



# Intelepeer SIP Trunking: Connecting Cisco Unified Communications Manager 8.5(1) via the Cisco Unified Border Element 1.3 using SIP

March 25, 2011

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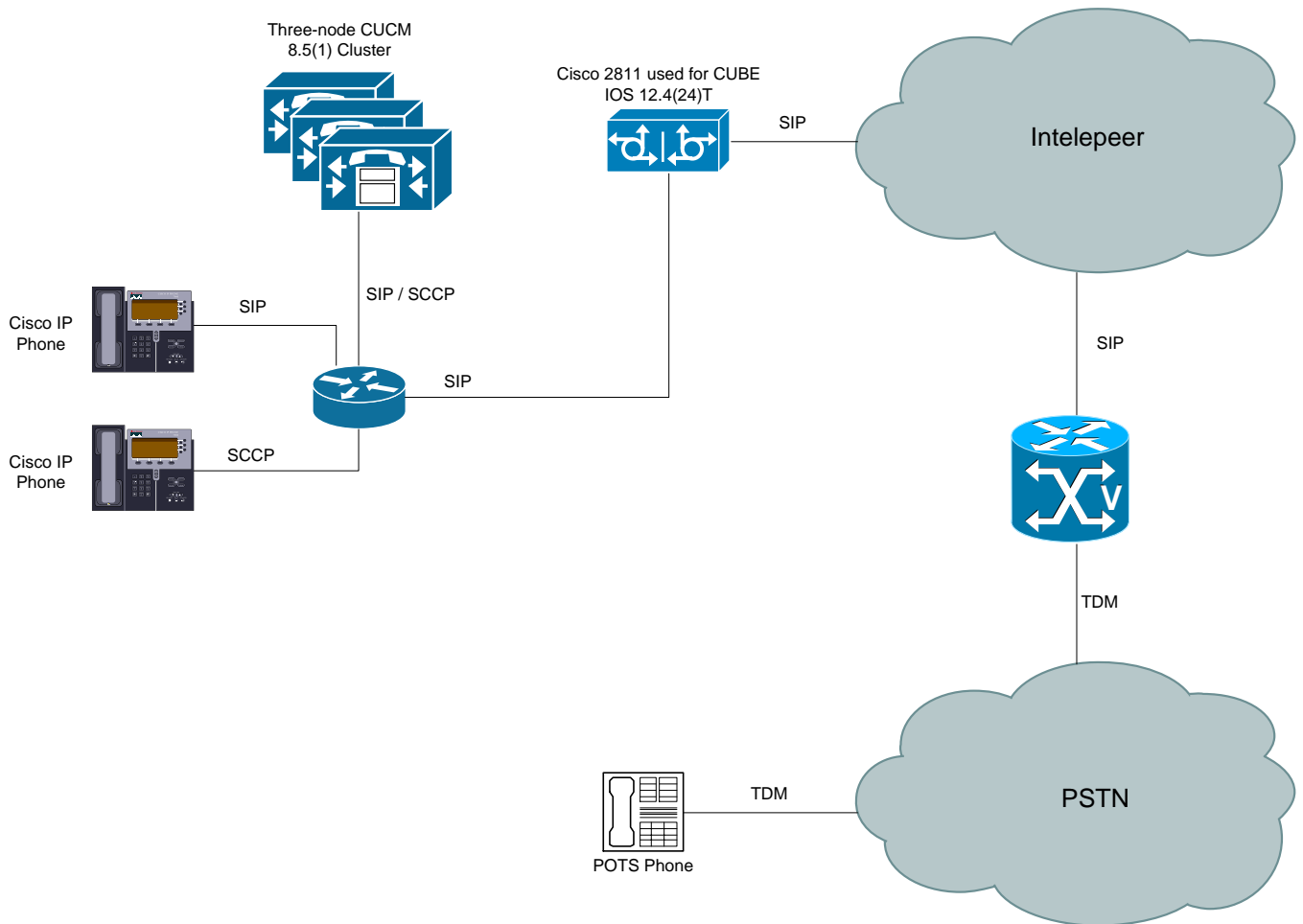
## Introduction

Service Providers today, such as Intelpeer Communications, 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. Intelpeer® SIP Trunking is a service provider 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 Cisco Unified Communications Manager (Cisco UCM) 8.5(1) with a Cisco Unified Border Element (Cisco UBE) for connectivity to the Intelpeer® SIP Trunking service. The deployment model covered in this application note is CPE (Cisco UCM 8.5(1)/Cisco UBE) to PSTN via Intelpeer® SIP Trunking. This document does not address 911 emergency outbound calls. For 911 feature service details contact Intelpeer Communications, directly.
- Testing was performed in accordance to Cisco's Service Provider SIP Trunk Validation Test Plan and all features were verified. Key features verified are:
  - Basic Calls
  - Basic Calls with Calling Name and Number as allowed or restricted
  - DTMF Relay
  - Call Conference (Intra-site, PSTN)
  - Call Transfer (Blind, Attended, Early Attended)
  - Hold and Resume
  - Voice Mail
  - T.38 Fax G3/SG3
  - Simultaneous Calls
  - Auto Attendant
  - International Calls
  - G.711 Fax G3/SG3
  - Call Forwarding – Find Me (Unconditional, Busy, No Reply)
  - Codec negotiation
  - Dial Plans
  - PRACK with SDP early-media cut-through
- 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 Intelpeer Communications' 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 Cisco UBE, to ensure these commands are set per each dial-peer requiring to interoperate to Intelpeer Communications' SIP network.
- This application note does not cover the use of calling search spaces (CSS) or partitions on Cisco Unified Communications Manager. To understand and learn how to apply CSS and partitions refer to the cisco.com link below:  
[http://www.cisco.com/en/US/partner/docs/voice\\_ip\\_comm/cucm/admin/8\\_0\\_2/ccmsys/a03ptcss.html](http://www.cisco.com/en/US/partner/docs/voice_ip_comm/cucm/admin/8_0_2/ccmsys/a03ptcss.html)

## Network Topology

Figure 1. Lab Network Topology



## System Components

### Hardware Components

- Cisco 2811 with VIC-4FXS/DID
- Cisco Unified Communications Manager (1-node cluster consisting of Cisco MCS 7800 Series servers)
- Cisco IP Phones

### Software Requirements

- Cisco Unified Communications Manager 8.5.1.10000-26
- Cisco Unified Border Element, IOS version 12.4(24)T (C2800NM-SPSERVICESK9-M)



## Features

### Features Supported

- Voice calls using G.729 and G.711 codecs
- RFC 3261 support
- Calling number presentation / restriction
- Call conferencing
- Call transfer (attended and unattended)
- Call hold and resume
- Call forwarding
- DTMF relay (RFC 2833)

### Features Not Supported

- Caller ID update via SIP UPDATE method
- T.38 fax
- G.711 pass-through fax
- Early media cut-through with DTMF relay before 200 OK



## Caveats

- Special consideration is required with the FROM header. Intelepeer only accepts INVITE requests with a FROM header that has a IntelepeerID. If caller id presentation is required, the information must be included in the P-asserted-identity, and synchronized with information provisioned on Intelepeer portal.
- When a PSTN to CPE call is transferred by the CPE to a second PSTN number, the Caller ID displayed on the transfer target is the CPE DID number. Intelepeer does not update to the original PSTN calling party number when the transfer is completed. Intelepeer does not support the UPDATE method.



## Call Flows

In the sample configuration presented here, CUCM is provisioned with four-digit directory numbers. And the last 2 digits corresponding to the last 2 DID digits..

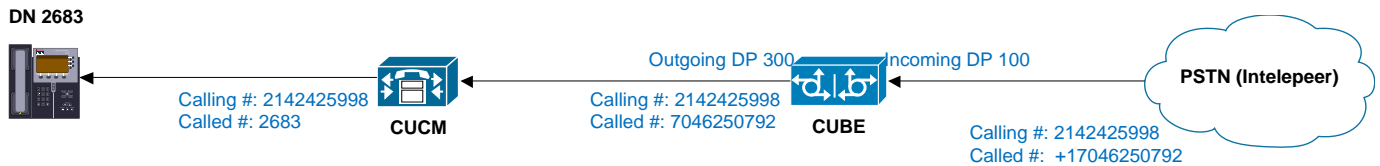
For incoming PSTN calls, the CUBE presents the full ten-digit DID number to CUCM. The CUCM trunk configuration strips all but the last four digits and routes the call based on those digits. Voice calls are routed to IP phones;

CPE callers make outbound PSTN calls by dialing a “9” prefix followed by the destination number. A “9.@” route pattern strips the prefix and routes the call with the remaining digits via a SIP trunk terminating on the CUBE.

Figure 2. Outbound Voice Call



Figure 3. Inbound Voice Call





## Configuration

### Configuring Cisco Unified Border Element

Critical commands are marked in **Bold** with footnotes at the bottom of the page

#### Version Information:

Cisco IOS Software, 2800 Software (C2800NM-SPSERVICESK9-M), Version 12.4(24)T, RELEASE SOFTWARE (fc1)

Technical Support: <http://www.cisco.com/techsupport>

Copyright (c) 1986-2009 by Cisco Systems, Inc.

Compiled Wed 25-Feb-09 17:55 by prod\_rel\_team

ROM: System Bootstrap, Version 12.4(13r)T5, RELEASE SOFTWARE (fc1)

Main2811 uptime is 2 weeks, 4 days, 21 hours, 37 minutes

System returned to ROM by power-on

System restarted at 10:50:20 CST Sun Mar 6 2011

System image file is "flash:c2800nm-spservicesk9-mz.124-24.T.bin"

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/wwl/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 2811 (revision 53.51) with 247808K/14336K bytes of memory.

Processor board ID FHK0923F0YT

2 FastEthernet interfaces

25 Serial interfaces

1 Channelized/Clear T1/PRI port

4 Voice FXS interfaces

DRAM configuration is 64 bits wide with parity enabled.

239K bytes of non-volatile configuration memory.

62720K bytes of ATA CompactFlash (Read/Write)

Configuration register is 0x2102

#### **Running Configuration:**

```
Current configuration : 4476 bytes
!
! Last configuration change at 17:04:36 CST Wed Feb 23 2011 by cisco
! NVRAM config last updated at 17:04:38 CST Wed Feb 23 2011 by cisco
!
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
```



```
hostname Main2811
!
boot-start-marker
boot system flash c2800nm-spservicesk9-mz.124-24.T.bin
boot-end-marker
!
card type t1 1 0
logging message-counter syslog
logging buffered 51200 warnings
no logging console
enable secret 5 *****
!
aaa new-model
!
!
!
!
aaa session-id common
clock timezone CST -6
clock summer-time CDT recurring
no network-clock-participate slot 1
!
voice-card 0
!
voice-card 1
  dspfarm
  dsp services dspfarm
!
dot11 syslog
ip source-route
!
!
ip cef
!
!
ip domain name lab.tekvizion.com1
ip name-server 10.64.1.32
no ipv6 cef
ntp server 10.10.10.5
multilink bundle-name authenticated
!
!
!
!
isdn switch-type primary-ni
!
!
!
voice rtp send-recv
!
voice service voip
  allow-connections sip to sip3
sip
  rellxx disable
  header-passing error-passthru
  early-offer forced4
```

<sup>1</sup> Optional, for use with a multiple-subscriber cluster. The domain name must match the CUCM enterprise parameter “Cluster Fully Qualified Domain Name” and must resolve to a DNS SRV. See “DNS Configuration” and “CUCM Configuration” below.

<sup>2</sup> IP address of DNS server

<sup>3</sup> Allow SIP to SIP call processing

<sup>4</sup> Configures CUBE to send a SIP INVITE with SDP on an outbound call leg (Delayed Offer to Early Offer)



```
midcall-signaling passthru

!
!
!
voice class codec 1
  codec preference 1 g729r8
  codec preference 2 g711ulaw
!

!
voice translation-rule 10
  rule 1 /^+1\(\.....\)$/ /\1/
!
!
voice translation-profile in_rmv_plus1
  translate called 10
!
!
!
!
!
!
username cisco password *****
archive
  log config
  hidekeys
!
!
controller T1 1/0/0
  cablelength long 0db
  pri-group timeslots 1-24
!
!
!
!
interface FastEthernet0/0
  description Internal LAN (CUCM-facing)
  ip address 10.64.1.88 255.255.0.0
  ip virtual-reassembly
  duplex full
  speed 100
!
interface FastEthernet0/1
  description External WAN (Service Provider facing)
  ip address 174.46.0.150 255.255.255.128
  ip virtual-reassembly
  duplex full
  speed 100
!
interface Serial0/0/0
  no ip address
  shutdown
!
interface Serial1/0/0:23
  no ip address
  encapsulation hdlc
  isdn switch-type primary-ni
  isdn timer T310 300000
  isdn incoming-voice voice
  isdn map address .* plan isdn type national
```



```
no cdp enable
!
ip forward-protocol nd
ip route 0.0.0.0 0.0.0.0 174.46.0.129
ip route 10.0.0.0 255.0.0.0 10.64.1.1
ip http server
no ip http secure-server
!
!
!
!
!
!
control-plane
!
!
!
voice-port 0/1/0
  caller-id enable
!
voice-port 0/1/1
!
voice-port 0/1/2
!
voice-port 0/1/3
!
voice-port 1/0/0:23
!
!
mgcp fax t38 ecm
!
!
!
!
dial-peer voice 100 voip
  description Incoming dialpeer and 1+10 digits to Intelepeer
  translation-profile incoming in_rmv_plus1
  destination-pattern ^1[2-9]..[2-9].....$
  voice-class codec 1
  voice-class sip early-offer forced
  session protocol sipv2
  session target sip-server
  session transport tcp
  incoming called-number .T
  dtmf-relay rtp-nte
!
dial-peer voice 101 voip
  description 10-digit local calls to Intelepeer
  destination-pattern ^[2-9]..[2-9].....$
  voice-class codec 1
  voice-class sip early-offer forced
  session protocol sipv2
  session target sip-server
  session transport tcp
  dtmf-relay rtp-nte
!
dial-peer voice 102 voip
  description International calls to Intelepeer
  destination-pattern ^011T
  voice-class codec 1
  voice-class sip early-offer forced
  session protocol sipv2
  session target sip-server
  session transport tcp
```



```
dtmf-relay rtp-nte
!
dial-peer voice 103 voip
description N11 calls to Intelepeer
destination-pattern ^[2-9]11$
voice-class codec 1
voice-class sip early-offer forced
session protocol sipv2
session target sip-server
session transport tcp
dtmf-relay rtp-nte
!
dial-peer voice 300 voip
description 70462507XX calls to primary CUCM
huntstop
preference 1
destination-pattern ^70462507..$5
voice-class codec 1
voice-class sip early-offer forced
session protocol sipv2
session target ipv4:10.70.1.26
dtmf-relay rtp-nte
!
dial-peer voice 301 voip
description 70462507XX calls to secondary CUCM
huntstop
preference 2
destination-pattern ^70462507..$7
voice-class codec 1
voice-class sip early-offer forced
session protocol sipv2
session target ipv4:10.70.1.38
dtmf-relay rtp-nte
!
dial-peer voice 302 voip
description Incoming calls from primary CUCM
huntstop
incoming called-number .%
voice-class codec 1
voice-class sip early-offer forced
session protocol sipv2
session target ipv4:10.70.1.2
dtmf-relay rtp-nte
!
dial-peer voice 303 voip
description Incoming calls from secondary CUCM
huntstop
incoming called-number .%
voice-class codec 1
voice-class sip early-offer forced
session protocol sipv2
session target ipv4:10.70.1.3
dtmf-relay rtp-nte
!
dial-peer voice 111 pots
```

<sup>5</sup> Outbound dial peer (to CUCM). The pattern here should match service provider-assigned DID numbers.

<sup>6</sup> Reference to CUCM primary IP address

<sup>7</sup> Outbound dial peer (to CUCM). The pattern here should match service provider-assigned DID numbers.

<sup>8</sup> Reference to CUCM secondary IP address



```
description used for analog phone/fax
huntstop
destination-pattern 7046250792
no digit-strip
port 0/1/0
!
!
gateway
timer receive-rtp 1200
!
sip-ua
retry invite 2
retry bye 2
retry cancel 2
retry register 10
sip-server dns:sip.intelepeer.com9
!
!
line con 0
exec-timeout 0 0
line aux 0
line vty 0 4
exec-timeout 0 0
privilege level 15
transport input telnet
line vty 5 15
privilege level 15
!
scheduler allocate 20000 1000
end
```

---

<sup>9</sup> Service provider signaling address.



## DNS Configuration

In a single-node cluster configuration (only one CUCM node running the CallManager service) the CUBE dial peers may simply point to the node's IP address, in which case no special DNS configuration is required. To refer to the single node by fully-qualified domain name (FQDN) a DNS A-record is required.

In a multi-node configuration, DNS SRV records are needed. The DNS configuration illustrated below is from a Microsoft® DNS server, although any similarly-configured, SRV-capable DNS server will work just as well.

**Figure 4.** DNS SRV Records

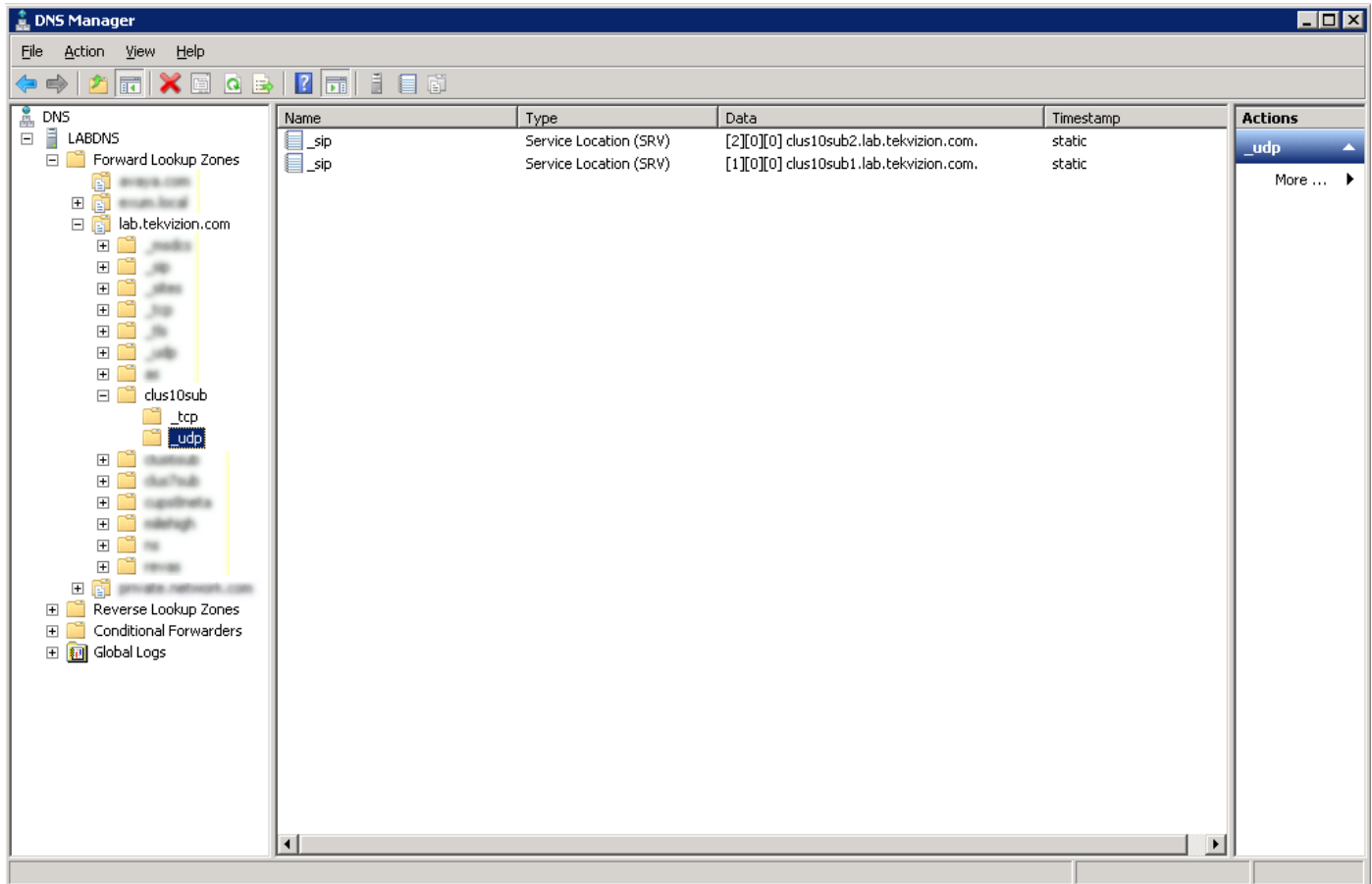




Figure 5. DNS A Records

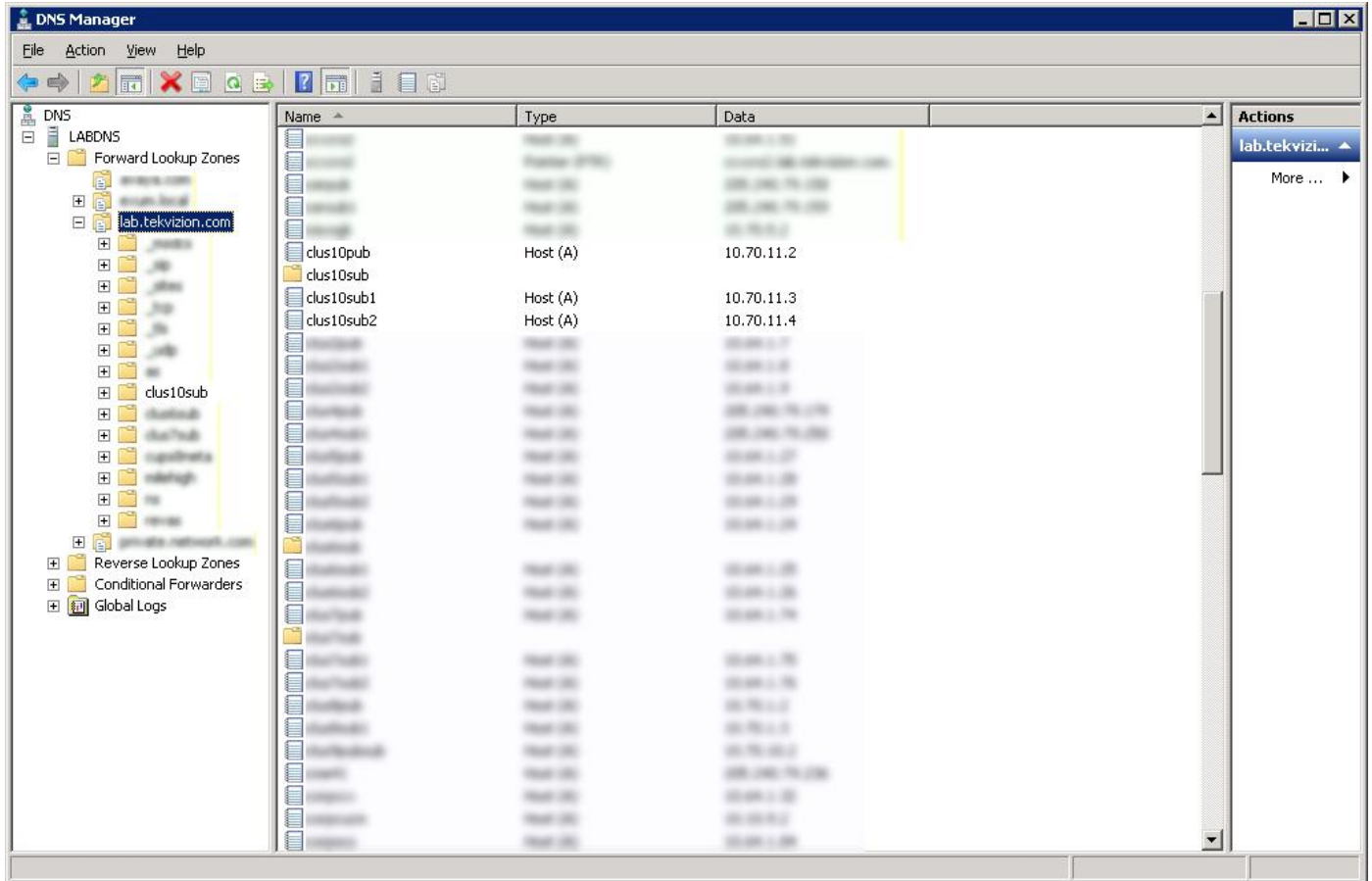
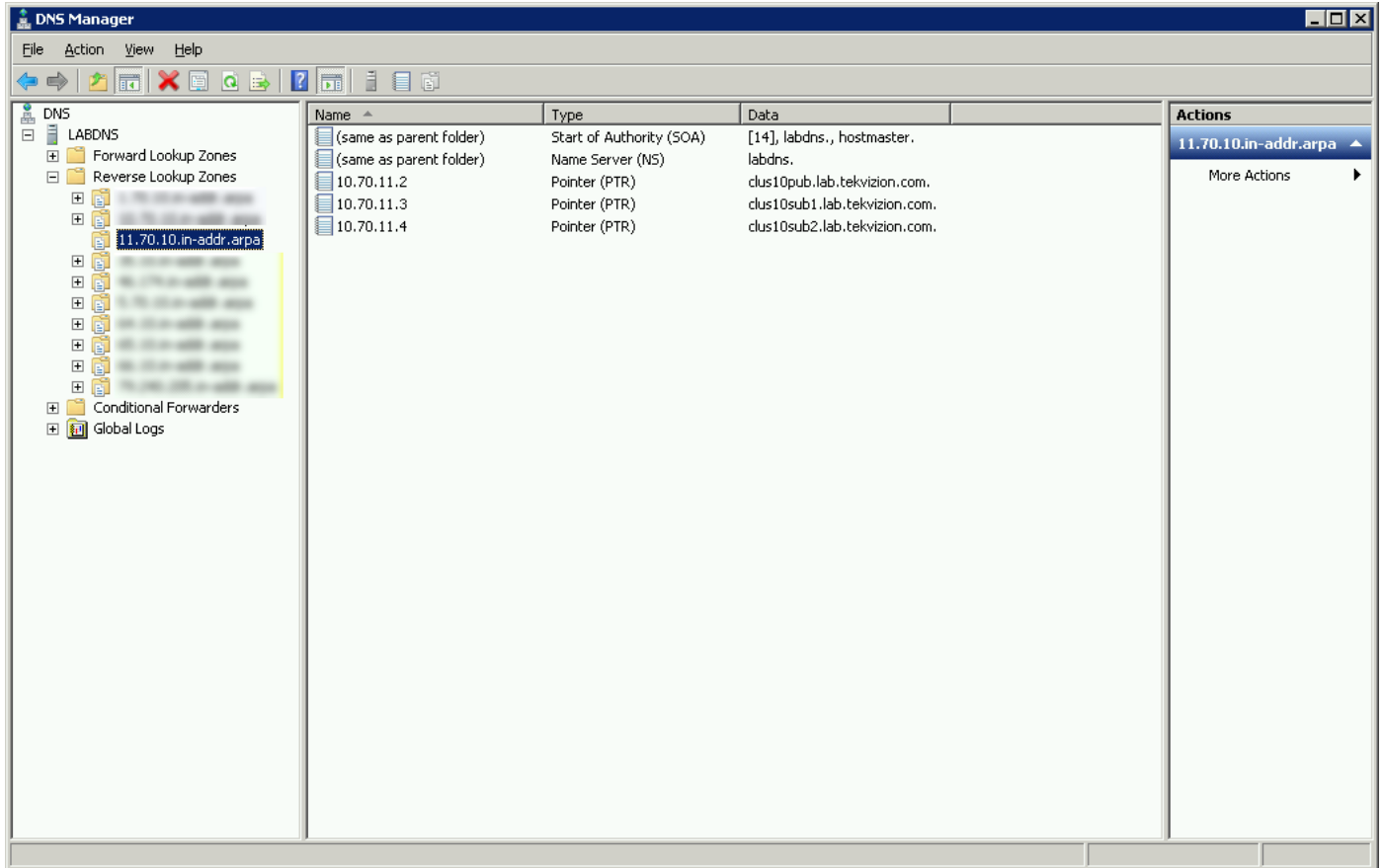




Figure 6. DNS PTR Records





## Configuring the Cisco Unified Communications Manager

Figure 7. Enterprise Parameters

Cisco Unified CM Administration  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go  
administrator | Search Documentation | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

### Enterprise Parameters Configuration

Save | Set to Default | Reset | Apply Config

|  |      |      |
|--|------|------|
| Service Manager TCP Server communication port number * | 8888 | 8888 |
| Service Manager TCP Client communication port number * | 8889 | 8889 |

**CRS Application Parameters**

|                            |       |
|----------------------------|-------|
| Auto Attendant Installed * | false |
| IPCC Express Installed *   | false |

**Clusterwide Domain Configuration**

|                                     |                             |  |
|-------------------------------------|-----------------------------|--|
| Organization Top Level Domain       |                             |  |
| Cluster Fully Qualified Domain Name | clus10sub.lab.tekvizion.com | Required when DNS SRV records are used. Must match the session-target in the CUBE dial peer. |

**Denial-of-Service Protection**

|                                |      |      |
|--------------------------------|------|------|
| Denial-of-Service Protection * | True | True |
|--------------------------------|------|------|

**TLS Handshake Timer**

|                       |    |    |
|-----------------------|----|----|
| TLS Handshake Timer * | 60 | 60 |
|-----------------------|----|----|

**Cisco Support Use**

|                     |  |  |
|---------------------|--|--|
| Cisco Support Use 1 |  |  |
| Cisco Support Use 2 |  |  |

**IPv6 configuration Modes**

|   |       |       |
|---|-------|-------|
| Enable IPv6 *                                 | False | False |
| IP Addressing Mode Preference for Media *     | IPv4  | IPv4  |
| IP Addressing Mode Preference for Signaling * | IPv4  | IPv4  |
| Allow Auto-Configuration for Phones *         | On    | On    |

**Cisco Syslog Agent**

|                           |  |  |
|---------------------------|--|--|
| Remote Syslog Server Name |  |  |
|---------------------------|--|--|

Internet | Protected Mode: On | 100%



Figure 8. SIP Trunk Security Profile

The screenshot shows the Cisco Unified CM Administration interface for configuring a SIP Trunk Security Profile. The page title is "SIP Trunk Security Profile Configuration". The status is "Ready". The configuration details are as follows:

| SIP Trunk Security Profile Information                              |   |
|---|---|
| Name*   | Non Secure SIP Trunk Profile                            |
| Description   | Non Secure SIP Trunk Profile authenticated by null Stri |
| Device Security Mode  | Non Secure  |
| Incoming Transport Type*  | TCP+UDP   |
| Outgoing Transport Type   | UDP   |
| <input type="checkbox"/> Enable Digest Authentication               |   |
| Nonce Validity Time (mins)*   | 600   |
| X.509 Subject Name  |   |
| Incoming Port*  | 5060  |
| <input type="checkbox"/> Enable Application Level Authorization     |   |
| <input type="checkbox"/> Accept Presence Subscription               |   |
| <input checked="" type="checkbox"/> Accept Out-of-Dialog REFER**    |   |
| <input checked="" type="checkbox"/> Accept Unsolicited Notification |   |
| <input checked="" type="checkbox"/> Accept Replaces Header          |   |
| <input type="checkbox"/> Transmit Security Status                   |   |

At the bottom of the configuration area, there are buttons for Save, Delete, Copy, Reset, Apply Config, and Add New. Below the configuration area, there are two informational messages:

- \*- indicates required item.
- \*\*If this profile is associated with an EMCC SIP trunk, Accept Out-of-Dialog REFER is enabled regardless of the setting on this page

The page footer shows "Done" and "Internet | Protected Mode: On".



Figure 9. SIP Trunk to Intelepeer via CUBE

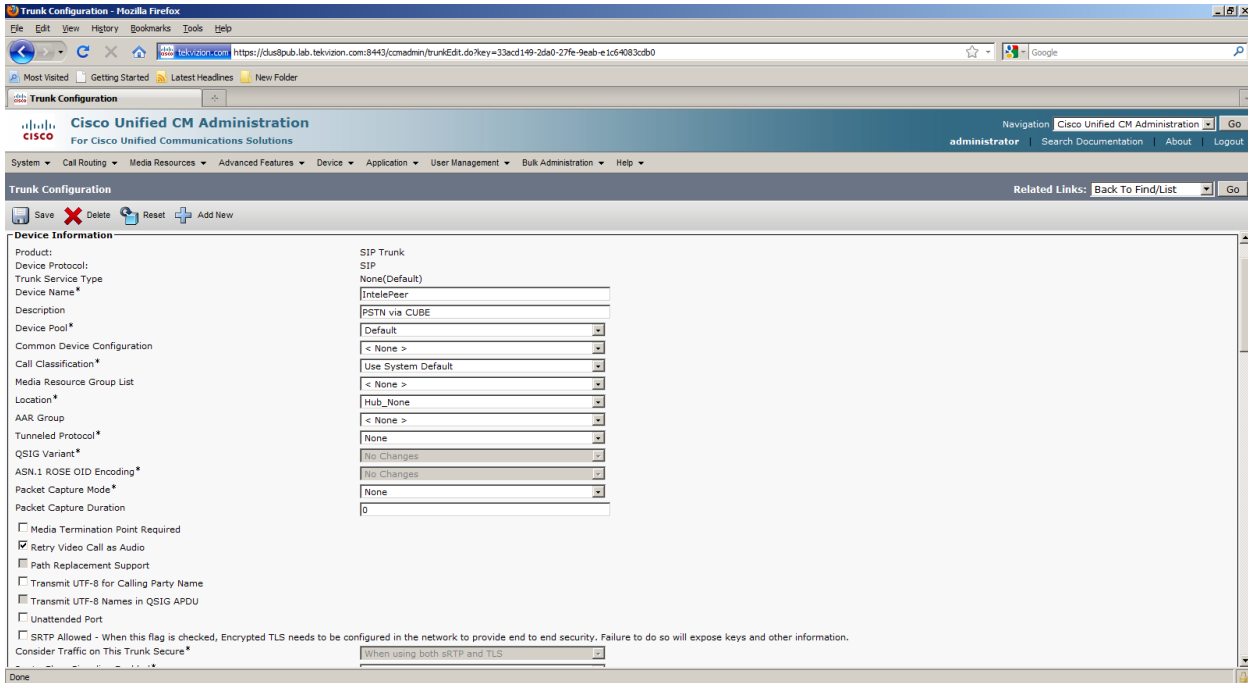


Figure 10. SIP Trunk to Intelepeer via CUBE (cont.)

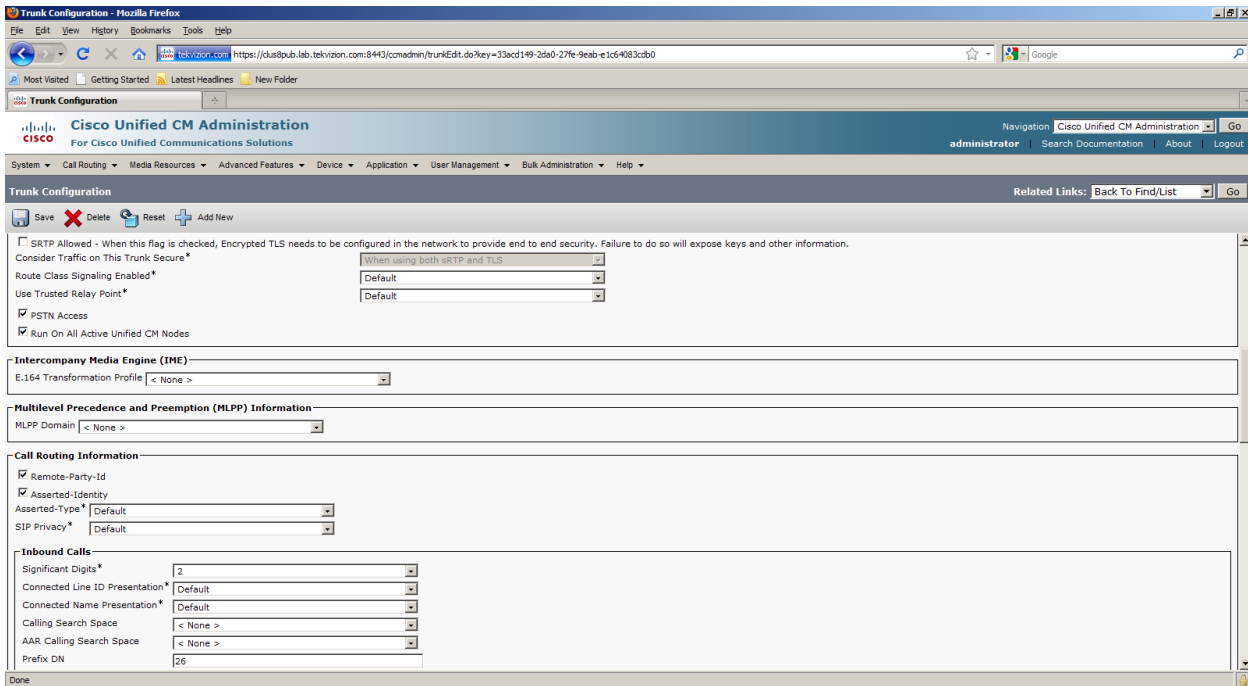




Figure 11. SIP Trunk to Intelepeer via CUBE (cont.)

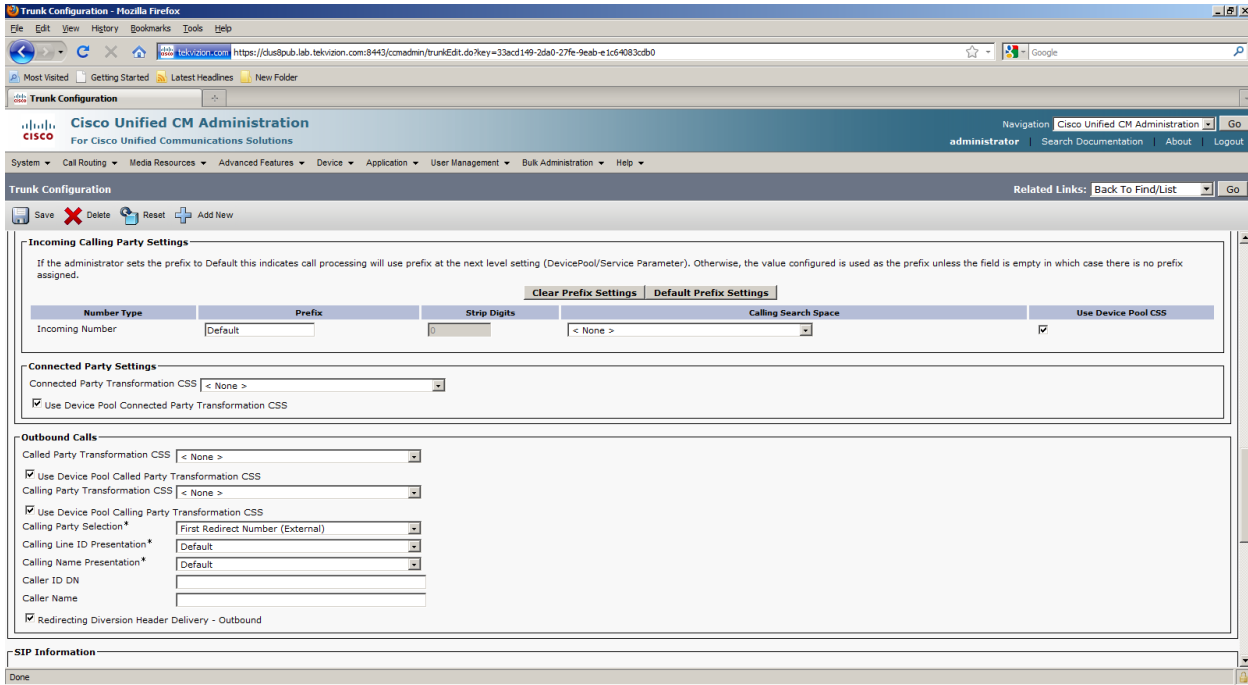


Figure 12. SIP Trunk to Intelepeer via CUBE (cont.)

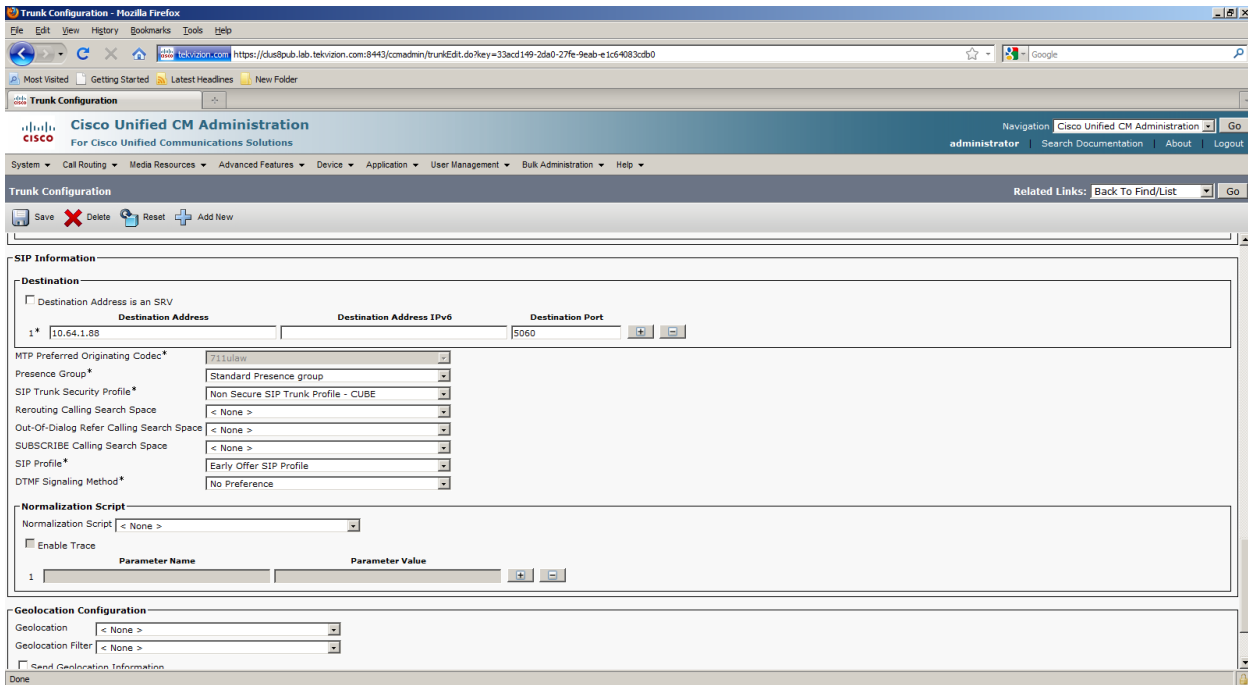




Figure 13. Route Pattern Configuration for SIP trunk to Intelepeer via CUBE

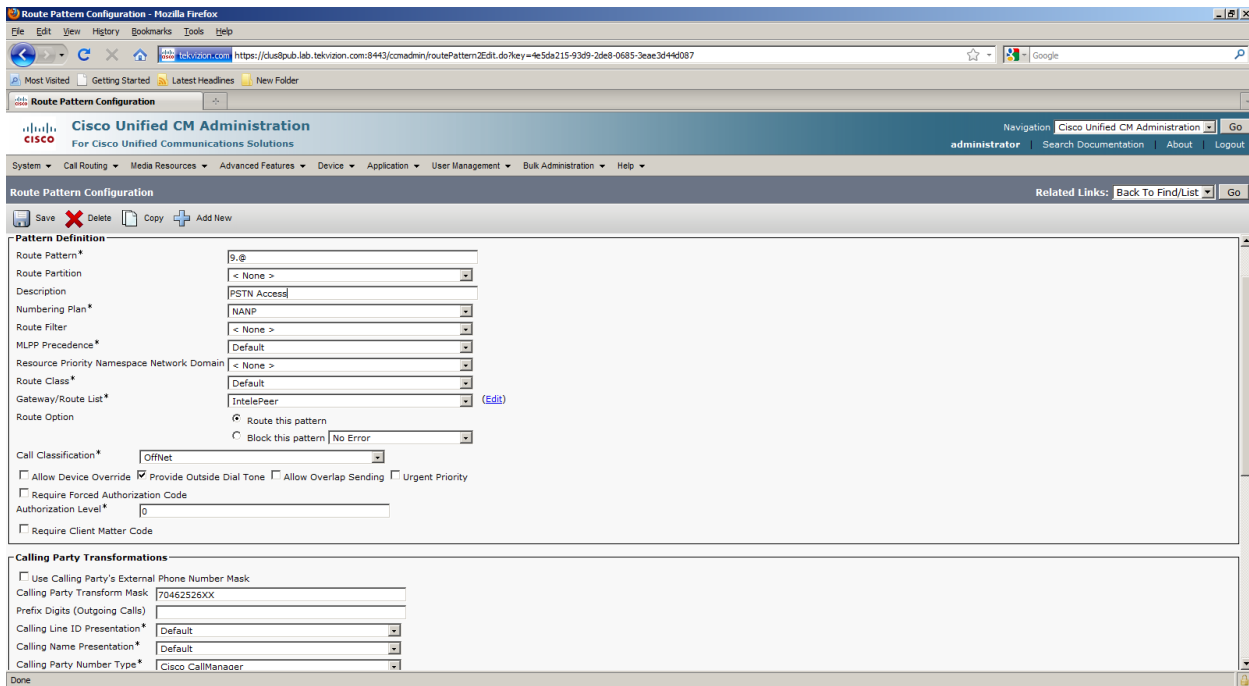




Figure 14. Route Pattern Configuration for SIP trunk to Intelpeer via CUBE (cont)

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### Route Pattern Configuration

Related Links: Back To Find/List Go

Save Delete Copy Add New

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation\* Allowed

Calling Name Presentation\* Allowed

Calling Party Number Type\* Cisco CallManager

Calling Party Numbering Plan\* Cisco CallManager

---

**Connected Party Transformations**

Connected Line ID Presentation\* Default

Connected Name Presentation\* Default

---

**Called Party Transformations**

Discard Digits **PreDot** Strip the leading "9" digit and transmit the remaining called digits to CUBE

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 Value |
|--------------------|------------------------|-------------------------|
| -- Not Selected -- | < Not Exist >          | <input type="text"/>    |

Save Delete Copy Add New

**i** \*- indicates required item.

Done Internet | Protected Mode: On 100%



**Acronyms**

| Acronym   | Definitions                          |
|-----------|--------------------------------------|
| SIP       | Session Initiation Protocol          |
| MGCP      | Media Gateway Control Protocol       |
| SCCP      | Skinny Client Control Protocol       |
| Cisco UCM | Cisco Unified Communications Manager |
| Cisco UBE | Cisco Unified Border Element         |
|           |                                      |
|           |                                      |
|           |                                      |
|           |                                      |
|           |                                      |
|           |                                      |
|           |                                      |
|           |                                      |
|           |                                      |
|           |                                      |



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EDCS# 1000350 Rev # Initial Revision

**Note:** Testing was conducted in TekVizion lab.