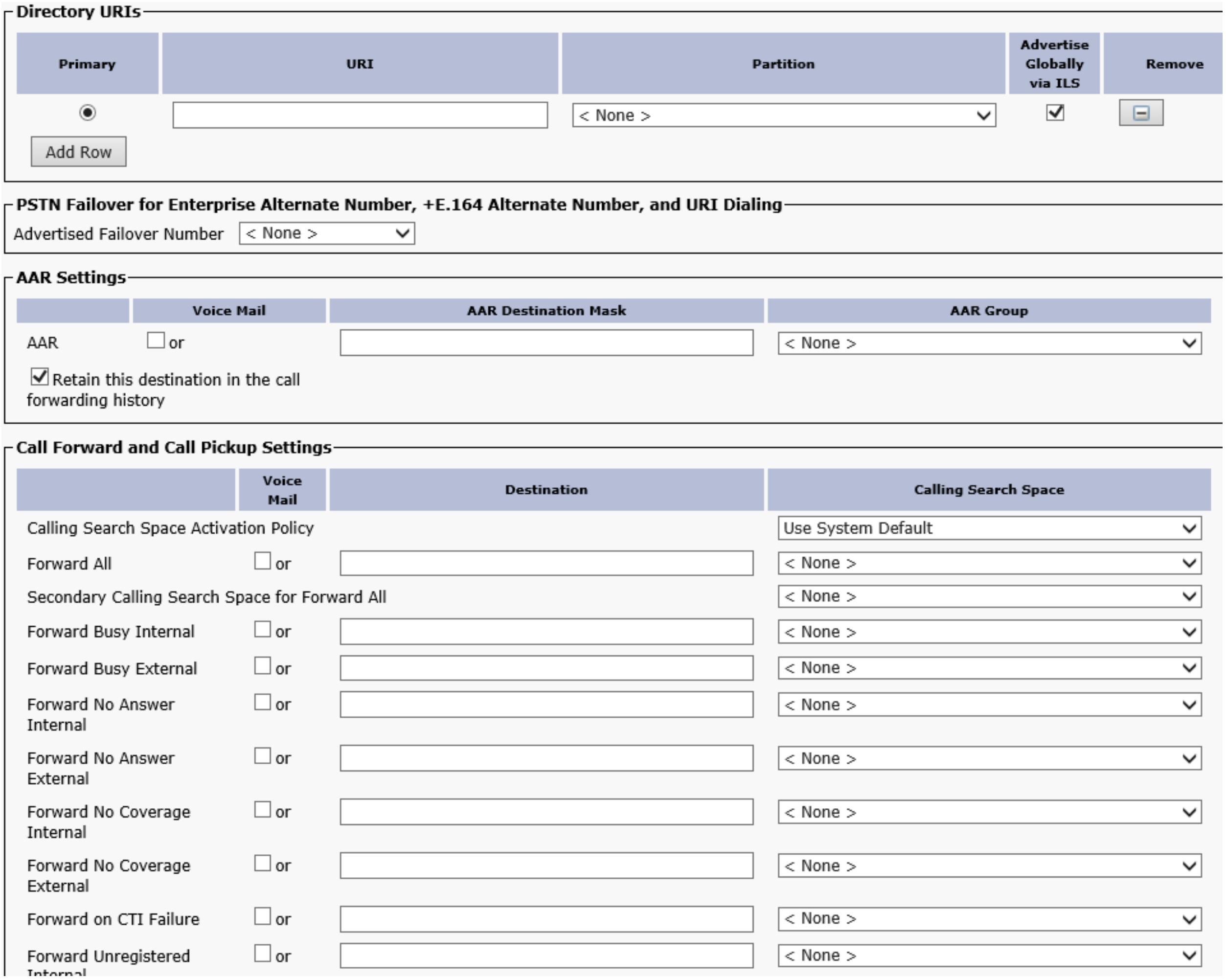


Figure 55: SCCP Phone Configuration-8

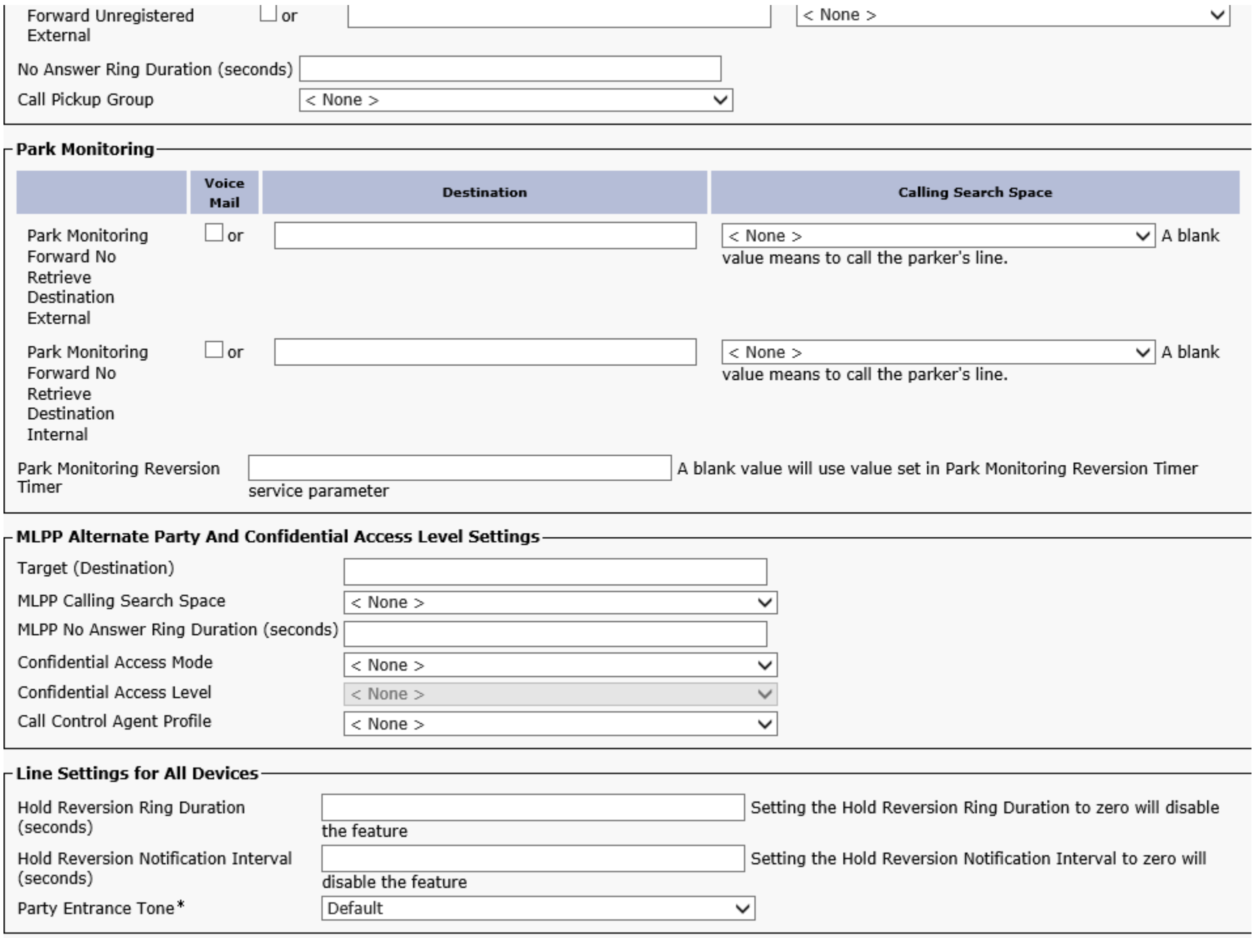


Figure 56: SCCP Phone Configuration-9

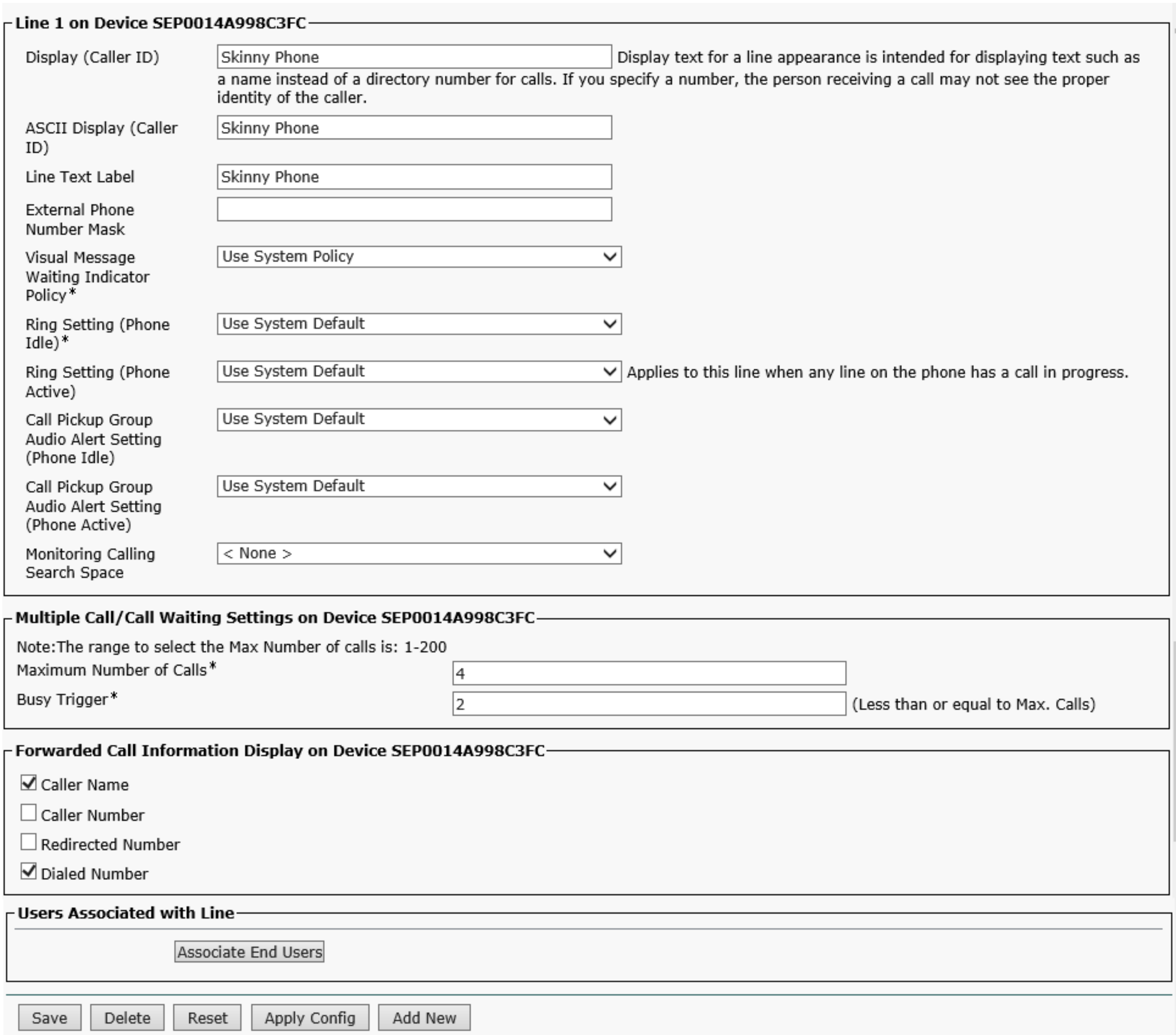
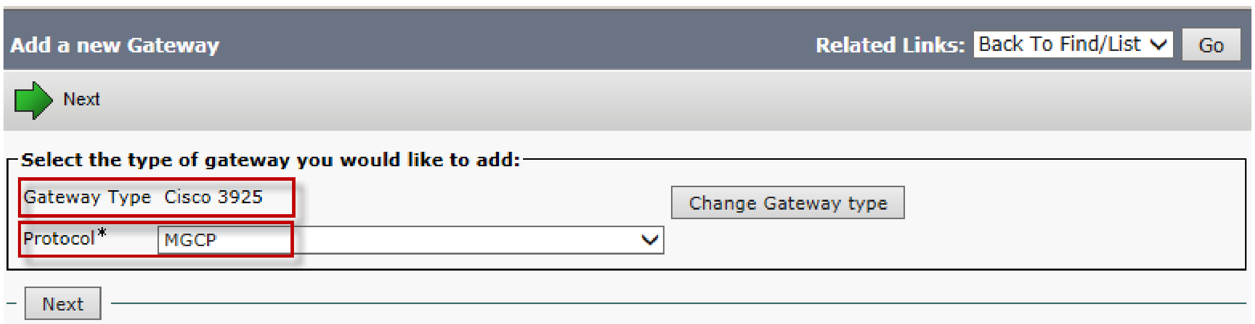


Figure 57: SCCP Phone Configuration-10

MGCP Gateway Configuration

Navigation: Device -> Gateway

1. Set **Gateway Type:** Select the gateway model you are using, from the available list of devices.
2. Set **Domain Name:** Enter the FQDN of the gateway device.
3. Set **Cisco Unified Communications Manager Group:** Default
4. Set **Module in Slot 0:** Select 'NM-4VWIC-MBRD'
   Upon saving the configuration, 'Subunit' configuration fields will appear
   a. Set **Subunit 0:** Select the model of T1 card (hardware), which is on your gateway.
5. Set Global ISDN Switch Type: primary-qsig



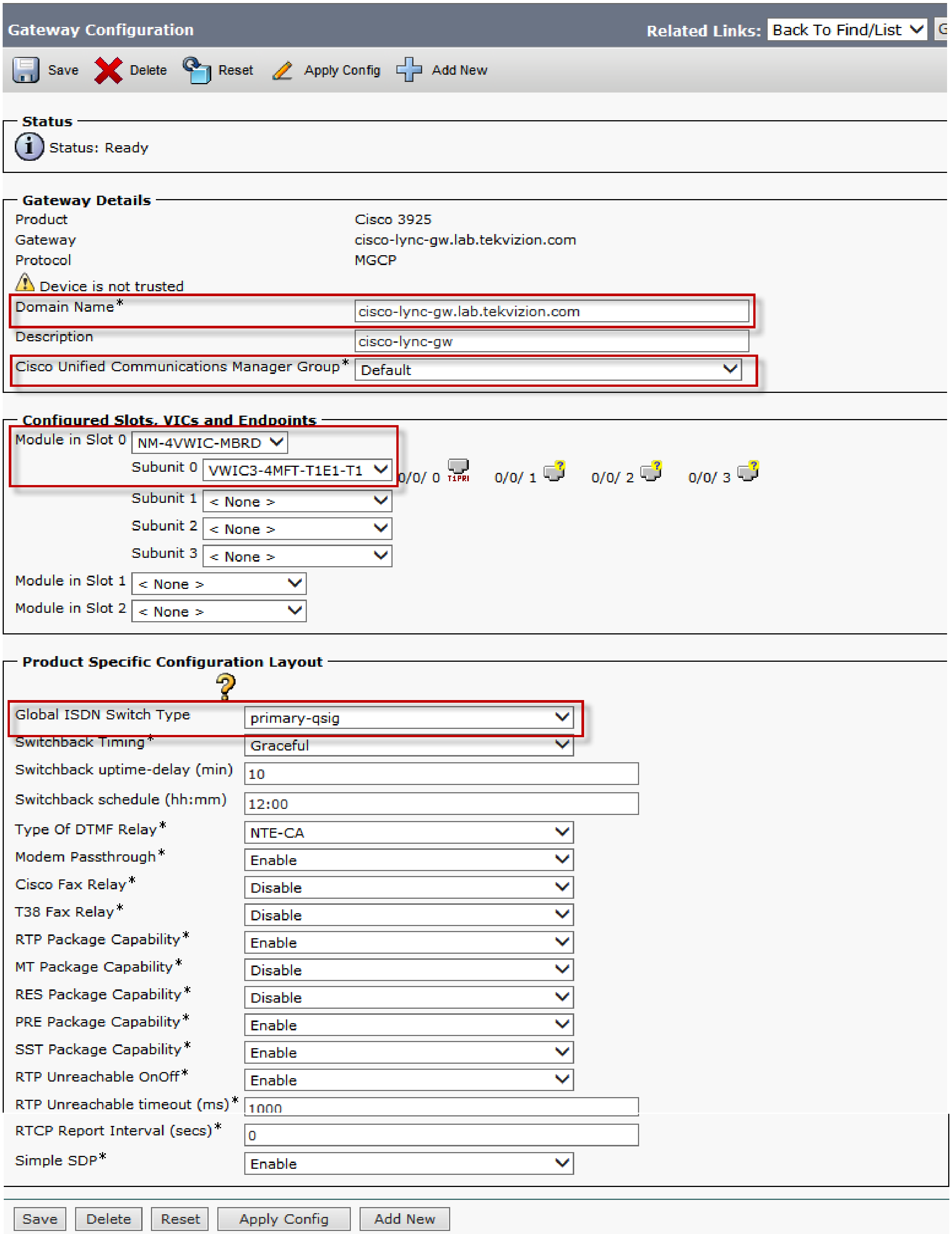


Figure 58: Trunk to Gateway-1

After saving the configuration page, beside the subunit (subunit 0 here) you have configured, click on the interface to which you have plugged the cable. A new configuration page appears, proceed with the configuration steps described below.

6. Set **Device Pool:** Default
7. Confirm **send Extra Leading Character in Display IE***:** is unchecked

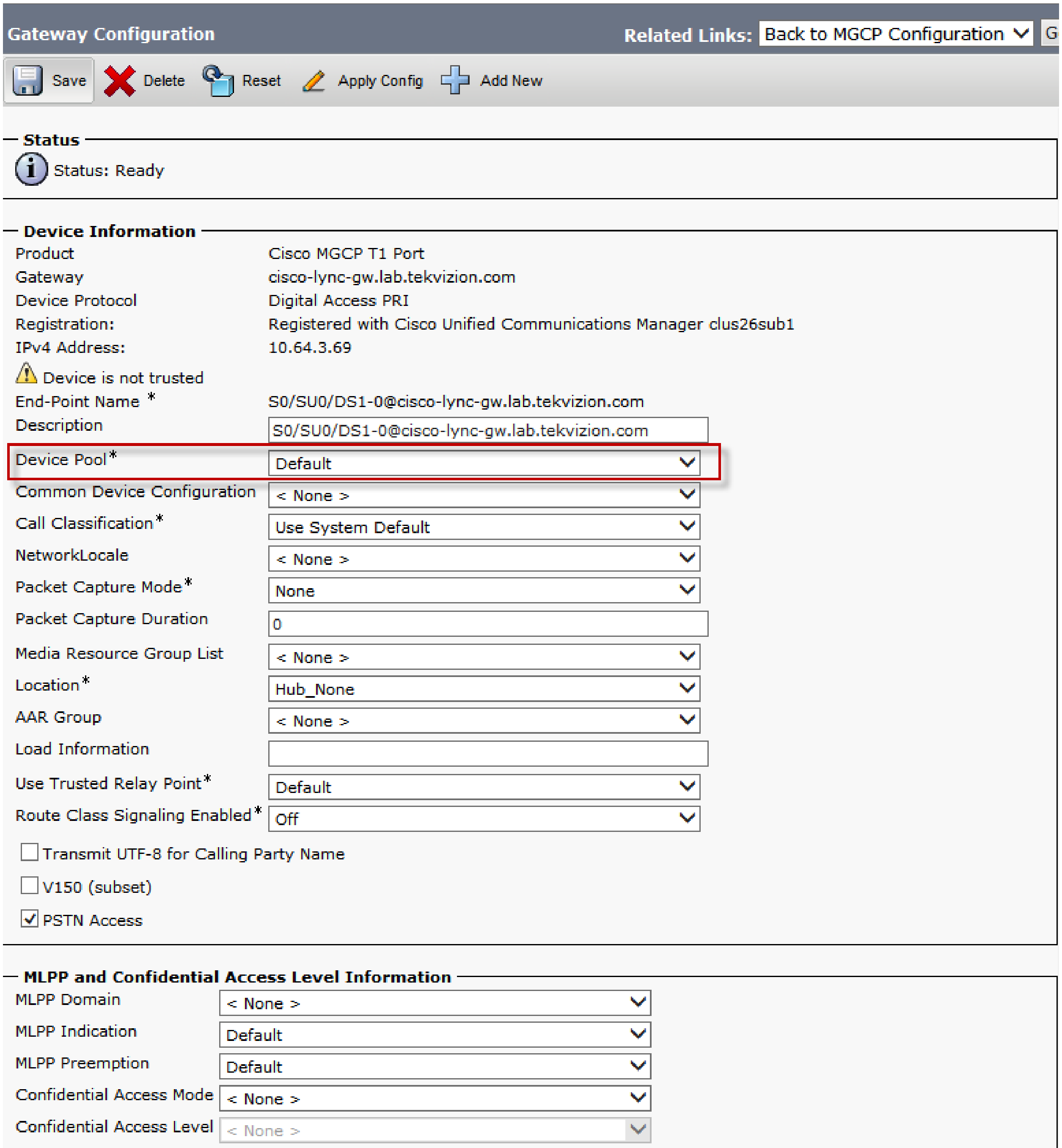


Figure 59: Trunk to Gateway-2

Interface Information

PRI Protocol Type*	PRI NI2
QSIG Variant*	No Changes
ASN.1 ROSE OID Encoding*	No Changes
Protocol Side*	User
Channel Selection Order*	Bottom Up
Channel IE Type*	Use Number when 1B
PCM Type*	µ-law
Delay for first restart (1/8 sec ticks)*	32
Delay between restarts (1/8 sec ticks)*	4

- [x] Inhibit restarts at PRI initialization
- [ ] Enable status poll
- [ ] Unattended Port
- [ ] Enable G.Clear

Call Routing Information - Inbound Calls

Significant Digits*	All
Calling Search Space	< None >
AAR Calling Search Space	< None >
Prefix DN	

Call Routing Information - Outbound Calls

Calling Party Presentation*	Default
Calling Party Selection*	Originator
Called party IE number type unknown*	Cisco CallManager
Calling party IE number type unknown*	Cisco CallManager
Called Numbering Plan*	Cisco CallManager
Calling Numbering Plan*	Cisco CallManager
Number of digits to strip*	0
Caller ID DN	
SMDI Base Port*	0
Called Party Transformation CSS	< None >

- [x] Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS	< None >

- [x] Use Device Pool Calling Party Transformation CSS

Figure 60: Trunk to Gateway-3

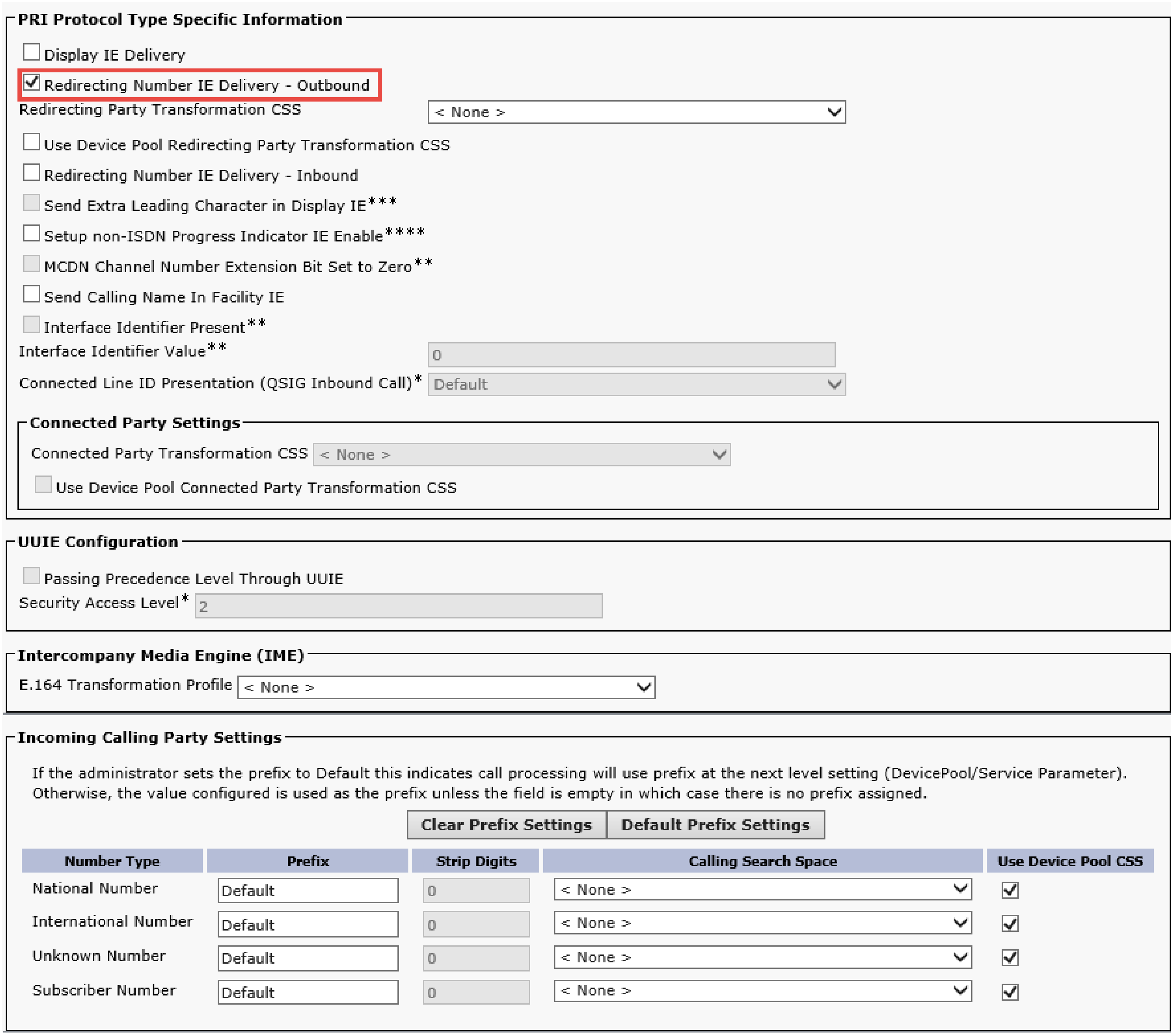


Figure 61: Trunk to Gateway-4

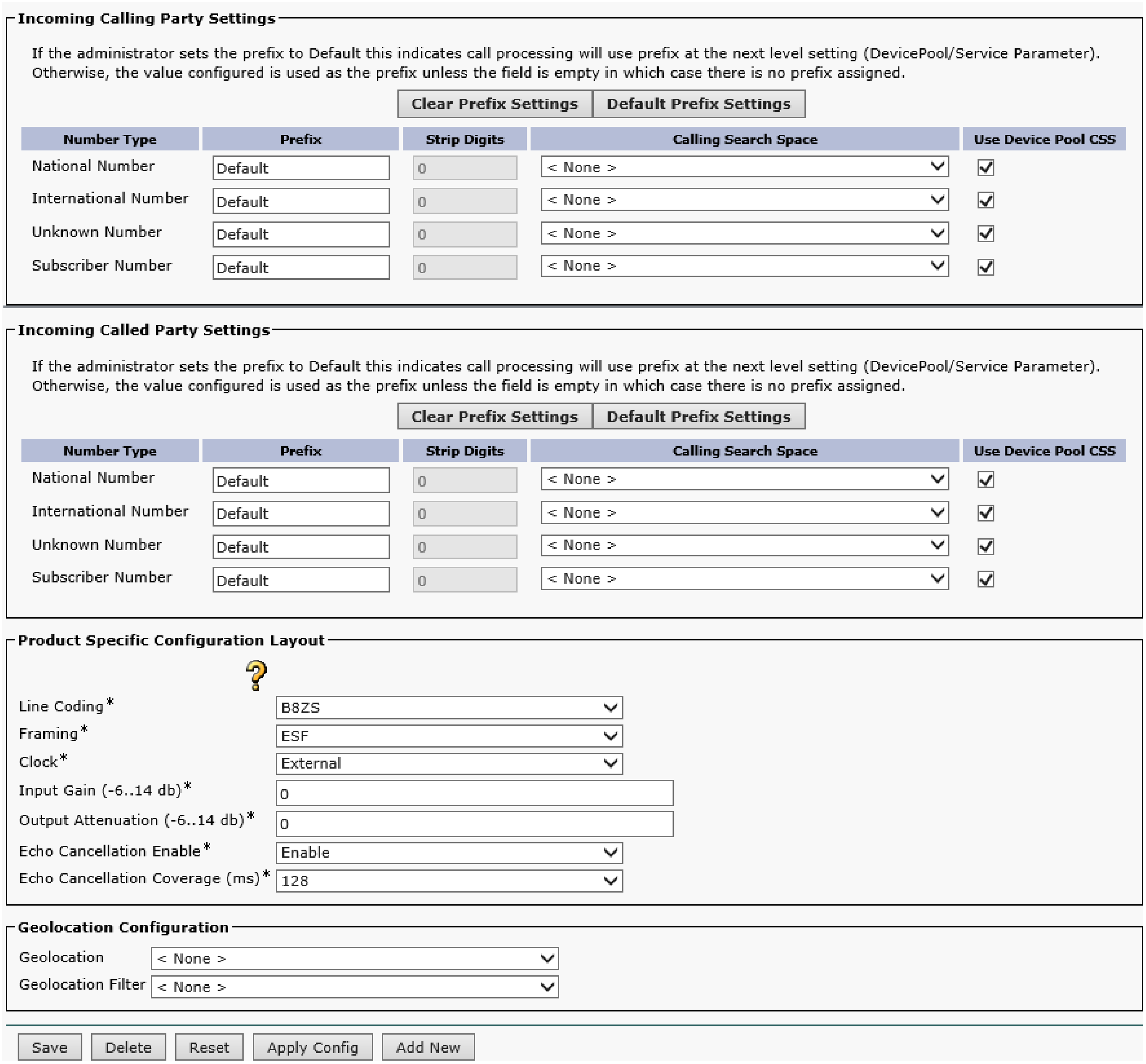


Figure 62: Trunk to Gateway-5

Voice Mail Pilot Configuration

Navigation: Advanced Features->Voice Mail->Voice Mail Profile

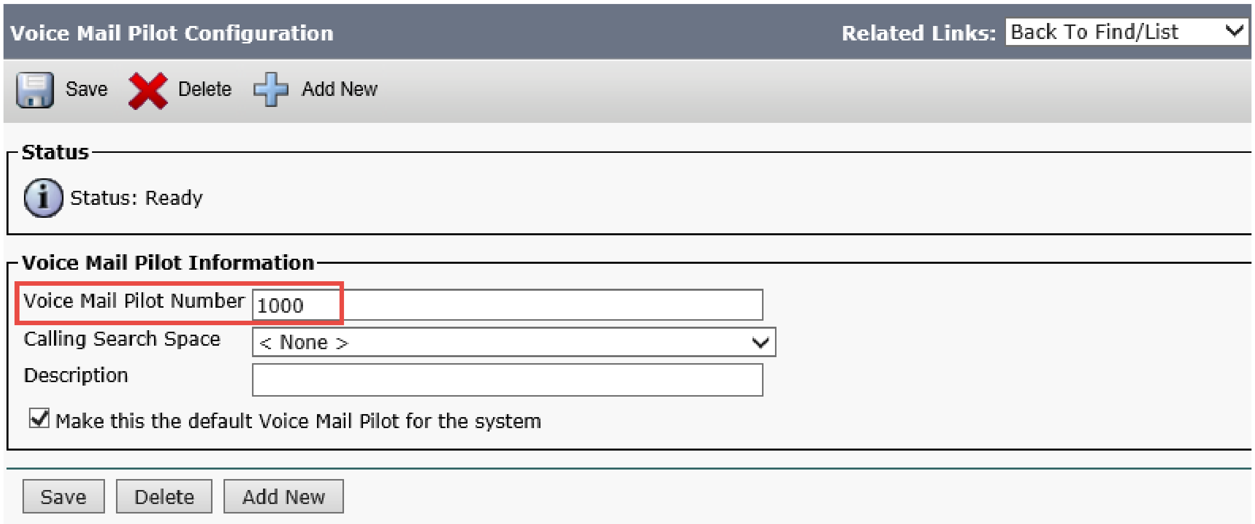


Figure 63: Voice Mail Port Configuration

Cisco Unity Voice Mail Server Configuration

Phone System Configuration

Navigation: Telephony Integrations -> Phone System

1. Set Phone System Name: Enter a name for the phone system
2. To navigate to the next step of configuration, move to related links 'Add Port Group' and click Go.

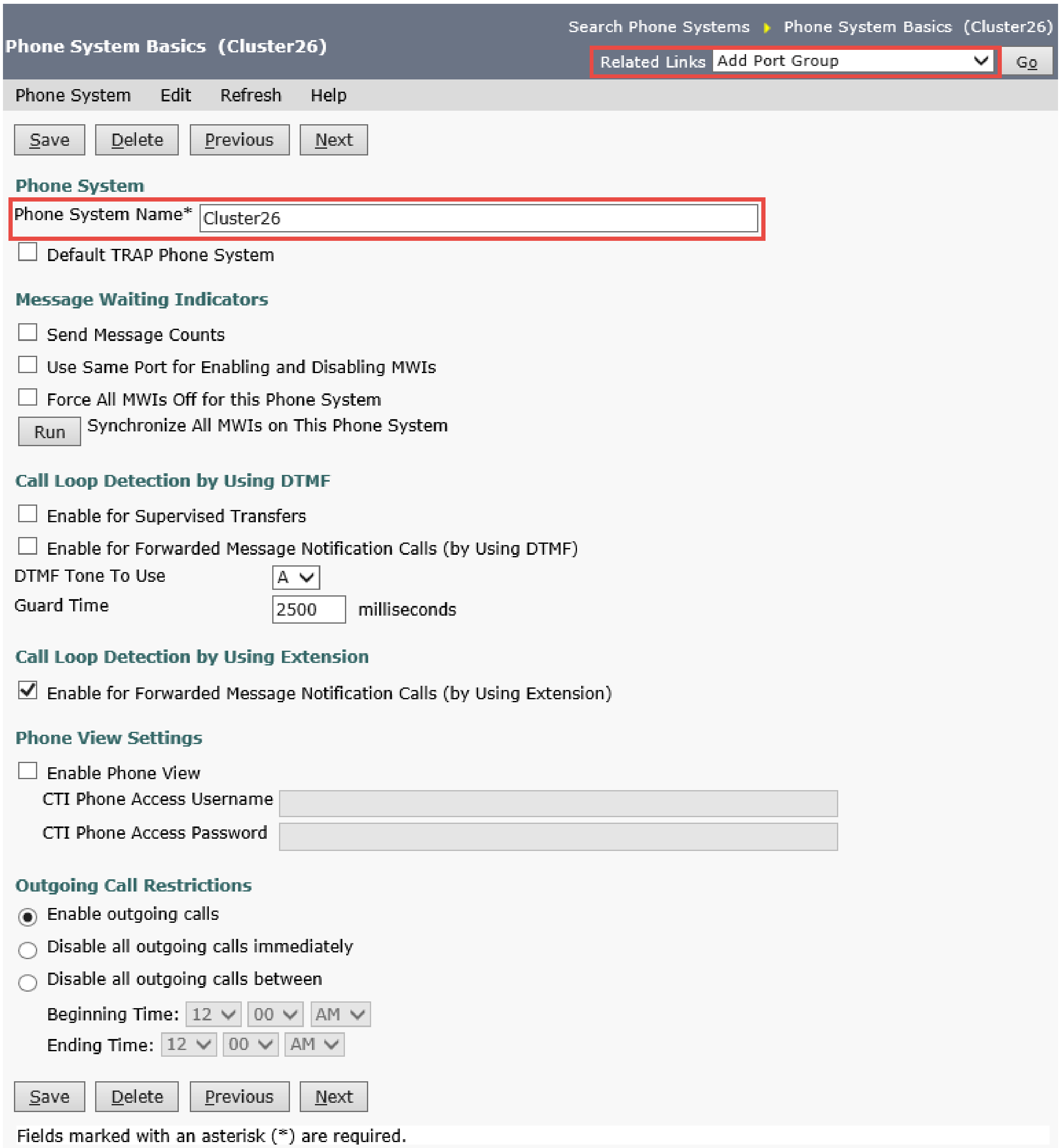


Figure 64: Unity Phone System Configuration

Port Group Configuration

Navigation: Telephony Integrations -> Port Group

1. Set Phone System: Select the phone system configured in the above section
2. Set Create From Port Group Type: SIP
3. Set SIP Security Profile: 5060
4. Set SIP Transport Protocol: TCP

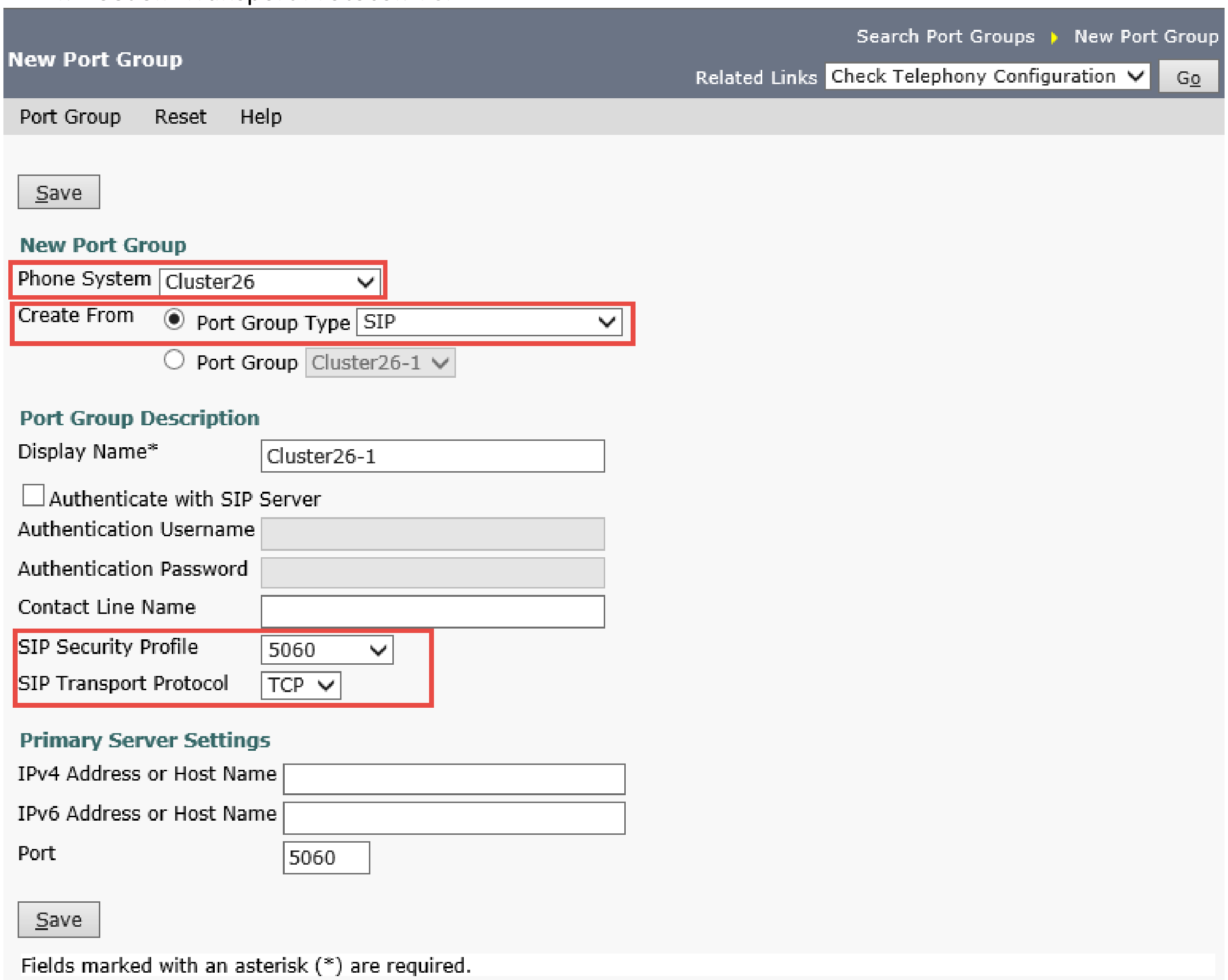


Figure 65: Port Group Configuration

Port Configuration

Navigation: Telephony Integrations -> Port

1. Set Port Name: Enter Name for the port
2. Set Phone System: Select the phone system which you are creating the ports to
3. Set Port Group: Select the port group configured to the phone system selected in the above step
4. Set Server: Select the unity server

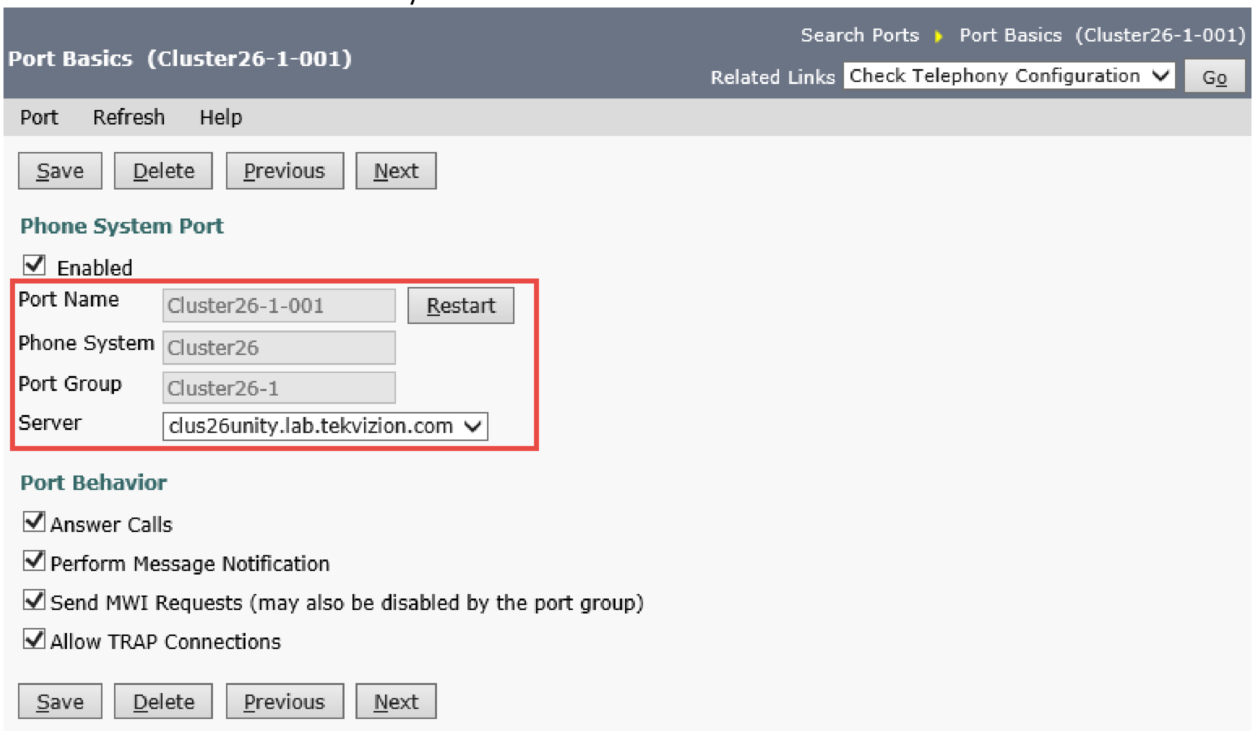


Figure 66: Port Configuration

Cisco Unity User Configuration

Navigation: Users -> Users

1. Set Alias: Enter a name for the user
2. Set Extension: Enter the extension you are configuring this user for
3. Set Phone System: Select the phone system you configured under *Telephony Integrations-> Phone System*

Cisco User Configuration

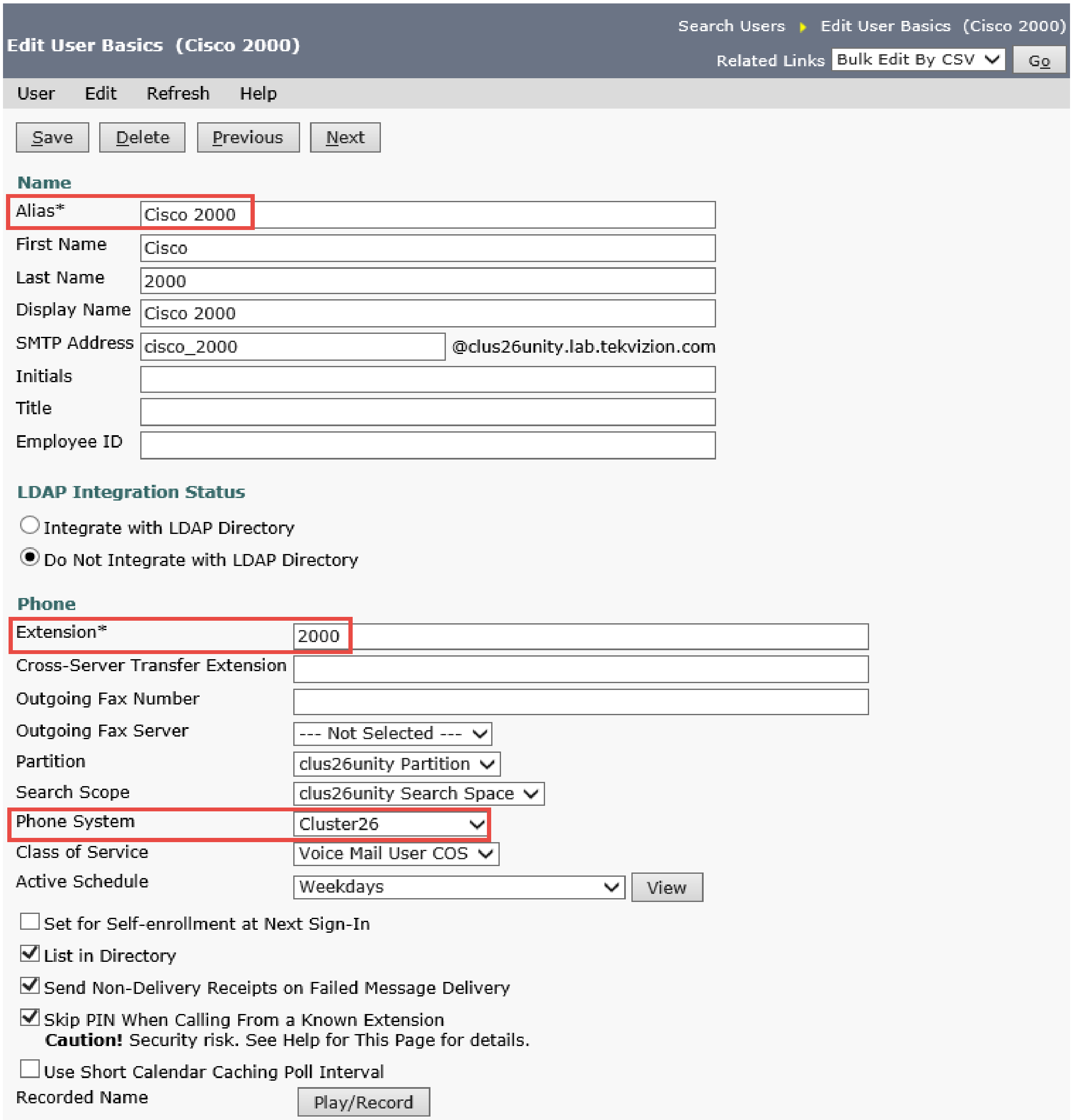


Figure 67: Unity Cisco User Configuration-1

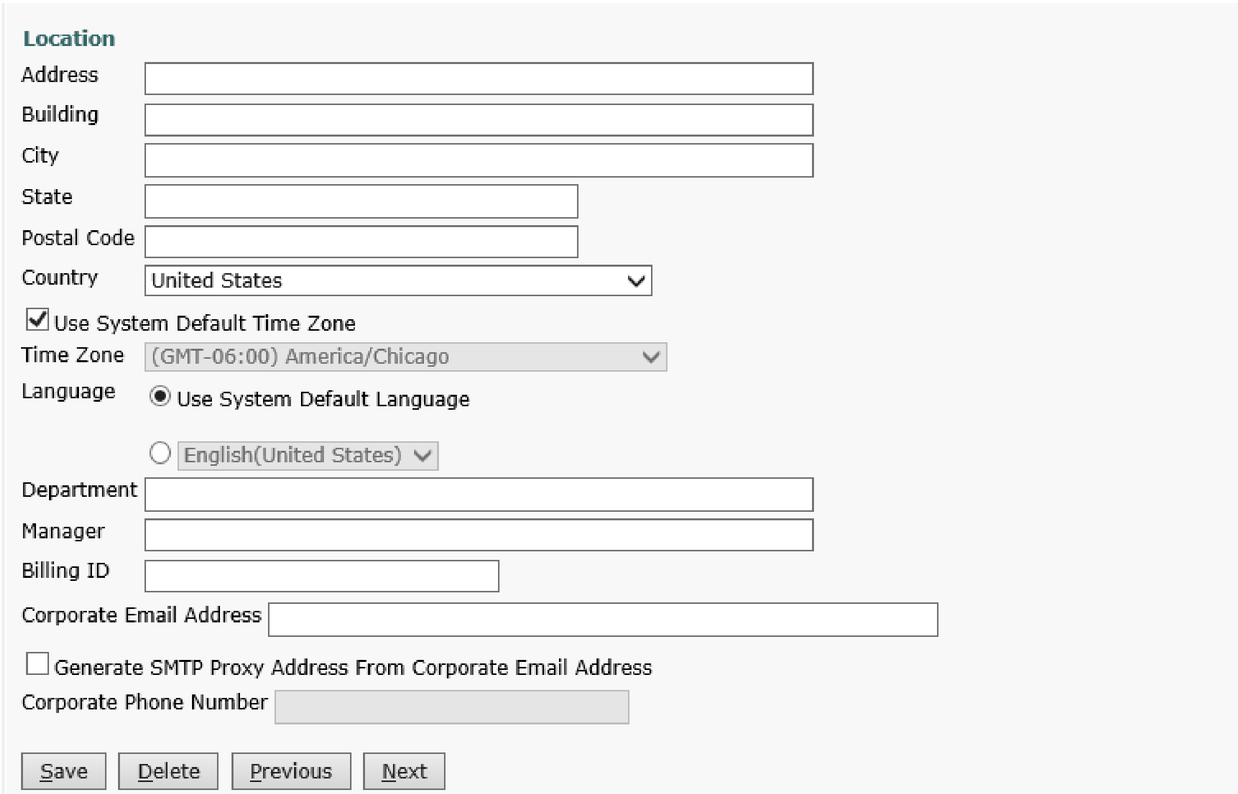


Figure 68: Unity Cisco User Configuration-2

Lync User Configuration

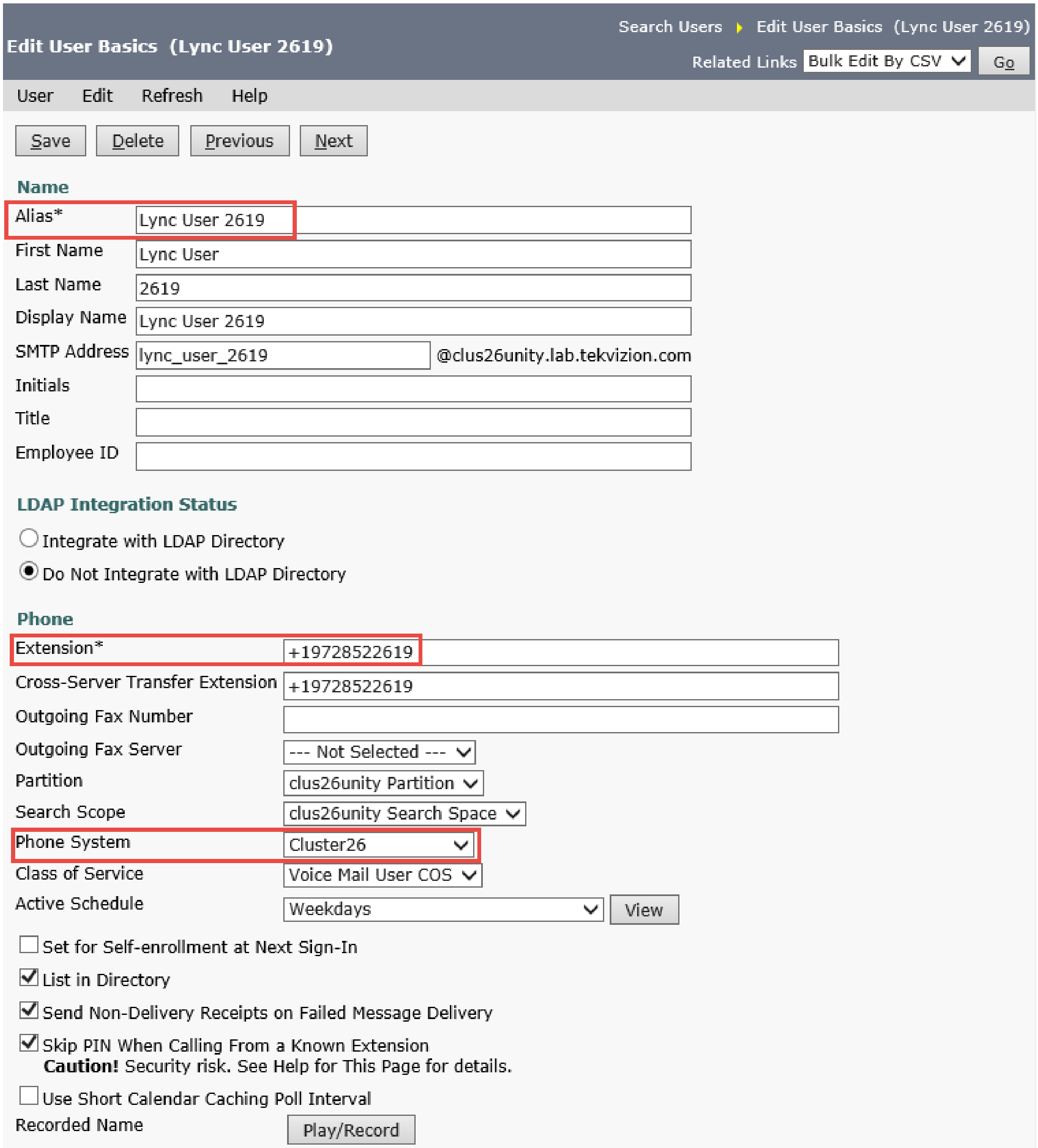


Figure 69: Unity Lync User Configuration-1

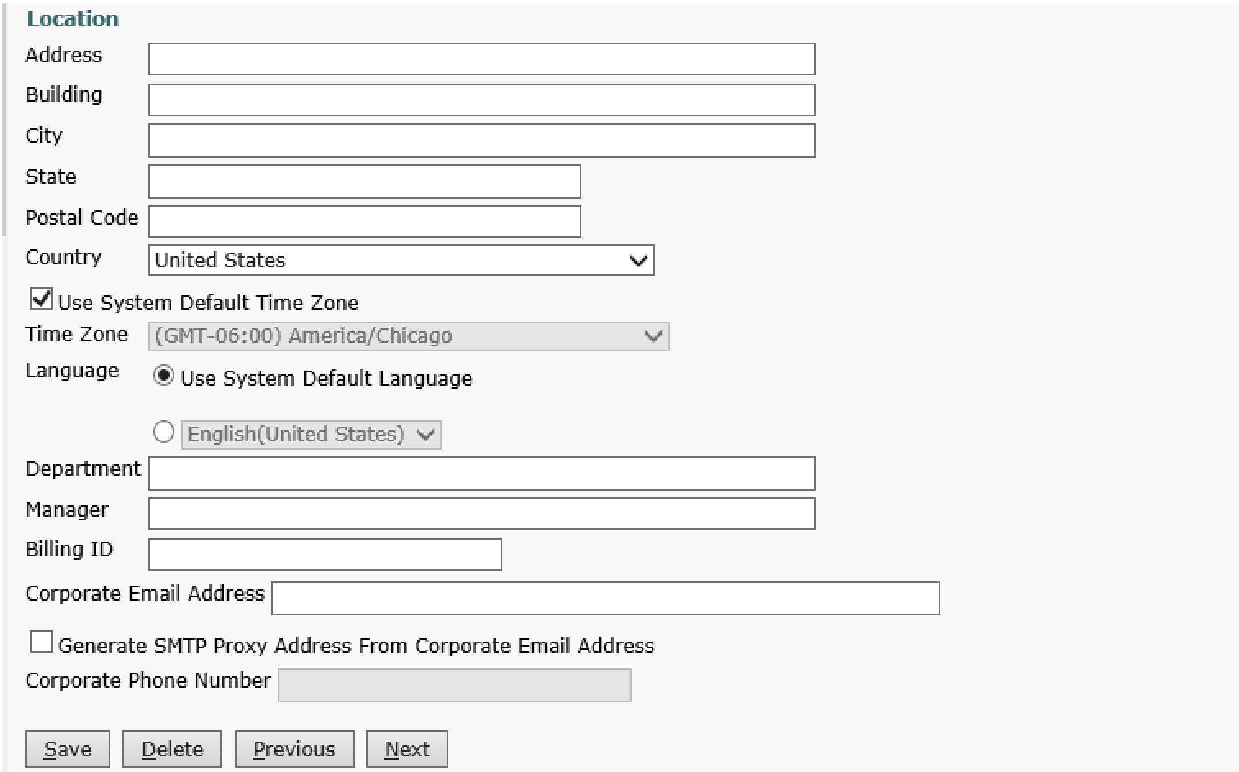


Figure 70: Unity Lync User Configuration-2

Gateway Configuration

```
version 15.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
service sequence-numbers
!
hostname cisco-lync-gw
!
boot-start-marker
boot system flash c3900e-universalk9-mz.SPA.154-1.T.bin
boot-end-marker
!
aqm-register-fnf
!
card type t1 0 0
logging buffered 999999999
no logging rate-limit
no logging console
enable secret 4 sKPgCY/XPea3wk8xoeSWo7UGFaNVwzXDEyXWhuDjeLk
!
no aaa new-model
!
network-clock-participate wic 0
network-clock-select 1 T1 0/0/0
!
ip domain name lab.tekvizion.com
ip name-server <name-server IP address>
ip cef
no ipv6 cef
!
multilink bundle-name authenticated
!
isdn switch-type primary-qsig
!
voice-card 0
 dsp services dspfarm
!
voice service voip
 ip address trusted list
  ipv4 0.0.0.0 0.0.0.0
 sip
  session transport tcp
  header-passing
  asserted-id pai
!
```

```
voice class uri 1 sip
 host cisco-lync-gw.lab.tekvizion.com
voice class codec 1
 codec preference 1 g711ulaw
 codec preference 2 g711alaw
!
controller T1 0/0/0
 cablelength long 0db
 pri-group timeslots 1-24 service mgcp
!
controller T1 0/0/1
 cablelength long 0db
!
controller T1 0/0/2
 cablelength long 0db
!
controller T1 0/0/3
 cablelength long 0db
!
interface GigabitEthernet0/0
 description $ETH-LAN$$ETH-SW-LAUNCH$$INTF-INFO-GE 0/0$
 ip address 10.64.3.69 255.255.0.0
 duplex auto
 speed auto
!
interface GigabitEthernet0/1
 no ip address
 shutdown
 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
!
interface Serial0/0/0:23
 no ip address
 encapsulation hdlc
 isdn switch-type primary-qsig
```

```
 isdn timer T310 120000
 isdn protocol-emulate network
 isdn incoming-voice voice
 isdn bind-l3 ccm-manager
 no cdp enable
!
ip default-gateway <default-gateway IP Address>
ip forward-protocol nd
!
ip http server
ip http access-class 23
ip http authentication local
ip http secure-server
ip http timeout-policy idle 60 life 86400 requests 10000
!
ip route 0.0.0.0 0.0.0.0 10.64.1.1
!
nls resp-timeout 1
cpd cr-id 1
!
control-plane
!
voice-port 0/0/0:23
!
voice-port 0/1/0
!
voice-port 0/1/1
!
voice-port 0/1/2
!
voice-port 0/1/3
 !
mgcp
mgcp call-agent clus26sub1 service-type mgcp version 0.1
mgcp dtmf-relay voip codec all mode nte-ca
mgcp rtp unreachable timeout 1000 action notify
mgcp modem passthrough voip mode nse
mgcp voice-quality-stats
mgcp package-capability rtp-package
mgcp package-capability sst-package
mgcp package-capability pre-package
mgcp package-capability fm-package
no mgcp package-capability res-package
no mgcp package-capability fxr-package
no mgcp timer receive-rtcp
mgcp sdp simple
mgcp fax t38 ecm
```

```
mgcp fax t38 inhibit
mgcp rtp payload-type g726r16 static
mgcp bind control source-interface GigabitEthernet0/0
mgcp bind media source-interface GigabitEthernet0/0
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 rtp payload-type nte 101
!
mgcp profile default
!
ccm-manager music-on-hold
!
ccm-manager mgcp
no ccm-manager fax protocol cisco
ccm-manager config server clus26sub1
ccm-manager config
!
sip-ua
 set pstn-cause 31 sip-status 480
 timers expires 1800000
!
gatekeeper
 shutdown
!
line con 0
 login local
line aux 0
line vty 0 4
 access-class 23 in
 exec-timeout 0 0
 privilege level 15
 login local
 transport input telnet ssh
line vty 5 15
 access-class 23 in
 exec-timeout 0 0
 privilege level 15
 login local
 transport input telnet ssh
!
scheduler allocate 20000 1000
!
end
```

Cisco Router[External IOS MTP] Configuration

```
sccp local GigabitEthernet0/1
sccp ccm 10.80.16.3 identifier 1 priority 1 version 7.0
sccp
!
sccp ccm group 40
 description EXT MTP
 bind interface GigabitEthernet0/1
 associate ccm 1 priority 1
 associate profile 40 register EXTMTP
!
dspfarm profile 40 mtp
 codec g711ulaw
 maximum sessions software 20
 associate application SCCP
!
```

Lync Server Configuration

Add CISCO UCM to Lync Topology

Lync recognizes CISCO UCM as a PSTN gateway connected by SIP trunk. So we need to add CISCO UCM 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 UCM to the PSTN gateway topology

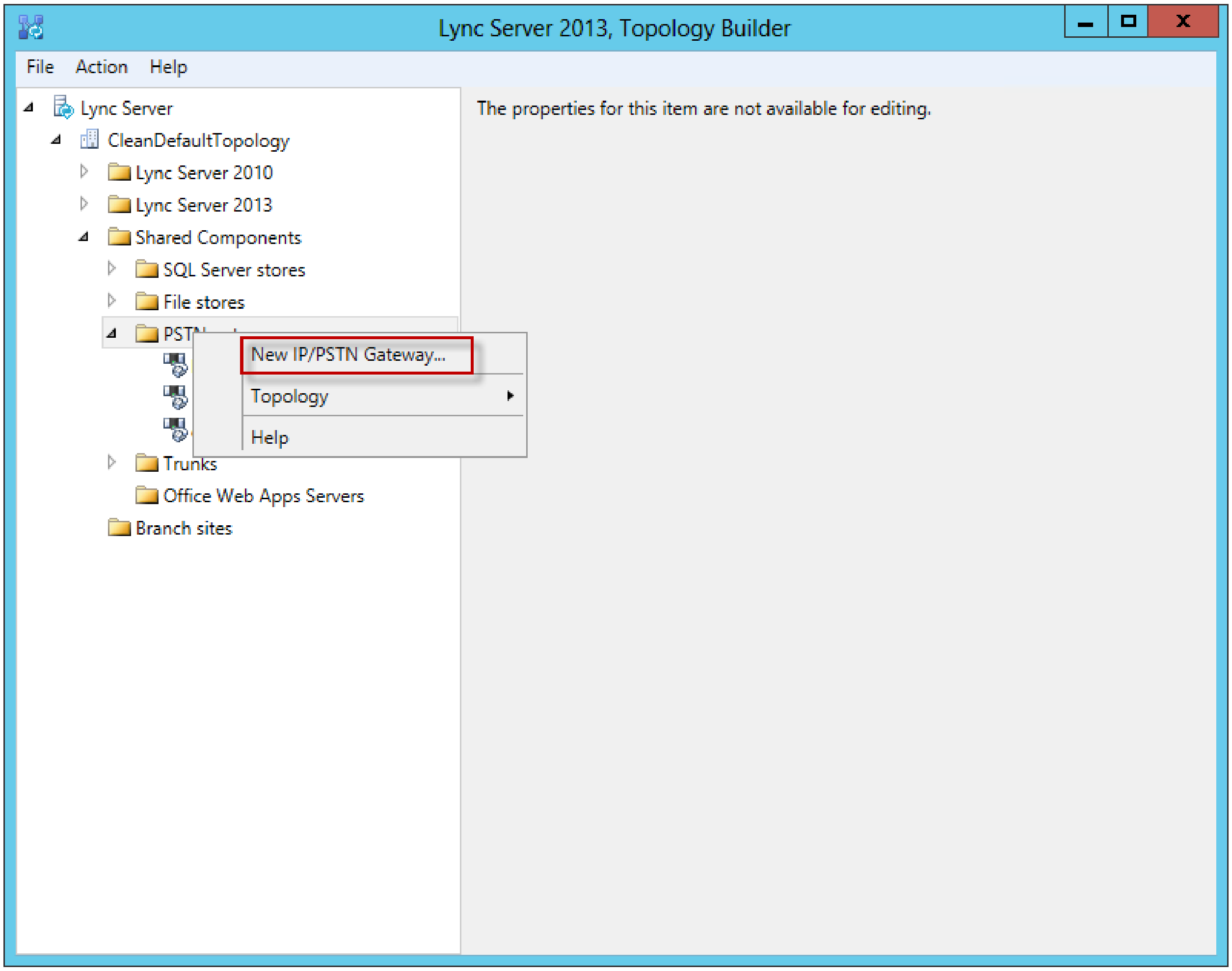


Figure 71: Configure PSTN Gateway -1

2. Set **FQDN**: This is the IP Address or FQDN of the CISCO UCM.

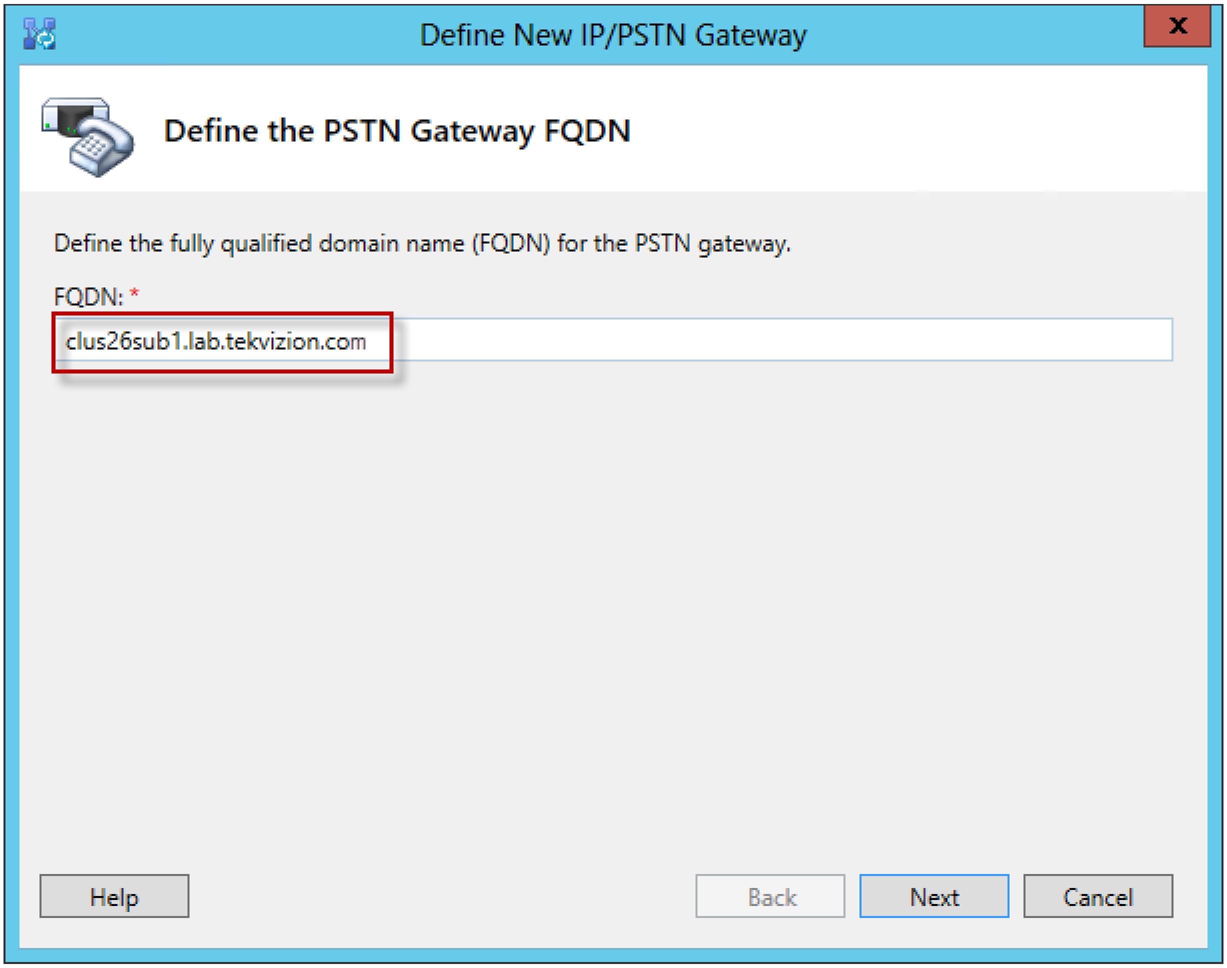
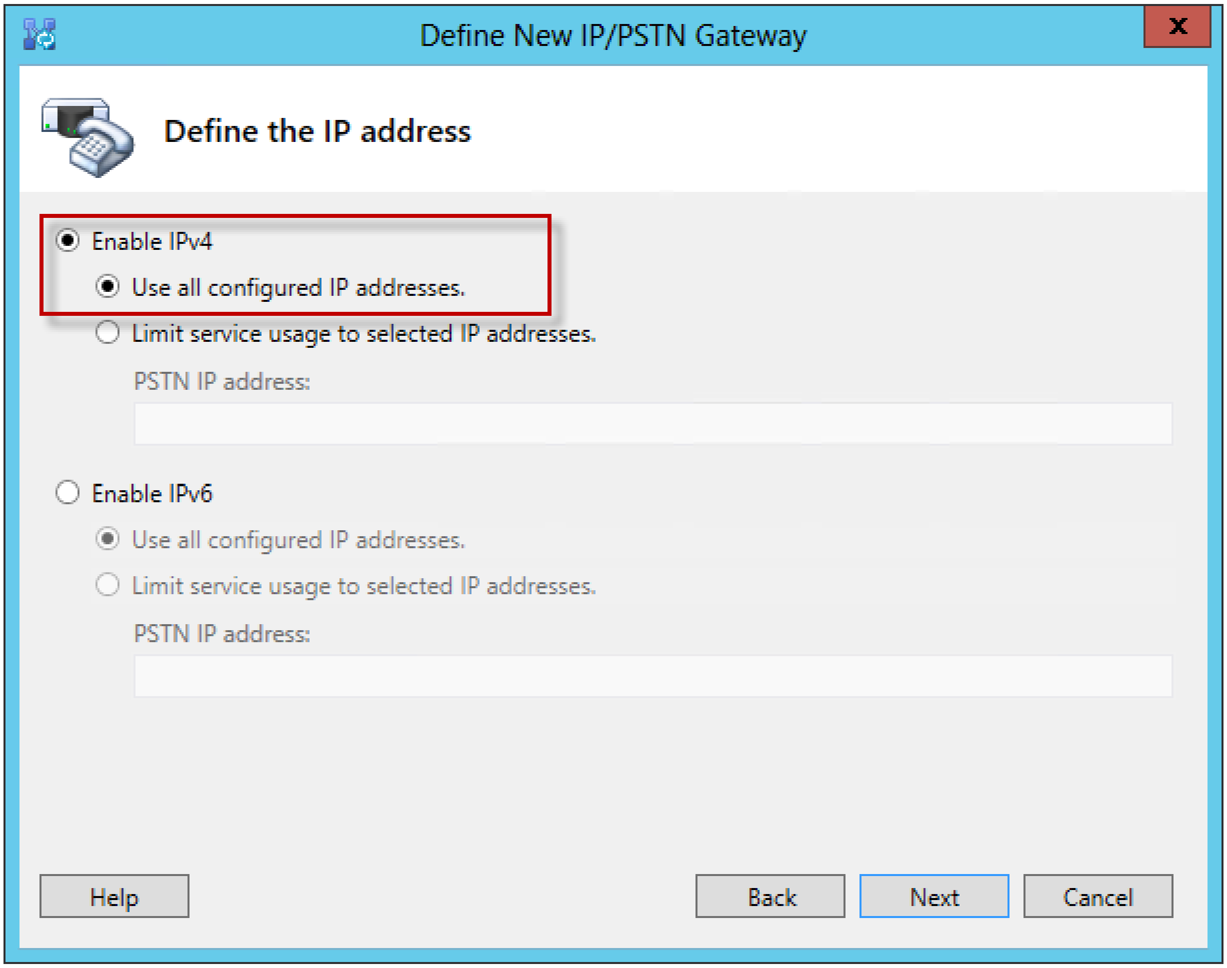


Figure 72: Configure PSTN Gateway -2


Define New IP/PSTN Gateway
Define the IP address
Enable IPv4
Use all configured IP addresses.
Limit service usage to selected IP addresses.
PSTN IP address:
Enable IPv6
Use all configured IP addresses.
Limit service usage to selected IP addresses.
PSTN IP address:
Help
Back
Next
Cancel

3. Set **Trunk Name**: This is the FQDN of the CISCO UCM
4. Set **Listening port for IP/PSTN gateway**: This **Listening port** should match the **Incoming Port** setting in the CISCO UCM's **SIP Trunk Security Profile**.
5. Set **SIP Transport Protocol**: TCP
6. Set **Associate Mediation Server**: Assign this PSTN gateway to the Mediation Server. Medpool.lynclabkm2013.local is used here for example.

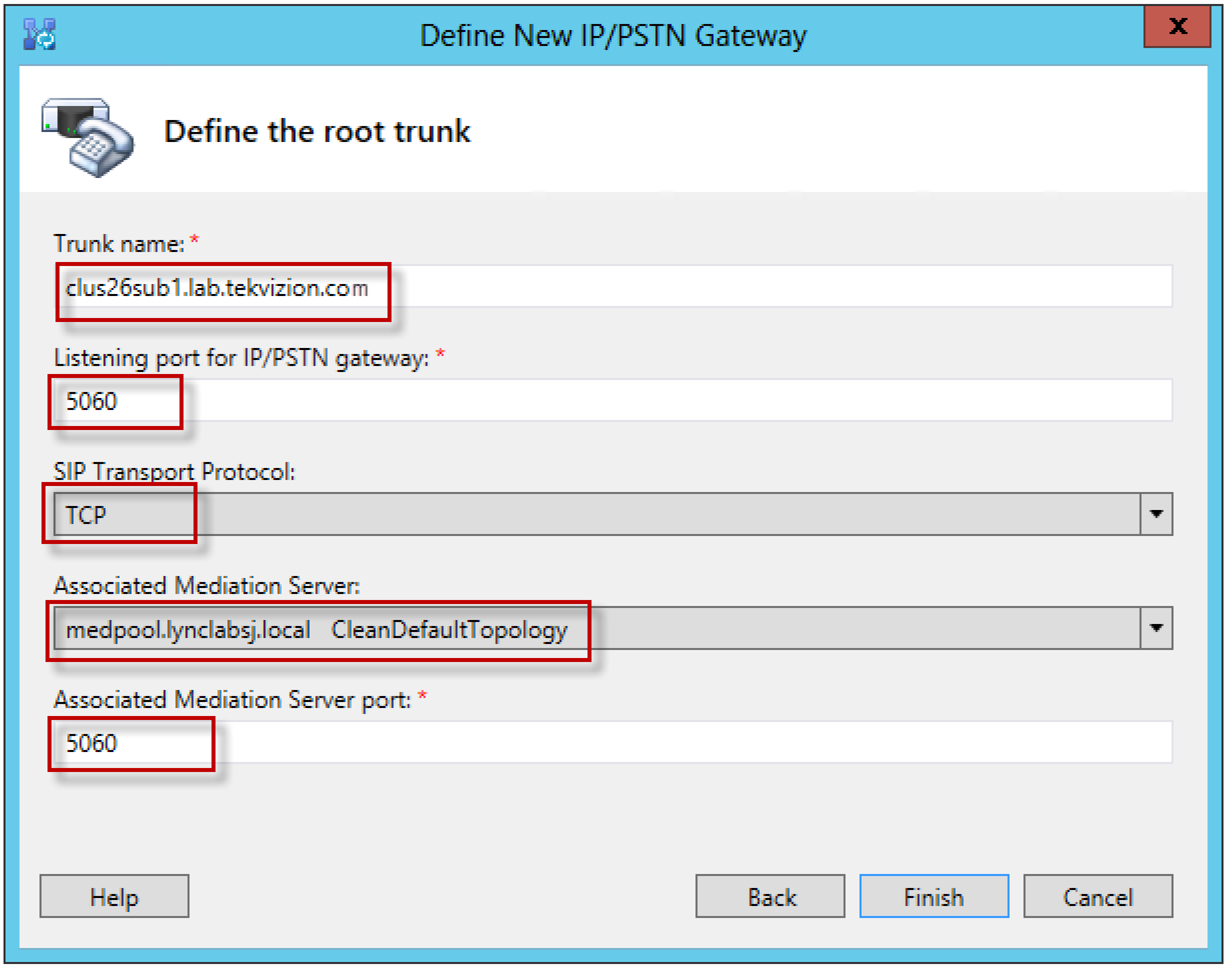


Figure 73: Configure PSTN Gateway -3

7. Publish topology to make the changes effective, refer to below scree capture for the process.

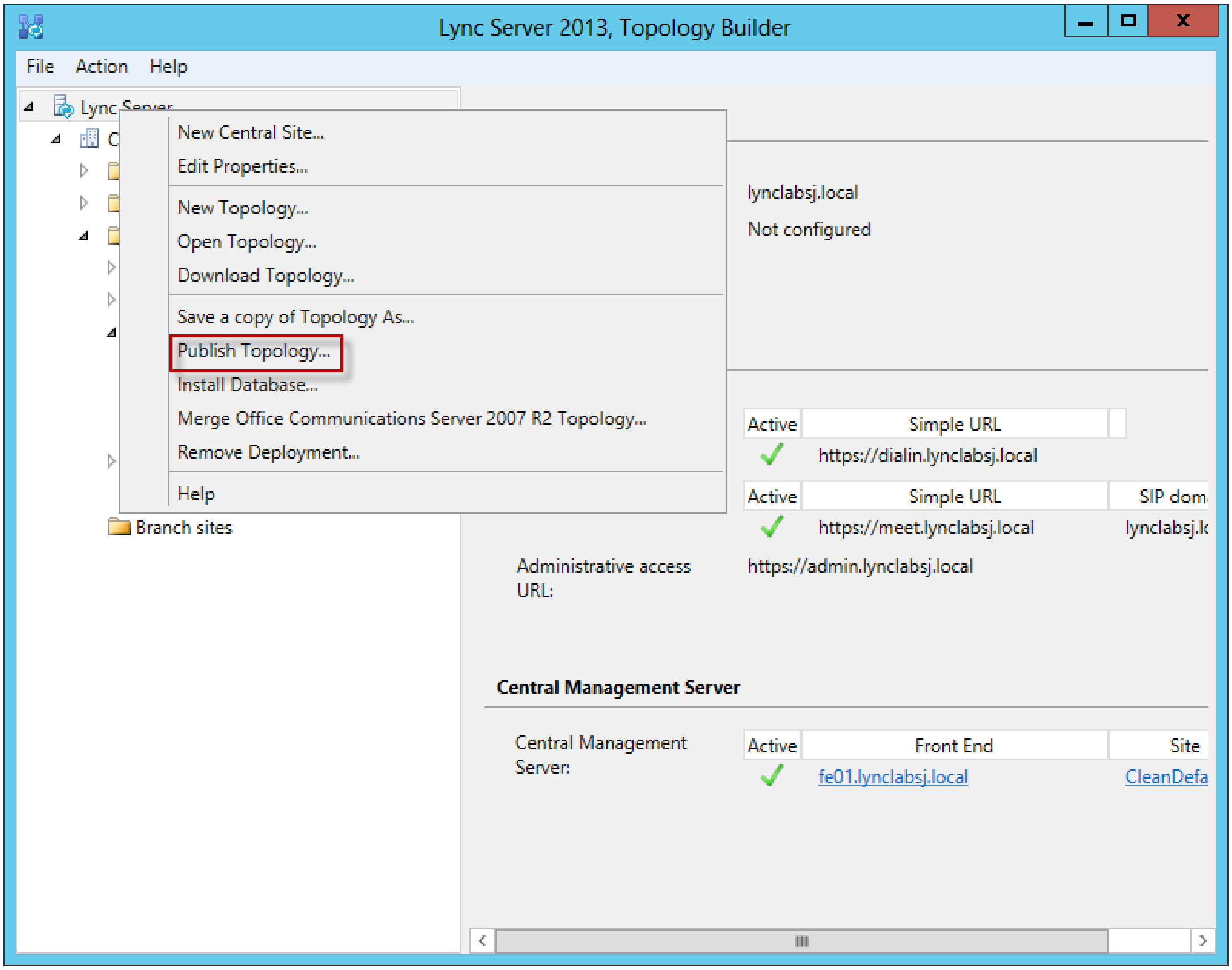


Figure 74: Publish Topology

Trunk Configuration

Navigation: Voice Routing -> Trunk Configuration

1. Create a **Pool Trunk** by selecting New
2. Select **Service**: Select the trunk to CISCO UCM you created in topology builder
3. Set **Maximum early dialogs supported**: 20
4. Set **Encryption support level**: Optional
5. Set **Refer Support**: Enable sending refer to the gateway
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

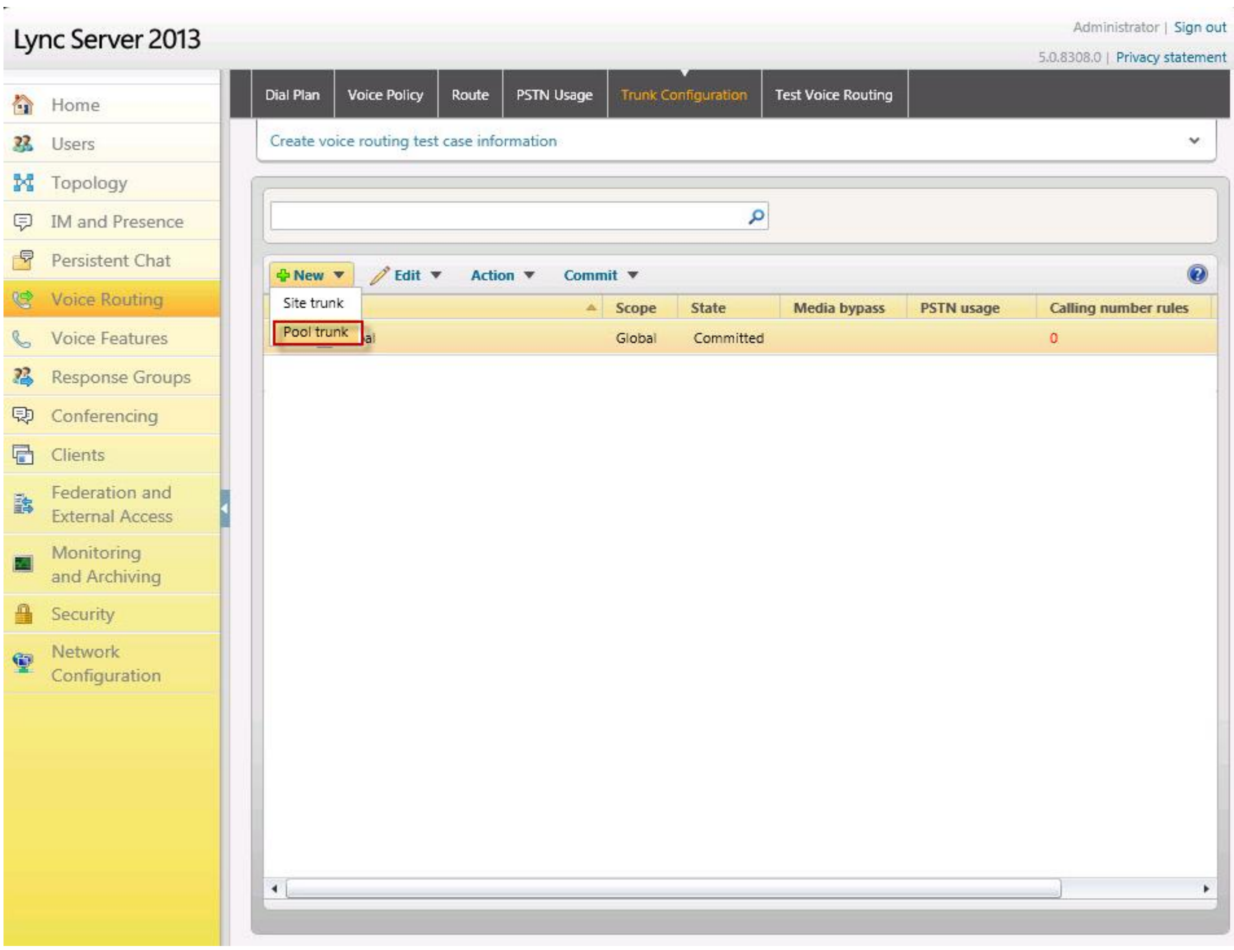


Figure 75: Trunk Configuration -1

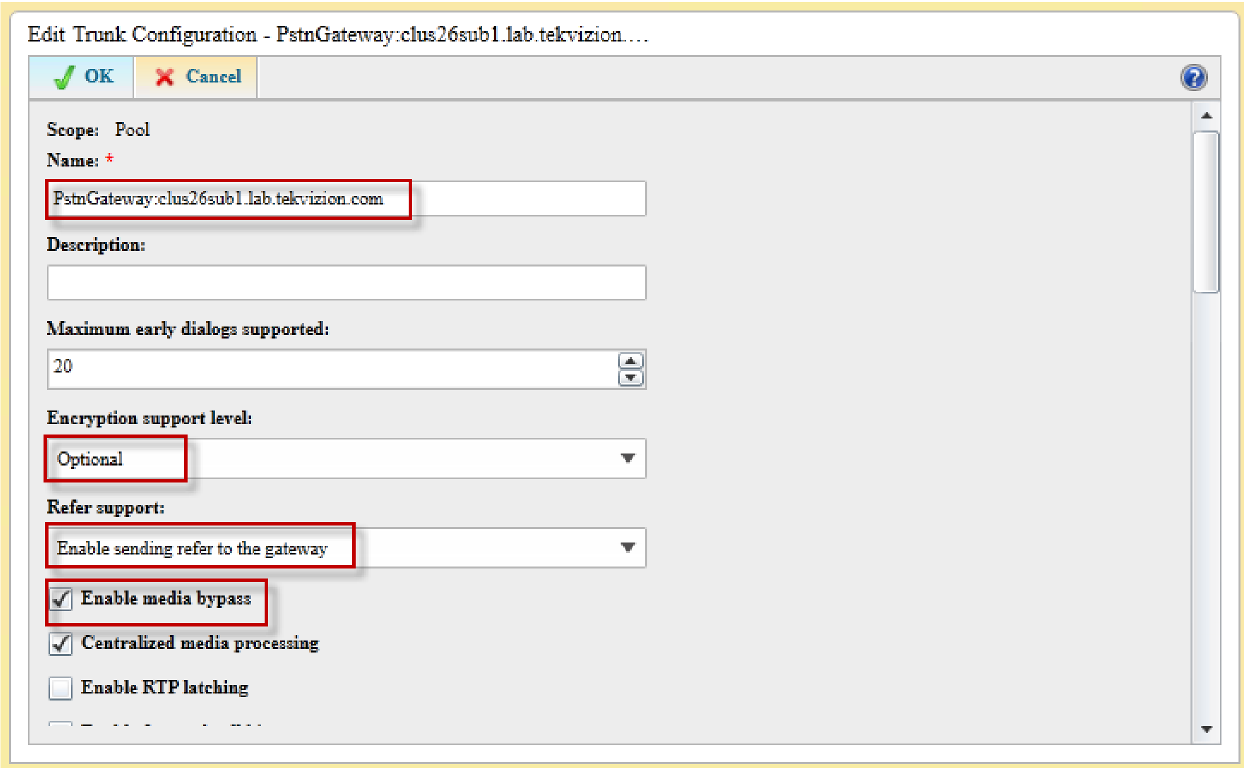


Figure 76: Trunk Configuration -2

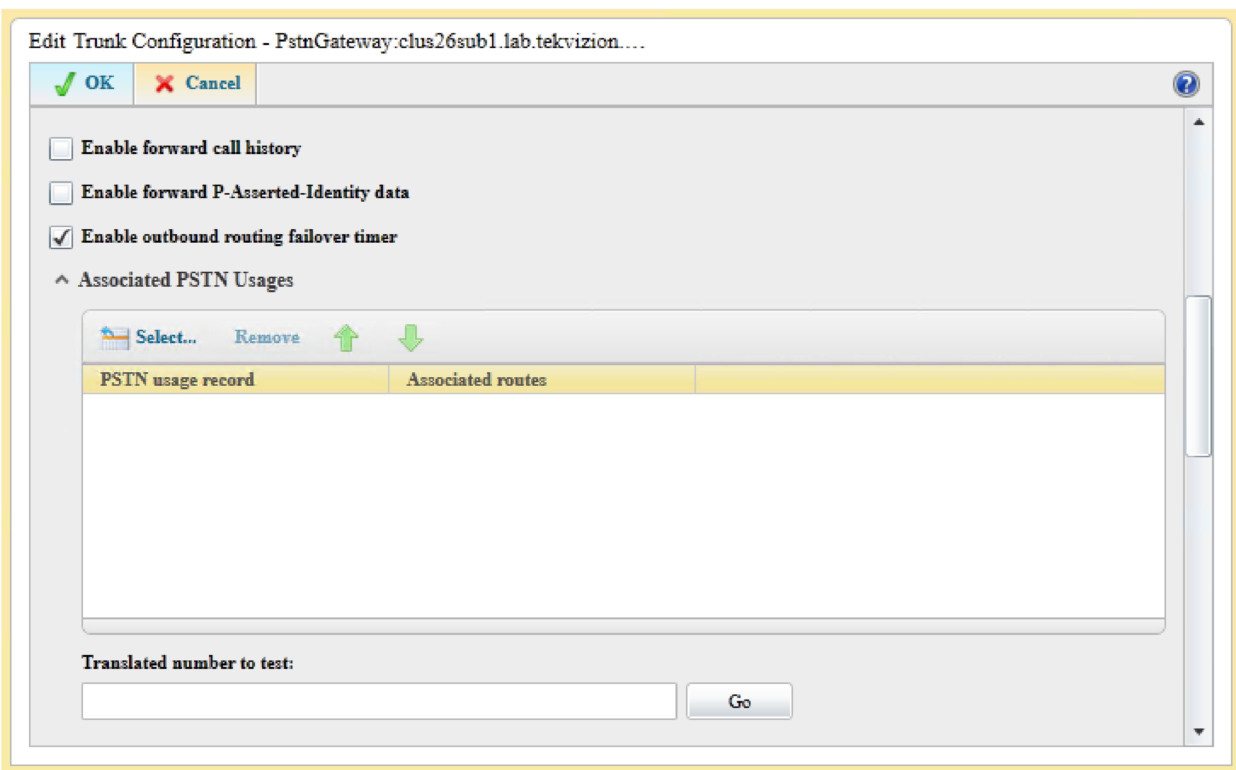


Figure 77: Trunk Configuration -3

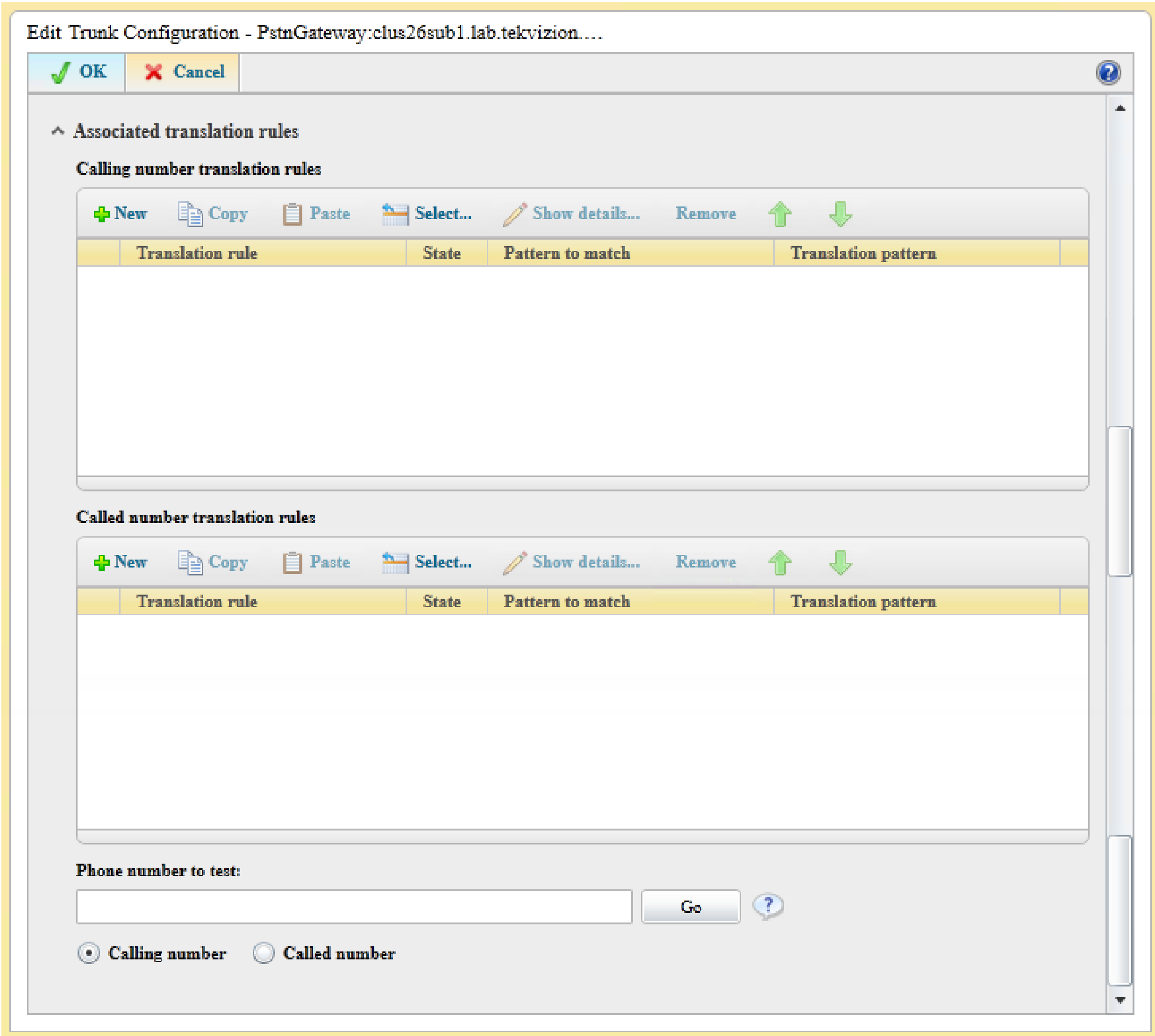


Figure 78: Trunk Configuration -4

Route

Navigation: Voice Routing -> Route

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

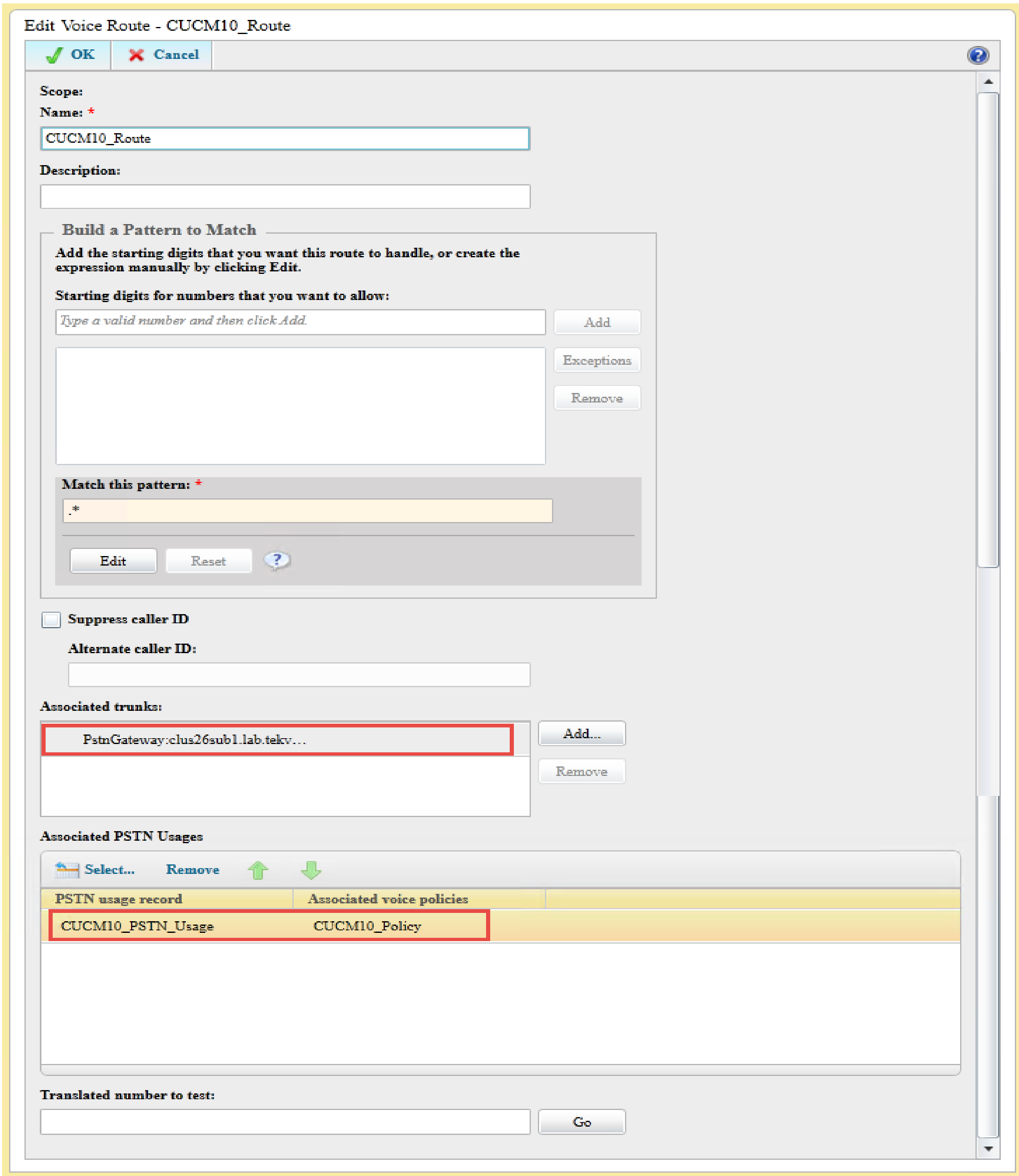


Figure 79: Route Configuration-1

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**:
    a. Enable call forwarding : Checked
    b. Enable delegation : Checked
    c. Enable call transfer : Checked
    d. Enable call park : Checked
    e. Enable simultaneous ringing of phones : Checked
    f. Enable team call : Checked
    g. Enable PSTN reroute : Checked
    h. Enable bandwidth policy override : Unchecked
    i. Enable malicious call tracing : Unchecked
4. Set **Associated PSTN Usages**:
    a. Select New to create a new PSTN Usage
    b. Set **Name**: Enter a name for this PSTN Usage
    c. Set **Associated Routes**: Select the route you created under Voice Routing -> Route

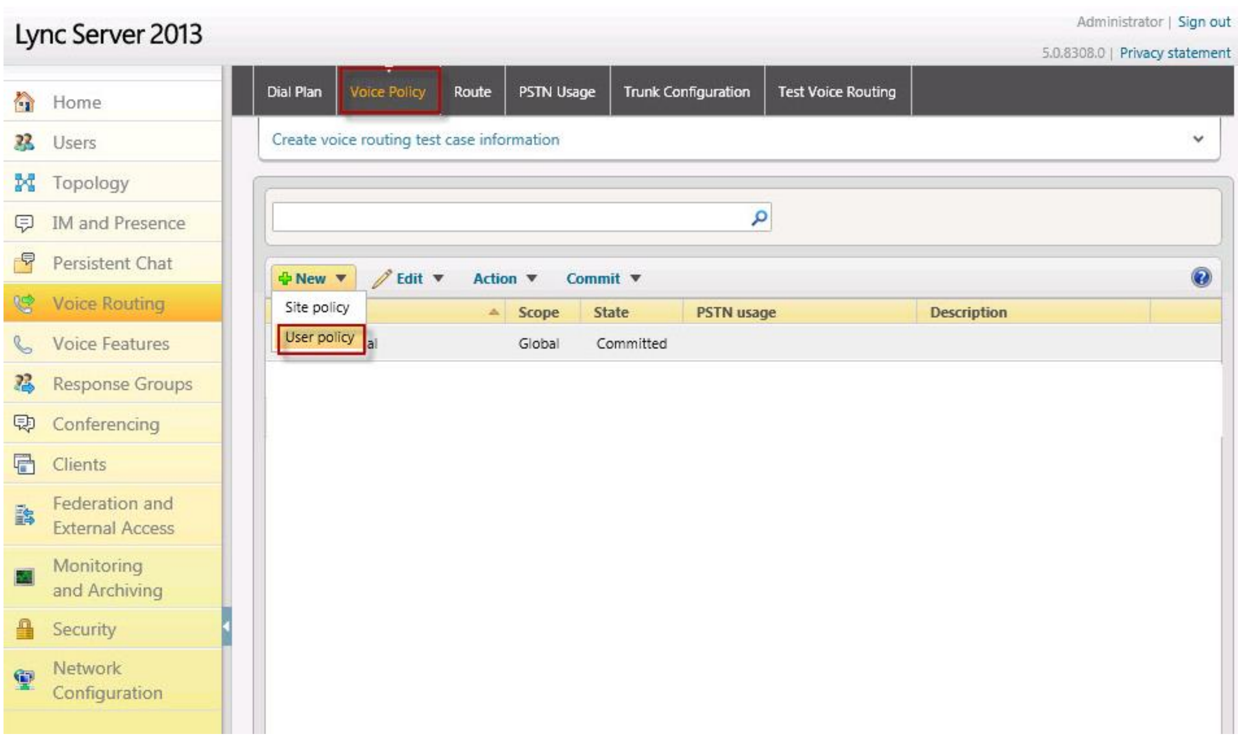


Figure 80: Voice Policy -1

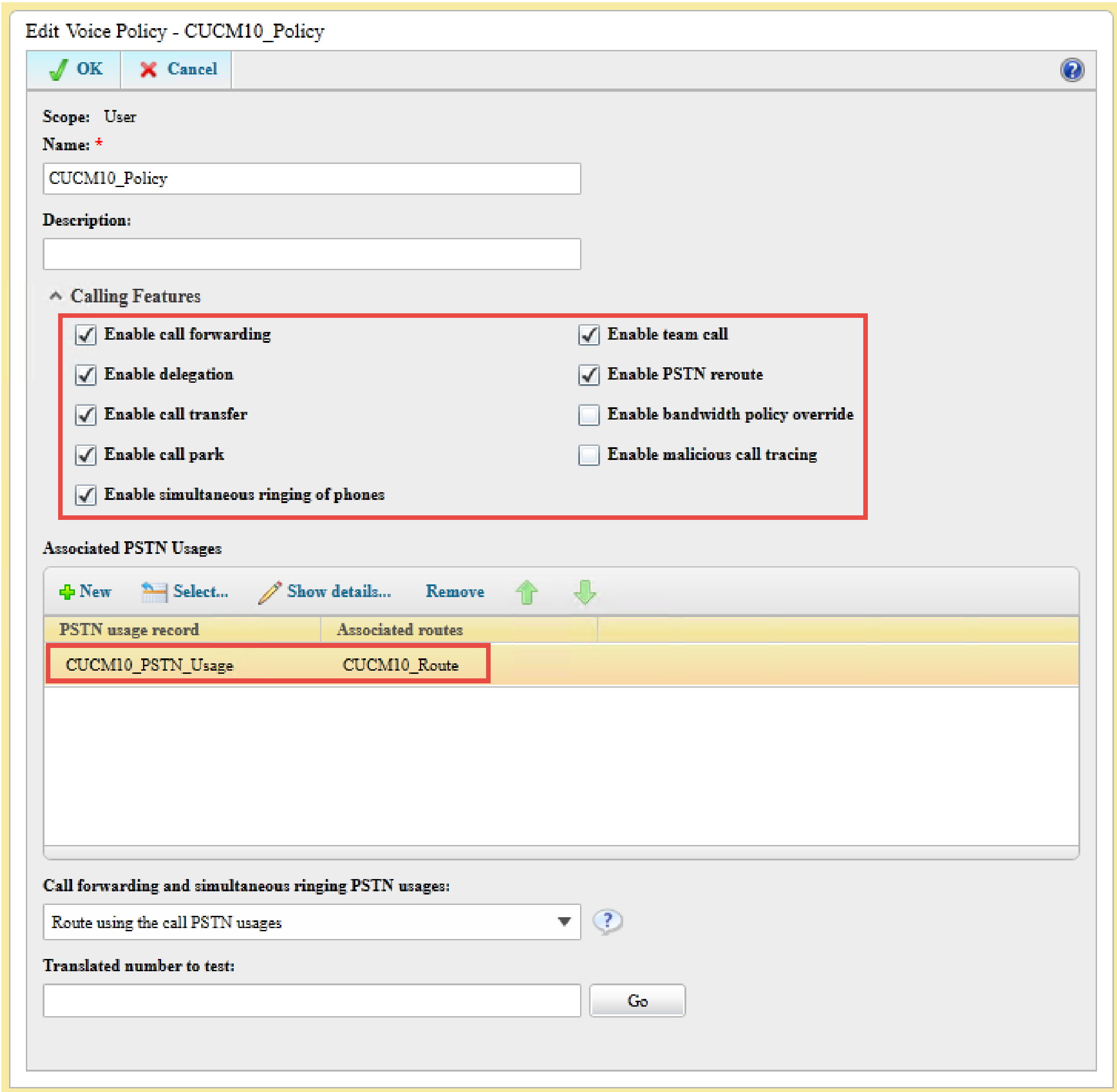


Figure 81: Voice Policy -2

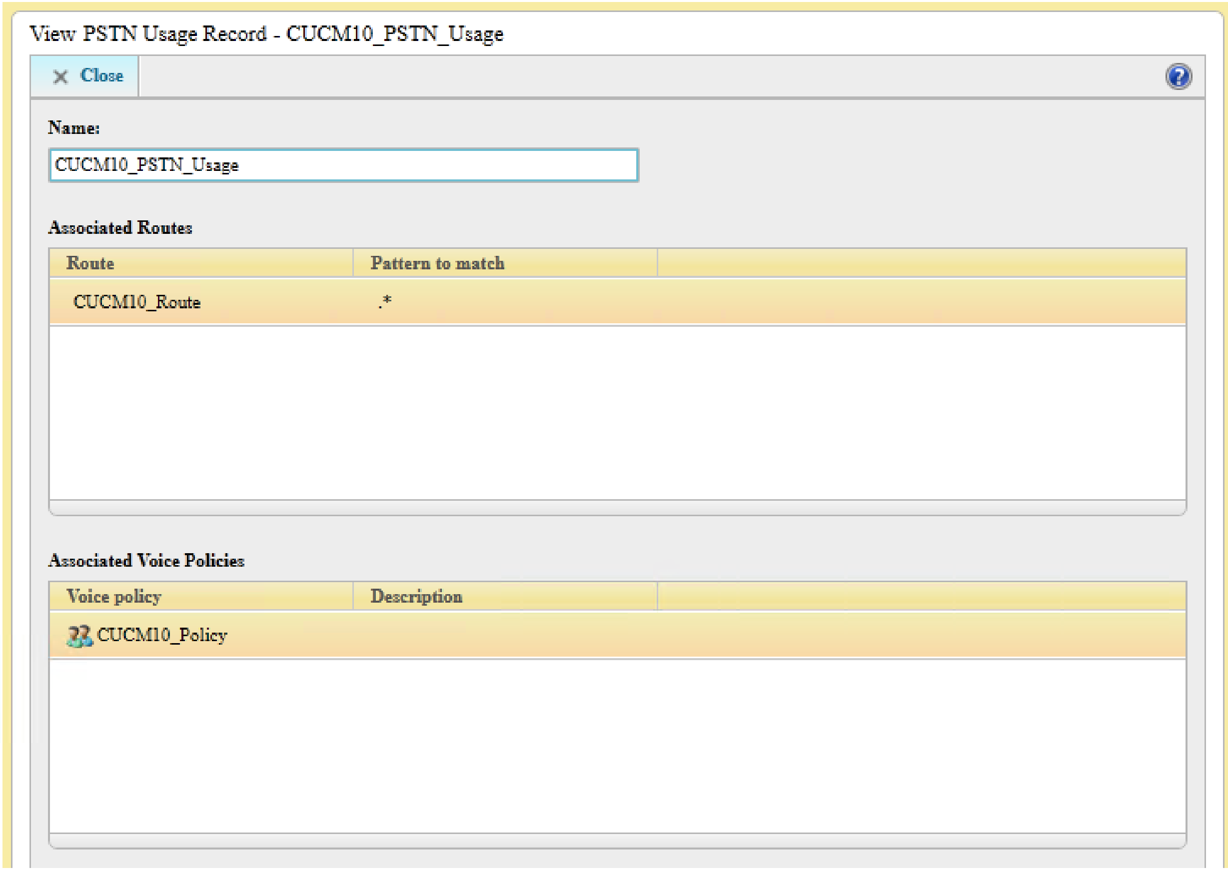


Figure 82: PSTN Usage

Dial Plan

Navigation: Voice Routing-> Dial Plan

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

1. Under Dial plan tab, select New (site/pool/user) dial plan or modify the existing Global dial plan
2. Configure the Normalization rules for 10-digit, 4-digit and 3-digit dialing as shown in the below screen shots.

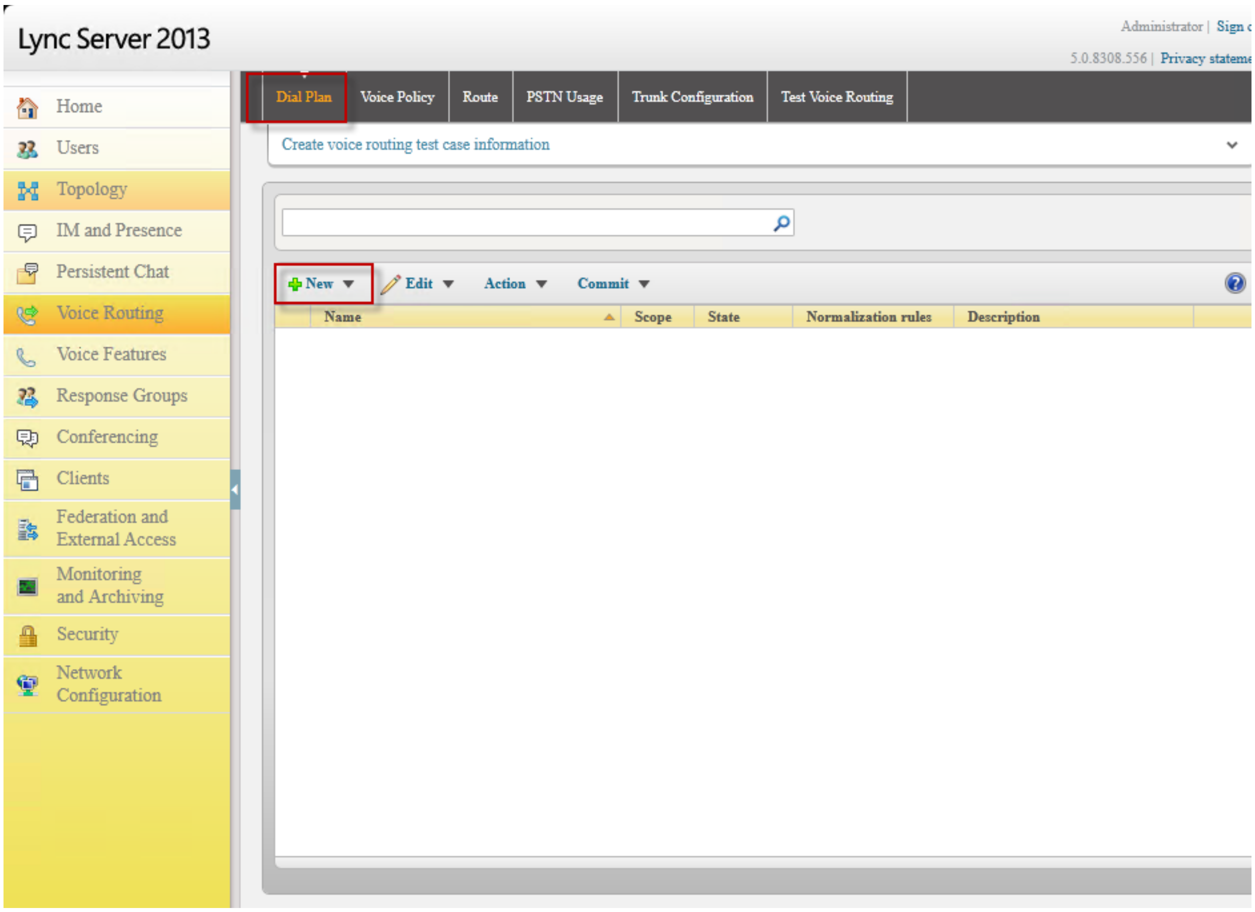


Figure 83: Dial Plan-1

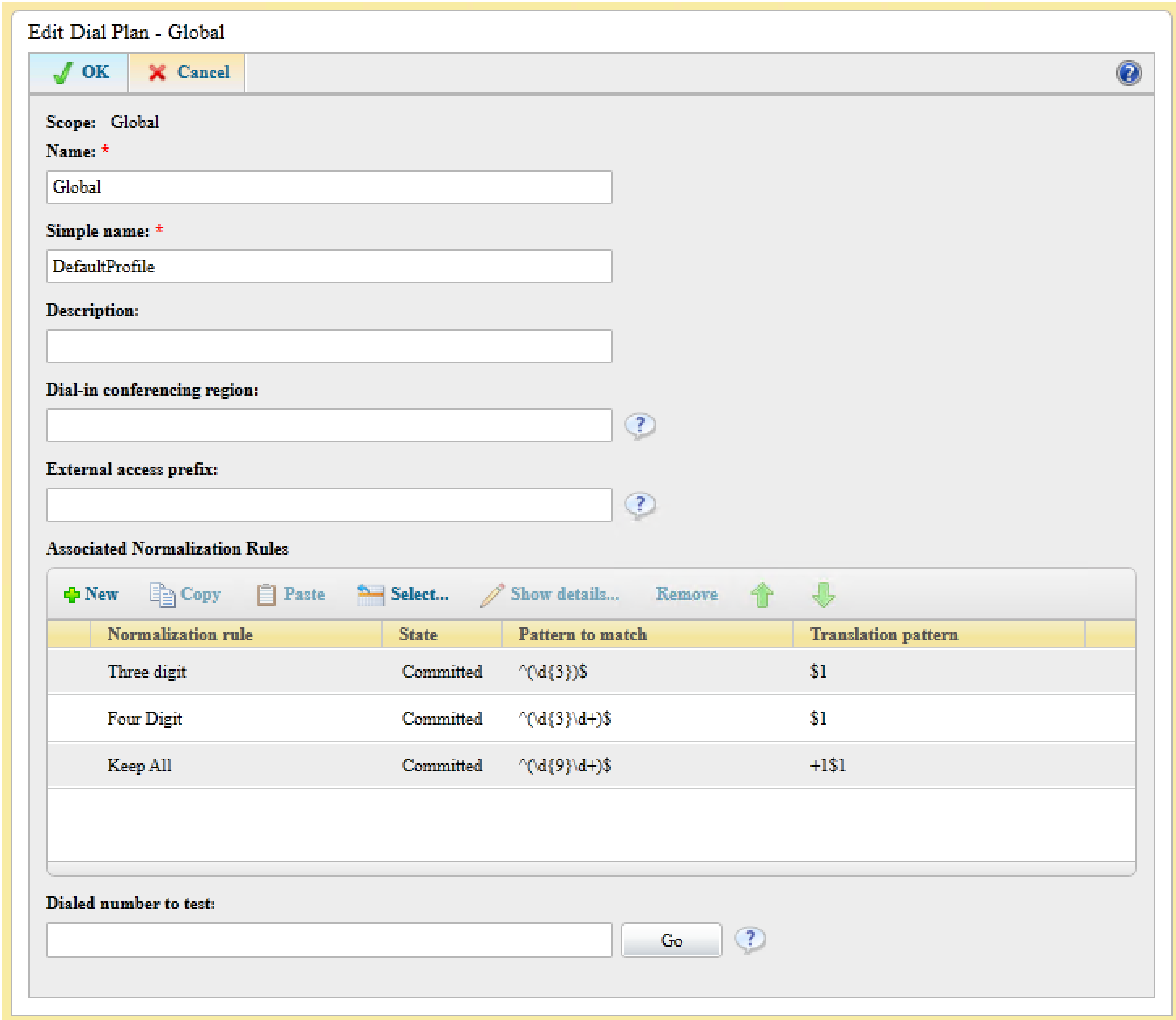


Figure 84: Dial Plan-2

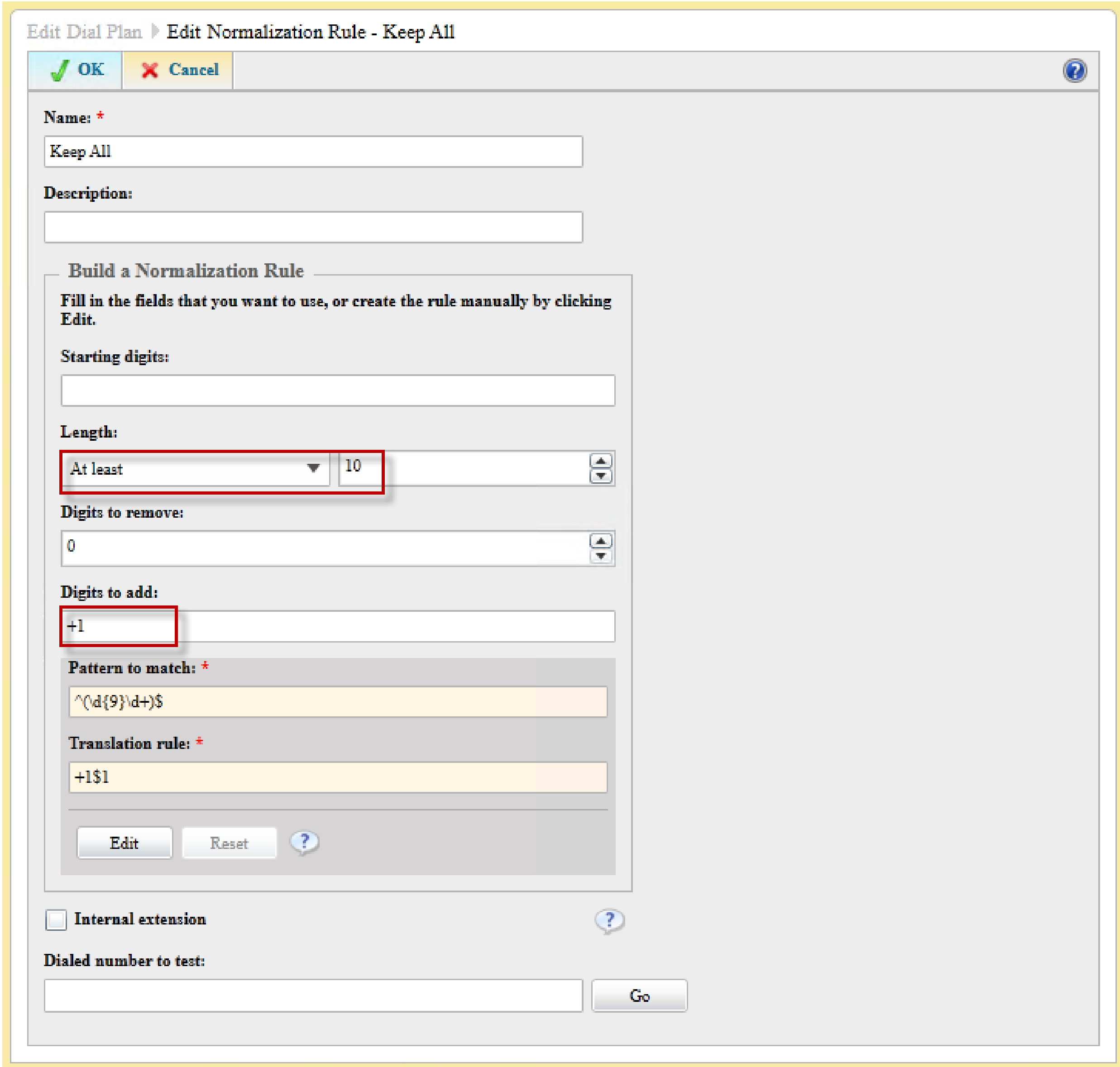


Figure 85: Normalization Rule for 10-digit dialing

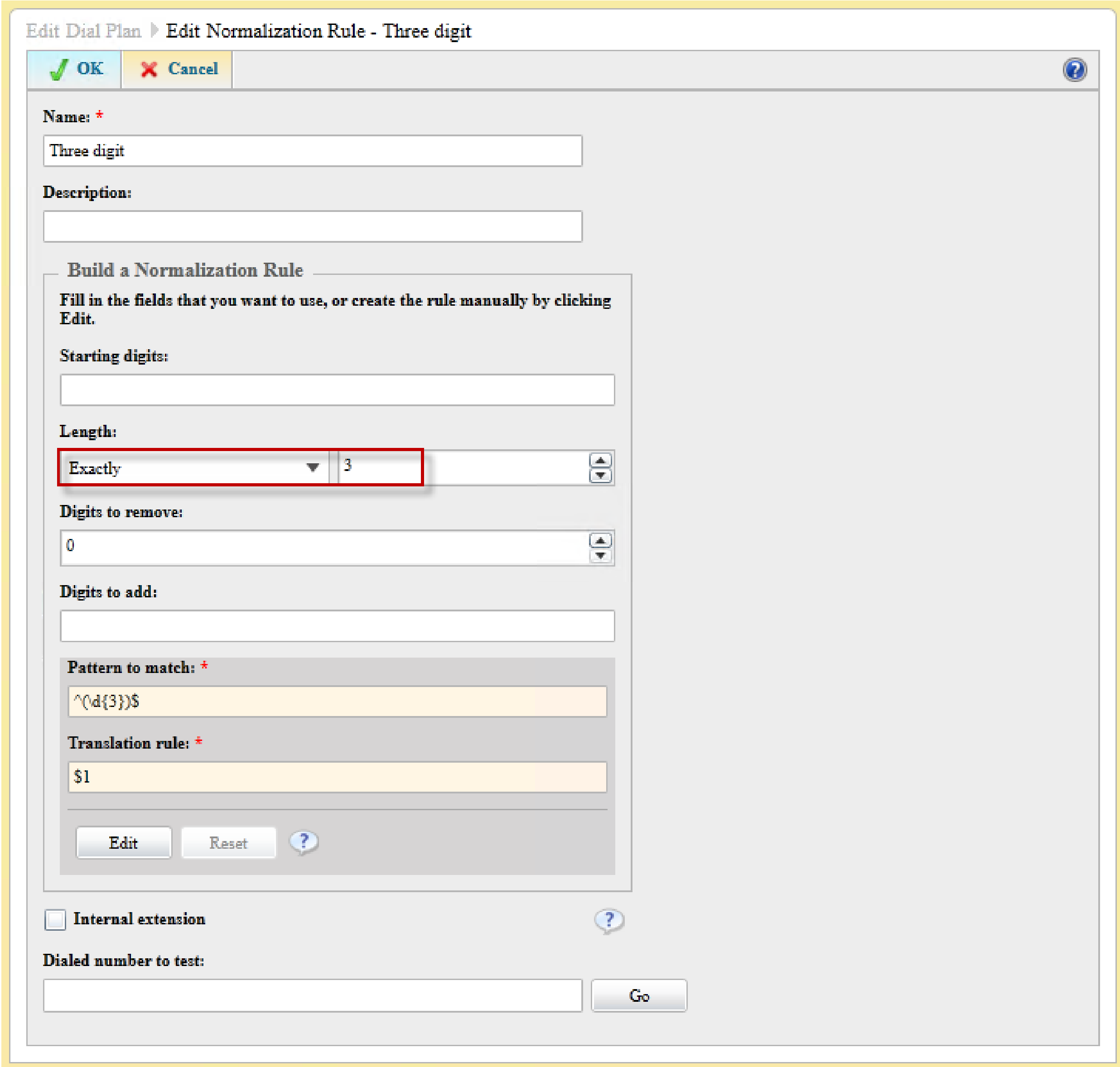


Figure 86: Normalization Rule for 3-digit dialing

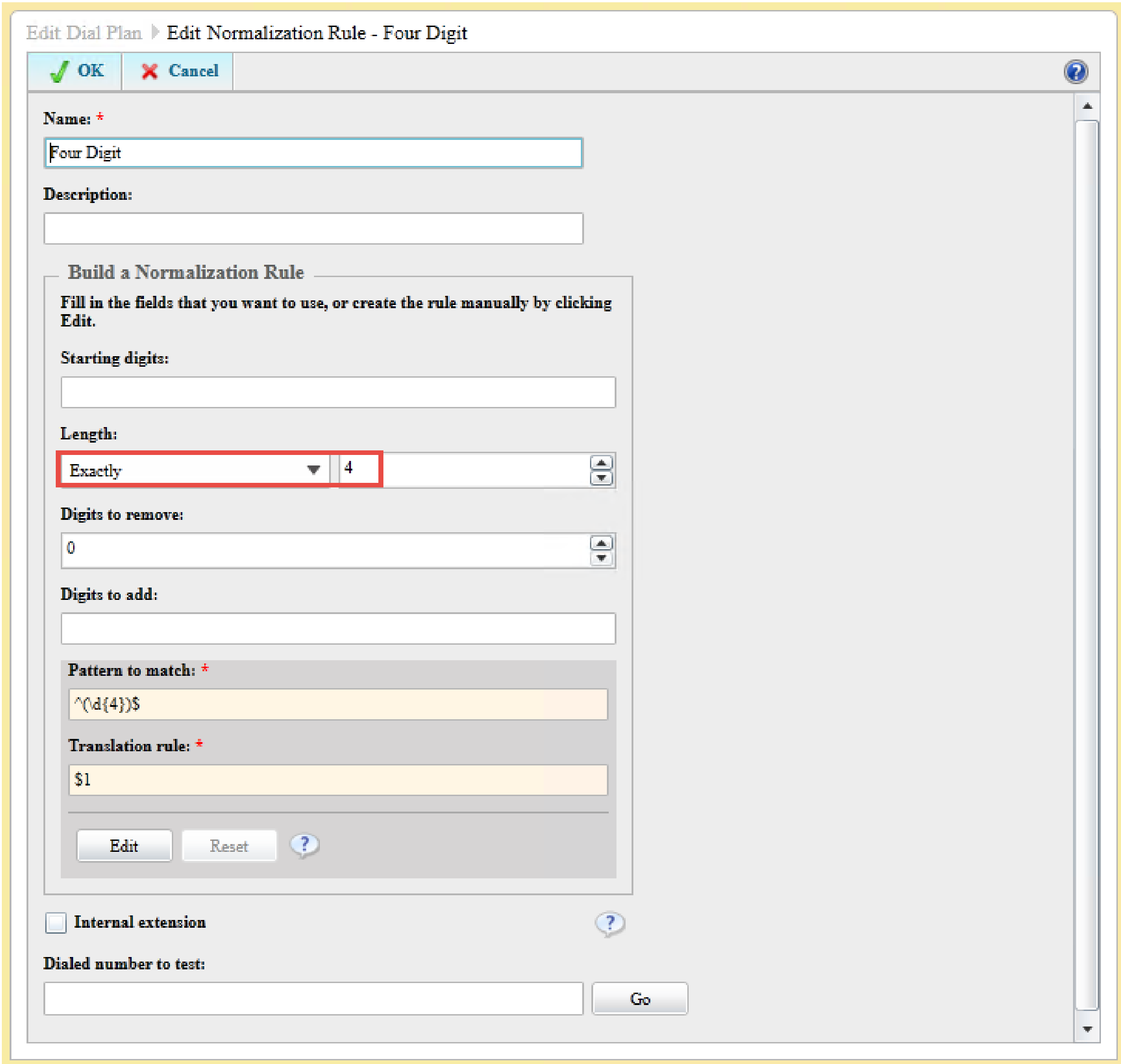


Figure 87: Normalization Rule for 4-digit dialing

Configure Media Bypass

Navigation: Network Configuration -> Global

1. Enable '**Enable media bypass**' in Global setting.
2. Confirm you have also selected **Always bypass**

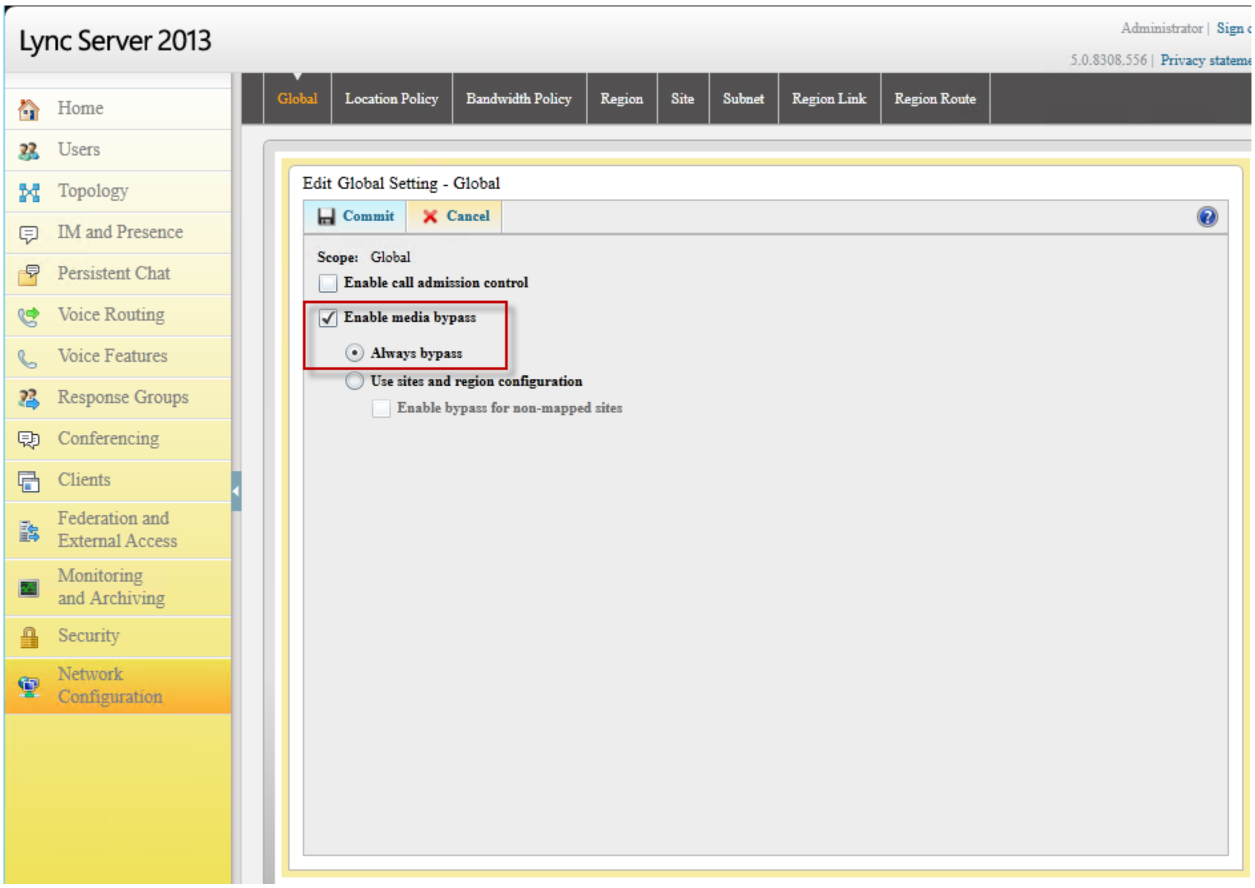


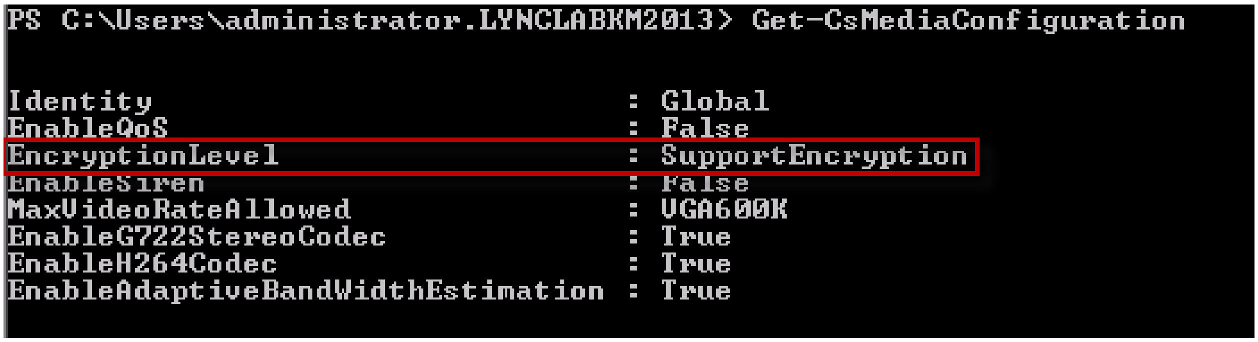
Figure 88: Media Bypass

Configure Encryption Level

Configure Encryption level parameters through the Windows PowerShell® command line interface because they are not configurable on Lync Server Control Panel

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



```
PS C:\Users\administrator.LYNCLABKM2013> Get-CsMediaConfiguration

Identity                             : Global
EnableQoS                            : False
EncryptionLevel                      : SupportEncryption
EnableSiren                          : False
MaxVideoRateAllowed                  : VGA600K
EnableG722StereoCodec                : True
EnableH264Codec                      : True
EnableAdaptiveBandWidthEstimation    : True
```