



# Telnet SIP Trunking: Connecting Cisco Unified Communications Manager 8.6(1) via the Cisco Unified Border Element 8.6 using SIP

February 25, 2012

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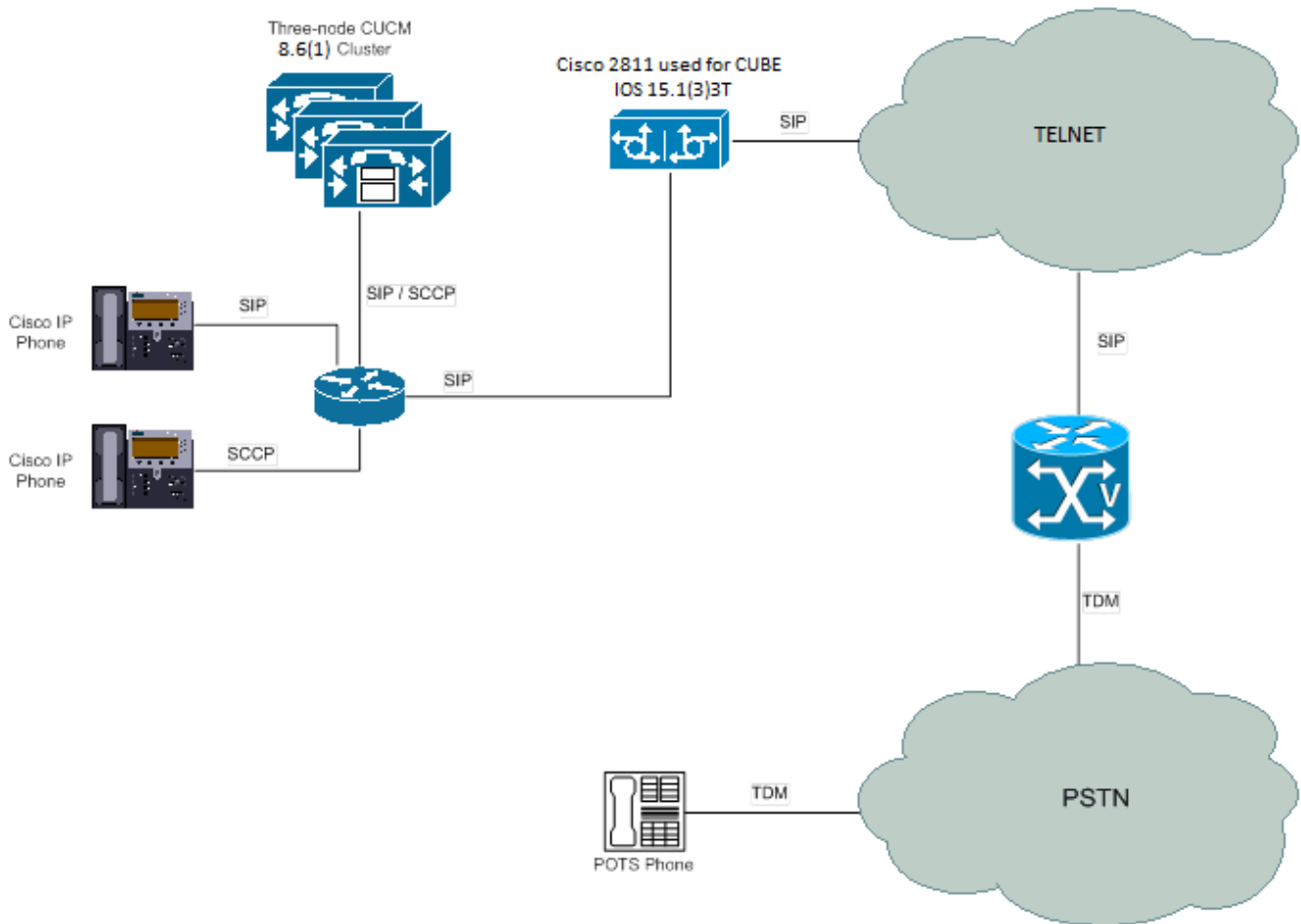
## Introduction

Service Providers today, such as Telnet, 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. Telnet 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) 8.6(1) with a Cisco Unified Border Element (Cisco UBE) for connectivity to the Telnet<sup>®</sup> SIP Trunking service. The deployment model covered in this application note is CPE (Cisco UCM 8.6(1)/Cisco UBE) to PSTN via Telnet<sup>®</sup> SIP Trunking. This document does not address 911 emergency outbound calls. For 911 feature service details contact Telnet, 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 Telnet 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 Telnet 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/cuem/admin/8\\_0\\_2/ccmsvs/a03ptcss.html](http://www.cisco.com/en/US/partner/docs/voice_ip_comm/cuem/admin/8_0_2/ccmsvs/a03ptcss.html)

## Network Topology

Figure 1. Lab Network Topology



## System Components

### Hardware Components

- Cisco 2811
- Cisco Unified Communications Manager (1-node cluster consisting of Cisco MCS 7800 Series servers)
- Cisco IP Phones
- TELNET - - GenBand C3 MGC
- TELNET - - GenBand G9 MG
- TELNET - - GenBand S3 SBC
- TELNET - - BroadSoft/BroadWorks

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EDCS# 247757 Rev# Initial version

**Note:** Testing was conducted at tekVizion Labs®.



## Software Requirements

- Cisco Unified Communications Manager 8.6.1.20000-1
- Cisco Unified Border Element, IOS version 15.1(3)3T (C2800NM-SPSERVICESK9-M)
- TELNET - - GenBand C3 MGC Rel. 8.1.58.05
- TELNET - - GenBand G9 MG Rel. 0801.05.03
- TELNET - - GenBand S3 SBC Rel. 7.1.6.8
- TELNET - - BroadSoft/BroadWorks Rel. 17.4



## 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)
- Early media cut-through with DTMF relay before 200 OK
- G.711 pass-through fax

### Features Not Supported

- Caller ID update via SIP UPDATE method
- T.38 fax



## Caveats

- Special consideration is required with the FROM header. Telnet only accepts INVITE requests with a FROM header that has a TelnetID. If caller id presentation is required, the information must be included in the P-asserted-identity, and synchronized with information provisioned on Telnet portal.
- Special consideration is required with the URI header on incoming call from Telnet. The Telnet network always send the pilot number registration on the request URI header and destination number in the TO header . CUCM use the URI header to route the call and not the TO header. The INVITE and CANCEL message require header manipulation that copy the TO header user part (Called number) to the URI header.
- 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. Telnet does not update to the original PSTN calling party number when the transfer is completed. Telnet 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 4 digits corresponding to the Extension.

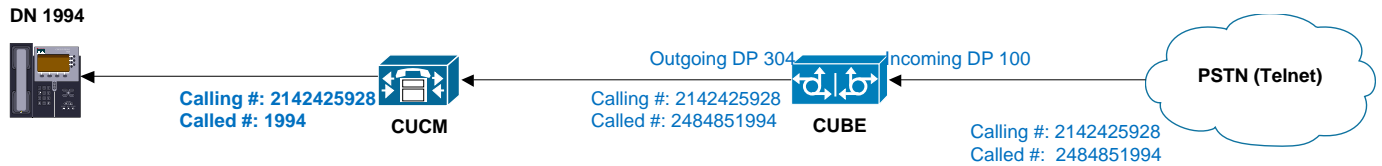
For incoming PSTN calls, the CUBE presents the full ten-digit DID number to CUCM. The CUCM trunk configuration strips all but the last four digits. Voice calls are routed to IP phones;

CPE callers make outbound PSTN calls by dialing a “91” prefix followed by the destination number. A “91.@” 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 15.1(3)T3, RELEASE SOFTWARE (fc1)

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

Copyright (c) 1986-2011 by Cisco Systems, Inc.

Compiled Wed 14-Dec-11 21:01 by prod\_rel\_team

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

Main2811 uptime is 5 days, 52 minutes

System returned to ROM by reload at 15:16:04 CST Fri Jan 27 2012

System restarted at 15:19:22 CST Fri Jan 27 2012

System image file is "usbflash0:c2800nm-spservicesk9-mz.151-3.T3.bin"

Last reload type: Normal Reload

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 T1/PRI port

DRAM configuration is 64 bits wide with parity enabled.

239K bytes of non-volatile configuration memory.

976880K bytes of USB Flash usbflash0 (Read/Write)

62720K bytes of ATA CompactFlash (Read/Write)

**Running Configuration:**

```
Current configuration : 7015 bytes
!
! Last configuration change at 15:41:08 CST Wed Feb 1 2012 by cisco
! NVRAM config last updated at 16:09:01 CST Tue Jan 31 2012 by cisco
!
version 15.1
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname Main2811
!
boot-start-marker
boot system usbflash0:c2800nm-spservicesk9-mz.151-3.T3.bin
boot-end-marker
```



```
!  
card type t1 1 0  
logging buffered 51200 warnings  
no logging console  
enable secret 5 $1$ELsJ$rMOpADTplX/DtwNNBvhY0  
!  
aaa new-model  
!  
aaa session-id common  
clock timezone CST -6 0  
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  
ip address trusted list3  
  ipv4 0.0.0.0 0.0.0.0  
  allow-connections sip to sip4  
  fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none5  
sip  
  rellxx disable  
  asserted-id pai  
  early-offer forced6  
  midcall-signaling passthru  
!  
voice class uri TRUNK sip  
  user-id 248485199.  
voice class codec 1  
  codec preference 1 g729r8  
  codec preference 2 g711ulaw
```

<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> IP address trusted list allowed to CUBE, only allow the network configure in this list.

<sup>4</sup> Allow SIP to SIP call processing

<sup>5</sup> Fax configuration for T38, to use G711u pass through change this line to: [fax protocol pass-through g711ulaw](#)

<sup>6</sup> Configures CUBE to send a SIP INVITE with SDP on an outbound call leg (Delayed Offer to Early Offer)



```
voice class sip-profiles 17
 request INVITE peer-header sip TO copy "sip:(.*)@" u01
 request CANCEL peer-header sip TO copy "sip:(.*)@" u02
 request INVITE sip-header SIP-Req-URI modify ".*@(.*)" "INVITE sip:\u01@\1"
 request INVITE sip-header To modify ".*@(.*)" "To: <sip:\u01@\1"
 request CANCEL sip-header SIP-Req-URI modify ".*@(.*)" "CANCEL sip:\u02@\1"
 request CANCEL sip-header To modify ".*@(.*)" "To: <sip:\u02@\1"
!
voice class sip-profiles 28
 request INVITE sip-header P-Asserted-Identity modify "174.46.0.150" "asmain.voip.telnetww.com"
 response ANY sdp-header Audio-Attribute modify "sendonly" "sendrecv"
 request ANY sdp-header Audio-Attribute modify "sendonly" "sendrecv"
!
voice class sip-copylist 1
 sip-header TO
!
crypto pki token default removal timeout 0
!
license udi pid CISCO2811 sn FHK0923F0YT
archive
 log config
  hidekeys
username cisco password 0 cisco
!
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
 duplex full
 speed 100
!
interface FastEthernet0/1
 description External WAN (Service Provider facing)
 ip address 174.46.0.150 255.255.255.128
 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
```

<sup>7</sup> **Sip-Profile 1** used to manipulate Invite and Cancel message coming from Telnet network. The Telnet network always send the pilot number registration on the request URI header and destination number in the TO header . CUCM use the URI header to route the call and not the TO header. These request manipulation copy the TO header user part (Called number) to the URI header user part.

<sup>8</sup> **Sip-Profile 2** used to manipulate Invite coming from CUCM. The first manipulation change the PAI header server part from IP address to FQDN and second manipulation change the sendonly attribute on SDP to sendrecv during hold to hear Music On Hold.



```
no cdp enable
!
ip forward-protocol nd
ip http server
no ip http secure-server
!
!
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!
!
!
!
!
control-plane
!
!
dial-peer voice 100 voip
description Incoming dialpeer and 1+10 digits to Telnet
destination-pattern ^1[2-9]..[2-9].....$
session protocol sipv2
session target sip-server
session transport udp
incoming called-number .T
incoming uri to TRUNK
voice-class codec 1
voice-class sip early-offer forced
voice-class sip profiles 2
voice-class sip copy-list 1
dtmf-relay rtp-nte
no vad
!
dial-peer voice 101 voip
description 10-digit local calls to Telnet
destination-pattern ^[2-9]..[2-9].....$
session protocol sipv2
session target sip-server
session transport udp
voice-class codec 1
voice-class sip early-offer forced
voice-class sip profiles 2
voice-class sip privacy id9
dtmf-relay rtp-nte
no vad
!
dial-peer voice 102 voip
description International calls to Telnet
destination-pattern ^011T
session protocol sipv2
session target sip-server
session transport udp
voice-class codec 1
voice-class sip early-offer forced
voice-class sip profiles 2
dtmf-relay rtp-nte
no vad
!
dial-peer voice 103 voip
description N11 calls to Telnet
destination-pattern ^[2-9]11$
```

---

<sup>9</sup> Enable Privacy in outgoing dial-peer using PAI header; remove this line to disable privacy.



```
session protocol sipv2
session target sip-server
session transport udp
voice-class codec 1
voice-class sip early-offer forced
voice-class sip profiles 2
dtmf-relay rtp-nte
no vad
!
dial-peer voice 302 voip
description Incoming calls from primary CUCM
huntstop
session protocol sipv2
session target ipv4:10.70.19.3
incoming called-number .%
voice-class codec 1
voice-class sip early-offer forced
dtmf-relay rtp-nte
no vad
!
dial-peer voice 303 voip
description Incoming calls from secondary CUCM
huntstop
session protocol sipv2
session target ipv4:10.70.19.4
incoming called-number .%
voice-class codec 1
voice-class sip early-offer forced
dtmf-relay rtp-nte
no vad!
!
dial-peer voice 304 voip
description 24848519.. calls to primary CUCM
huntstop
preference 1
destination-pattern ^24848519..10
session protocol sipv2
session target ipv4:10.70.19.311
voice-class codec 1
voice-class sip early-offer forced
voice-class sip profiles 1
dtmf-relay rtp-nte
no vad
!
dial-peer voice 305 voip
description 24848519.. calls to secondary CUCM
huntstop
preference 2
destination-pattern ^24848519..12
session protocol sipv2
session target ipv4:10.70.19.413
voice-class codec 1
```

<sup>10</sup> Outbound dial peer (to CUCM). The pattern here should match service provider-assigned DID numbers.

<sup>11</sup> Reference to CUCM primary IP address

<sup>12</sup> Outbound dial peer (to CUCM). The pattern here should match service provider-assigned DID numbers.

<sup>13</sup> Reference to CUCM secondary IP address



```
voice-class sip early-offer forced
voice-class sip profiles 1
dtmf-relay rtp-nte
no vad
!

gateway
timer receive-rtp 1200
!
sip-ua
credentials username 2484851990 password 7 014357530A5E51 realm BroadWorks14
authentication username 2484851990 password 7 014357530A5E51 realm BroadWorks15
no remote-party-id
retry invite 2
retry bye 2
retry cancel 2
retry register 10
registrar dns:asmain.voip.telnetww.com expires 360016
sip-server dns:asmain.voip.telnetww.com17
!
!
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>14</sup> Service Provider credentials for Invite challenge (User, Password and Realm)

<sup>15</sup> Service Provider authentication for registration challenge (User, Password and Realm)

<sup>16</sup> Service Provider registration signaling address and expiration timer

<sup>17</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

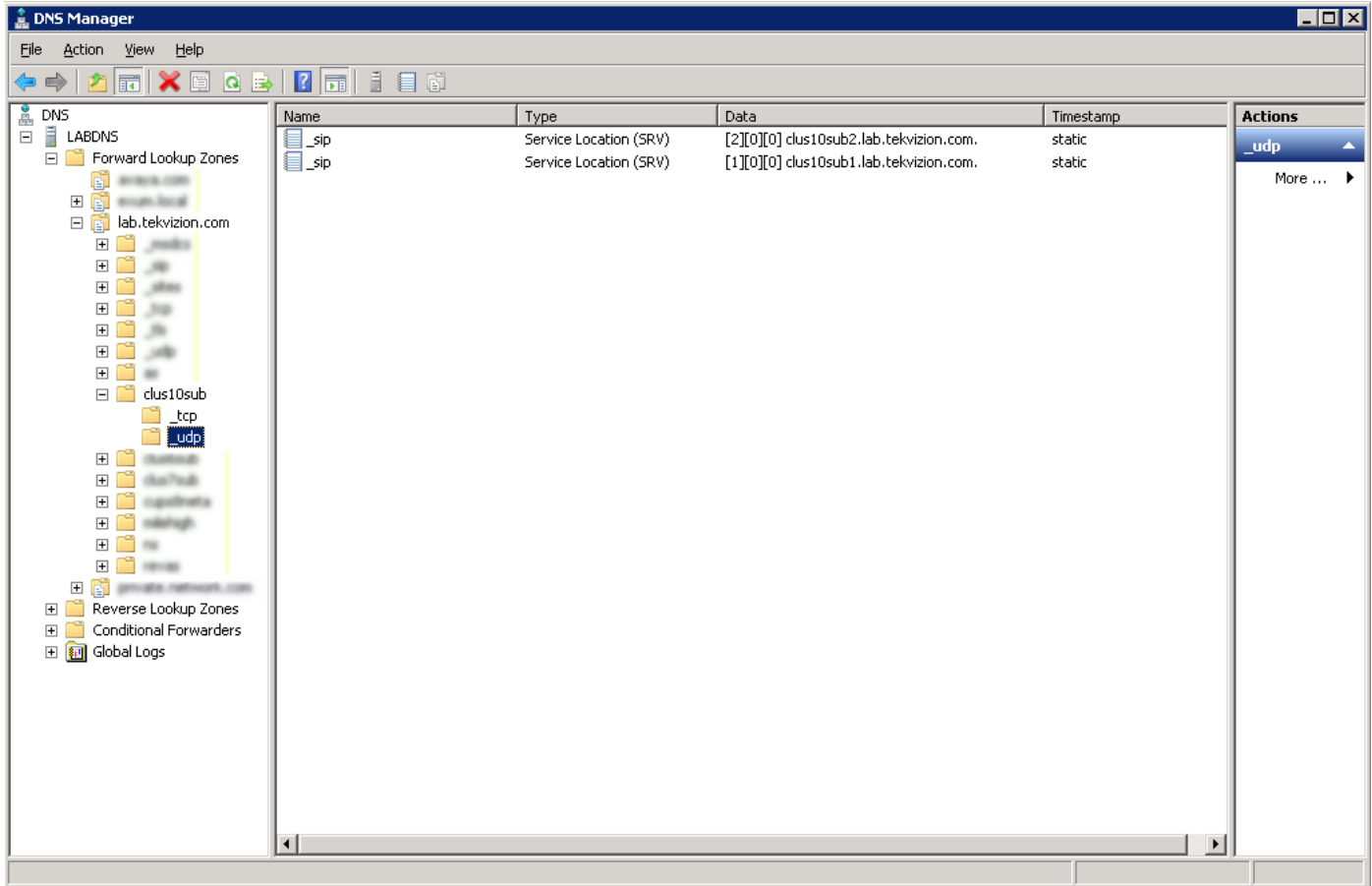
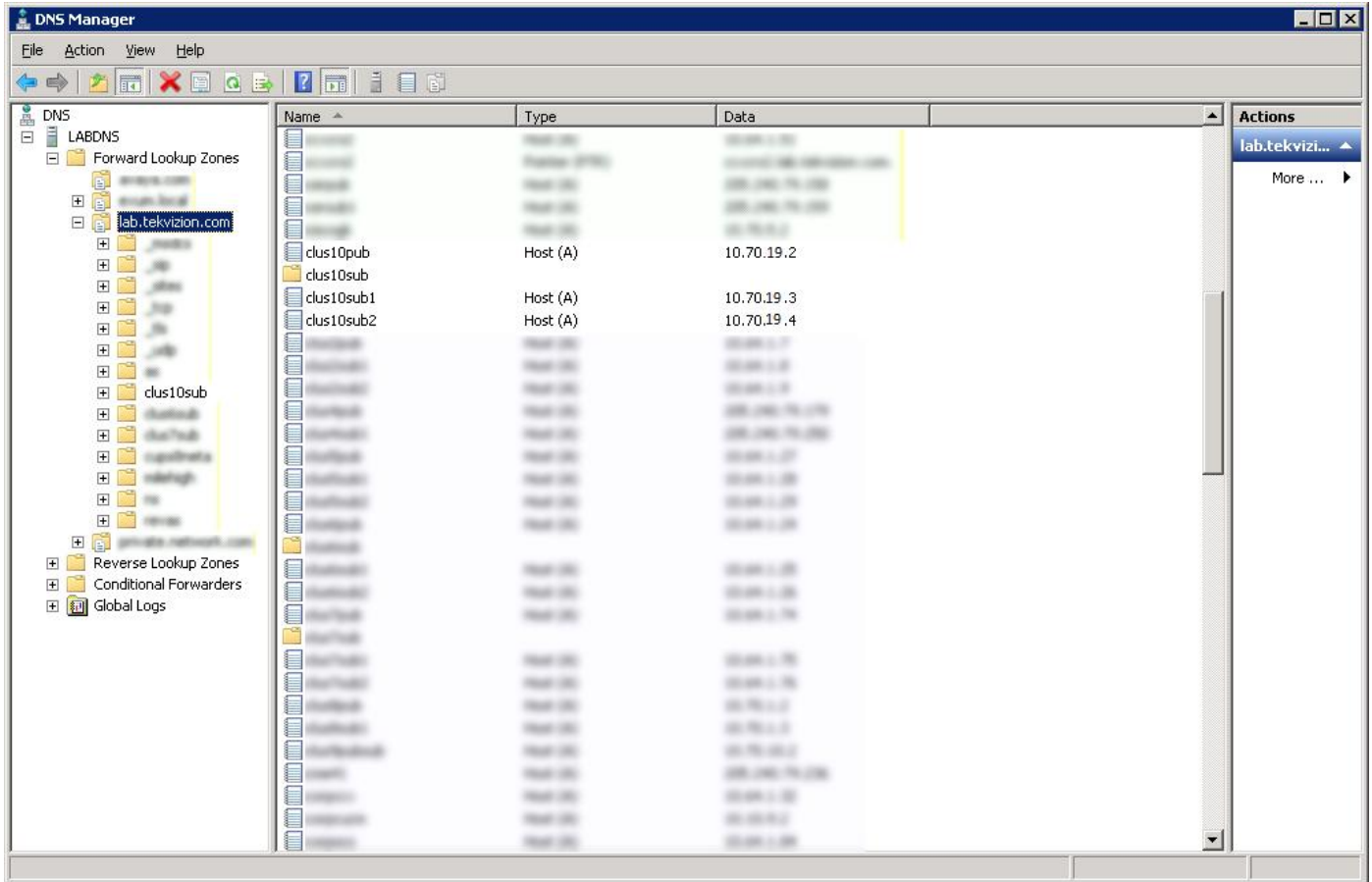




Figure 5. DNS ARecords

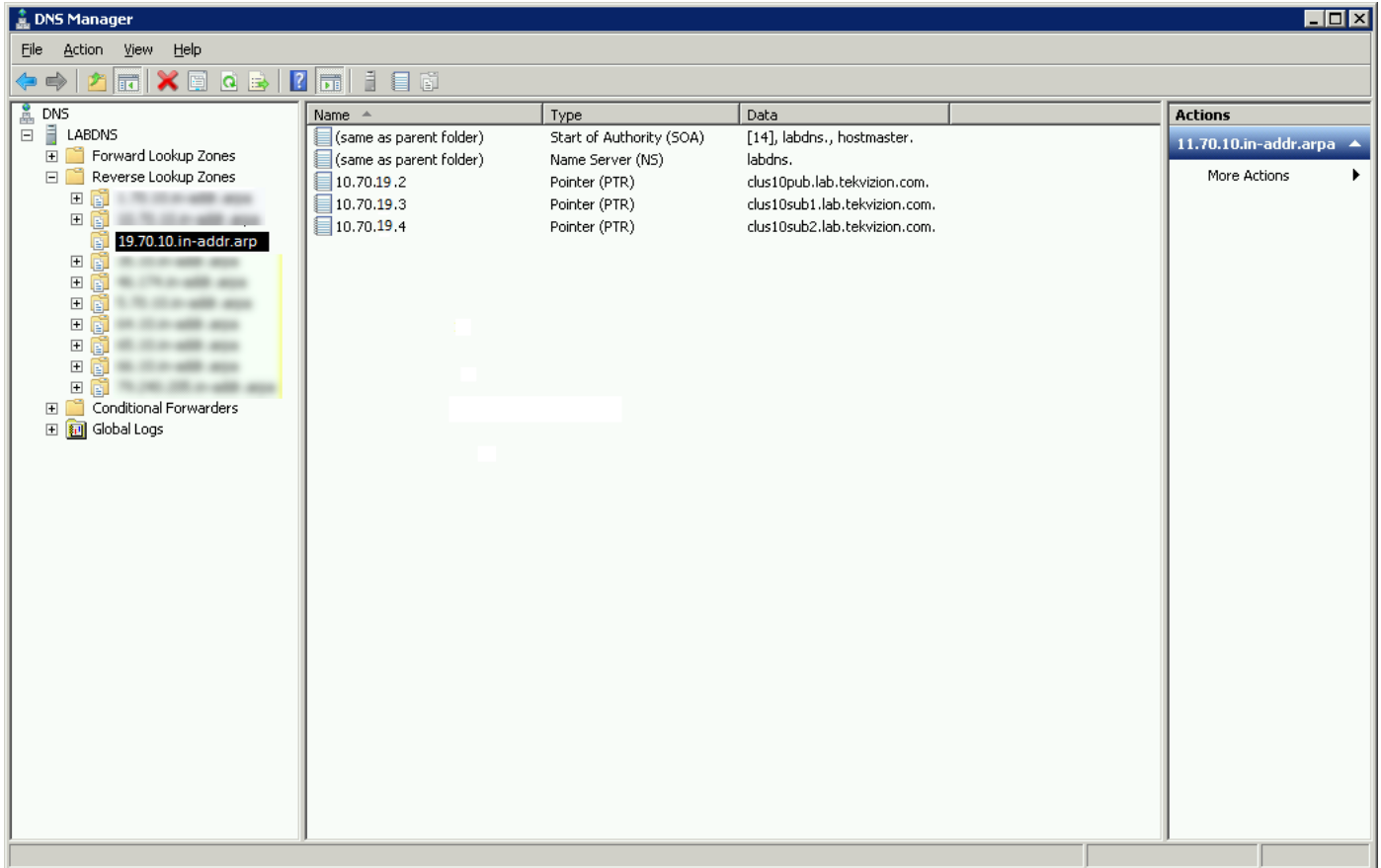


**Note:** Testing was conducted at tekVizion Labs®.





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

**SIP Trunk Security Profile Configuration** Related Links: [Back To Find/List](#)

---

**Status**

Status: Ready

---

**SIP Trunk Security Profile Information**

Name*	<input type="text" value="Non Secure SIP Trunk Profile Telnet"/>
Description	<input type="text" value="Non Secure SIP Trunk Profile authenticated by null Stri"/>
Device Security Mode	<input type="text" value="Non Secure"/>
Incoming Transport Type*	<input type="text" value="TCP+UDP"/>
Outgoing Transport Type	<input type="text" value="UDP"/>
<input type="checkbox"/> Enable Digest Authentication	
Nonce Validity Time (mins)*	<input type="text" value="600"/>
X.509 Subject Name	<input type="text"/>
Incoming Port*	<input type="text" value="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	
SIP V.150 Outbound SDP Offer Filtering*	<input type="text" value="Use Default Filter"/>



Figure 9. SIP Trunk to Telnet via CUBE

Trunk Configuration Related Links: [Back To Find/List](#)

**Device Information**

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	<input type="text" value="ServiceProvider"/>
Description	<input type="text" value="PSTN via CUBE"/>
Device Pool*	<input type="text" value="G729"/>
Common Device Configuration	<input type="text" value="&lt; None &gt;"/>
Call Classification*	<input type="text" value="Use System Default"/>
Media Resource Group List	<input type="text" value="&lt; None &gt;"/>
Location*	<input type="text" value="Hub_None"/>
AAR Group	<input type="text" value="&lt; None &gt;"/>
Tunneled Protocol*	<input type="text" value="None"/>
QSIG Variant*	<input type="text" value="No Changes"/>
ASN.1 ROSE OID Encoding*	<input type="text" value="No Changes"/>
Packet Capture Mode*	<input type="text" value="None"/>
Packet Capture Duration	<input type="text" value="0"/>

Media Termination Point Required

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\*

Route Class Signaling Enabled\*

Use Trusted Relay Point\*

PSTN Access

Run On All Active Unified CM Nodes



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

<b>Intercompany Media Engine (IME)</b>				
E.164 Transformation Profile < None >				
<b>Multilevel Precedence and Preemption (MLPP) Information</b>				
MLPP Domain < None >				
<b>Call Routing Information</b>				
<input type="checkbox"/> Remote-Party-Id				
<input checked="" type="checkbox"/> Asserted-Identity				
Asserted-Type* PAI				
SIP Privacy* Default				
<b>Inbound Calls</b>				
Significant Digits* 4				
Connected Line ID Presentation* Default				
Connected Name Presentation* Default				
Calling Search Space < None >				
AAR Calling Search Space < None >				
Prefix DN				
<input type="checkbox"/> Redirecting Diversion Header Delivery - Inbound				
<b>Incoming Calling Party Settings</b>				
If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.				
<input type="button" value="Clear Prefix Settings"/> <input type="button" value="Default Prefix Settings"/>				
<b>Number Type</b>	<b>Prefix</b>	<b>Strip Digits</b>	<b>Calling Search Space</b>	<b>Use Device Pool CSS</b>
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>
<b>Connected Party Settings</b>				
Connected Party Transformation CSS < None >				
<input checked="" type="checkbox"/> Use Device Pool Connected Party Transformation CSS				



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

**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\* First Redirect Number (External)

Calling Line ID Presentation\* Default

Calling Name Presentation\* Default

Caller ID DN

Caller Name

Redirecting Diversion Header Delivery - Outbound

---

**SIP Information**

**Destination**

Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1*	10.64.1.88		5060

MTP Preferred Originating Codec\* 711ulaw

Presence Group\* Standard Presence group

SIP Trunk Security Profile\* Non Secure SIP Trunk Profile Telnet

Rerouting Calling Search Space < None >

Out-Of-Dialog Refer Calling Search Space < None >

SUBSCRIBE Calling Search Space < None >

SIP Profile\* Standard SIP Profile

DTMF Signaling Method\* No Preference



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

**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\* First Redirect Number (External)

Calling Line ID Presentation\* Default

Calling Name Presentation\* Default

Caller ID DN

Caller Name

Redirecting Diversion Header Delivery - Outbound

---

**SIP Information**

**Destination**

Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1*	10.64.1.88		5060

MTP Preferred Originating Codec\* 711ulaw

Presence Group\* Standard Presence group

SIP Trunk Security Profile\* Non Secure SIP Trunk Profile Telnet

Rerouting Calling Search Space < None >

Out-Of-Dialog Refer Calling Search Space < None >

SUBSCRIBE Calling Search Space < None >

SIP Profile\* Standard SIP Profile

DTMF Signaling Method\* No Preference



Figure 13. RoutePattern Configuration for SIP trunk to Telnet via CUBE

**Route Pattern Configuration** Related Links: [Back To Find/List](#)

---

**Status**

Status: Ready

---

**Pattern Definition**

Route Pattern*	<input type="text" value="91.@"/>
Route Partition	< None >
Description	<input type="text" value="PSTN Access"/>
Numbering Plan*	NANP
Route Filter	< None >
MLPP Precedence*	Default
<input type="checkbox"/> Apply Call Blocking Percentage	<input type="text"/>
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Gateway/Route List*	ServiceProvider <a href="#">(Edit)</a>
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern <input type="text" value="No Error"/>
Call Classification*	OffNet
<input type="checkbox"/> Allow Device Override	<input checked="" type="checkbox"/> Provide Outside Dial Tone
<input type="checkbox"/> Allow Overlap Sending	<input type="checkbox"/> Urgent Priority
<input type="checkbox"/> Require Forced Authorization Code	
Authorization Level*	<input type="text" value="0"/>
<input type="checkbox"/> Require Client Matter Code	

---

**Calling Party Transformations**

<input type="checkbox"/> Use Calling Party's External Phone Number Mask	
Calling Party Transform Mask	<input type="text" value="248485XXXX "/>
Prefix Digits (Outgoing Calls)	<input type="text"/>
Calling Line ID Presentation*	Default
Calling Name Presentation*	Default
Calling Party Number Type*	Cisco CallManager
Calling Party Numbering Plan*	Cisco CallManager





Figure 14. RoutePattern Configuration for SIP trunk to Telnet via CUBE (cont)

<b>Connected Party Transformations</b>		
Connected Line ID Presentation*	Default	
Connected Name Presentation*	Default	
<b>Called Party Transformations</b>		
Discard Digits	PreDot	Strip the leading "91" 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	
<b>ISDN Network-Specific Facilities Information Element</b>		
Network Service Protocol	-- Not Selected --	
Carrier Identification Code		
Network Service	Service Parameter Name	Service Parameter Value
-- Not Selected --	< Not Exist >	



**Acronyms**

<b>Acronym</b>	<b>Definitions</b>
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

**Note:** Testing was conducted at tekVizion Labs®.



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**Corporate  
Headquarters**

Cisco Systems, Inc.  
170 West Tasman Drive  
San Jose, CA 95134-1706  
USA  
www.cisco.com  
Tel: 408 526-4000  
800 553-NETS (6387)  
Fax: 408 526-4100

**European  
Headquarters**

Cisco Systems International  
BV  
Haarlerbergpark  
Haarlerbergweg 13-19  
1101 CH Amsterdam  
The Netherlands  
www-europe.cisco.com  
Tel: 31 0 20 357 1000  
Fax: 31 0 20 357 1100

**Americas  
Headquarters**

Cisco Systems, Inc.  
170 West Tasman Drive  
San Jose, CA 95134-1706  
USA  
www.cisco.com  
Tel: 408 526-7660  
Fax: 408 527-0883

**Asia Pacific  
Headquarters**

Cisco Systems, Inc.  
Capital Tower  
168 Robinson Road  
#22-01 to #29-01  
Singapore 068912  
www.cisco.com  
Tel: +65 317 7777  
Fax: +65 317 7799

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