



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

June 3, 2011

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



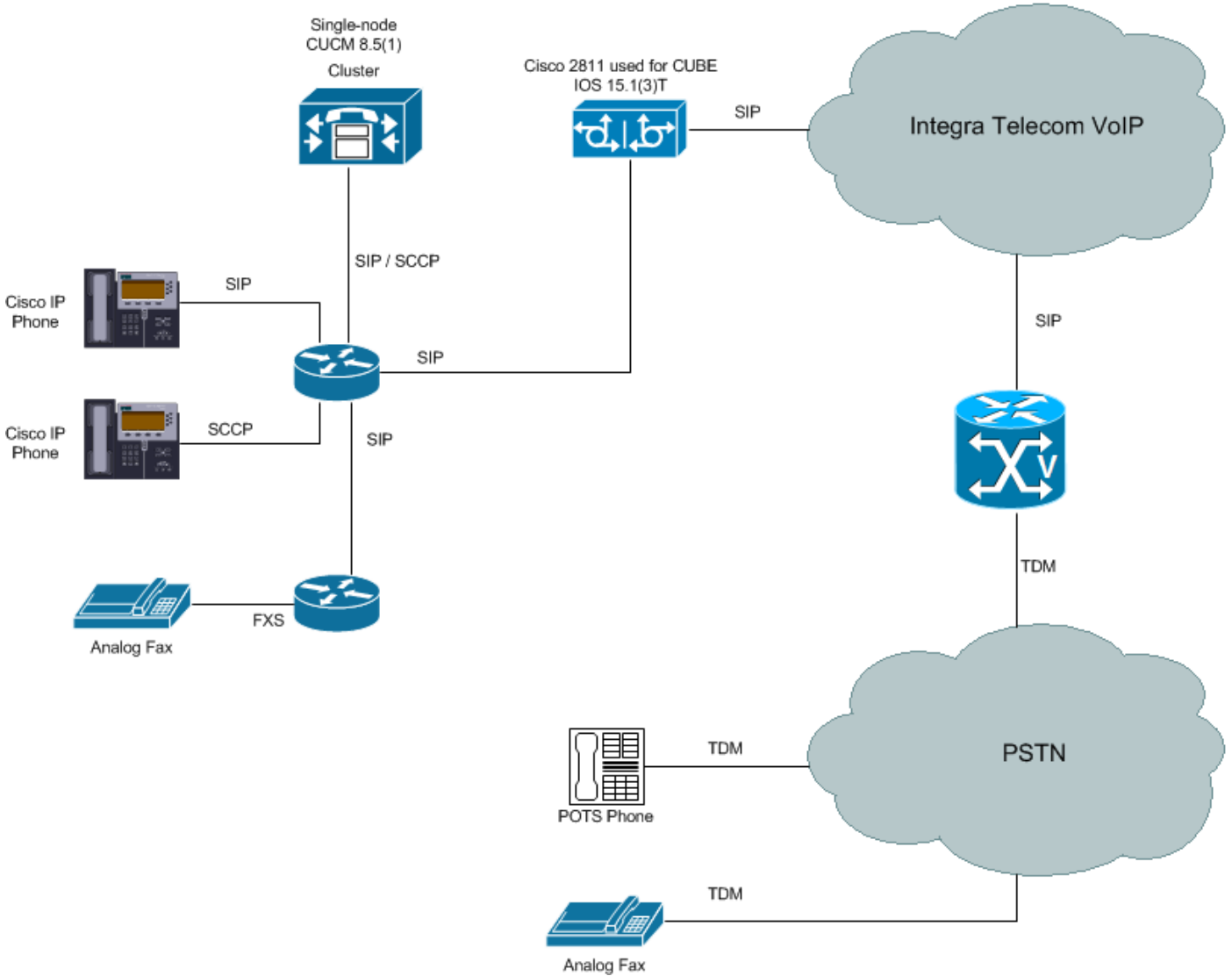
## Introduction

Service Providers today, such as Integra Telecom, 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. Integra Telecom 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 Integra Telecom SIP Trunking service. The deployment model covered in this application note is CPE (Cisco UCM 8.5(1)/Cisco UBE) to PSTN via Integra Telecom SIP Trunking. This document does not address 911 emergency outbound calls. For 911 feature service details contact Integra Telecom, 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 Integra Telecom's 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 Integra Telecom's 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 (used as Cisco Unified Border Element)
- Cisco 3845 with VIC-4FXS/DID (used for analog fax and hardware MTP)
- Cisco Unified Communications Manager (single-node cluster consisting of one Cisco MCS 7800 Series server)
- Cisco IP Phones



## Software Requirements

- Cisco Unified Communications Manager 8.5.1.10000-26
- Cisco Unified Border Element, IOS version 15.1(3)T (c2800nm-spservicesk9-mz.151-3.T.bin)
- Cisco 3845 router, IOS version 15.1(3)T (c3845-adventerprisek9-mz.151-3.T.bin)



## Features

### Features Supported

- Voice calls using G.729 and G.711 codecs
- RFC 3261 support
- Early media cut-through with DTMF relay before 200 OK
- Calling number presentation / restriction
- Call conferencing
- Call transfer (attended and unattended)
- Call hold and resume
- Call forwarding
- DTMF relay (RFC 2833)
- T.38 fax
- G.711 pass-through fax

### Features Not Supported

- Caller ID update via SIP UPDATE method

### Caveats

- When a PSTN to CPE call is transferred by the CPE to a second PSTN number, the outgoing call leg fails due to CUBE bug in which CUBE fails to offer the correct codec list to the service provider.



## Call Flows

In the sample configuration presented here, CUCM is provisioned with four-digit directory numbers corresponding to the last four DID digits. No digit manipulation is performed on the CUBE.

For incoming PSTN calls, the CUBE presents the full ten-digit DID number to CUCM. A CUCM Translation Pattern strips all but the last four digits and routes the call based on those digits. Voice calls are routed to IP phones; fax calls are routed via a 4-digit route pattern to a SIP trunk that terminates on the router hosting the analog fax endpoint.

CPE callers make outbound PSTN calls by dialing a “90” prefix followed by the destination number. For outbound fax calls from the analog fax endpoint, the router hosting that endpoint delivers all digits including the “90” prefix. A “90.@” 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



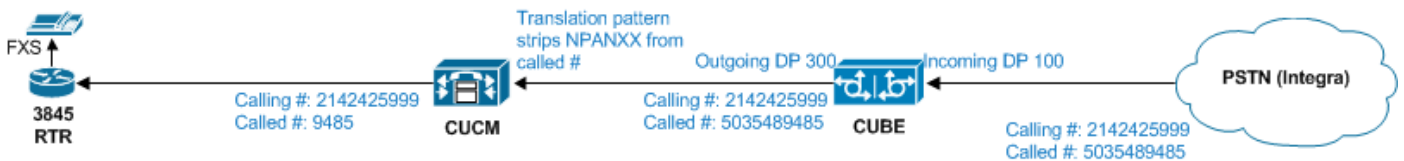
Figure 3. Inbound Voice Call



Figure 4. Outbound fax call



Figure 5. Inbound fax call



**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)T, RELEASE SOFTWARE (fc1)  
Technical Support: <http://www.cisco.com/techsupport>  
Copyright (c) 1986-2010 by Cisco Systems, Inc.  
Compiled Mon 15-Nov-10 21:41 by prod\_rel\_team

ROM: System Bootstrap, Version 12.4(13r)T11, RELEASE SOFTWARE (fc1)

MTPnetB uptime is 7 weeks, 4 days, 11 hours, 38 minutes  
System returned to ROM by power-on  
System image file is "flash:c2800nm-spservicesk9-mz.151-3.T.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 512000K/12288K bytes of memory.  
Processor board ID FTX1411ALT7  
2 FastEthernet interfaces  
DRAM configuration is 64 bits wide with parity enabled.  
239K bytes of non-volatile configuration memory.  
126000K bytes of ATA CompactFlash (Read/Write)

License Info:

License UDI:

```
-----  
Device#   PID                SN  
-----  
*0        CISCO2811          FTX1411ALT7
```

Configuration register is 0x2102



## Running Configuration:

```
version 15.1
service config
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname MTPnetB
!
boot-start-marker
boot system flash c2800nm-spservicesk9-mz.151-3.T.bin
boot-end-marker
!
!
logging buffered 51200 warnings
no logging console
!
aaa new-model
!
!
!
!
!
!
aaa session-id common
clock timezone CST -6 0
clock summer-time CDT recurring
!
dot11 syslog
ip source-route
!
!
ip cef
!
!
!
ip domain name lab.tekvizion.com1
ip name-server 10.64.1.32
no ipv6 cef
multilink bundle-name authenticated
!
!
!
!
!
!
voice rtp send-recv3
!
voice service voip
  ip address trusted list4
    ipv4 207.173.7.100
    ipv4 10.70.18.2
  allow-connections sip to sip5
```

<sup>1</sup> Optional. The domain name must match the CUCM enterprise parameter “Cluster Fully Qualified Domain Name” and must resolve to a DNS A-record. See “DNS Configuration” and “CUCM Configuration” below

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

<sup>3</sup> Establish a 2-way voice path when a Real Time Protocol channel is opened

<sup>4</sup> Required for toll fraud prevention. IP addresses listed here are those of trusted VoIP peers.



```
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711ulaw6
sip
  early-offer forced7
  midcall-signaling passthru8
!
voice class codec 19
  codec preference 1 g729br8
  codec preference 2 g729r8
  codec preference 3 g711ulaw
!
!
!
!
!
voice-card 0
!
crypto pki token default removal timeout 0
!
!
!
!
license udi pid CISCO2811 sn FTX1411ALT7
username cisco privilege 15 password 0 cisco
!
!
!
!
!
!
interface FastEthernet0/0
  ip address 10.70.18.15 255.255.255.0
  duplex full
  speed 100
!
interface FastEthernet0/1
  description This IP address is associated with IntegraTelecom.
  ip address 174.46.0.150 255.255.255.128
  duplex auto
  speed auto
!
ip forward-protocol nd
no ip http server
no ip http secure-server
!
!
ip route 0.0.0.0 0.0.0.0 174.46.0.129
!
logging esm config
!
!
!
!
control-plane
!
!
```

<sup>5</sup> Allow SIP to SIP call processing.

<sup>6</sup> Enable T.38 fax relay. Remaining parameters are optional; this example disables redundancy for both low-speed T.30 messaging and high-speed page data, and enables G.711 fallback if T.38 negotiation fails.

<sup>7</sup> Configures CUBE to send a SIP INVITE with SDP on an outbound call leg (Delayed Offer to Early Offer)

<sup>8</sup> Enable support for SIP re-INVITE supplementary services

<sup>9</sup> Codec preference list, referenced from the dial-peers with the “voice-class codec *n*” configuration command



```
!  
!  
mgcp profile default  
!  
sccp local FastEthernet0/0  
!  
dial-peer voice 100 voip  
description 1+10 digits to IntegraTelecom  
destination-pattern ^1[2-9]..[2-9].....$10  
session protocol sipv2  
session target sip-server11  
voice-class codec 112  
voice-class sip early-offer forced13  
dtmf-relay rtp-nte14  
!  
dial-peer voice 102 voip  
description International Calls to IntegraTelecom  
destination-pattern ^011T15  
session protocol sipv2  
session target sip-server  
voice-class codec 1  
voice-class sip early-offer forced  
dtmf-relay rtp-nte  
!  
dial-peer voice 300 voip  
description 503548XXXX calls to CUCM  
huntstop  
destination-pattern ^503548....$16  
session protocol sipv2  
session target dns:clus3pubsub.lab.tekvizion.com17  
voice-class codec 1  
voice-class sip early-offer forced  
dtmf-relay rtp-nte  
!  
dial-peer voice 20 voip  
description all calls from CUCM  
huntstop  
session protocol sipv2  
session target dns:clus3pubsub.lab.tekvizion.com  
incoming called-number .T18  
voice-class codec 1  
voice-class sip early-offer forced  
dtmf-relay rtp-nte  
!  
dial-peer voice 10 voip  
description calls from Service Provider  
session protocol sipv2  
session target sip-server  
incoming called-number .T  
voice-class codec 1
```

<sup>10</sup> Outbound dial peer for 1+10-digit called numbers

<sup>11</sup> Sip-server resolves to the Integra Telecom SBC IP address, and is defined in the sip-ua section below.

<sup>12</sup> Reference to codec preference list “voice class codec 1”, above

<sup>13</sup> Enable delayed-offer to early-offer

<sup>14</sup> Forward DTMF tones by using RTP with the Named Telephone Event (NTE) payload type (RFC 2833).

<sup>15</sup> Outbound dial peer for international calls (US market)

<sup>16</sup> Outbound dial peer (to CUCM). The pattern here should match service provider-assigned DID numbers.

<sup>17</sup> Reference to DNS A record. In a single-subscriber deployment, this could either be a DNS reference, or it could be an IP address: “ipv4:xxx.xxx.xxx.xxx”. In a multiple-node cluster, this would be a reference to a DNS SRV record.

<sup>18</sup> This is the default inbound dial peer



```
voice-class sip early-offer forced
dtmf-relay rtp-nte
!
!
gateway
 timer receive-rtp 1200
!
sip-ua
 authentication username 5035489480 password 7 131504411E08543A733C3C6060364219
 retry invite 7
 retry bye 7
 retry cancel 7
 retry register 10
 sip-server ipv4:207.173.7.100:506020
!
!
!
line con 0
 exec-timeout 0 0
 logging synchronous
line aux 0
line vty 0 4
 exec-timeout 0 0
 password *****
 transport input all
!
scheduler allocate 20000 1000
end
```

---

<sup>19</sup> Authentication credentials provided by service provider

<sup>20</sup> Service provider signaling address.



## Configuring the Cisco Unified Communications Manager

Figure 6. Enterprise Parameters

Cisco Unified CM Administration  
For Cisco Unified Communications Solutions

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### Enterprise Parameters Configuration

Save | Set to Default | Reset | Apply Config

<b>User Search Parameters</b>		
Enable All User Search *	True	True
User Search Limit *	64	64
<b>CCM Web Services Parameters</b>		
Allowed Performance Queries Per Minute *	50	50
Allowed Device Queries Per Minute *	15	15
Performance Queue Limit *	100	100
Allowed CDRonDemand_get_file Queries Per Minute *	10	10
Allowed CDRonDemand_get_file_list Queries Per Minute *	20	20
<b>Trace Parameters</b>		
File Close Thread Flag *	True	True
FileCloseThreadQueueWatermark *	100	100
<b>User Management Parameters</b>		
Effective Access Privileges For Overlapping User Groups and roles *	Maximum	Maximum
<b>Service Manager TCP ports parameters</b>		
Service Manager TCP Server communication port number *	8888	8888
Service Manager TCP Client communication port number *	8889	8889
<b>CRS Application Parameters</b>		
Auto Attendant Installed *	false	
IPCC Express Installed *	false	
<b>Clusterwide Domain Configuration</b>		
Organization Top Level Domain		
Cluster Fully Qualified Domain Name	clus3pubsub.lab.tekvizion.com	Required when DNS is used to route to CUCM. Must match the session-target in the CUBE dial peer.
<b>Denial-of-Service Protection</b>		
Denial-of-Service Protection *	True	True
<b>TLS Handshake Timer</b>		
TLS Handshake Timer *	60	60



Figure 7. SIP Trunk Security Profile

**Cisco Unified CM Administration**  
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### SIP Trunk Security Profile Configuration

Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

**Status:** Ready

#### SIP Trunk Security Profile Information

Name*	IntegraTelecom-Non Secure SIP Trunk Profile
Description	Non Secure SIP Trunk Profile authenticated by null Strii
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	

Save Delete Copy Reset Apply Config Add New

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

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



Figure 8. SIP Trunk to Integra Telecom via CUBE

The screenshot shows the Cisco Unified CM Administration interface for configuring a SIP Trunk. The page title is "Trunk Configuration" and the status is "Ready". The configuration is for a SIP Trunk named "IntegraTelecom".

Device Information	Value
Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type:	None(Default)
Device Name*:	IntegraTelecom
Description:	IntegraTelecom via cube
Device Pool*:	external
Common Device Configuration:	< None >
Call Classification*:	Use System Default
Media Resource Group List:	IntegraMRGL
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
<input type="checkbox"/> Media Termination Point Required	
<input checked="" type="checkbox"/> Retry Video Call as Audio	
<input type="checkbox"/> Path Replacement Support	
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name	
<input type="checkbox"/> Transmit UTF-8 Names in QSIG APDU	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.	
Consider Traffic on This Trunk Secure*:	When using both sRTP and TLS
Route Class Signaling Enabled*:	Default
Use Trusted Relay Point*:	Default
<input checked="" type="checkbox"/> PSTN Access	
<input type="checkbox"/> Run On All Active Unified CM Nodes	

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



Figure 9. SIP Trunk to Integra Telecom via CUBE (cont.)

**Cisco Unified CM Administration**  
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### Trunk Configuration

Related Links: Back To Find/List Go

Save Delete Reset Add New

Route Class Signaling Enabled\* Default  
Use Trusted Relay Point\* Default

PSTN Access  
 Run On All Active Unified CM Nodes

**Intercompany Media Engine (IME)**  
E.164 Transformation Profile: < None >

**Multilevel Precedence and Preemption (MLPP) Information**  
MLPP Domain: < None >

**Call Routing Information**

Remote-Party-Id  
 Asserted-Identity  
Asserted-Type\* Default  
SIP Privacy\* Default

**Inbound Calls**

Significant Digits\* All  
Connected Line ID Presentation: Default  
Connected Name Presentation\*: Default  
Calling Search Space: < None >  
AAR Calling Search Space: < None >  
Prefix DN:  
 Redirecting Diversion Header Delivery - Inbound

*In this sample configuration, all ten digits are delivered by CUBE to CUCM.*

**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.

Clear Prefix Settings Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>

**Connected Party Settings**  
Connected Party Transformation CSS: < None >



Figure 10. SIP Trunk to Integra Telecom via CUBE (cont.)

Cisco Unified CM Administration  
For Cisco Unified Communications Solutions

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### Trunk Configuration

Related Links: Back To Find/List Go

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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\* **First Redirect Number (External)** *Required for call-forward to PSTN scenario in order for Caller ID to be presented correctly.*  
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	<b>10.70.18.15</b> <i>CUBE private IP address</i>		5060

MTP Preferred Originating Codec\* 711ulaw  
Presence Group\* **Standard Presence group**  
SIP Trunk Security Profile\* **IntegraTelecom-Non Secure SIP Trunk Profile** *Refer to SIP Trunk Security Profile, above*  
Rerouting Calling Search Space < None >  
Out-Of-Dialog Refer Calling Search Space < None >  
SUBSCRIBE Calling Search Space < None >  
SIP Profile\* **Cube Standard SIP Profile** *Refer to SIP Profile, below*  
DTMF Signaling Method\* No Preference



Figure 11. SIP Profile

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**SIP Profile Configuration** Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

**Status**

- Status: Ready
- All SIP devices using this profile must be restarted before any changes will take affect.

**SIP Profile Information**

Name\* Cube Standard SIP Profile

Description Default SIP Profile

Default MTP Telephony Event Payload Type\* 101

Resource Priority Namespace List < None >

Early Offer for G.Clear Calls\* Disabled

Redirect by Application

Disable Early Media on 180

Outgoing T.38 INVITE include audio mline

Enable ANAT

Require SDP Inactive Exchange for Mid-Call Media Change

**Parameters used in Phone**

Timer Invite Expires (seconds)*	180
Timer Register Delta (seconds)*	5
Timer Register Expires (seconds)*	3600
Timer T1 (msec)*	500
Timer T2 (msec)*	4000
Retry INVITE*	6
Retry Non-INVITE*	10
Start Media Port*	16384
Stop Media Port*	32766
Call Pickup URI*	x-cisco-serviceuri-pickup
Call Pickup Group Other URI*	x-cisco-serviceuri-opickup
Call Pickup Group URI*	x-cisco-serviceuri-gpickup
Meet Me Service URI*	x-cisco-serviceuri-meetme
User Info*	None



Figure 12. SIP Profile (cont.)

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### SIP Profile Configuration

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Call Pickup Group Other URI*	x-cisco-serviceuri-opickup
Call Pickup Group URI*	x-cisco-serviceuri-gpickup
Meet Me Service URI*	x-cisco-serviceuri-meetme
User Info*	None
DTMF DB Level*	Nominal
Call Hold Ring Back*	Off
Anonymous Call Block*	Off
Caller ID Blocking*	Off
Do Not Disturb Control*	User
Telnet Level for 7940 and 7960*	Disabled
Timer Keep Alive Expires (seconds)*	120
Timer Subscribe Expires (seconds)*	120
Timer Subscribe Delta (seconds)*	5
Maximum Redirections*	70
Off Hook To First Digit Timer (milliseconds)*	15000
Call Forward URI*	x-cisco-serviceuri-cfdwall
Speed Dial (Abbreviated Dial) URI*	x-cisco-serviceuri-abbrdial

Conference Join Enabled  
 RFC 2543 Hold  
 Semi Attended Transfer  
 Enable VAD  
 Stutter Message Waiting

#### Trunk Specific Configuration

Reroute Incoming Request to new Trunk based on\* Never

RSVP Over SIP\* Local RSVP

Fall back to local RSVP

SIP Rel1XX Options\* Disabled

Deliver Conference Bridge Identifier  
 Early Offer support for voice and video calls (insert MTP if needed)  
 Send send-receive SDP in mid-call INVITE



Figure 13. SIP Profile (cont.)

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### SIP Profile Configuration

Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

Timer Subscribe Expires (seconds)\* 120  
Timer Subscribe Delta (seconds)\* 5  
Maximum Redirections\* 70  
Off Hook To First Digit Timer (milliseconds)\* 15000  
Call Forward URI\* x-cisco-serviceuri-cfwdall  
Speed Dial (Abbreviated Dial) URI\* x-cisco-serviceuri-abbrdial

Conference Join Enabled  
 RFC 2543 Hold  
 Semi Attended Transfer  
 Enable VAD  
 Stutter Message Waiting

**Trunk Specific Configuration**

Reroute Incoming Request to new Trunk based on\* Never  
RSVP Over SIP\* Local RSVP  
 Fall back to local RSVP  
SIP Rel1XX Options\* Disabled  
 Deliver Conference Bridge Identifier  
 Early Offer support for voice and video calls (insert MTP if needed)  
 Send send-receive SDP in mid-call INVITE

**SIP OPTIONS Ping**

Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"  
Ping Interval for In-service and Partially In-service Trunks (seconds)\* 60  
Ping Interval for Out-of-service Trunks (seconds)\* 120  
Ping Retry Timer (milliseconds)\* 500  
Ping Retry Count\* 6

Save Delete Copy Reset Apply Config Add New

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



Figure 14. Translation Pattern for incoming calls from Integra Telecom

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**Translation Pattern Configuration** Related Links: Back To Find/List Go

Save Delete Copy Add New

**Status**  
Status: Ready

**Pattern Definition**

Translation Pattern	503548.XXXX	Match a 10-digit with NPA-NXX of 503548. The digits before the "." will be stripped by the Strip Digits configuration below.
Partition	< None >	
Description		
Numbering Plan	< None >	
Route Filter	< None >	
MLPP Precedence*	Default	
Resource Priority Namespace Network Domain	< None >	
Route Class*	Default	
Calling Search Space	< None >	
External Call Control Profile	< None >	
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error	

Provide Outside Dial Tone  
 Urgent Priority  
 Route Next Hop By Calling Party Number

**Calling Party Transformations**

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask:   
Prefix Digits (Outgoing Calls):

Calling Line ID Presentation\*: Default  
Calling Name Presentation\*: Default  
Calling Party Number Type\*: Cisco CallManager  
Calling Party Numbering Plan\*: Cisco CallManager

**Connected Party Transformations**

Connected Line ID Presentation\*: Default  
Connected Name Presentation\*: Default

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



Figure 15. Translation Pattern for incoming calls from Integra Telecom (cont.)

**Cisco Unified CM Administration**  
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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

### Translation Pattern Configuration

Related Links: Back To Find/List Go

Save Delete Copy Add New

Route Class\* Default  
Calling Search Space < None >  
External Call Control Profile < None >  
Route Option  
 Route this pattern  
 Block this pattern No Error

Provide Outside Dial Tone  
 Urgent Priority  
 Route Next Hop By Calling Party Number

#### Calling Party Transformations

Use Calling Party's External Phone Number Mask  
Calling Party Transform Mask  
Prefix Digits (Outgoing Calls)  
Calling Line ID Presentation\* Default  
Calling Name Presentation\* Default  
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 all called-number digits prior to the "." in the translation pattern above (i.e. strip the NPA-NXX). Calls are routed based on the last four digits only.  
Called Party Transform Mask  
Prefix Digits (Outgoing Calls)  
Called Party Number Type\* Cisco CallManager  
Called Party Numbering Plan\* Cisco CallManager

Save Delete Copy Add New

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



Figure 16. Route Pattern Configuration for SIP trunk to Integra Telecom via CUBE

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Route Pattern Configuration Related Links: Back To Find/List Go

Save Delete Copy Add New

Status: Ready

**Pattern Definition**

Route Pattern\* 90.@ This pattern routes calls with a "90" followed by an NANP number

Route Partition < None >

Description PSTN via 3845

Numbering Plan\* NANP

Route Filter < None >

MLPP Precedence\* Default

Resource Priority Namespace Network Domain < None >

Route Class\* Default

Gateway/Route List\* IntegraTelecom (Edit) Calls are routed to Integra Telecom via CUBE

Route Option  
 Route this pattern  
 Block this pattern No Error

Call Classification\* OFFNET

Allow Device Override  Provide Outside Dial Tone  Allow Overlap Sending  Urgent Priority Dial tone provided to caller after the "90" digits are dialed

Require Forced Authorization Code

Authorization Level\* 0

Require Client Matter Code

**Calling Party Transformations**

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation\* Default

Calling Name Presentation\* Default

Calling Party Number Type\* Cisco CallManager

Calling Party Numbering Plan\* Cisco CallManager

**Connected Party Transformations**

Connected Line ID Presentation\* Default



Figure 17. Route Pattern Configuration for SIP trunk to Integra Telecom via CUBE (cont)

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

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

### Route Pattern Configuration

Related Links: Back To Find/List Go

Save Delete Copy Add New

Require Forced Authorization Code  
Authorization Level\* 0  
 Require Client Matter Code

**Calling Party Transformations**

Use Calling Party's External Phone Number Mask  
Calling Party Transform Mask  
Prefix Digits (Outgoing Calls)  
Calling Line ID Presentation\* Default  
Calling Name Presentation\* Default  
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 "90" digits 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 -- Not Selected -- Service Parameter Name < Not Exist > Service Parameter Value

Save Delete Copy Add New

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

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



Figure 18. SIP Trunk to Cisco 3845 (for analog fax)

The screenshot displays the Cisco Unified CM Administration web interface. At the top, the navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and the subtitle "For Cisco Unified Communications Solutions". The user is logged in as "administrator". The main menu includes "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", "Bulk Administration", and "Help". The current page is "Trunk Configuration", with a "Related Links" section containing "Back To Find/List".

Below the navigation bar, there are buttons for "Save", "Delete", "Reset", and "Add New". A "Status" section shows "Status: Ready".

The main configuration area is titled "Device Information" and contains the following fields and options:

- Product: SIP Trunk
- Device Protocol: SIP
- Trunk Service Type: None(Default)
- Device Name\*: FaxSipTrunk
- Description: Fax trunk to 3845
- Device Pool\*: FAX
- Common Device Configuration: < None >
- Call Classification\*: Use System Default
- Media Resource Group List: < None >
- Location\*: Hub\_None
- AAR Group: < None >
- Tunneled Protocol\*: None
- QSIG Variant\*: No Changes
- ASN.1 ROSE OID Encoding\*: No Changes
- Packet Capture Mode\*: None
- Packet Capture Duration: 0

Additional configuration options include:

- 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\*: When using both sRTP and TLS
- Route Class Signaling Enabled\*: Default
- Use Trusted Relay Point\*: Default
- PSTN Access
- Run On All Active Unified CM Nodes

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



Figure 19. SIP Trunk to Cisco 3845 (for analog fax) (cont)

**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

### Trunk Configuration

Related Links: Back To Find/List Go

Save Delete Reset Add New

Route Class Signaling Enabled\* Default  
Use Trusted Relay Point\* Default

PSTN Access  
 Run On All Active Unified CM Nodes

**Intercompany Media Engine (IME)**  
E.164 Transformation Profile: < None >

**Multilevel Precedence and Preemption (MLPP) Information**  
MLPP Domain: < None >

**Call Routing Information**

Remote-Party-Id  
 Asserted-Identity  
Asserted-Type\* Default  
SIP Privacy\* Default

**Inbound Calls**

Significant Digits\* All  
Connected Line ID Presentation Default  
Connected Name Presentation\* Default  
Calling Search Space < None >  
AAR Calling Search Space < None >  
Prefix DN  
 Redirecting Diversion Header Delivery - Inbound

**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.

Clear Prefix Settings Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>

**Connected Party Settings**  
Connected Party Transformation CSS: < None >

*In this sample configuration, the router hosting the fax device transmits all called digits, including the "90" prefix for PSTN calls.*



Figure 20. SIP Trunk to Cisco 3845 (for analog fax) (cont)

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

### Trunk Configuration

Related Links: Back To Find/List Go

Save Delete Reset Add New

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  
Caller ID DN  
Caller Name  
 Redirecting Diversion Header Delivery - Outbound

#### SIP Information

Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1	10.70.18.14 Router address		5060

MTP Preferred Originating Codec\* 711ulaw  
Presence Group\* Standard Presence group  
SIP Trunk Security Profile\* FAX 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\* FAX SIP Profile  
DTMF Signaling Method\* No Preference



Figure 21. SIP Trunk to Cisco 3845 (for analog fax) (cont)

**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

### Trunk Configuration

Related Links: Back To Find/List Go

Save Delete Reset Add New

Carrier name: \_\_\_\_\_

Redirecting Diversion Header Delivery - Outbound

---

#### SIP Information

Destination Address is an SRV

Destination Address	Destination Address IPv6	Destination Port
1* 10.70.18.14		5060

MTP Preferred Originating Codec\* 711ulaw

Presence Group\* Standard Presence group

SIP Trunk Security Profile\* FAX 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\* FAX SIP Profile

DTMF Signaling Method\* No Preference

---

#### Normalization Script

Normalization Script < None >

Enable Trace

Parameter Name	Parameter Value
1	

---

#### Geolocation Configuration

Geolocation < None >

Geolocation Filter < None >

Send Geolocation Information

Save Delete Reset Add New

**i** \* indicates required item.  
**i** \*\* Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.

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



Figure 22. Route Pattern Configuration (fax)

**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

**Route Pattern Configuration** Related Links: Back To Find/List Go

Save Delete Copy Add New

Status: Ready

**Pattern Definition**

Route Pattern\* **9485** *Route to fax device based on 4-digit directory number. (CUCM strips NPA-NXX from incoming trunk calls from Integra.*

Route Partition: < None >  
Description:  
Numbering Plan: -- Not Selected --  
Route Filter: < None >  
MLPP Precedence\*: Default  
Resource Priority Namespace Network Domain: < None >  
Route Class\*: Default  
Gateway/Route List\*: FaxSipTrunk (Edit)  
Route Option:  
 Route this pattern  
 Block this pattern No Error

Call Classification\*: OffNet  
 Allow Device Override  Provide Outside Dial Tone  Allow Overlap Sending  Urgent Priority  
 Require Forced Authorization Code  
Authorization Level\*: 0  
 Require Client Matter Code

**Calling Party Transformations**

Use Calling Party's External Phone Number Mask  
Calling Party Transform Mask:  
Prefix Digits (Outgoing Calls):  
Calling Line ID Presentation\*: Default  
Calling Name Presentation\*: Default  
Calling Party Number Type\*: Cisco CallManager  
Calling Party Numbering Plan\*: Cisco CallManager

**Connected Party Transformations**

Connected Line ID Presentation\*: Default

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



Figure 23. Route Pattern Configuration (fax) (cont)

**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

### Route Pattern Configuration

Related Links: Back To Find/List Go

Save Delete Copy Add New

Require Forced Authorization Code  
Authorization Level\* 0  
 Require Client Matter Code

**Calling Party Transformations**

Use Calling Party's External Phone Number Mask  
Calling Party Transform Mask  
Prefix Digits (Outgoing Calls)  
Calling Line ID Presentation\* Default  
Calling Name Presentation\* Default  
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 < None >  
Called Party Transform Mask  
Prefix Digits (Outgoing Calls)  
Called Party Number Type\* Cisco CallManager  
Called Party Numbering Plan\* Cisco CallManager

**ISDN Network-Specific Facilities Information Element**

Network Service Protocol -- Not Selected --  
Carrier Identification Code  
Network Service -- Not Selected -- Service Parameter Name Service Parameter Value  
< Not Exist >

Save Delete Copy Add New

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



## (Optional) Configuring a Cisco 3845 for analog fax

### Cisco 3845 configuration for analog fax endpoint

Note: only the configuration elements relevant to the fax endpoint and the SIP trunk to CUCM are shown

```
dial-peer voice 400 pots
  description POTS dial peer for analog fax endpoint
  destination-pattern 948521
  fax rate voice
  port 0/0/0
!
dial-peer voice 201 voip
  description Outbound dial peer for PSTN fax calls via CUCM
  destination-pattern 90T22
  session protocol sipv2
  session target dns:clus3pubsub.lab.tekvizion.com
  dtmf-relay rtp-nte
  codec g711ulaw
  fax-relay sg3-to-g323
  fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711ulaw24
  no vad
!
dial-peer voice 300 voip
  description incoming dial peer for fax calls
  incoming called-number 9485
  codec g711ulaw
  fax-relay sg3-to-g3
  fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback pass-through g711ulaw
```

---

<sup>21</sup> CUCM sends 4-digit extension for inbound fax calls to this endpoint.

<sup>22</sup> For fax calls to PSTN caller dials 9+PSTN number. CUCM strips the “9” before forwarding to CUBE.

<sup>23</sup> Enable sg3 spoofing.

<sup>24</sup> Enable T.38 for fax. Fall back to G.711 if T.38 is unsuccessful.



**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

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



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