Nexmo SIP Trunking Configuration Guide

CUCM 11.5.1.12900-21 With CUBE 16.05.01b

May 2017

Contents

1 Introduction

This document is intended for Nexmo SIP trunk customer's technical staff and Value Added Retailer (VAR) having installation and operational responsibilities. This configuration guide provides steps for configuring Cisco Unified Communications Manager (Cisco UCM) 11.5.1.12900-21 and Cisco Unified Boarder Element (Cisco UBE) 16.05.01b to Nexmo SIP Trunking services.

2 SIP Trunking Network Components

The network for the SIP trunk reference configuration is illustrated below and is representative of a Cisco UCM and Cisco UBE configuration to Nexmo SIP trunking.

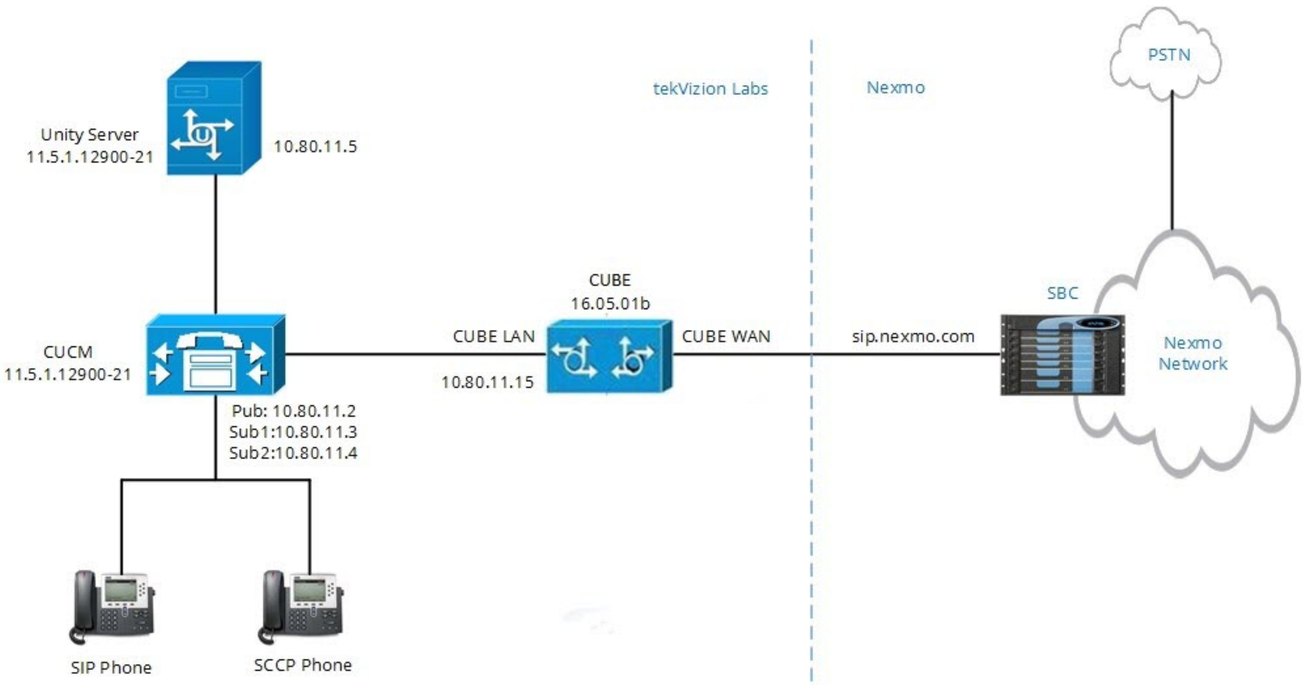


Figure 1: Topology Diagram

2.1 Hardware Components

- Cisco UCS-C240-M3S VMWare host running ESXi 5.5.0 Standard
- Cisco ISR4321/K9 router as CUBE
- Cisco ISR4321/K9 (1RU) processor with 1684579K/6147K bytes of memory with 3 Gigabit Ethernet interfaces
- Processor board ID FLM1925W0X0
- IP Phones 7942(SCCP), 7841(SIP)

2.2 Software Requirements

- Cisco Unified Communications Manager 11.5.1.12900-21
- Cisco Unity Connection 11.5.1.12900-21
- IOS 16.05.01b for ISR4321/K9 Cisco Unified Border Element
- Cisco IOS Software [Everest], ISR Software (X86_64_LINUX_IOSD-UNIVERSALK9-M), Version 16.5.1b, RELEASE SOFTWARE (fc1)
- Cisco IOS XE Software, Version 16.05.01b

3 Features

3.1.1 Features Supported

- Incoming and outgoing off-net calls using G711ULAW & G711ALAW voice codecs
- Calling Line (number) Identification Presentation
- Calling Line (number) Identification Restriction
- Call hold and resume
- Call transfer (unattended and attended)
- Call Conference
- Call forward (all, no answer)
- DTMF relay both directions (RFC2833)
- Media flow-through on Cisco UBE

3.1.2 Features Not Supported by PBX

- None

3.1.3 Caveats and Limitations

- Caller ID is not updated after attended or semi-attended transfers to off-net phones. This is due to a limitation on Cisco UBE. The issue does not impact the calls.

4 Configuration

4.1 IP Address Worksheet

The specific values listed in the table below and in subsequent sections are used in the lab configuration described in this document, and are for **illustrative purposes only**. The customer must obtain and use the values for your deployment.

Table 1 – IP Addresses

Component	Lab Value	Customer Value
Cisco UBE		
LAN IP Address	10.80.11.15	
LAN Subnet Mask	255.255.255.0	
WAN IP Address	192.65.79.XXX	
WAN Subnet Mask	255.255.255.128	
Cisco UCM IP PBX		
System IP Address	10.80.11.2	

4.2 Configuring Cisco Unified Communications Manager

4.2.1 Cisco UCM Version

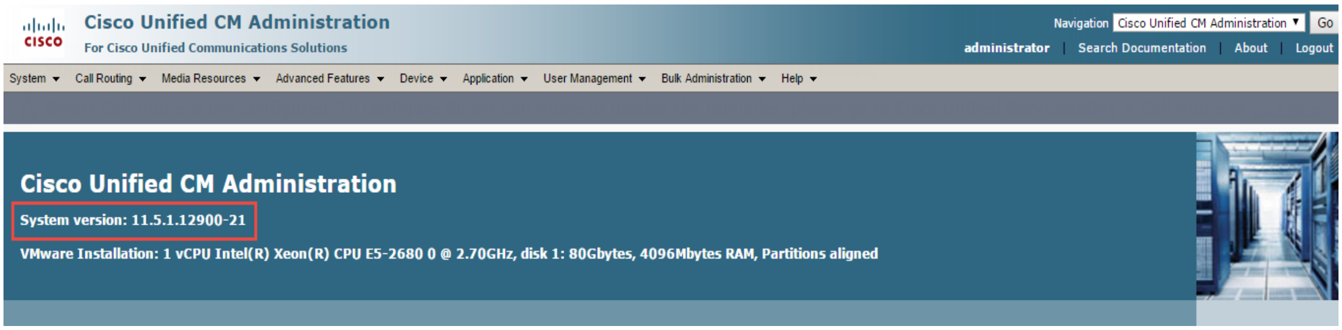


Figure 2: Cisco UCM Version

4.2.2 Cisco Unified Call Manager Service Parameters

Navigation: System → Service Parameters

1. Select **Server**: clus21pub--CUCM Voice/Video (Active)
2. Select **Service**: Cisco CallManager (Active)
3. All other fields are set to default values

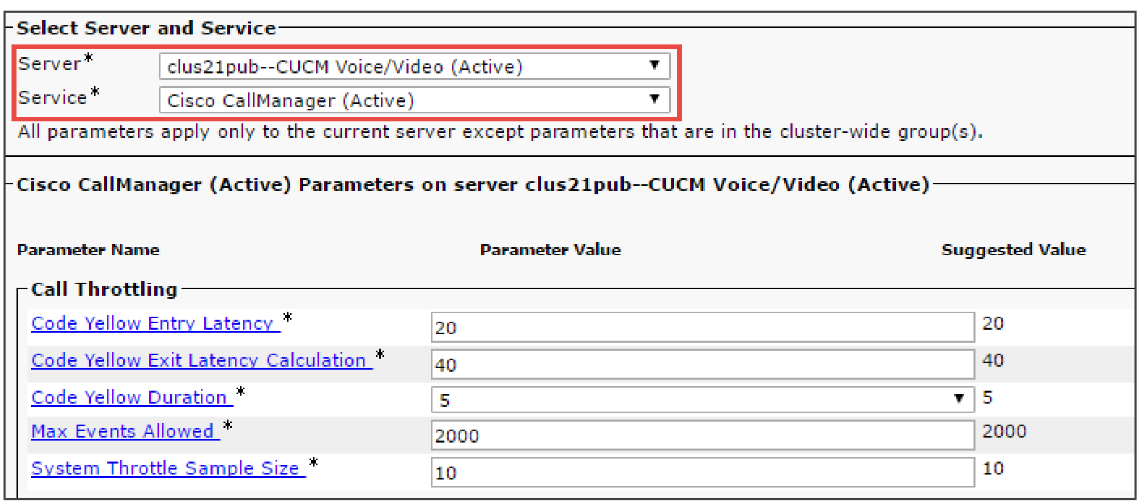


Figure 3: Service Parameters

4.2.3 Offnet Calls via Nexmo SIP Trunk

Off-net calls are served by SIP trunks configured between Cisco UCM and Nexmo Network and calls are routed via Cisco UBE. From Cisco UBE, we have pointed the trunk to sip.nexmo.com and opened the firewall for the list of IPs in the portal provided by Nexmo.

4.2.3.1 SIP Trunk Security Profile

Navigation: System → Security → SIP Trunk Security Profile

1. Set **Name**: *Non Secure SIP Trunk Profile* is used as an example
2. Set **Outgoing Transport Type**: UDP in this example
3. SIP trunks to Nexmo should use UDP as a transport protocol for SIP. This is configured using SIP Trunk Security profile, which is later assigned to the SIP trunk itself.

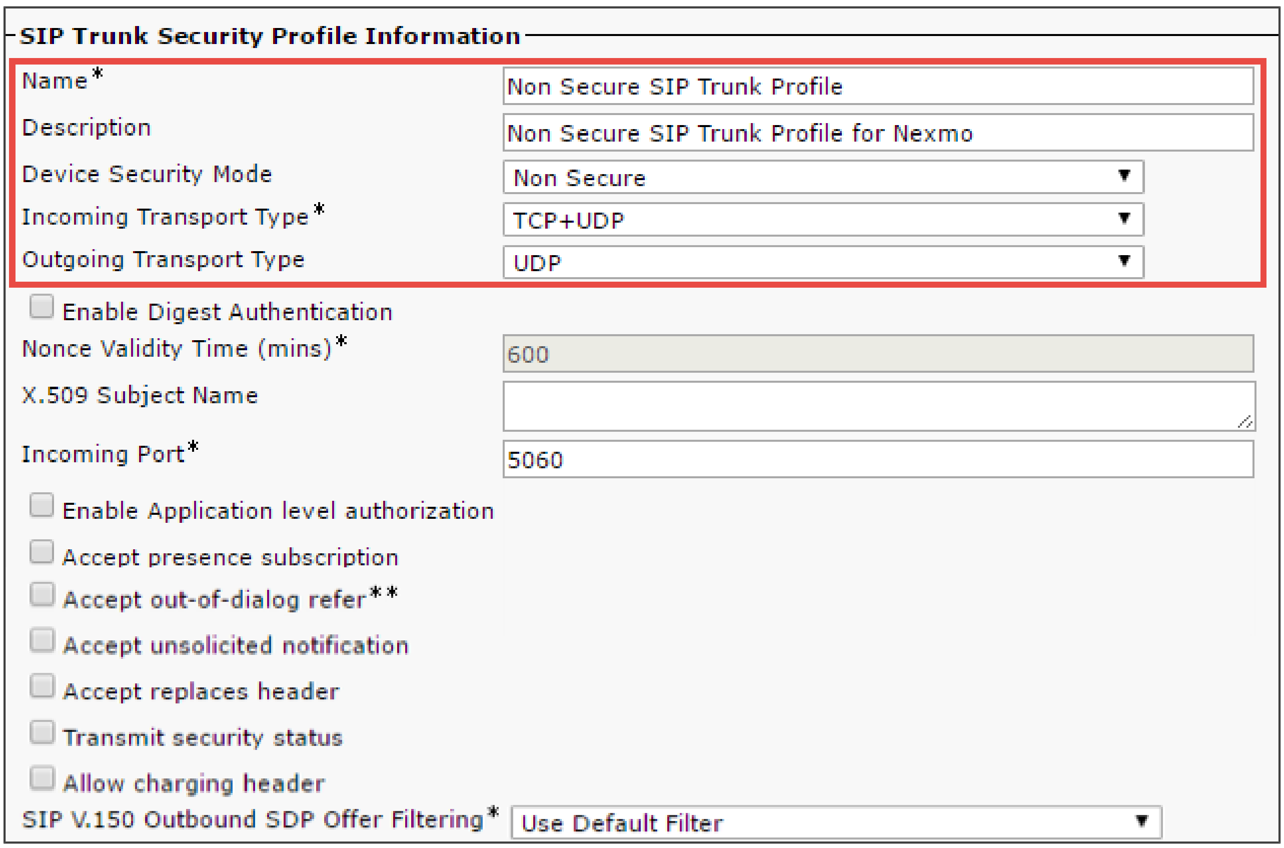
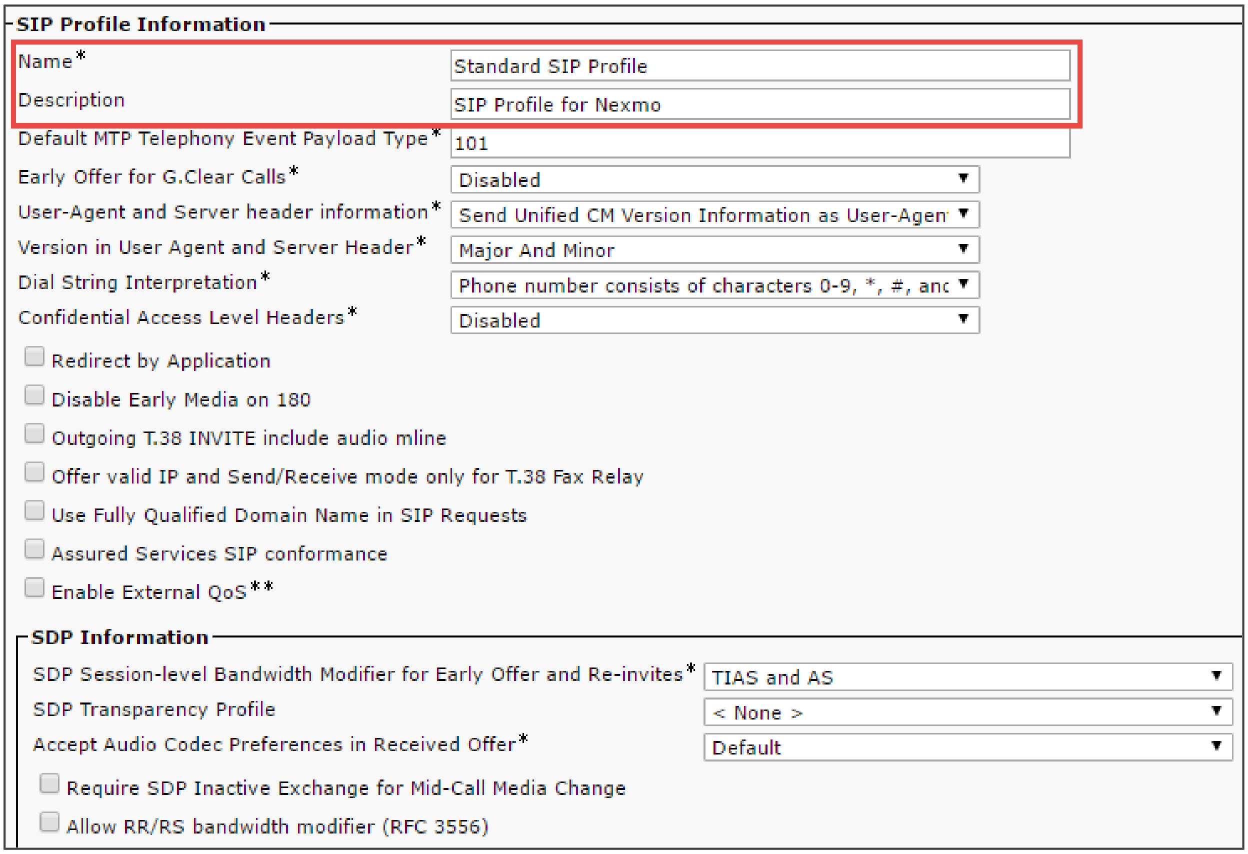


Figure 4: SIP Trunk Security Profile

4.2.3.2 SIP Profile Configuration

Navigation: Device → Device Settings → SIP Profile

1. Set **Name**: *Standard SIP Profile* is used as an example



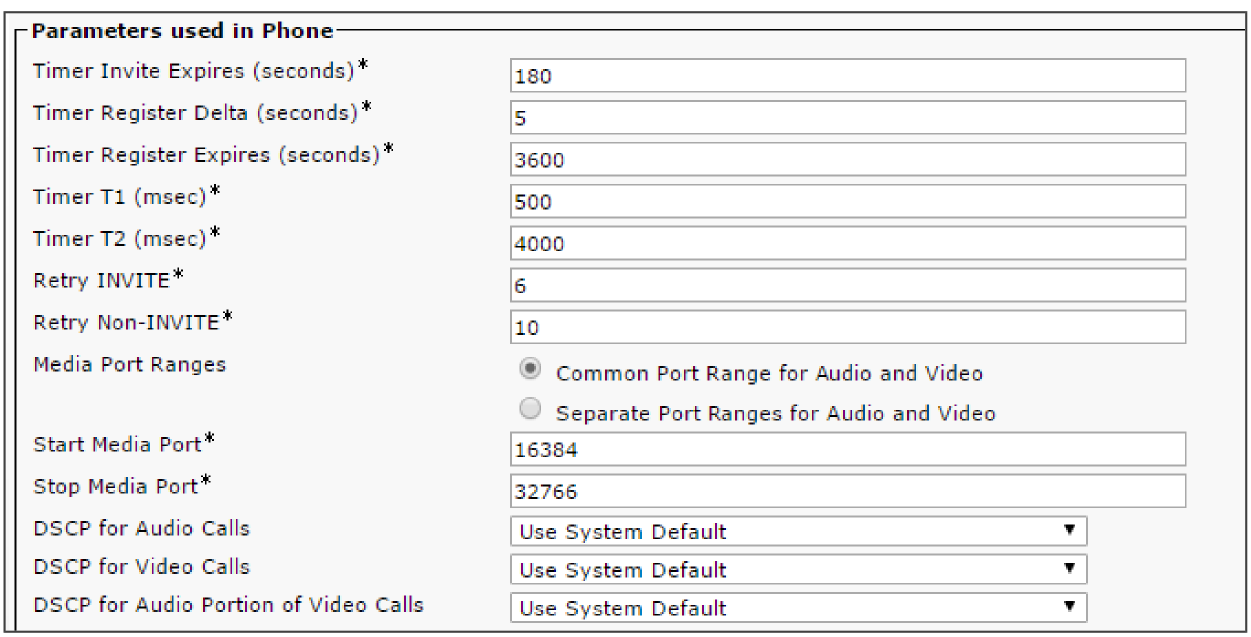


Figure 5: SIP Profile Configuration

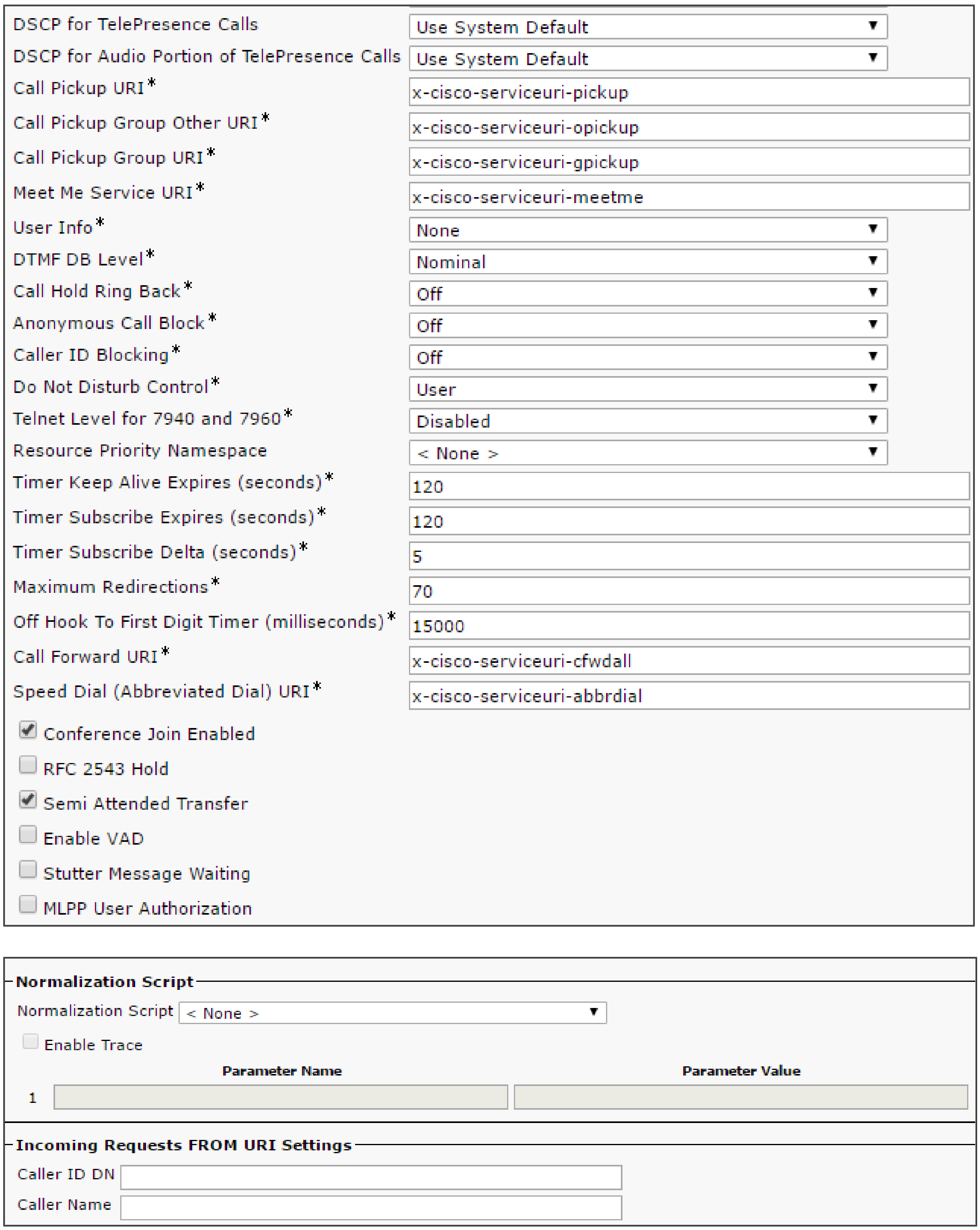


Figure 6: SIP Profile Configuration – Cont.

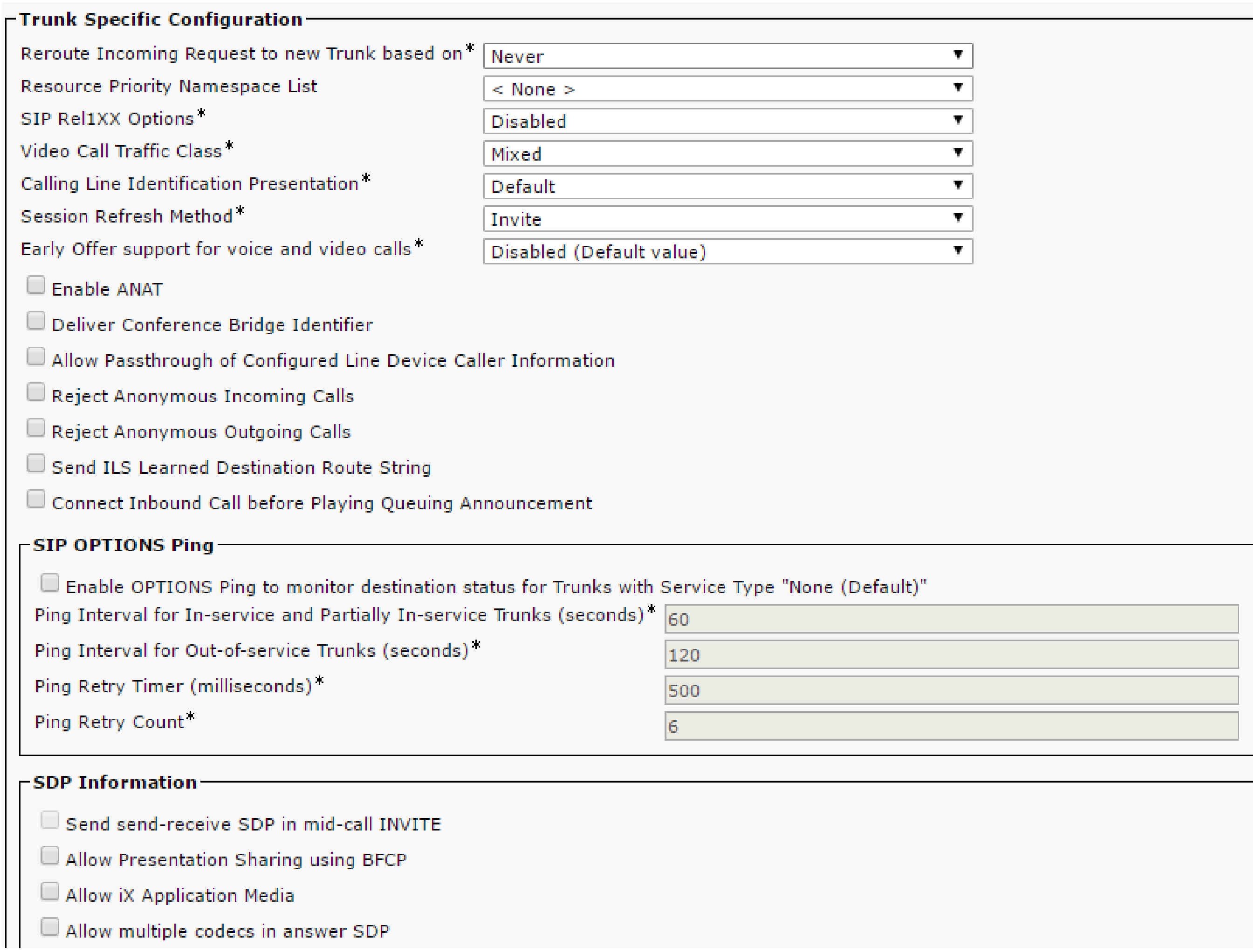


Figure 7: SIP Profile Configuration (cont.)

4.2.3.3 SIP Trunk Configuration

Create SIP trunk to Cisco UBE

Navigation: Device → Trunk

Name ▲	Description	Calling Search Space	Device Pool	Route Pattern	Partition	Route Group	Priority	Trunk Type	SIP Trunk Status	SIP Trunk Duration	SIP Trunk Security Profile
Trunk_to_CUBE_10.80.11.15	Nexmo CUBE		Default	8.@				SIP Trunk	Unknown - OPTIONS Ping not enabled		Non Secure SIP Trunk Profile

Figure 8: SIP Trunk List

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	Trunk_to_CUBE_10.80.11.15
Description	Nexmo CUBE
Device Pool*	Default
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	MRGL_MTP
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

- [ ] 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

Figure 9: SIP Trunk to Cisco UBE

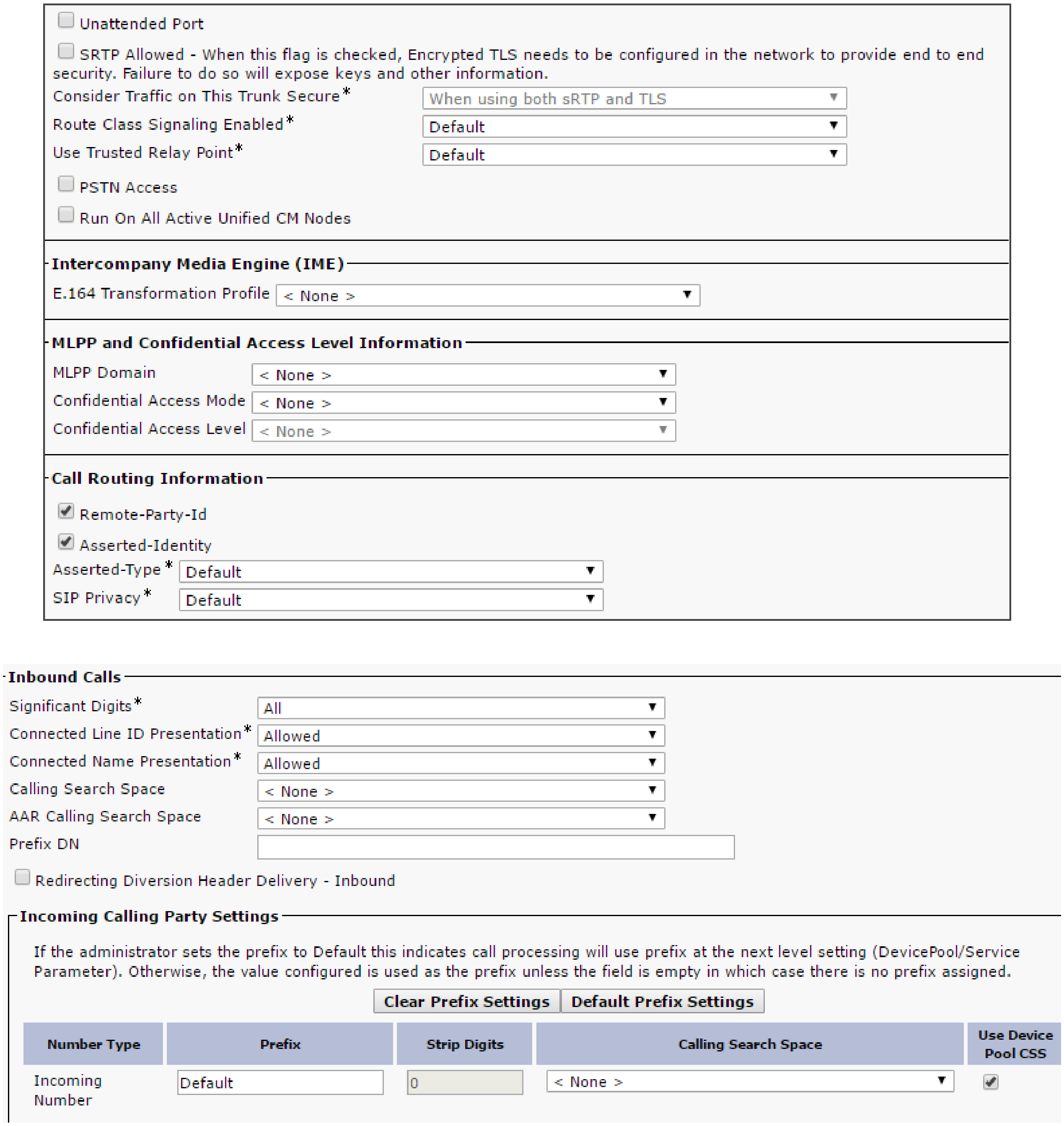
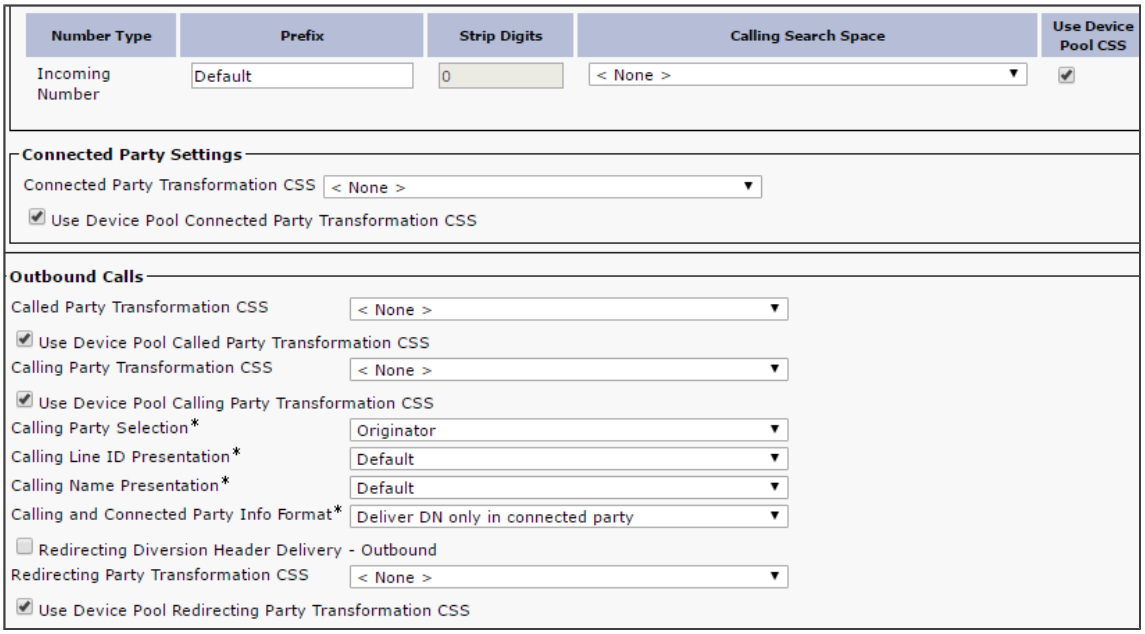
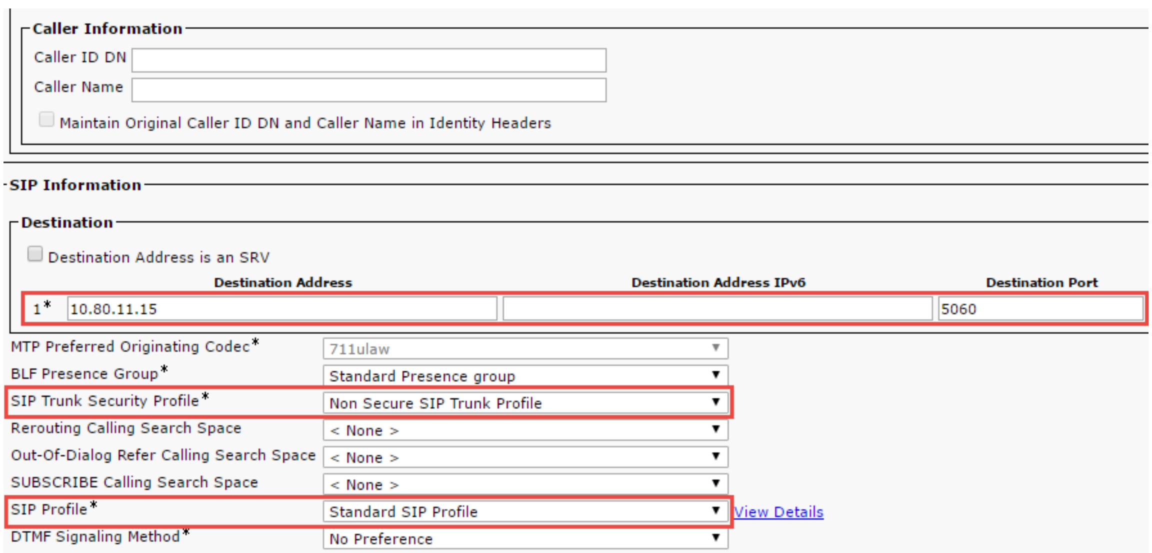


Figure 10: SIP Trunk to Cisco UBE – Cont.


Number Type
Prefix
Strip Digits
Calling Search Space
Use Device Pool CSS
Incoming Number
Default
0
< None >
Connected Party Settings
Connected Party Transformation CSS
< None >
Use Device Pool Connected Party Transformation CSS
Outbound Calls
Called Party Transformation CSS
< None >
Use Device Pool Called Party Transformation CSS
Calling Party Transformation CSS
< None >
Use Device Pool Calling Party Transformation CSS
Calling Party Selection*
Originator
Calling Line ID Presentation*
Default
Calling Name Presentation*
Default
Calling and Connected Party Info Format*
Deliver DN only in connected party
Redirecting Diversion Header Delivery - Outbound
Redirecting Party Transformation CSS
< None >
Use Device Pool Redirecting Party Transformation CSS


Caller Information
Caller ID DN
Caller Name
Maintain Original Caller ID DN and Caller Name in Identity Headers
SIP Information
Destination
Destination Address is an SRV
Destination Address
Destination Address IPv6
Destination Port
1*
10.80.11.15
5060
MTP Preferred Originating Codec*
711ulaw
BLF Presence Group*
Standard Presence group
SIP Trunk Security Profile*
Non Secure SIP Trunk Profile
Rerouting Calling Search Space
< None >
Out-Of-Dialog Refer Calling Search Space
< None >
SUBSCRIBE Calling Search Space
< None >
SIP Profile*
Standard SIP Profile
View Details
DTMF Signaling Method*
No Preference

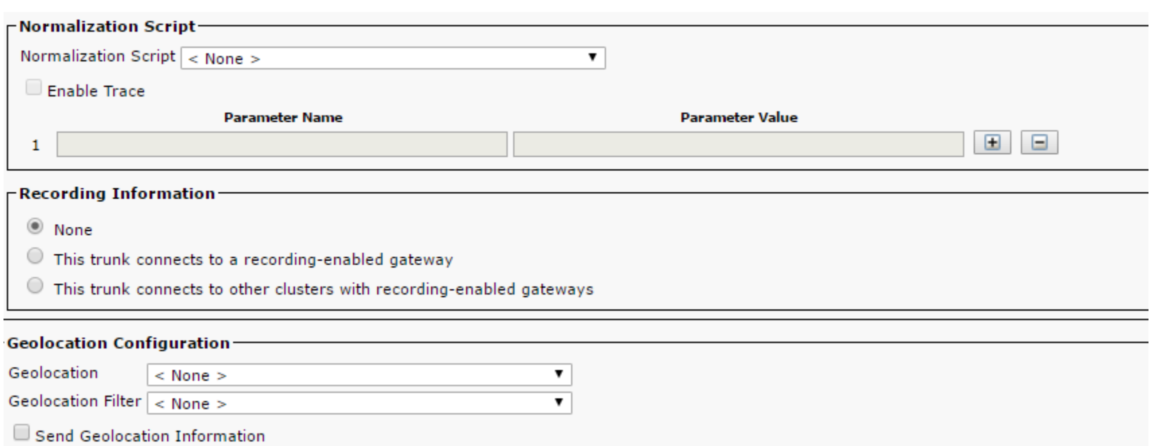

Normalization Script
Normalization Script
< None >
Enable Trace
Parameter Name
Parameter Value
1
Recording Information
None
This trunk connects to a recording-enabled gateway
This trunk connects to other clusters with recording-enabled gateways
Geolocation Configuration
Geolocation
< None >
Geolocation Filter
< None >
Send Geolocation Information

Figure 11: SIP Trunk to Cisco UBE – Cont.

4.2.4 Dial Plan

Navigation: Call Routing → Route/Hunt → Route Pattern

Route patterns are configured as below:

Cisco IP phone dial “8”+11 digit number to access PSTN via Cisco UBE. “8“ is removed before sending to Cisco UBE.

Find Route Patterns where Pattern ▾ begins with ▾ | Find | Clear Filter

	Pattern ▲	Description	Partition	Route Filter	Associated Device
☐	8.@	To Nexmo CUBE 10_80_11_15			Trunk_to_CUBE_10.80.11.15

Figure 12 Route Pattern List

4.3 **Configuring Cisco Unified Border Element**

4.3.1 Network Interface

Configure Ethernet IP address and sub interface. The IP address and VLAN encapsulation used are for illustration only, the actual IP address can vary. For SIP trunks two IP addresses must be configured—LAN and WAN.

```
interface GigabitEthernet0/0/0
 ip address 192.65.79.XXX 255.255.255.128
 negotiation auto
interface GigabitEthernet0/0/1
 ip address 10.80.11.15 255.255.255.0
 negotiation auto
```

4.3.2 Global Cisco UBE Settings

In order to enable Cisco UBE IP2IP gateway functionality, enter the following:

```
voice service voip
 ip address trusted list
  ipv4 173.193.199.24
  ipv4 174.37.245.34
  ipv4 5.10.112.121
  ipv4 5.10.112.122
  ipv4 119.81.44.6
  ipv4 119.81.44.7
```

```
 address-hiding
 mode border-element license capacity 20
 allow-connections sip to sip
 sip
  session refresh
  asserted-id pai
  early-offer forced
  midcall-signaling passthru
  g729 annexb-all
```

4.3.3 Codecs

G711ulaw and G711alaw voice codecs are used for this testing. Codec preferences used to change according to the test plan description

```
voice class codec 1
 codec preference 1 g711ulaw
 codec preference 2 g711alaw
```

4.3.4 Dial Peer

Cisco UBE uses dial-peer to route the call based on the digit to route the call accordingly.

```
dial-peer voice 1 voip
 description incoming dial-peer from CUCM to CUBE
 session protocol sipv2
 session transport udp
 incoming called-number .T
 voice-class codec 1
 dtmf-relay rtp-nte
 no vad
dial-peer voice 2 voip
 description outgoing dial-peer from CUBE to CUCM
 destination-pattern 120........
 session protocol sipv2
 session target ipv4:10.80.11.2
 session transport udp
 voice-class codec 1
 voice-class sip options-keepalive
```

```
 dtmf-relay rtp-nte
 no vad
dial-peer voice 3 voip
 description incoming dial-peer from NEXMO to CUBE
 session protocol sipv2
 session transport udp
 incoming called-number 120........
 voice-class codec 1
 dtmf-relay rtp-nte
 no vad
dial-peer voice 4 voip
 description outgoing dial-peer from CUBE to NEXMO
 destination-pattern .T
 session protocol sipv2
 session target sip-server
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip options-keepalive
 dtmf-relay rtp-nte
 no vad
```

4.3.5 Configuration Example

```
User Access Verification
Username: cisco
Password:
nexmo#
nexmo#sh run
Building configuration...
Current configuration : 5992 bytes
version 16.5
service timestamps debug datetime msec
service timestamps log datetime msec
service password-encryption
platform qfp utilization monitor load 80
no platform punt-keepalive disable-kernel-core
hostname nexmo
boot-start-marker
```

```
boot system flash isr4300-universalk9.16.05.01b.SPA.bin
boot-end-marker
vrf definition Mgmt-intf

address-family ipv4
 exit-address-family
 address-family ipv6
 exit-address-family
enable secret 5
no aaa new-model
ip name-server 8.8.8.8
subscriber templating
multilink bundle-name authenticated
crypto pki trustpoint TP-self-signed-1017057749
 enrollment selfsigned
 subject-name cn=IOS-Self-Signed-Certificate-1017057749
 revocation-check none
 rsakeypair TP-self-signed-1017057749
crypto pki certificate chain TP-self-signed-1017057749
 certificate self-signed 01
  30820330 30820218 A0030201 02020101 300D0609 2A864886 F70D0101 05050030
  31312F30 2D060355 04031326 494F532D 53656C66 2D536967 6E65642D 43657274
  69666963 6174652D 31303137 30353737 3439301E 170D3137 30353130 31353233
  34315A17 0D323030 31303130 30303030 305A3031 312F302D 06035504 03132649
  4F532D53 656C662D 5369676E 65642D43 65727469 66696361 74652D31 30313730
  35373734 39308201 22300D06 092A8648 86F70D01 01010500 0382010F 00308201
  0A028201 0100BF99 0B3B8C33 835DC696 011A6384 ACF8B705 E34D0B17 9BF7A355
  BAB68AED 970A3529 C4780464 92AD7408 96C38292 F286685A 0C3A285C 614EC7A7
  E0D3F7B3 D38037E0 C828DBB8 F08F5474 8A453D68 D3FAAB83 004BA2F3 55201661
  1E4F6DBD 9C0771B4 E8EF4B08 C70CDAD1 8C5F8B00 3C07FEC2 375FE2E3 73BD4F47
  FD1B4F88 D6D19FAB C23069E0 F91E6099 FB7B00D4 0D7D5419 F5570F93 EFBB5C79
  EE86DC0B 72043F04 C7F2B07E 0E681425 705762BF 8B7A0360 25C1077A 2A2BC17A
  68F75A15 7E2439F7 770D90F1 0E8C00F3 65AA0D65 6B891C32 BA19C16E 3B902974
  4A296DB1 8E3E7AD3 694A03AF FA3B5051 D1762F4E 26CBCF74 57DEA2B8 35FDAA31
  44E65C43 76B30203 010001A3 53305130 0F060355 1D130101 FF040530 030101FF
  301F0603 551D2304 18301680 144171AB 9DC3C6B5 F0CA2C01 78ADDAA8 FB66024B
  70301D06 03551D0E 04160414 4171AB9D C3C6B5F0 CA2C0178 ADDAA8FB 66024B70
  300D0609 2A864886 F70D0101 05050003 82010100 003606AE 1AFB9104 447F53BB
  71338C17 F4848B40 9F4A9AA7 9CB791AE 44B73856 241CB923 FD0B0109 2F51F91B
```

```
  B5CD1660 D54BEF67 354213D4 2A442000 B0662481 36D063B3 9BD7D567 46A85C9A
  9AC3E4CD 4B373ECB C8F91089 AF698DCD 37002793 AE1B645A 5F5C1EA2 CBEF72D5
  0763A01E D25FC6C1 A06AF364 47AC82E4 134C463B 176D32CD 16A0AD15 383FB164
  D62134E5 218478F0 5B389D19 75A2C399 C1CC40B5 6AC3DAB2 8AA5D21D 25728B12
  6696650C 5220DB5F A22A304C 8F37EA5C A1C2C37B 7C58F5D2 4B214B5E A1C99E67
  A741E30D 798A7C2F 92F15D55 D8E74340 3A3AF3EB 048EE669 85B8F7FD 5B607C98
  AB1BB24D 0C8B76C4 FAC45B66 52CF5BC0 9CCDFE0B
voice service voip
 ip address trusted list
  ipv4 173.193.199.24
  ipv4 174.37.245.34
  ipv4 5.10.112.121
  ipv4 5.10.112.122
  ipv4 119.81.44.6
  ipv4 119.81.44.7
 address-hiding
 mode border-element license capacity 20
 allow-connections sip to sip
 sip
  session refresh
  asserted-id pai
  early-offer forced
  midcall-signaling passthru
  g729 annexb-all
voice class codec 1
 codec preference 1 g711ulaw
 codec preference 2 g711alaw
license udi pid ISR4321/K9 sn FDO19220MQ9
license boot suite AdvUCSuiteK9
license boot level uck9
diagnostic bootup level minimal
spanning-tree extend system-id
username cisco privilege 15 password 7
redundancy
 mode none
interface GigabitEthernet0/0/0
 ip address 192.65.79.160 255.255.255.128
 negotiation auto
interface GigabitEthernet0/0/1
```

```
 ip address 10.80.11.15 255.255.255.0
 negotiation auto
interface GigabitEthernet0/1/0
 no ip address
 negotiation auto
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
 no ip address
 negotiation auto
threat-visibility
ip forward-protocol nd
ip http server
ip http authentication local
ip http secure-server
ip route 0.0.0.0 0.0.0.0 192.65.79.129
ip route 10.64.0.0 255.255.0.0 10.80.11.1
ip route 172.16.24.0 255.255.248.0 10.80.11.1
ip ssh server algorithm encryption aes128-ctr aes192-ctr aes256-ctr
ip ssh client algorithm encryption aes128-ctr aes192-ctr aes256-ctr
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 4 voip
 description outgoing dial-peer from CUBE to NEXMO
 destination-pattern .T
 session protocol sipv2
 session target sip-server
 session transport udp
 voice-class codec 1
 voice-class sip asserted-id pai
 voice-class sip options-keepalive
 dtmf-relay rtp-nte
 no vad
dial-peer voice 1 voip
 description incoming dial-peer from CUCM to CUBE
 session protocol sipv2
```

```
 session transport udp
 incoming called-number .T
 voice-class codec 1
 dtmf-relay rtp-nte
 no vad
dial-peer voice 2 voip
 description outgoing dial-peer from CUBE to CUCM
 destination-pattern 120........
 session protocol sipv2
 session target ipv4:10.80.11.2
 session transport udp
 voice-class codec 1
 voice-class sip options-keepalive
 dtmf-relay rtp-nte
 no vad
dial-peer voice 3 voip
 description incoming dial-peer from NEXMO to CUBE
 session protocol sipv2
 session transport udp
 incoming called-number 120........
 voice-class codec 1
 dtmf-relay rtp-nte
 no vad
sip-ua
 credentials number 12014647035 username 911236e3 password 7 realm sip.nexmo.com
 authentication username 911236e3 password 7
 sip-server dns:sip.nexmo.com:5060
line con 0
 transport input none
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 login local
no network-clock synchronization automatic
```

4.4 Configure Numbers in Nexmo Account

1. Login to the Nexmo account using the credentials provided at the time of registration. A **Key** and **Secret** will be displayed on the dashboard and this can be used as the username and password for Registration SIP Trunks.

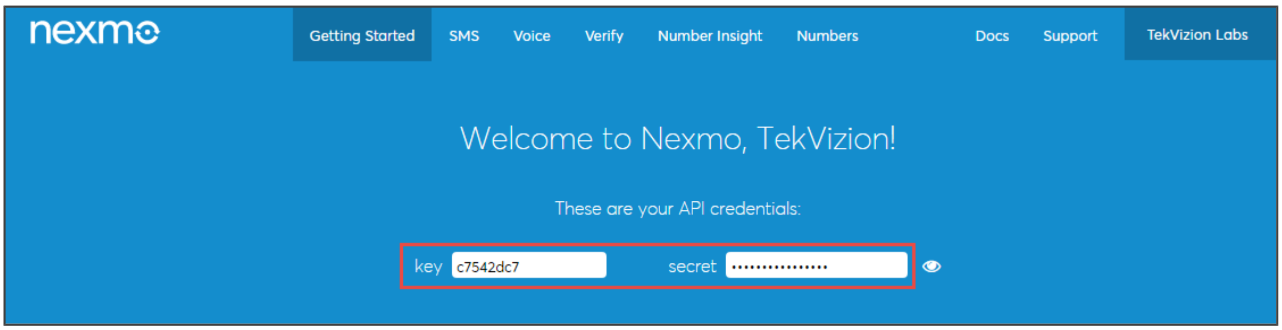


Figure 13: Nexmo Dashboard

2. In order to provide the URL to which the call has to be routed from Nexmo, navigate to the **Numbers** tab
3. Click **Edit** against each number as shown below

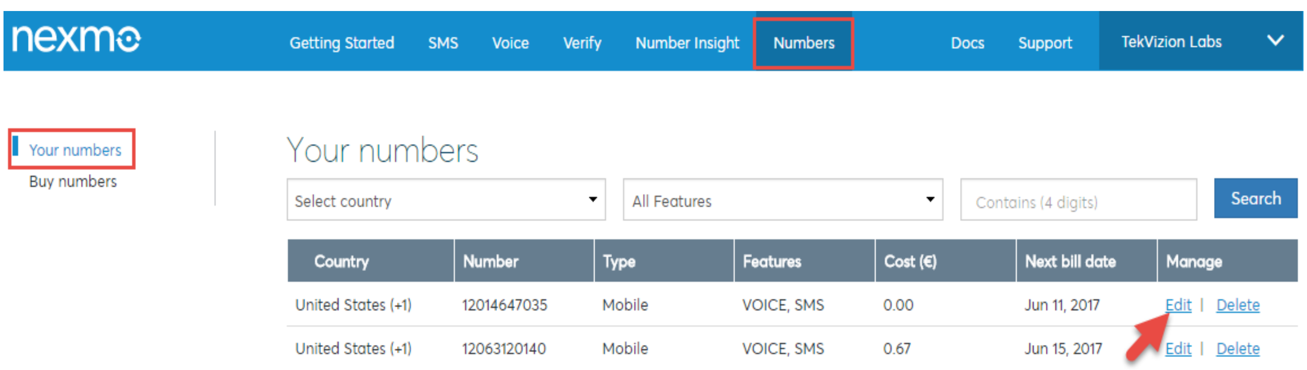


Figure 14: Your Numbers

1. A pop-up will be displayed
2. Select the “**Forward to**” and provide the URL to which the calls route
3. Click **Update** to save the changes

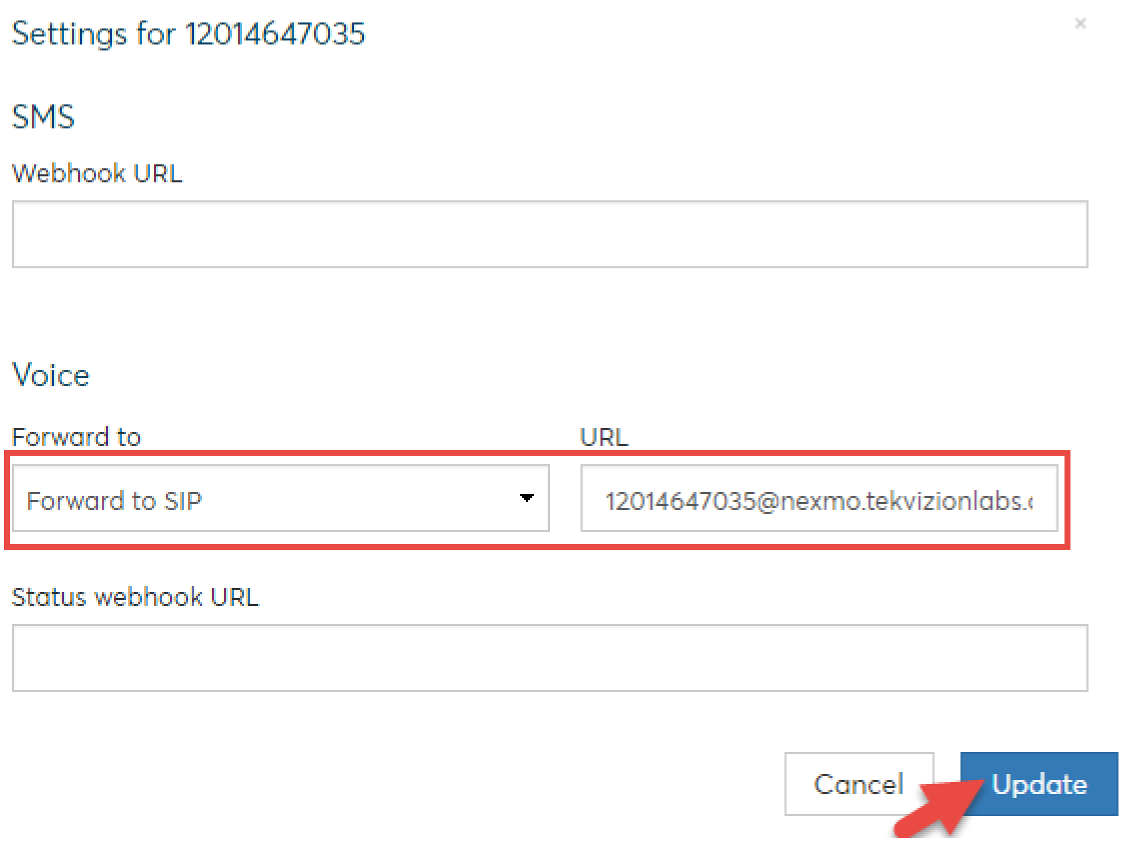


Figure 15: Your Numbers – Cont.