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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Note: Testing was conducted in tekVizion labs.
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® SIP Trunking service. The deployment model covered in this application note is CPE (Cisco UCM 8.6(1)/Cisco UBE) to PSTN via Telnet® 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:
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

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 MG C 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

Note: Testing was conducted at tekVizion Labs®.
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

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

<table>
<thead>
<tr>
<th>DN 1994</th>
<th>User dials 912142425928</th>
<th>CUCM</th>
<th>Called #: 2142425928</th>
<th>Calling #: 2484851994</th>
<th>CUBE</th>
<th>Called #: 2142425928</th>
<th>Calling #: 2484851994</th>
<th>PSTN (Telnet)</th>
</tr>
</thead>
</table>

**Figure 3. Inbound Voice Call**

<table>
<thead>
<tr>
<th>DN 1994</th>
<th>Calling #: 2142425928</th>
<th>CUCM</th>
<th>Called #: 2142425928</th>
<th>Calling #: 2484851994</th>
<th>CUBE</th>
<th>Called #: 2142425928</th>
<th>Calling #: 2484851994</th>
<th>PSTN (Telnet)</th>
</tr>
</thead>
</table>

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

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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

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

If you require further assistance please contact us by sending email to 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

Note: Testing was conducted at tekVizion Labs®.
! card type t1 1 0
logging buffered 51200 warnings
no logging console
enable secret 5 $1$SELSJ$rMOpADTplX/DtBNhY0
!

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.com
ip name-server 10.64.1.3
no ipv6 cef
ntp server 10.10.10.5
multilink bundle-name authenticated
!
isd switch-type primary-ni
!
voice rtp send-recv
!
voice service voip
ip address trusted list
 ipv4 0.0.0.0 0.0.0.0
allow-connections sip to sip
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
sip
rel1xx disable
asserted-id pai
early-offer forced
midcall-signaling passthru
!
voice class uri TRUNK sip
 user-id 248485199.
voice class codec 1
codec preference 1 g729r8
codec preference 2 g711ulaw

1 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.

2 IP address of DNS server

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

4 Allow SIP to SIP call processing

5 Fax configuration for T38, to use G711u pass through change this line to: fax protocol pass-through g711ulaw

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

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Note: Testing was conducted at tekVizion Labs ®.
voice class sip-profiles 1
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 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 2
request INVITE sip-header PAI Asserted-Identity modify "174.46.0.150" "asmain.voip.telnetww.com"
response ANY sip-header Audio-Attribute modify "sendonly" "sendrecv"
request ANY sip-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 174.46.0.150 255.255.128.128
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

1 SipProfile 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.

2 SipProfile 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.

Note: Testing was conducted at tekVizion Labs®.
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 id
#戸
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$ 

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

Note: Testing was conducted at tekVizion Labs®.
session protocol sipv2
session target sip-server
session transport udp
voice-class codec 1
voice-class sip early-offer forced
dtmf-relay rtp-npe
!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-npe
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-npe
no vad!
!
!dial-peer voice 304 voip
description 24848519.. calls to primary CUCM
huntstop
preference 1
destination-pattern ”24848519.”
session protocol sipv2
session target ipv4:10.70.19.3
voice-class codec 1
voice-class sip early-offer forced
dtmf-relay rtp-npe
no vad
!
!
!dial-peer voice 305 voip
description 24848519.. calls to secondary CUCM
huntstop
preference 2
destination-pattern ”24848519.”
session protocol sipv2
session target ipv4:10.70.19.4
voice-class codec 1

Outbound dial peer (to CUCM). The pattern here should match service provider-assigned DID numbers.

Reference to CUCM primary IP address

Outbound dial peer (to CUCM). The pattern here should match service provider-assigned DID numbers.

Reference to CUCM secondary IP address
voice-class sip early-offer forced
voice-class sip profiles 1
dtmf-relay rtp-nate
no vad
!
gateway
timer receive-rtp 1200
!
sip ua
  credentials username 2484851990 password 7 014357530A5E51 realm BroadWorks®
  authentication username 2484851990 password 7 014357530A5E51 realm BroadWorks®
  no remote-party-id
  retry invite 2
  retry bye 2
  retry cancel 2
  retry register 10
  registrar dns:asmain.voip.telnets.com expires 3600
  sip server dns:asmain.voip.telnets.com
!
!
line con 0
exec-timeout 0 0
line aux 0
line vty 0 4
exec-timeout 0 0
transport input telnet
line vty 5 15
  privilege level 15
!
scheduler allocate 20000 1000
end

---

14 Service Provider credentials for Invite challenge (User, Password and Realm)

15 Service Provider authentication for registration challenge (User, Password and Realm)

16 Service Provider registration signaling address and expiration timer

17 Service Provider signaling address

---

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Note: Testing was conducted at tekVizion Labs®.
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

![DNS SRV Records](image-url)
Figure 5. DNS A Records
Figure 6. DNS PTR Records

Note: Testing was conducted at tekVizion Labs®.
### Configuring the Cisco Unified Communications Manager

**Figure 7. Enterprise Parameters**

![Enterprise Parameters Configuration](image)

**Note:** Testing was conducted at tekVizion Labs®.
Figure 8. SIP Trunk Security Profile

![SIP Trunk Security Profile Configuration](image)

**Note:** Testing was conducted at tekVizion Labs®.
**Figure 9.** SIP Trunk to Telnet via CUBE

<table>
<thead>
<tr>
<th>Trunk Configuration</th>
<th>Related Links: Back To Find/List</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
</tr>
<tr>
<td><strong>Device Information</strong></td>
<td><strong>Trunk Configuration</strong></td>
</tr>
<tr>
<td>Product</td>
<td>SIP Trunk</td>
</tr>
<tr>
<td>Device Protocol</td>
<td>SIP</td>
</tr>
<tr>
<td>Trunk Service Type</td>
<td>None (Default)</td>
</tr>
<tr>
<td>Device Name</td>
<td>Service Provider</td>
</tr>
<tr>
<td>Description</td>
<td>PSTN via CUBE</td>
</tr>
<tr>
<td>Device Pool</td>
<td>G729</td>
</tr>
<tr>
<td>Common Device Configuration</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Call Classification</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Location</td>
<td>Hub, None</td>
</tr>
<tr>
<td>AAR Group</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Tunneled Protocol</td>
<td>None</td>
</tr>
<tr>
<td>QSIG Variant</td>
<td>No Changes</td>
</tr>
<tr>
<td>ASN.1 ROSE OID Encoding</td>
<td>No Changes</td>
</tr>
<tr>
<td>Packet Capture Mode</td>
<td>None</td>
</tr>
<tr>
<td>Packet Capture Duration</td>
<td>0</td>
</tr>
<tr>
<td>Media Termination Point Required</td>
<td>No Changes</td>
</tr>
<tr>
<td>Retry Video Call as Audio</td>
<td>No Changes</td>
</tr>
<tr>
<td>Path Replacement Support</td>
<td>No Changes</td>
</tr>
<tr>
<td>Transmit UTF-8 for Calling Party Name</td>
<td>No Changes</td>
</tr>
<tr>
<td>Transmit UTF-8 Names in QSIG APDU</td>
<td>No Changes</td>
</tr>
<tr>
<td>Unattended Port</td>
<td>No Changes</td>
</tr>
<tr>
<td>SRTP Allowed</td>
<td>When using both SRTP and TLS</td>
</tr>
<tr>
<td>Consider Traffic On This Trunk Secure</td>
<td>Default</td>
</tr>
<tr>
<td>Route Class Signaling Enabled</td>
<td>Default</td>
</tr>
<tr>
<td>Use Trusted Relay Point</td>
<td>Default</td>
</tr>
<tr>
<td>PSTN Access</td>
<td></td>
</tr>
<tr>
<td>Run On All Active Unified CM Nodes</td>
<td></td>
</tr>
</tbody>
</table>

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**Note:** Testing was conducted at tekVizion Labs®.
Figure 10. SIP Trunk to Telnet via CUBE (cont.)

### Intercompany Media Engine (IME)
- E.164 Transformation Profile: None

### Multilevel Precedence and Preemption (MLPP) Information
- MLPP Domain: None

### Call Routing Information
- Remote Party Id: Checked
- Asserted-Identity: Checked
  - Asserted-Type: PAI
- SIP Privacy: Default

### Inbound Calls
- Significant Digits: 4
- Connected Line ID Presentation: Default
- Connected Name Presentation: Default
- Calling Search Space: None
- AAR Calling Search Space: None
- Prefix DN: 

### Incoming Calling Party Settings
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.

<table>
<thead>
<tr>
<th>Number Type</th>
<th>Prefix</th>
<th>Strip Digits</th>
<th>Calling Search Space</th>
<th>Use Device Pool CSS</th>
</tr>
</thead>
<tbody>
<tr>
<td>Incoming Number</td>
<td>Default</td>
<td>0</td>
<td>None</td>
<td>✓</td>
</tr>
</tbody>
</table>

### Connected Party Settings
- Connected Party Transformation CSS: None
- Use Device Pool Connected Party Transformation CSS: ✓
Figure 11.  SIP Trunk to Telnet via CUBE (cont.)

<table>
<thead>
<tr>
<th>Outbound Calls</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Called Party Transformation CSS</td>
<td>&quot;None&quot;</td>
</tr>
<tr>
<td>Use Device Pool Called Party Transformation CSS</td>
<td></td>
</tr>
<tr>
<td>Calling Party Transformation CSS</td>
<td>&quot;None&quot;</td>
</tr>
<tr>
<td>Use Device Pool Calling Party Transformation CSS</td>
<td></td>
</tr>
<tr>
<td>Calling Party Selection</td>
<td>First Redirect Number [External]</td>
</tr>
<tr>
<td>Calling Line ID Presentation</td>
<td>Default</td>
</tr>
<tr>
<td>Calling Name Presentation</td>
<td>Default</td>
</tr>
<tr>
<td>Caller ID DN</td>
<td></td>
</tr>
<tr>
<td>Caller Name</td>
<td></td>
</tr>
<tr>
<td>Redirecting Diversion Header Delivery - Outbound</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Destination Address</th>
<th>Destination Address IPv6</th>
<th>Destination Port</th>
</tr>
</thead>
<tbody>
<tr>
<td>10.64.1.88</td>
<td></td>
<td>5060</td>
</tr>
</tbody>
</table>

| 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 |

Note: Testing was conducted at tekVizion Labs®.
Figure 12. SIP Trunk to Telnet via CUBE (cont.)

<table>
<thead>
<tr>
<th><strong>Outbound Calls</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td>Called Party Transformation CSS</td>
</tr>
<tr>
<td>None</td>
</tr>
<tr>
<td>Use Device Pool Called Party Transformation CSS</td>
</tr>
<tr>
<td>Calling Party Transformation CSS</td>
</tr>
<tr>
<td>None</td>
</tr>
<tr>
<td>Use Device Pool Calling Party Transformation CSS</td>
</tr>
<tr>
<td>Calling Party Selection</td>
</tr>
<tr>
<td>First Redirect Number (External)</td>
</tr>
<tr>
<td>Calling Line ID Presentation</td>
</tr>
<tr>
<td>Default</td>
</tr>
<tr>
<td>Calling Name Presentation</td>
</tr>
<tr>
<td>Default</td>
</tr>
<tr>
<td>Caller DN</td>
</tr>
<tr>
<td></td>
</tr>
<tr>
<td>Caller Name</td>
</tr>
<tr>
<td></td>
</tr>
<tr>
<td>Redirecting Diversion Header Delivery - Outbound</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th><strong>SIP Information</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td>Destination Address is an SRV</td>
</tr>
<tr>
<td>---------------------------------------</td>
</tr>
<tr>
<td>Destination Address</td>
</tr>
<tr>
<td>10.64.1.88</td>
</tr>
<tr>
<td>Destination Address IPv6</td>
</tr>
<tr>
<td></td>
</tr>
<tr>
<td>Destination Port</td>
</tr>
<tr>
<td>5060</td>
</tr>
<tr>
<td>MTP Preferred Originating Codec</td>
</tr>
<tr>
<td>711law</td>
</tr>
<tr>
<td>Presence Group</td>
</tr>
<tr>
<td>Standard Presence group</td>
</tr>
<tr>
<td>SIP Trunk Security Profile</td>
</tr>
<tr>
<td>Non Secure SIP Trunk Profile.Telnet</td>
</tr>
<tr>
<td>Rerouting Calling Search Space</td>
</tr>
<tr>
<td>None</td>
</tr>
<tr>
<td>Out-Of-Dialog Refer Calling Search Space</td>
</tr>
<tr>
<td>None</td>
</tr>
<tr>
<td>SUBSCRIBE Calling Search Space</td>
</tr>
<tr>
<td>None</td>
</tr>
<tr>
<td>SIP Profile</td>
</tr>
<tr>
<td>Standard SIP Profile</td>
</tr>
<tr>
<td>DTMF Signaling Method</td>
</tr>
<tr>
<td>No Preference</td>
</tr>
</tbody>
</table>

**Note:** Testing was conducted at tekVizion Labs®.
Figure 13. Route Pattern Configuration for SIP trunk to Telnet via CUBE

<table>
<thead>
<tr>
<th>Route Pattern Configuration</th>
<th>Related Links:</th>
<th>Back To Find/List</th>
</tr>
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<tbody>
<tr>
<td><strong>Status</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Status: Ready</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td><strong>Pattern Definition</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Route Pattern</td>
<td>910</td>
<td></td>
</tr>
<tr>
<td>Route Partition</td>
<td>&lt; None &gt;</td>
<td>Route Partition</td>
</tr>
<tr>
<td>Description</td>
<td>PSTN Access</td>
<td>Description</td>
</tr>
<tr>
<td>Numbering Plan</td>
<td>NANP</td>
<td>Numbering Plan</td>
</tr>
<tr>
<td>Route Filter</td>
<td>&lt; None &gt;</td>
<td>Route Filter</td>
</tr>
<tr>
<td>MLPP Precedence</td>
<td>Default</td>
<td>MLPP Precedence</td>
</tr>
<tr>
<td>Apply Call Blocking Percentage</td>
<td></td>
<td>Apply Call</td>
</tr>
<tr>
<td>Resource Priority Namespace Network Domain</td>
<td></td>
<td>Blocking Percentage</td>
</tr>
<tr>
<td>Route Class</td>
<td>Default</td>
<td>Route Class</td>
</tr>
<tr>
<td>Gateway/Route List</td>
<td>ServiceProvider</td>
<td>Gateway/Route</td>
</tr>
<tr>
<td>Route Option</td>
<td>Route this pattern</td>
<td>Route Option</td>
</tr>
<tr>
<td>Call Classification</td>
<td>OffNet</td>
<td>Call Classification</td>
</tr>
<tr>
<td>Allow Device Override</td>
<td>Yes</td>
<td>Allow Device</td>
</tr>
<tr>
<td>Provide Outside Dial Tone</td>
<td>Yes</td>
<td>Provide Outside</td>
</tr>
<tr>
<td>Allow Overlap Sending</td>
<td>Yes</td>
<td>Allow Overlap</td>
</tr>
<tr>
<td>Urgent Priority</td>
<td>Yes</td>
<td>Urgent Priority</td>
</tr>
<tr>
<td>Require Forced Authorization Code</td>
<td>No</td>
<td>Require Forced</td>
</tr>
<tr>
<td>Authorization Level</td>
<td>0</td>
<td>Authorization Level</td>
</tr>
<tr>
<td>Require Client Matter Code</td>
<td></td>
<td>Require Client</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Matter Code</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td><strong>Calling Party Transformations</strong></td>
<td></td>
<td>Calling Party</td>
</tr>
<tr>
<td>Use Calling Party’s External Phone Number Mask</td>
<td>Yes</td>
<td>Use Calling Party’s</td>
</tr>
<tr>
<td>Calling Party Transform Mask</td>
<td>246485XXXXX</td>
<td>Calling Party</td>
</tr>
<tr>
<td>Prefix Digits (Outgoing Calls)</td>
<td></td>
<td>Prefix Digits</td>
</tr>
<tr>
<td>Calling Line ID Presentation</td>
<td>Default</td>
<td>Calling Line ID</td>
</tr>
<tr>
<td>Calling Name Presentation</td>
<td>Default</td>
<td>Calling Name</td>
</tr>
<tr>
<td>Calling Party Number Type</td>
<td>Cisco CallManager</td>
<td>Calling Party</td>
</tr>
<tr>
<td>Calling Party Numbering Plan</td>
<td>Cisco CallManager</td>
<td>Calling Party</td>
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</table>

**Note:** Testing was conducted at tekVizion Labs®.
Figure 14. Route Pattern Configuration for SIP trunk to Telnet via CUBE (cont)

<table>
<thead>
<tr>
<th>Connected Party Transformations</th>
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<tbody>
<tr>
<td>Connected Line ID Presentation</td>
<td>Default</td>
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<tr>
<td>Connected Name Presentation</td>
<td>Default</td>
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</table>

<table>
<thead>
<tr>
<th>Called Party Transformations</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Discard Digits</td>
<td>PreDot</td>
</tr>
<tr>
<td>Called Party Transform Mask</td>
<td>Strip the leading “91” digit and transmit the remaining called digits to CUBE</td>
</tr>
<tr>
<td>Prefix Digits (Outgoing Calls)</td>
<td></td>
</tr>
<tr>
<td>Called Party Number Type</td>
<td>Cisco CallManager</td>
</tr>
<tr>
<td>Called Party Numbering Plan</td>
<td>Cisco CallManager</td>
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</table>

<table>
<thead>
<tr>
<th>EISN Network-Specific Facilities Information Element</th>
<th></th>
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</thead>
<tbody>
<tr>
<td>Network Service Protocol</td>
<td>Not Selected</td>
</tr>
<tr>
<td>Network Service Identification Code</td>
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</tr>
<tr>
<td>Network Service</td>
<td>Service Parameter Name</td>
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<td>---</td>
<td>---</td>
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<tr>
<td>Not Selected</td>
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### Acronyms

<table>
<thead>
<tr>
<th>Acronym</th>
<th>Definitions</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIP</td>
<td>Session Initiation Protocol</td>
</tr>
<tr>
<td>MGCP</td>
<td>Media Gateway Control Protocol</td>
</tr>
<tr>
<td>SCCP</td>
<td>Skinny Client Control Protocol</td>
</tr>
<tr>
<td>Cisco UCM</td>
<td>Cisco Unified Communications Manager</td>
</tr>
<tr>
<td>Cisco UBE</td>
<td>Cisco Unified Border Element</td>
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</table>

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