

## Cox SIP Trunking:

Cisco Unified Communications Manager 11.0.1 with Cisco Unified Border Element (CUBE 11.1.0) on ISR 4K [IOS-XE 3.16] using SIP

November 11, 2015



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## Introduction

Service Providers today, such as Cox, 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 centralized IP to TDM POP gateways to provide on-net and off-net services.

Cox is a service provider that allows connection to the PSTN and may offer the end customer a viable alternative to traditional PSTN connectivity. A demarcation device between these services and customer owned services is recommended. As an intermediary device between Cisco Unified Communications Manager and Cox Session Border Controller (EdgeMarc), Cisco Unified Border Element (CUBE) ISR 15.5(3)S can be used. The Cisco Unified Border Element ISR 15.5(3)S provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 11.0.1 connected to Cox IP network.

This document assumes the reader is knowledgeable with the terminology and configuration of CUCM (Cisco Unified Communications Manager). Only configuration settings specifically required for Cox interoperability are presented. Feature configuration, and most importantly the dial plan, are customer specific and need an individual approach.

- This application note describes how to configure a Cisco Unified Communications Manager (Cisco UCM) 11.0.1 and Cisco Unified Border Element (CUBE 11.1.0) on ISR 4K [IOS-XE3.16] for connectivity to Cox SIP trunking service. The deployment model covered in this application note is CPE (Cisco UCM 11.0.1) to PSTN (Cox).
- Testing was performed in accordance to Cox generic SIP trunking test methodology and among features verified were – basic calls, DTMF transport, Music on Hold, Semi-attendant and attendant transfers, call forward, conferences, and interoperability with Cisco Unity Connection
- The CUCM configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between Cox SIP network and Cisco Unified Communications. The configuration described in this document details the important configuration settings to have enabled for interoperability to be successful and care must be taken by the network administrator deploying Cisco UCM to interoperate to Cox SIP trunking 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/docs/voice\\_ip\\_comm/cucm/srnd/collab09/dialplan.html#wpmkr1044275](http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/srnd/collab09/dialplan.html#wpmkr1044275)

This application note does not cover the configuration of the EdgeMarc E-SBC. The Cox E-SBC is the Edgewater Networks EdgeMarc appliance. The EdgeMarc is the service demarcation point between customer's LAN network and Cox's WAN network and provides firewall/NAT traversal, B2BUA and SIP Application-level gateway. The EdgeMarc has diverse routes to a primary and secondary Acme SBC. For more info regarding the EdgeMarc E-SBC visit [www.edgewaternetworks.com](http://www.edgewaternetworks.com).

## Network Topology

### Basic Call Setup

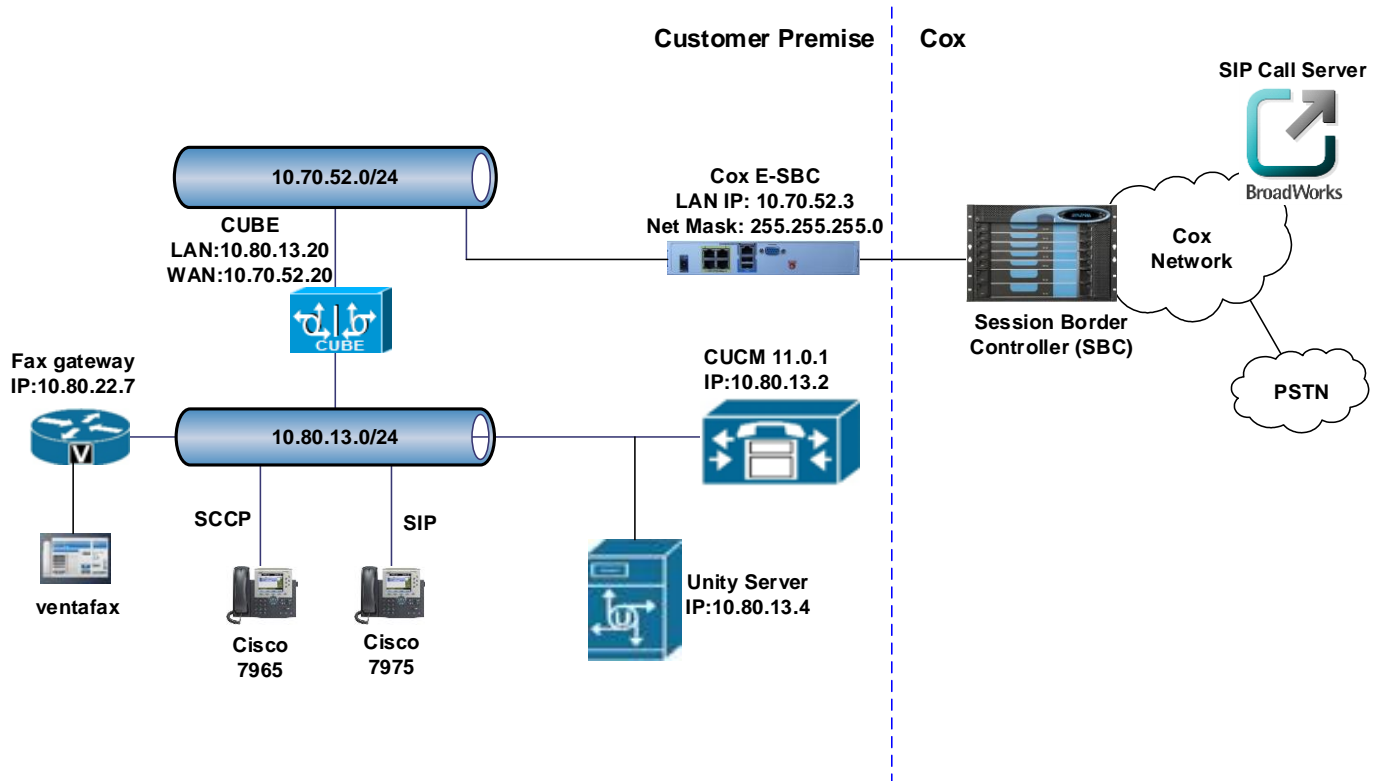


Figure 1: Network Topology



## **System Components**

### ***Hardware Components***

- Cisco UCM and Unity Connection run on VMware
- ISR G2 2901/K9 router as Fax Gateway
- ISR 4321/K9 router as CUBE
- IP phones 7975(SIP) and 7965(SCCP)( please consult “Features not supported” for restrictions)

### ***Software Requirements***

- Cisco Unified Communications Manager 11.0.1
- IOS-XE 3.16 for Cisco Unified Border Element on ISR4321
- IOS 15.4(3)M1 for Fax Gateways on ISR2901
- Cisco Unity Connection 11.0.1

### ***Features Supported***

- Incoming and outgoing off-net calls using G711Ulaw (Cox only offer G711Ulaw) with 20ms packetization
- Call hold
- Call transfer (Semi-Attendant and Attendant)
- Call conference
- Call forward (all, busy, no answer)
- Calling line (number) identification presentation (CLIP)
- Calling line (number) identification restriction (CLIR)
- DTMF (RFC2833)
- Media flow-through on CUBE
- Fax G.711 pass-through

### ***Features Not Supported***

- Outbound SIP REFER with Replaces. Cisco UCM does not currently support generation of an outbound SIP REFER with Replaces
- Cisco IP phones used in this test do not support Blind Transfer, only Semi-attendant and Attendant transfers were tested

### ***Caveats***

- CUBE High Availability (HA) was not tested in this setup due to lack of hardware
- The caller ID of the DUT is being seen instead of the originator of the call that is transferred or forwarded
- Defect ID: PAI/PPI support for INVITE/UPDATE Request/Response in CUBE - CSCua03687



## Configuration

### *Configuring the Cisco Unified Border Element*

#### **Network Interface**

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

```
interface GigabitEthernet0/0/0

description COX CUBE1 LAN

ip address 10.80.13.20 255.255.255.0

negotiation auto

!

interface GigabitEthernet0/0/1

description COX CUBE1 WAN

ip address 10.70.52.20 255.255.255.0

negotiation auto
```

#### **Global CUBE Settings**

In order to enable CUBE IP2IP gateway functionality, following command has to be entered:

```
voice service voip

no ip address trusted authenticate

address-hiding

mode border-element

allow-connections sip to sip

no supplementary-service sip handle-replaces

fax protocol pass-through g711ulaw

sip

session refresh

asserted-id pai
```



early-offer forced  
midcall-signaling passthru  
privacy-policy passthru  
privacy-policy send-always  
g729 annexb-all

#### Explanation

Command	Description
allow-connections sip to sip	Allow IP2IP connections between two SIP call legs
fax protocol	Specifies the fax protocol
asserted-id	Specifies the type of privacy header in the outgoing SIP requests and response messages
early-offer forced	Enables SIP Delayed-Offer to Early-Offer globally
midcall-signaling passthru	Passes SIP messages from one IP leg to another IP leg

#### Media Passing Through CUBE (media flow-through vs. media flow-around)

Default CUBE configuration enables CUBE to work in flow-through mode (this test use flow-through mode). In order to enable flow-around mode, please perform the following actions:

voice service voip  
media flow-around

#### Codecs

Cox offer only G.711ulaw codec for voice call, it allows codecs other than G.711ulaw but will only accept G.711ulaw.

For customers using **G.711 ulaw** codec:

voice class codec 1  
codec preference 1 g711ulaw

#### Dial Peer

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

dial-peer voice 10 voip  
description "Outgoing To edgemark"-edgemark facing side





```
huntstop
destination-pattern [0-9]T
no modem passthrough
session protocol sipv2
session target ipv4:10.70.52.3:5060
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 11 voip
description "Outgoing To edgemark .IP PBX facing side"
no modem passthrough
session protocol sipv2
incoming called-number [0-9]
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip bind control source-interface GigabitEthernet0/0/0
```



```
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 20 voip
description " Incoming edgemark to IP-PBX - IP-PBX facing side "
huntstop
destination-pattern 402502....
no modem passthrough
session protocol sipv2
session target ipv4:10.80.13.2:5060
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
fax-relay ecm disable
fax-relay sg3-to-g3
fax rate disable
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
```



```
!  
dial-peer voice 21 voip  
description " Incoming edgemark to IP-PBX . AT&T facing side "  
huntstop  
no modem passthrough  
session protocol sipv2  
incoming called-number 40250.....  
voice-class codec 1  
voice-class sip asymmetric payload full  
voice-class sip asserted-id pai  
voice-class sip privacy-policy passthru  
voice-class sip bind control source-interface GigabitEthernet0/0/1  
voice-class sip bind media source-interface GigabitEthernet0/0/1  
dtmf-relay rtp-nte  
fax-relay ecm disable  
fax rate disable  
fax protocol pass-through g711ulaw  
no vad
```

## Call Flow

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. The 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 fax gateway

CPE callers make outbound PSTN calls by dialing a "9" prefix followed by the destination number. For outbound fax calls from the analog fax endpoint, fax gateway sends to Cisco UCM the DID with leading access code "9". A "9.@" route pattern strips the prefix and routes the call with the remaining digits via a SIP trunk terminating on the CUBE for voice call or outbound fax.



## Configuration Example

The following configuration snippet contains a sample configuration of Cisco Unified Border Element with all parameters mentioned previously.

```
COX_CUBE1#sh running-config
```

```
Building configuration...
```

```
Current configuration : 7526 bytes
```

```
!
```

```
version 15.5
```

```
service timestamps debug datetime msec
```

```
service timestamps log datetime msec
```

```
no platform punt-keepalive disable-kernel-core
```

```
!
```

```
hostname COX_CUBE1
```

```
!
```

```
boot-start-marker
```

```
boot system flash:isr4300-universalk9.03.16.00.S.155-3.S-ext.SPA.bin
```

```
boot system bootflash:isr4300-universalk9.03.16.00.S.155-3.S-ext.SPA.bin
```

```
boot-end-marker
```

```
!
```

```
!
```

```
vrf definition Mgmt-intf
```

```
!
```

```
address-family ipv4
```

```
exit-address-family
```

```
!
```

```
address-family ipv6
```

```
exit-address-family
```



```
!  
enable secret 5 $1$4Gfa$O/1WQEcuut.YXLcn3acUP1  
enable password tekV1z10n  
!  
no aaa new-model  
!  
!  
subscriber templating  
!  
multilink bundle-name authenticated  
!  
!  
voice service voip  
no ip address trusted authenticate  
address-hiding  
mode border-element  
allow-connections sip to sip  
no supplementary-service sip handle-replaces  
fax protocol pass-through g711ulaw  
sip  
session refresh  
asserted-id pai  
early-offer forced  
midcall-signaling passthru  
privacy-policy passthru  
privacy-policy send-always  
g729 annexb-all  
!
```



```
voice class codec 1
  codec preference 1 g711ulaw
!
voice class codec 2
  codec preference 1 g711ulaw
  codec preference 2 g729r8
!
!
license udi pid ISR4321/K9 sn FDO19220MQ8
license boot level appxk9
license boot level uck9
!
spanning-tree extend system-id
!
redundancy
  mode none
!
vlan internal allocation policy ascending
!
!
interface GigabitEthernet0/0/0
  description COX CUBE1 LAN
  ip address 10.80.13.20 255.255.255.0
  negotiation auto
!
interface GigabitEthernet0/0/1
  description COX CUBE1 WAN
  ip address 10.70.52.20 255.255.255.0
```



```
negotiation auto
```

```
!
```

```
interface GigabitEthernet0
```

```
vrf forwarding Mgmt-intf
```

```
no ip address
```

```
negotiation auto
```

```
!
```

```
interface Vlan1
```

```
no ip address
```

```
shutdown
```

```
!
```

```
ip forward-protocol nd
```

```
no ip http server
```

```
no ip http secure-server
```

```
ip tftp source-interface GigabitEthernet0
```

```
ip route 0.0.0.0 0.0.0.0 10.70.52.1
```

```
ip route 172.16.0.0 255.255.0.0 10.80.13.1
```

```
!
```

```
!
```

```
control-plane
```

```
!
```

```
!
```

```
mgcp behavior rsip-range tgcp-only
```

```
mgcp behavior comedia-role none
```

```
mgcp behavior comedia-check-media-src disable
```

```
mgcp behavior comedia-sdp-force disable
```

```
!
```

```
mgcp profile default
```



```
!  
!  
dial-peer voice 10 voip  
description "Outgoing To edgemark"-edgemark facing side  
huntstop  
destination-pattern [0-9]T  
no modem passthrough  
session protocol sipv2  
session target ipv4:10.70.52.3:5060  
voice-class codec 1  
voice-class sip asymmetric payload full  
voice-class sip asserted-id pai  
voice-class sip privacy-policy passthru  
voice-class sip early-offer forced  
voice-class sip bind control source-interface GigabitEthernet0/0/1  
voice-class sip bind media source-interface GigabitEthernet0/0/1  
dtmf-relay rtp-nte  
fax-relay ecm disable  
fax rate disable  
fax protocol pass-through g711ulaw  
no vad  
!  
dial-peer voice 11 voip  
description "Outgoing To edgemark .IP PBX facing side"  
no modem passthrough  
session protocol sipv2  
incoming called-number [0-9]  
voice-class codec 1
```





```
voice-class sip asymmetric payload full
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 20 voip
description " Incoming edgemark to IP-PBX - IP-PBX facing side "
huntstop
destination-pattern 402502....
no modem passthrough
session protocol sipv2
session target ipv4:10.80.13.2:5060
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
fax-relay ecm disable
fax-relay sg3-to-g3
```



```
fax rate disable
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 21 voip
description " Incoming edgemark to IP-PBX . AT&T facing side "
huntstop
no modem passthrough
session protocol sipv2
incoming called-number 40250.....
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax protocol pass-through g711ulaw
no vad
!
!
sip-ua
credentials username 4023159990 password 7 15465B5E577B7E7D716A65 realm 10.70.52.3
authentication username 4023159990 password 7 01475656085A535678151E
registrar ipv4:10.70.52.3 expires 3600
```




```
!  
!  
line con 0  
  stopbits 1  
line aux 0  
  stopbits 1  
line vty 0 4  
  password tekV1z10n  
  login  
  transport input telnet ssh  
!  
!  
End
```

### ***Configuring the Cisco Unified Communications Manager***

#### **Cisco Call Manager Service Parameters**




1. Navigate to **System > Service Parameters**
2. Set **Duplex Streaming**: enabled = true


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

Navigation Cisco Unified CM Administration  
**administrator** | [Search Documentation](#) | [About](#) | [Log Out](#)

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

**Service Parameter Configuration**
Related Links: Parameters for All Servers ▾ Go

 Save
 Set to Default
 Advanced

**Select Server and Service**  
Server\* clus23pubsub--CUCM Voice/Video (Active) ▾  
Service\* Cisco CallManager (Active) ▾  
All parameters apply only to the current server except parameters that are in the cluster-wide group(s).

**Cisco CallManager (Active) Parameters on server clus23pubsub--CUCM Voice/Video (Active)**

Parameter Name	Parameter Value	Suggested Value
<b>Call Throttling</b>		
<a href="#">Code Yellow Entry Latency</a> *	<input type="text" value="20"/>	20
<a href="#">Code Yellow Exit Latency Calculation</a> *	<input type="text" value="40"/>	40
<a href="#">Code Yellow Duration</a> *	<input type="text" value="5"/>	5
<a href="#">Max Events Allowed</a> *	<input type="text" value="2000"/>	2000
<a href="#">System Throttle Sample Size</a> *	<input type="text" value="10"/>	10
<b>Memory Throttling</b>		
<a href="#">Enable Memory Throttling</a> *	<input type="text" value="True"/>	True

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

**System**

<a href="#">CDR Enabled Flag</a> *	<input type="text" value="False"/>	False
<a href="#">CDR Log Calls with Zero Duration Flag</a> *	<input type="text" value="False"/>	False
<a href="#">Digit Analysis Complexity</a> *	<input type="text" value="StandardAnalysis"/>	StandardAnalysis
<a href="#">Database Debounce Timer</a> *	<input type="text" value="0"/>	0
<a href="#">Maximum Phone Fallback Queue Depth</a> *	<input type="text" value="10"/>	10
<a href="#">Maximum Number of Registered Devices</a> *	<input type="text" value="5000"/>	5000
<a href="#">System Initialization Timer</a> *	<input type="text" value="60"/>	60

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

Figure 2: Service Parameters

SDL Trace		
<a href="#">SDL Trace Data Flags</a> *	<input type="text" value="0x00000111"/>	0x00000111
<a href="#">SDL Trace Flush Immediately</a> *	<input type="text" value="False"/>	False
<a href="#">SDL Trace Data Size</a> *	<input type="text" value="0"/>	0
<a href="#">SDL Trace Flag</a> *	<input type="text" value="True"/>	True
<a href="#">SDL TraceType Flags</a> *	<input type="text" value="0x8000EB15"/>	0x8000EB15

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

Clusterwide Parameters (Device - General)		
<a href="#">Call Diagnostics Enabled</a> *	<input type="text" value="Disabled"/>	Disabled
<a href="#">Show Line Group Member DN in finalCalledPartyNumber CDR Field</a> *	<input type="text" value="False"/>	False
<a href="#">Show Line Group Member Non Masked DN in finalCalledPartyNumber CDR Field</a> *	<input type="text" value="False"/>	False
<a href="#">CTI New Call Accept Timer</a> *	<input type="text" value="4"/>	4
<a href="#">CTI Generate Digits Interval</a> *	<input type="text" value="250"/>	250
<a href="#">CTI Dial Digits Interval</a> *	<input type="text" value="250"/>	250
<a href="#">CTI Await Further Digits</a> *	<input type="text" value="False"/>	False
<a href="#">CTI Use Wildcard Pattern as calledPartyDN</a> *	<input type="text" value="False"/>	False
<a href="#">CTI Report Ringback on SIP 183 with SDP</a> *	<input type="text" value="True"/>	True
<a href="#">Retain Media on Disconnect with PI for Active Call</a> *	<input type="text" value="False"/>	False
<a href="#">Station and Backup Server KeepAlive Interval</a> *	<input type="text" value="60"/>	60
<a href="#">Station KeepAlive Interval</a> *	<input type="text" value="30"/>	30
<a href="#">Status Enquiry Poll Flag</a> *	<input type="text" value="False"/>	False
<a href="#">Strip # Sign from Called Party Number</a> *	<input type="text" value="True"/>	True
<a href="#">Session Handoff Alerting Timer</a> *	<input type="text" value="10"/>	10
<a href="#">T301 Timer</a> *	<input type="text" value="180000"/>	180000
<a href="#">T302 Timer</a> *	<input type="text" value="15000"/>	15000
<a href="#">T303 Timer</a> *	<input type="text" value="4000"/>	4000
<a href="#">T304 Timer</a> *	<input type="text" value="30000"/>	30000
<a href="#">T305 Timer</a> *	<input type="text" value="30000"/>	30000
<a href="#">T306 Timer</a> *	<input type="text" value="30000"/>	30000

Figure 3: Service Parameters - Cont.

<a href="#">T308 Timer</a> *	4000	4000
<a href="#">T309 Timer</a> *	90000	90000
<a href="#">T310 Timer</a> *	60000	60000
<a href="#">T313 Timer</a> *	4000	4000
<a href="#">T316 Timer</a> *	120000	120000
<a href="#">T317 Timer</a> *	100000	100000
<a href="#">T321 Timer</a> *	30000	30000
<a href="#">T322 Timer</a> *	4000	4000
<a href="#">Tone on Hold Timer</a> *	10	10
<a href="#">Unknown Caller ID Flag</a> *	True	True
<a href="#">Call Classification</a> *	OffNet	OffNet
<a href="#">Always Display Original Dialed Number</a> *	False	False
<a href="#">Name Display for Original Dialed Number When Translated</a> *	Show the Display Name for Original Dialed Number even if Translated	Show the Display Name for Original Dialed Number even if Translated
<a href="#">Always Use PIs With Original Dialed Number</a> *	False	False
<a href="#">Fail Call If Trusted Relay Point Allocation Fails</a> *	True	True
<a href="#">Display Calling/Called ID When PI is Not Available</a> *	False	False
<a href="#">Enable Transit Counter Processing on QSIG Trunks</a> *	False	False
<a href="#">Egress Facility IE Count</a> *	6	6

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

---

**Clusterwide Parameters (Device - Phone)**

<a href="#">Always Use Prime Line</a> *	False	False
<a href="#">Always Use Prime Line for Voice Message</a> *	False	False
<a href="#">Builtin Bridge Enable</a> *	Off	Off
<a href="#">Device Mobility Mode</a> *	Off	Off
<a href="#">Display Device Mobility Location During Phone Registration</a> *	True	True
<a href="#">Auto Answer Timer</a> *	1	1
<a href="#">Extension Display on Cisco IP Phone Model 7910</a> *	False	False
<a href="#">Alternate Idle Phone Auto-Answer Behavior Enabled</a> *	False	False
<a href="#">Hold Type</a> *	False	False
<a href="#">Line State Update Enabled</a> *	True	True
<a href="#">Off-hook to First Digit Timer</a> *	15000	15000

Figure 4: Service Parameters - Cont.

<a href="#">Override Auto Answer If Speaker Is Disabled *</a>	True	True
<a href="#">Out-of-Bandwidth Text *</a>	Not Enough Bandwidth	Not Enough Bandwidth
<a href="#">Forced Authorization Code Prompt Text *</a>	Enter Authorization Code	Enter Authorization Code
<a href="#">Client Matter Code Prompt Text *</a>	Enter Client Matter Code	Enter Client Matter Code
<a href="#">AAR Network Congestion Rerouting Text *</a>	Network Congestion. Rerouting.	Network Congestion. Rerouting.
<a href="#">Ring Setting of Busy Station Policy *</a>	Only Apply Ring Setting of Busy Station When Incoming Call Arrives	Only Apply Ring Setting of Busy Station When Incoming Call Arrives
<a href="#">Transfer On-hook Enabled *</a>	False	False
<a href="#">Ring Setting of Busy Station *</a>	Beep Only	Beep Only
<a href="#">Ring Setting of Idle Station *</a>	Ring	Ring
<a href="#">Call Pickup Group Audio Alert Setting of Idle Station *</a>	Ring Once	Ring Once
<a href="#">Call Pickup Group Audio Alert Setting of Busy Station *</a>	Beep Only	Beep Only
<a href="#">BLF Pickup Audio Alert Setting of Idle Station *</a>	Disable	Disable
<a href="#">BLF Pickup Audio Alert Setting of Busy Station *</a>	Disable	Disable
<a href="#">Privacy Setting *</a>	True	True
<a href="#">Enforce Privacy Setting on Held Calls *</a>	False	False
<a href="#">SIP Station KeepAlive Interval *</a>	120	120
<a href="#">SIP Station Realm *</a>	ccmsipline	ccmsipline
<a href="#">Hunt Group Logoff Notification *</a>	None	None
<a href="#">Speed Dial Await Further Digits *</a>	False	False
<a href="#">Display CTI Route Point Name or DN *</a>	False	False
<a href="#">Display Original Calling Number on Transfer from Cisco Unity *</a>	False	False
<a href="#">URI Dialing Display Preference *</a>	DN	DN
<a href="#">Insert Hyphens in 12-Digit Numbers *</a>	False	False
<a href="#">Allow Call Waiting During an In-Progress Outbound Analog Call *</a>	True	True

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

**Figure 5: Service Parameters - Cont.**

Clusterwide Parameters (Device - PRI and MGCP Gateway)		
<a href="#">Calling Party Number Screening Indicator</a> *	CallManager sets the screening indicator value - Defau	CallManager sets the screening indicator value - Default setting
<a href="#">Enable Outbound NetworkTrunk CallingParty Restriction</a> *	False	False
<a href="#">Clear Calls Flag When Datalink Is Down</a> *	True	True
<a href="#">Device Status Poll Interval</a> *	3000	3000
<a href="#">Disable Alerting Progress Indicator</a> *	False	False
<a href="#">Discard Non Inband Progress in Overlap Sending</a> *	False	False
<a href="#">Disable Resume from Shared-line MGCP FXS Port</a> *	True	True
<a href="#">DTMF Silence Tone Flag</a> *	False	False
<a href="#">Enable Display IE in Codeset 6</a> *	False	False
<a href="#">Enable Sending PRI NI2 Service Message</a> *	False	False
<a href="#">Flash Hook Duration</a> *	500	500
<a href="#">Gateway Poll Timer</a> *	10	10
<a href="#">Location In PRI Progress Indicator IE (User Side Only)</a> *	Use the Network Side PRI progress indicator IE	Use the Network Side PRI progress indicator IE
<a href="#">Matching Calling Party with Attendant Flag</a> *	False	False
<a href="#">MGCP Database Query Delay Timer</a> *	1000	1000
<a href="#">MGCP FXS On-Hook Pending Timer</a> *	3	3
<a href="#">MGCP Response Timer</a> *	30	30
<a href="#">MGCP Timer</a> *	3	3
<a href="#">Numbering Plan Info</a> *	1	1
<a href="#">Overlap Receiving Flag for PRI</a> *	True	True
<a href="#">Outgoing Media Connect Time for PRI</a> *	Connect ASAP	Connect ASAP
<a href="#">Port Release Timer</a> *	0	0
<a href="#">SMDI Call Delay Timer</a> *	0	0
<a href="#">Stable in State 4 Flag</a> *	False	False

Figure 6: Service Parameters - Cont.



<a href="#">Optimize MGCP Registration</a> *	True	True
<a href="#">Suppress Out-of-Channels Alarms</a> *	True	True
<a href="#">I-Frame Timer</a> *	2000	2000
<a href="#">User-to-User IE Status</a> *	False	False
<a href="#">Convert European Progress Message to Alerting</a> *	False	False
<a href="#">Enable DMS PRI Notify Message from User to Network</a> *	True	True
<a href="#">Audit OOS Channels Interval</a> *	10	10
<a href="#">Digital and Analog Ports Enabled</a> *	True	True

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

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**Clusterwide Parameters (Device - H323)**

<a href="#">Accept Unknown TCP Connection</a> *	False	False
<a href="#">BRQ Enabled</a> *	False	False
<a href="#">Call Present Disconnect Flag</a> *	False	False
<a href="#">Check Progress Indicator Before Establishing Media</a> *	False	False
<a href="#">H225 Block Setup Destination</a> *	False	False
<a href="#">H225 DB Retry Timer</a> *	0	0
<a href="#">H225 Device Connect Timer</a> *	0	0
<a href="#">H225 DTMF Duration</a> *	100	100
<a href="#">H225 TspReq Retry</a> *	2	2
<a href="#">H225 Intercluster Call Throttle Timer</a> *	30	30
<a href="#">H225 T301 Timer</a> *	180000	180000
<a href="#">H225 T302 Timer</a> *	15000	15000
<a href="#">H225 T303 Timer</a> *	4000	4000
<a href="#">H225 T304 Timer</a> *	30000	30000
<a href="#">H225 T305 Timer</a> *	30000	30000
<a href="#">H225 T310 Timer</a> *	60000	60000
<a href="#">H225 TCP Timer</a> *	5	5
<a href="#">H245 TCS Timeout</a> *	10	10
<a href="#">H323 Calling Party Number Screening Indicator</a> *	Calling number screened and passed	Calling number screened and passed
<a href="#">Apply External Phone Number Mask for H.323 Calls</a> *	False	False
<a href="#">Tone on Connect</a> *	False	False
<a href="#">Wait Time for SDP with SR/RO Mode</a> *	3	3

Figure 7: Service Parameters - Cont.

<a href="#">RAS ARQ Timer</a> *	3	3
<a href="#">RAS BRQ Timer</a> *	3	3
<a href="#">RAS DRQ Timer</a> *	3	3
<a href="#">RAS RRQ Timer</a> *	3	3
<a href="#">Ras URQ Timer</a> *	3	3
<a href="#">Retry Count for ARQ</a> *	2	2
<a href="#">Retry Count for BRQ</a> *	2	2
<a href="#">Retry Count for DRQ</a> *	2	2
<a href="#">Retry Count for RRQ</a> *	2	2
<a href="#">Retry Count for URQ</a> *	1	1
<a href="#">Send Product ID and Version ID</a> *	False	False
<a href="#">Send Unified CM Version as Version ID in H225Setup</a> *	False	False
<a href="#">Send Progress Timer</a> *	3000	3000
<a href="#">Send H225 User Info Message</a> *	User Info for Call Progress Tone	User Info for Call Progress Tone
<a href="#">Status Enquiry Poll Timer</a> *	10000	10000
<a href="#">Device Name of GK-controlled Trunk That Will Use Port 1720</a> *	None	None
<a href="#">Host Name/IP Address of GK That Will Use RAS UDP Port 1719</a> *	None	None
<a href="#">Fail Call If MTP Allocation Fails</a> *	False	False
<a href="#">Overlap Receiving Flag for H323</a> *	False	False
<a href="#">Allocate Transcoder for H.323 on Early Offer SIP Trunk for Calls with Early Media</a> *	False	False
There are hidden parameters in this group. Click on Advanced button to see hidden parameters.		
<b>Clusterwide Parameters (Device - SIP)</b>		
<a href="#">SIP Interoperability Enabled</a> *	True	True
<a href="#">Retry Count for SIP Bye</a> *	10	10
<a href="#">Retry Count for SIP Cancel</a> *	10	10
<a href="#">Retry Count for SIP Invite</a> *	6	6
<a href="#">Retry Count for SIP PRACK</a> *	6	6
<a href="#">Retry Count for SIP Rel1XX</a> *	10	10
<a href="#">Retry Count for SIP Publish</a> *	6	6
<a href="#">Retry Count for SIP Response</a> *	6	6
<a href="#">SIP Connect Timer</a> *	500	500
<a href="#">SIP Disconnect Timer</a> *	500	500

Figure 8: Service Parameters - Cont.

<a href="#">SIP Expires Timer</a> *	180000	180000
<a href="#">SIP PRACK Timer</a> *	500	500
<a href="#">SIP Rel1XX Timer</a> *	500	500
<a href="#">SIP Trying Timer</a> *	500	500
<a href="#">SIP Publish Timer</a> *	500	500
<a href="#">SIP Min-SE Value</a> *	1800	1800
<a href="#">SIPS URI Handling</a> *	Reject	Reject
<a href="#">SIP statistics Periodic update Timer</a> *	2	2
<a href="#">SIP Session Expires Timer</a> *	1800	1800
<a href="#">SIP Trunk TspReq Retry</a> *	2	2
<a href="#">SIP TCP Unused Connection Timer</a> *	14	14
<a href="#">SIP TCP Timer</a> *	5	5
<a href="#">SIP Station TCP Port Throttle Threshold</a> *	100	100
<a href="#">SIP Trunk TCP Port Throttle Threshold</a> *	500	500
<a href="#">SIP V.150 Outbound SDP Offer Filtering</a> *	No Filtering	No Filtering
<a href="#">Send SIP Multicast TTL in SDP</a> *	False	False
<a href="#">Default PUBLISH Expiration Timer</a> *	3600	3600
<a href="#">Minimum PUBLISH Expiration Timer</a> *	60	60
<a href="#">IM and Presence Publish Trunk</a>	IMPTrunk	
<a href="#">Send 181 Call Is Being Forwarded</a> *	False	False
<a href="#">Delay Sending 181 until 180/183 message is received</a> *	True	True
<a href="#">Fail Call Over SIP Trunk if MTP Allocation Fails</a> *	False	False
<a href="#">Log Call-Related REFER/NOTIFY /SUBSCRIBE SIP Messages for Session Trace</a> *	True	True
<a href="#">Port Received Timer for Outbound Call Setup</a> *	2	2

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

Figure 9: Service Parameters - Cont.

Clusterwide Parameters (Feature - General)		
<a href="#">Call Park Display Timer</a> *	<input type="text" value="10"/>	10
<a href="#">Caller ID Display Priority Enabled</a> *	<input type="text" value="True"/>	True
<a href="#">Call Park Reversion Timer</a> *	<input type="text" value="60"/>	60
<a href="#">Park Monitoring Reversion Timer</a> *	<input type="text" value="60"/>	60
<a href="#">Park Monitoring Periodic Reversion Timer</a> *	<input type="text" value="30"/>	30
<a href="#">Park Monitoring Forward No Retrieve Timer</a> *	<input type="text" value="300"/>	300
<a href="#">Preserve globalCallId for Parked Calls</a> *	<input type="text" value="True"/>	True
<a href="#">Maximum Call Duration Timer</a> *	<input type="text" value="720"/>	720
<a href="#">Maximum Hold Duration Timer</a> *	<input type="text" value="360"/>	360
<a href="#">Party Entrance Tone</a> *	<input type="text" value="True"/>	True
<a href="#">Message Waiting Lamp Policy</a> *	<input type="text" value="Primary Line - Light and Prompt"/>	Primary Line - Light and Prompt
<a href="#">Audible Message Waiting Indication Policy</a> *	<input type="text" value="OFF"/>	OFF
<a href="#">Message Waiting Indicator Inbound Calling Search Space</a>	<input type="text" value="&lt; None &gt;"/>	
<a href="#">Multiple Tenant MWI Modes</a> *	<input type="text" value="False"/>	False
<a href="#">MWI Non Message Center Signaling Call Duration</a> *	<input type="text" value="0"/>	0
<a href="#">Message Waiting Indicator APDU Digit Translation CSS</a>	<input type="text" value="&lt; None &gt;"/>	
<a href="#">Block OffNet To OffNet Transfer</a> *	<input type="text" value="False"/>	False
<a href="#">Use Original Call Classification for Transferred Calls</a> *	<input type="text" value="False"/>	False
<a href="#">Use Restriction attribute of ID/Name Presentation of Transferring Party</a> *	<input type="text" value="True"/>	True
<a href="#">Local route group for redirected calls</a> *	<input type="text" value="Local route group of calling party"/>	Local route group of calling party
<a href="#">Block Unencrypted Calls</a> *	<input type="text" value="False"/>	False
There are hidden parameters in this group. Click on Advanced button to see hidden parameters.		
Clusterwide Parameters (Feature - Conference)		
<a href="#">Suppress MOH to Conference Bridge</a> *	<input type="text" value="True"/>	True
<a href="#">Drop Ad Hoc Conference</a> *	<input type="text" value="Never"/>	Never
<a href="#">Maximum Ad Hoc Conference</a> *	<input type="text" value="4"/>	4
<a href="#">Maximum MeetMe Conference Unicast</a> *	<input type="text" value="4"/>	4
<a href="#">Advanced Ad Hoc Conference Enabled</a> *	<input type="text" value="False"/>	False
<a href="#">Choose Encrypted Audio Conference Instead Of Video Conference</a> *	<input type="text" value="True"/>	True

Figure 10: Service Parameters - Cont.

<a href="#">Minimum Video Capable Participants To Allocate Video Conference *</a>	<input type="text" value="2"/>	<input type="text" value="2"/>
<a href="#">Enable Click-to-Conference for Third-Party Applications *</a>	<input type="text" value="False"/>	<input type="text" value="False"/>
<a href="#">IMS Conference Factory URI *</a>	<input type="text" value="cucm-conference-factory@cucm1.company.com"/>	<input type="text" value="cucm-conference-factory@cucm1.company.com"/>
<a href="#">Cluster Conferencing Prefix Identifier</a>	<input type="text"/>	
There are hidden parameters in this group. Click on Advanced button to see hidden parameters.		

<b>Clusterwide Parameters (Feature - Call Secure Status Policy)</b>		
<a href="#">Secure Call Icon Display Policy *</a>	<input type="text" value="All media except BFCP and iX transports must be encry"/>	<input type="text" value="All media except BFCP and iX transports must be encrypted"/>

<b>Clusterwide Parameters (Feature - Forward)</b>		
<a href="#">Forward Maximum Hop Count *</a>	<input type="text" value="12"/>	<input type="text" value="12"/>
<a href="#">Forward No Answer Timer *</a>	<input type="text" value="12"/>	<input type="text" value="12"/>
<a href="#">Max Forward Hops to DN *</a>	<input type="text" value="12"/>	<input type="text" value="12"/>
<a href="#">Retain Forward Information *</a>	<input type="text" value="False"/>	<input type="text" value="False"/>
<a href="#">Forward By Reroute Enabled *</a>	<input type="text" value="False"/>	<input type="text" value="False"/>
<a href="#">Transform Forward by Reroute Destination *</a>	<input type="text" value="True"/>	<input type="text" value="True"/>
<a href="#">Always Forward Switch Voice Mail Calls *</a>	<input type="text" value="True"/>	<input type="text" value="True"/>
<a href="#">Forward By Reroute T1 Timer *</a>	<input type="text" value="10"/>	<input type="text" value="10"/>
<a href="#">Include Original Called Info for Q.SIG Call Diversions *</a>	<input type="text" value="Only after the first diversion"/>	<input type="text" value="Only after the first diversion"/>
<a href="#">Set Private Numbering Plan for Call Forward *</a>	<input type="text" value="False"/>	<input type="text" value="False"/>
<a href="#">Set Type of Number for Call Forward *</a>	<input type="text" value="Level1RegionalNumber"/>	<input type="text" value="Level1RegionalNumber"/>
<a href="#">Max Forward UnRegistered Hops to DN *</a>	<input type="text" value="0"/>	<input type="text" value="0"/>
<a href="#">CFA CSS Activation Policy *</a>	<input type="text" value="With Configured CSS"/>	<input type="text" value="With Configured CSS"/>
<a href="#">Cause Code When Maximum Forward Hop Count is Triggered *</a>	<input type="text" value="Normal Unspecified"/>	<input type="text" value="Normal Unspecified"/>
There are hidden parameters in this group. Click on Advanced button to see hidden parameters.		

<b>Clusterwide Parameters (Feature - Hold Reversion)</b>		
<a href="#">Hold Reversion Duration *</a>	<input type="text" value="0"/>	<input type="text" value="0"/>
<a href="#">Hold Reversion Notification Interval *</a>	<input type="text" value="30"/>	<input type="text" value="30"/>
<a href="#">CFA Destination Override *</a>	<input type="text" value="False"/>	<input type="text" value="False"/>

Figure 11: Service Parameters - Cont.

Clusterwide Parameters (Feature - Call Pickup)		
<a href="#">Auto Call Pickup Enabled</a> *	False	False
<a href="#">Call Pickup Locating Timer</a> *	1	1
<a href="#">Call Pickup No Answer Timer</a> *	12	12

Clusterwide Parameters (Feature - Refer)		
<a href="#">Validate Refer-to URI</a> *	Validate Except for Anonymous Users	Validate Except for Anonymous Users

Clusterwide Parameters (Feature - Replaces)		
<a href="#">Block OffNet To OffNet Replaces</a> *	False	False

Clusterwide Parameters (Feature - Redirection [3xx])		
<a href="#">Redirection Ring No Answer</a>	24	24
<a href="#">Reversion Timer</a> *		
<a href="#">Maximum Redirection Count</a> *	70	70

Clusterwide Parameters (Feature - Multilevel Precedence and Preemption)		
<a href="#">Locations-based MLPP Enable</a> *	False	False
<a href="#">Executive Override Call Preemptable</a> *	False	False
<a href="#">Location-based Maximum Bandwidth Enforcement Level for MLPP Calls</a> *	Lenient	Lenient
<a href="#">Non-Preemption Pattern CSS</a>	< None >	
<a href="#">MLPP Exception Level</a> *	Executive Override	Executive Override

Clusterwide Parameters (Feature - Path Replacement)		
<a href="#">Path Replacement Enabled</a> *	False	False
<a href="#">Path Replacement on Tromboned Calls</a> *	True	True
<a href="#">Start Path Replacement Minimum Delay Time</a> *	0	0
<a href="#">Start Path Replacement Maximum Delay Time</a> *	0	0
<a href="#">Path Replacement T1 Timer</a> *	30	30
<a href="#">Path Replacement T2 Timer</a> *	15	15
<a href="#">Path Replacement PINX ID</a>		
<a href="#">Path Replacement Calling Search Space</a>	< None >	

Figure 12: Service Parameters - Cont.

**Clusterwide Parameters (Feature - Call Back)**

<a href="#">Call Back Enabled Flag</a> *	True	True
<a href="#">Call Back Notification Audio File Name</a> *	CallBack.raw	CallBack.raw
<a href="#">Connection Proposal Type</a> *	Connection Retention	Connection Retention
<a href="#">Connection Response Type</a> *	Default to Connection Retention	Default to Connection Retention
<a href="#">Call Back Request Protection T1 Timer</a> *	10	10
<a href="#">Call Back Recall T3 Timer</a> *	20	20
<a href="#">Call Back Calling Search Space</a>	< None >	
<a href="#">No Path Reservation</a> *	True	True
<a href="#">Set Private Numbering Plan for Call Back</a> *	False	False
<a href="#">Set Type of Number for Call Back</a> *	Level1RegionalNumber	Level1RegionalNumber

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

**Clusterwide Parameters (Feature - Call Recording)**

<a href="#">Play Recording Notification Tone To Observed Target</a> *	False	False
<a href="#">Play Recording Notification Tone To Observed Connected Parties</a> *	False	False

**Clusterwide Parameters (Feature - Monitoring)**

<a href="#">Play Monitoring Notification Tone To Observed Target</a> *	False	False
<a href="#">Play Monitoring Notification Tone To Observed Connected Parties</a> *	False	False

**Clusterwide Parameters (Feature - Join Across Lines And Single Button Barge Feature Set)**

<a href="#">Join Across Lines Policy</a> *	Off	Off
<a href="#">Single Button Barge/CBarge Policy</a> *	Off	Off
<a href="#">Allow Barging When Ringing</a> *	False	False

**Clusterwide Parameters (Feature - Secure Tone)**

<a href="#">Play Tone to Indicate Secure/Non-Secure Call Status</a> *	False	False
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There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

Figure 13: Service Parameters - Cont.

Clusterwide Parameters (Feature - External Call Control)		
<a href="#">External Call Control Diversion Maximum Hop Count</a> *	12	12
<a href="#">Maximum External Call Control Diversion Hops to Pattern or DN</a> *	12	12
<a href="#">External Call Control Routing Request Timer</a> *	2000	2000
<a href="#">External Call Control Fully Qualified Role And Resource</a> *	CISCO:UC:UCMPolicy:VoiceOrVideoCall	CISCO:UC:UCMPolicy:VoiceOrVideoCall
<a href="#">External Call Control Initial Connection Count To PDP</a> *	2	2
<a href="#">External Call Control Maximum Connection Count To PDP</a> *	4	4
<a href="#">Always use External Call Control-specified Called/Calling Party Names</a> *	True	True

Clusterwide Parameters (Route Plan)		
<a href="#">Stop Routing on Out of Bandwidth Flag</a> *	False	False
<a href="#">Stop Routing on Unallocated Number Flag</a> *	True	True
<a href="#">Stop Routing on User Busy Flag</a> *	True	True
There are hidden parameters in this group. Click on Advanced button to see hidden parameters.		

Clusterwide Parameters (Route Class Signaling)		
<a href="#">Route Class Trunk Signaling Enabled</a> *	True	True
<a href="#">SIP Route Class Naming Authority</a> *	cisco.com	cisco.com
There are hidden parameters in this group. Click on Advanced button to see hidden parameters.		

Clusterwide Parameters (Hunt List)		
<a href="#">Stop Hunting on Out of Bandwidth Flag</a> *	False	False
<a href="#">Use Pickup Group Of Line Group Member DN</a> *	False	False

Clusterwide Parameters (External QoS)		
<a href="#">External QoS Enabled</a> *	False	False

Figure 14: Service Parameters - Cont.



Clusterwide Parameters (Service)		
<a href="#">Default Network Hold MOH Audio Source ID</a> *	1	1
<a href="#">Default User Hold MOH Audio Source ID</a> *	1	1
<a href="#">Duplex Streaming Enabled</a> *	True	False
<a href="#">Media Exchange Interface Capability Timer</a> *	8	8
<a href="#">Send Multicast MOH in H.245 OLC Message</a> *	True	True
<a href="#">Media Exchange Timer</a> *	12	12
<a href="#">Media Exchange Stop Streaming Timer</a> *	8	8
<a href="#">Open Video Channel Response Timer for SIP Interop</a> *	500	500
<a href="#">Port Received Timer After Call Connection</a> *	500	500
<a href="#">Media Resource Allocation Timer</a> *	12	12
<a href="#">MTP and Transcoder Resource Throttling Percentage</a> *	95	95
<a href="#">Intercluster Capabilities Mismatch Timer</a> *	1000	1000
<a href="#">Silence Suppression</a> *	False	False
<a href="#">Silence Suppression for Gateways</a> *	False	False
<a href="#">Strip G.729 Annex B (Silence Suppression) from Capabilities</a> *	False	False
<a href="#">Enable Source IP Address Verification for Software Media Devices</a> *	True	True

Clusterwide Parameters (System - General)		
<a href="#">Always Use Dial Tone Setting</a> *	Default	Default
<a href="#">Restart Cisco CallManager on Initialization Exception</a> *	True	True
<a href="#">Digit Analysis Timer</a> *	6	6
<a href="#">Statistics Enabled</a> *	True	True

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.

Figure 15: Service Parameters - Cont.

### Clusterwide Parameters (System - QoS)

<a href="#">Priority Class</a> *	Normal Priority	Normal Priority
<a href="#">DSCP for Audio Calls</a> *	46 (101110)	46 (101110)
<a href="#">DSCP for Video Calls</a> *	34 (100010)	34 (100010)
<a href="#">DSCP for Audio Portion of Video Calls</a> *	34 (100010)	34 (100010)
<a href="#">DSCP for TelePresence Calls</a> *	32 (100000)	32 (100000)
<a href="#">DSCP for Audio Portion of TelePresence Calls</a> *	32 (100000)	32 (100000)
<a href="#">DSCP for Priority Audio Calls</a> *	45 (101101)	45 (101101)
<a href="#">DSCP for Immediate Audio Calls</a> *	44 (101100)	44 (101100)
<a href="#">DSCP for Flash Audio Calls</a> *	41 (101001)	41 (101001)
<a href="#">DSCP for Flash Override Audio Calls</a> *	42 (101010)	42 (101010)
<a href="#">DSCP for Executive Override Audio Calls</a> *	42 (101010)	42 (101010)
<a href="#">DSCP for Priority Video Calls</a> *	39 (100111)	39 (100111)
<a href="#">DSCP for Immediate Video Calls</a> *	37 (100101)	37 (100101)
<a href="#">DSCP for Flash Video Calls</a> *	35 (100011)	35 (100011)
<a href="#">DSCP for Flash Override Video Calls</a> *	33 (100001)	33 (100001)
<a href="#">DSCP for Executive Override Video Calls</a> *	33 (100001)	33 (100001)
<a href="#">DSCP for G.Clear Calls</a> *	46 (101110)	46 (101110)
<a href="#">DSCP for Priority G.Clear Calls</a> *	45 (101101)	45 (101101)
<a href="#">DSCP for Immediate G.Clear Calls</a> *	44 (101100)	44 (101100)
<a href="#">DSCP for Flash G.Clear Calls</a> *	41 (101001)	41 (101001)
<a href="#">DSCP for Flash Override G.Clear Calls</a> *	42 (101010)	42 (101010)
<a href="#">DSCP for Executive Override G.Clear Calls</a> *	42 (101010)	42 (101010)
<a href="#">DSCP for Audio Calls when RSVP Fails</a> *	0 (000000)	0 (000000)
<a href="#">DSCP for Video Calls when RSVP Fails</a> *	0 (000000)	0 (000000)
<a href="#">DSCP for ICCP Protocol Links</a> *	24 (011000)	24 (011000)

### Clusterwide Parameters (System - SDL)

<a href="#">SDL Listening Port Number</a> *	8002	8002
<a href="#">SDL Max Router Latency</a> *	20	20
<a href="#">Suppress Debug Info for Router Death</a> *	0	0
<a href="#">Asynchronous SDL Logging Enabled</a> *	False	False

Figure 16: Service Parameters - Cont.

Clusterwide Parameters (System - Location and Region)		
<a href="#">Enforce Millisecond Packet Size</a> *	True	True
<a href="#">Locations Trace Details Enabled</a> *	False	False
<a href="#">Preferred G.711 Millisecond Packet Size</a> *	20	20
<a href="#">Preferred G.722 Millisecond Packet Size</a> *	20	20
<a href="#">Preferred G.723.1 Millisecond Packet Size</a> *	30	30
<a href="#">Preferred G.729 Millisecond Packet Size</a> *	20	20
<a href="#">Always Use Preferred G.729 Packet Size For SIP Trunk Answers</a> *	False	False
<a href="#">Preferred GSM EFR Bytes Packet Size</a> *	31	31
<a href="#">G.711 A-law Codec Enabled</a> *	Enabled for All Devices	Enabled for All Devices
<a href="#">G.711 mu-law Codec Enabled</a> *	Enabled for All Devices	Enabled for All Devices
<a href="#">G.722 Codec Enabled</a> *	Enabled for All Devices	Enabled for All Devices
<a href="#">iLBC Codec Enabled</a> *	Enabled for All Devices	Enabled for All Devices
<a href="#">iSAC Codec Enabled</a> *	Enabled for All Devices	Enabled for All Devices
<a href="#">Opus Codec Enabled</a> *	Enabled for All Devices	Enabled for All Devices
<a href="#">Default Intraregion Max Audio Bit Rate</a> *	64 kbps (G.722, G.711)	64 kbps (G.722, G.711)
<a href="#">Default Interregion Max Audio Bit Rate</a> *	8 kbps (G.729)	8 kbps (G.729)
<a href="#">Default Intraregion Max Video Call Bit Rate (Includes Audio)</a> *	384	384
<a href="#">Default Interregion Max Video Call Bit Rate (Includes Audio)</a> *	384	384
<a href="#">Default Intraregion Max Immersive Video Call Bit Rate (Includes Audio)</a> *	2000000000	2000000000
<a href="#">Default Interregion Max Immersive Video Call Bit Rate (Includes Audio)</a> *	2000000000	2000000000
<a href="#">Use Video BandwidthPool for Immersive Video Calls</a> *	True	True
<a href="#">Default Intraregion and Interregion Link Loss Type</a> *	Low Loss	Low Loss
<a href="#">Default Audio Codec List between Regions</a> *	Factory Default low loss	Factory Default low loss
<a href="#">Default Audio Codec List within Region</a> *	Factory Default low loss	Factory Default low loss
<a href="#">Accept Audio Codec Preferences in Received Offer</a> *	Off	Off
<a href="#">G.Clear Bandwidth Override</a> *	False	False

Figure 17: Service Parameters - Cont.

Clusterwide Parameters (System - CCM Automated Alternate Routing)		
<a href="#">Automated Alternate Routing Enable</a> *	False	False

Clusterwide Parameters (System - RSVP)		
<a href="#">Default inter-location RSVP Policy</a> *	No Reservation	No Reservation
<a href="#">RSVP Retry Timer</a> *	60	60
<a href="#">Mandatory RSVP Mid-call Retry Counter</a> *	1	1
<a href="#">Mandatory RSVP mid call error handle option</a> *	Call becomes best effort	Call becomes best effort
<a href="#">RSVP Video Tspec Burst Size Factor</a> *	5	5
<a href="#">MLPP EXECUTIVE_OVERRIDE To RSVP Priority Mapping</a> *	65535	65535
<a href="#">MLPP FLASH_OVERRIDE To RSVP Priority Mapping</a> *	65534	65534
<a href="#">MLPP FLASH To RSVP Priority Mapping</a> *	65533	65533
<a href="#">MLPP IMMEDIATE To RSVP Priority Mapping</a> *	65532	65532
<a href="#">MLPP PL_PRIORITY To RSVP Priority Mapping</a> *	65531	65531
<a href="#">MLPP PL_ROUTINE To RSVP Priority Mapping</a> *	65530	65530
<a href="#">RSVP Audio Application ID</a> *	AudioStream	AudioStream
<a href="#">RSVP Video Application ID</a> *	VideoStream	VideoStream
<a href="#">RSVP Response Timer</a> *	2	2

TLS Packet Capture Configurations		
<a href="#">Packet Capture Enable</a> *	False	False
<a href="#">Packet Capture Max File Size (MB)</a> *	2	2

Clusterwide Parameters (System - Presence)		
<a href="#">Presence Subscription Throttling Threshold</a> *	60000	60000
<a href="#">Presence Subscription Resume Threshold</a> *	80	80
<a href="#">Default Inter-Presence Group Subscription</a> *	Disallow Subscription	Disallow Subscription
<a href="#">BLF Status Depicts DND</a> *	False	False

Figure 18: Service Parameters - Cont.

Clusterwide Parameters (System - Mobility)		
<a href="#">Enterprise Feature Access Code for Hold *</a>	<input type="text" value="*81"/>	*81
<a href="#">Enterprise Feature Access Code for Exclusive Hold *</a>	<input type="text" value="*82"/>	*82
<a href="#">Enterprise Feature Access Code for Resume *</a>	<input type="text" value="*83"/>	*83
<a href="#">Enterprise Feature Access Code for Transfer *</a>	<input type="text" value="*84"/>	*84
<a href="#">Enterprise Feature Access Code for Conference *</a>	<input type="text" value="*85"/>	*85
<a href="#">Enterprise Feature Access Code for Session Handoff *</a>	<input type="text" value="*74"/>	*74
<a href="#">Enterprise Feature Access Code for Starting Selective Recording *</a>	<input type="text" value="*86"/>	*86
<a href="#">Enterprise Feature Access Code for Stopping Selective Recording *</a>	<input type="text" value="*87"/>	*87
<a href="#">Smart Mobile Phone Interdigit Timer *</a>	<input type="text" value="500"/>	500
<a href="#">Non-Smart Mobile Phone Interdigit Timer *</a>	<input type="text" value="2000"/>	2000
<a href="#">Send Call to Mobile Menu Timer *</a>	<input type="text" value="60"/>	60
<a href="#">SIP Dual Mode Alert Timer *</a>	<input type="text" value="1500"/>	1500
<a href="#">Call Screening Timer *</a>	<input type="text" value="4000"/>	4000
<a href="#">Session Resumption Await Timer *</a>	<input type="text" value="180"/>	180
<a href="#">Inbound Calling Search Space for Remote Destination *</a>	Trunk or Gateway Inbound Calling Search Space	Trunk or Gateway Inbound Calling Search Space
<a href="#">Enable Enterprise Feature Access *</a>	False	False
<a href="#">Dial-via-Office Forward Service Access Number</a>	<input type="text"/>	
<a href="#">Enable Mobile Voice Access *</a>	False	False
<a href="#">Mobile Voice Access Number</a>	<input type="text"/>	
<a href="#">Matching Caller ID with Remote Destination *</a>	Complete Match	Complete Match
<a href="#">Number of Digits for Caller ID Partial Match *</a>	<input type="text" value="10"/>	10
<a href="#">System Remote Access Blocked Numbers</a>	<input type="text"/>	
<a href="#">Enable Use of Called Party Transformed Number for Mobile-terminated Calls *</a>	False	False
<a href="#">Honor Gateway or Trunk Outbound Calling Party Selection for Mobile Connect Calls *</a>	False	False

Figure 19: Service Parameters - Cont.

Clusterwide Parameters (System - Mobility Single Number Reach Voicemail)		
<a href="#">Single Number Reach Voicemail Policy</a> *	Timer Control	Timer Control
<a href="#">Dial-via-Office Reverse Voicemail Policy</a> *	Timer Control	Timer Control
<a href="#">User Control Delayed Announcement Timer</a> *	1000	1000
<a href="#">User Control Confirmed Answer Indication Timer</a> *	10000	10000

Clusterwide Parameters (Feature - Reroute Remote Destination Calls to Enterprise Number)		
<a href="#">Reroute Remote Destination Calls to Enterprise Number</a> *	False	False
<a href="#">Ring All Shared Lines</a> *	False	False
<a href="#">Ignore Call Forward All on Enterprise DN</a> *	True	True

Clusterwide Parameters (Feature - Immediate Divert)		
<a href="#">Use Legacy Immediate Divert</a> *	True	True
<a href="#">Allow QSIG during iDivert</a> *	False	False
<a href="#">Immediate Divert User Response Timer</a> *	5	5

Clusterwide Parameters (Call Admission Control)		
<a href="#">Call Counting CAC Enabled</a> *	False	False
<a href="#">Audio Bandwidth For Call Counting CAC</a> *	102	102
<a href="#">Video Bandwidth For Call Counting CAC</a> *	500	500
<a href="#">UCM to LBM Periodic Reservation Refresh Timer</a> *	5	5
<a href="#">Maximum Bandwidth Deduction Duration</a> *	720	720
<a href="#">Call Treatment When No LBM Available</a> *	Allow Calls	Allow Calls
<a href="#">Locations Media Resource Audio Bit Rate Policy</a> *	Lowest Bit Rate	Lowest Bit Rate
<a href="#">Video Call QoS Marking Policy</a> *	Default	Default
<a href="#">Deduct Audio Bandwidth Portion from Audio Pool for a Video Call</a> *	False	False

Clusterwide Parameters (Emergency Calling for Require Off-premise Location)		
<a href="#">Alternate Destination for Emergency Call</a>		
<a href="#">Alternate Calling Search Space for Emergency Call</a>	< None >	

Figure 20: Service Parameters - Cont.

## Offnet Calls via Cox SIP Trunk

Off-net calls are served by SIP trunks configured between CUCM and the Cox EdgeMarc E-SBC. Calls are routed via the CUBE.

### SIP Trunk Security Profile

1. Navigate to **System > Security > SIP Trunk Security Profile**
2. Click **Add New**

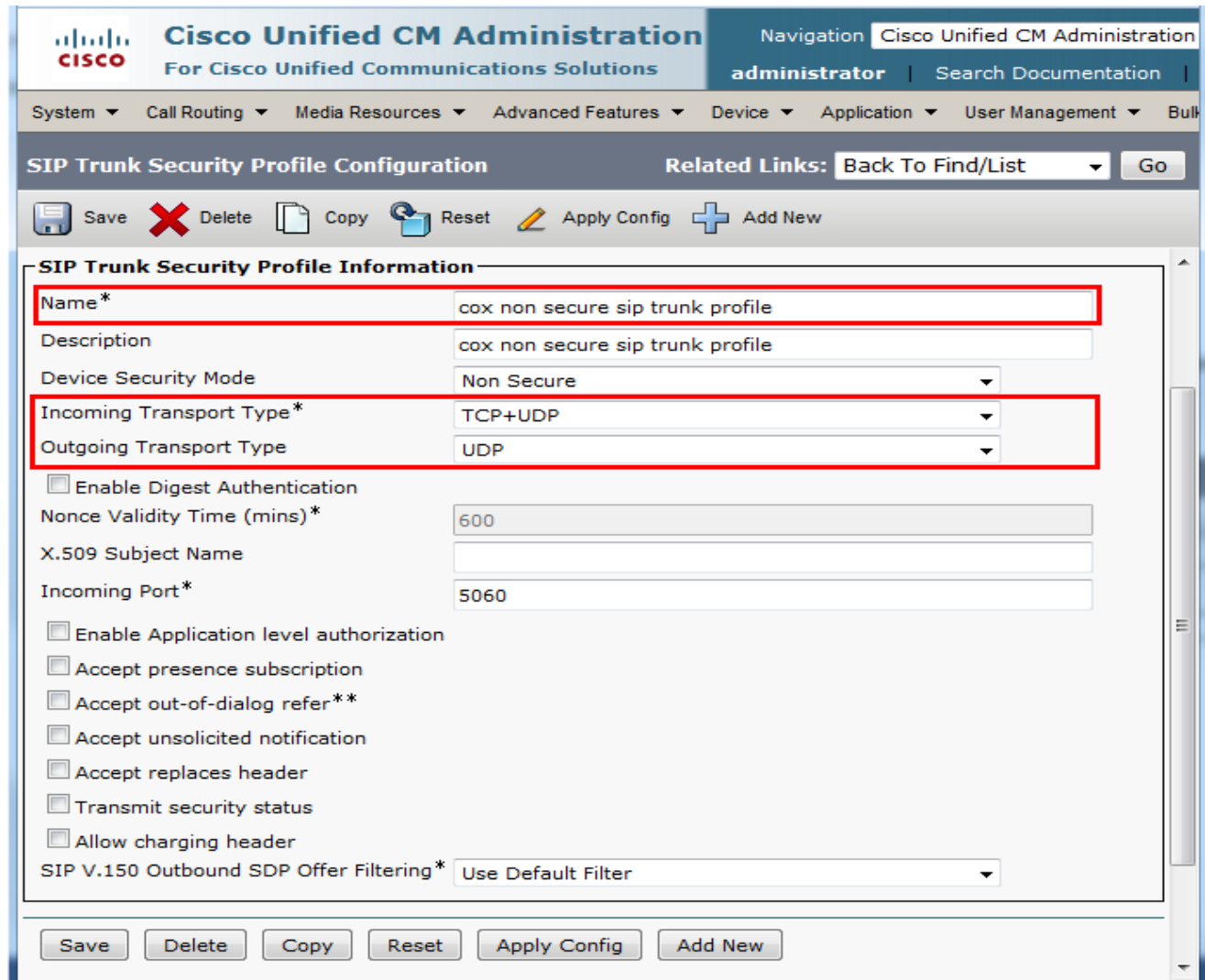


Figure 21: SIP Trunk Security Profile

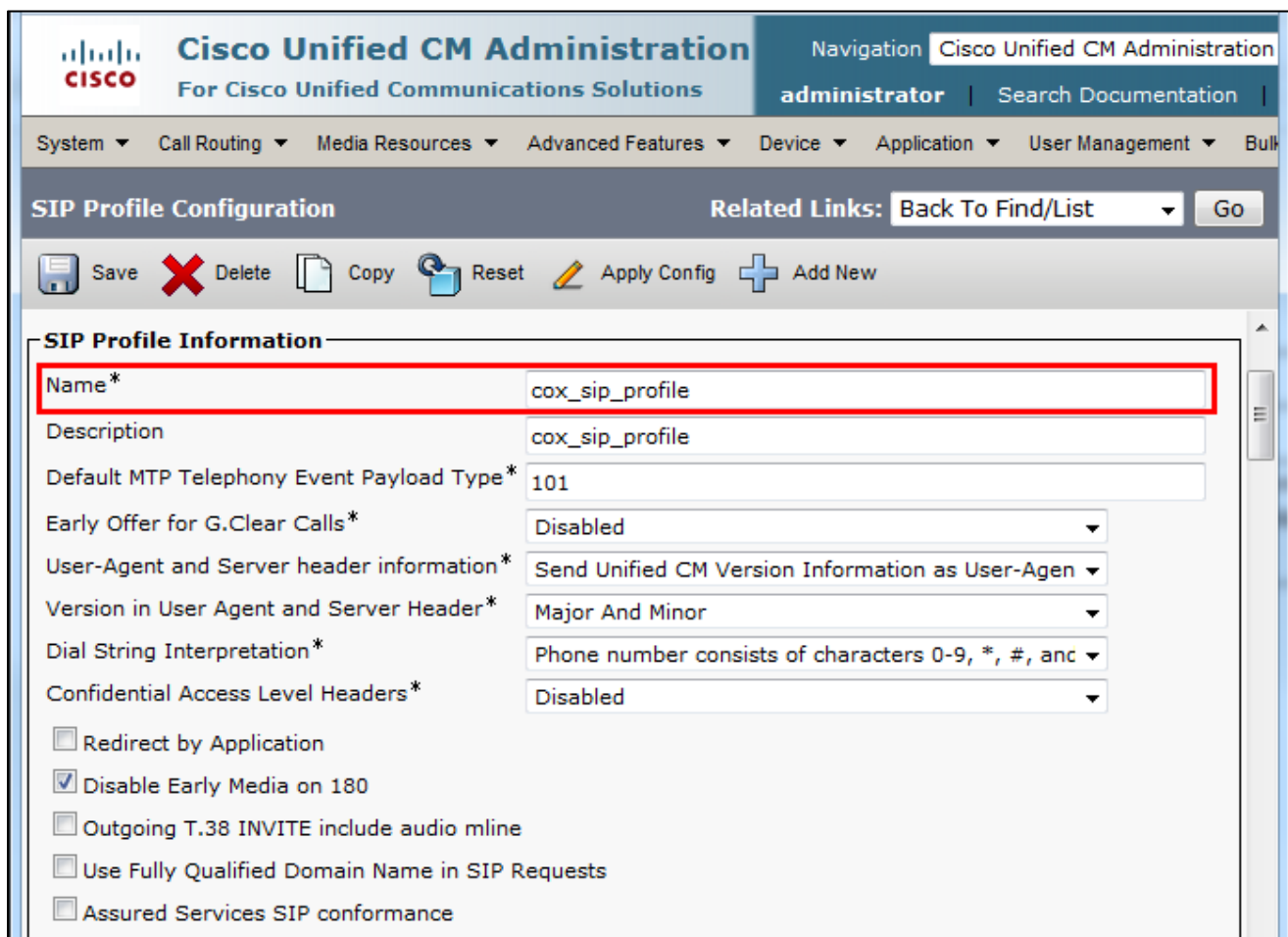
Parameter	Value	Description
-----------	-------	-------------

Incoming Transport Type	TCP + UDP	
Outgoing Transport Type	UDP	SIP trunks to Cox E-SBC should use UDP as a transport protocol for SIP. This is configured using the SIP Trunk Security profile, which is later assigned to the SIP trunk itself.

### SIP Profile

The SIP Profile will be associated later with the SIP trunk

1. Navigate to **Device > Device Settings > SIP Profile**
2. Click **Copy** to modify the default SIP Profile



The screenshot shows the Cisco Unified CM Administration interface. The top navigation bar includes the Cisco logo, the title 'Cisco Unified CM Administration', and the role 'administrator'. Below the navigation bar is a menu with options like System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, and Bulk. The main content area is titled 'SIP Profile Configuration' and includes a 'Related Links' section with a 'Back To Find/List' button. Below this is a toolbar with icons for Save, Delete, Copy, Reset, Apply Config, and Add New. The 'SIP Profile Information' section contains several fields and checkboxes. The 'Name\*' field is highlighted with a red box and contains the value 'cox\_sip\_profile'. Other fields include 'Description' (cox\_sip\_profile), 'Default MTP Telephony Event Payload Type\*' (101), 'Early Offer for G.Clear Calls\*' (Disabled), 'User-Agent and Server header information\*' (Send Unified CM Version Information as User-Agen), 'Version in User Agent and Server Header\*' (Major And Minor), 'Dial String Interpretation\*' (Phone number consists of characters 0-9, \*, #, and), and 'Confidential Access Level Headers\*' (Disabled). There are also several checkboxes at the bottom, including 'Redirect by Application', 'Disable Early Media on 180' (checked), 'Outgoing T.38 INVITE include audio mline', 'Use Fully Qualified Domain Name in SIP Requests', and 'Assured Services SIP conformance'.

Figure 22: SIP Profile



### SDP Information

SDP Session-level Bandwidth Modifier for Early Offer and Re-invites*	TIAS and AS
SDP Transparency Profile	Pass all unknown SDP attributes
Accept Audio Codec Preferences in Received Offer*	Default
<input checked="" type="checkbox"/> Require SDP Inactive Exchange for Mid-Call Media Change	
<input type="checkbox"/> Allow RR/RS bandwidth modifier (RFC 3556)	

### 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
Media Port Ranges	<input checked="" type="radio"/> Common Port Range for Audio and Video <input type="radio"/> Separate Port Ranges for Audio and Video
Start Media Port*	16384
Stop Media Port*	32766
DSCP for Audio Calls	Use System Default
DSCP for Video Calls	Use System Default
DSCP for Audio Portion of Video Calls	Use System Default
DSCP for TelePresence Calls	Use System Default
DSCP for Audio Portion of TelePresence Calls	Use System Default

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
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



**Figure 23: SIP Profile - Cont.**

Resource Priority Namespace	< None >
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-cfwdall
Speed Dial (Abbreviated Dial) URI*	x-cisco-serviceuri-abbrdial
<input checked="" type="checkbox"/> Conference Join Enabled <input type="checkbox"/> RFC 2543 Hold <input checked="" type="checkbox"/> Semi Attended Transfer <input type="checkbox"/> Enable VAD <input type="checkbox"/> Stutter Message Waiting <input type="checkbox"/> MLPP User Authorization	

<b>Normalization Script</b>							
Normalization Script	< None >						
<input type="checkbox"/> Enable Trace							
	<table border="1"> <thead> <tr> <th></th> <th>Parameter Name</th> <th>Parameter Value</th> </tr> </thead> <tbody> <tr> <td>1</td> <td></td> <td></td> </tr> </tbody> </table>		Parameter Name	Parameter Value	1		
	Parameter Name	Parameter Value					
1							
<b>Incoming Requests FROM URI Settings</b>							
Caller ID DN							
Caller Name							

<b>Trunk Specific Configuration</b>	
Reroute Incoming Request to new Trunk based on*	Never
Resource Priority Namespace List	< None >
SIP Rel1XX Options*	Send PRACK if 1xx Contains SDP
Video Call Traffic Class*	Mixed
Calling Line Identification Presentation*	Default
Session Refresh Method*	Invite
Early Offer support for voice and video calls*	Best Effort (no MTP inserted)
<input type="checkbox"/> Enable ANAT <input type="checkbox"/> Deliver Conference Bridge Identifier <input type="checkbox"/> Allow Passthrough of Configured Line Device Caller Information <input type="checkbox"/> Reject Anonymous Incoming Calls <input type="checkbox"/> Reject Anonymous Outgoing Calls <input type="checkbox"/> Send ILS Learned Destination Route String	

**Figure 24: SIP Profile - Cont.**

**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

**SDP Information**

☒ Send send-receive SDP in mid-call INVITE
   
☐ Allow Presentation Sharing using BFCP
   
☐ Allow iX Application Media
   
☐ Allow multiple codecs in answer SDP

Figure 25: SIP Profile – Cont.

Parameter	Value	Description
Default MTP Telephony Event Payload Type	101	RFC2833 DTMF payload type
Require SDP Inactive Exchange for Mid-Call Media Change	Checked	Send SDP with Inactive when call on hold
SIP Rel1XX Options	Send PRACK for 1xx Messages	Enable Provisional Acknowledgements (Reliable 100 messages)
Ping Interval for In-service and Partially In-service Trunks (seconds)	300	OPTIONS message parameters- interval time
Ping Interval for Out-of-service Trunks (seconds)	5	OPTIONS message parameters- interval time

## SIP Trunk Configuration

To create SIP trunks to Cox

1. Navigate to **Device > Trunk**
2. Click **Add New**
3. Apply to create SIP trunks to Cisco Unity Connection and Fax Gateway






Cisco Unified CM Administration									
For Cisco Unified Communications Solutions									
Navigation		Cisco Unified CM Administration							
administrator		Search Documentation   About   Logout							
System	Call Routing	Media Resources	Advanced Features	Device	Application	User Management	Bulk Administration	Help	
Find and List Trunks									
<input type="button" value="Add New"/> <input type="button" value="Select All"/> <input type="button" value="Clear All"/> <input type="button" value="Delete Selected"/> <input type="button" value="Reset Selected"/>									
<input type="checkbox"/>		<a href="#">cox_trunk</a>		<a href="#">G711 pool</a>	<a href="#">9.0</a>		SIP Trunk	Full Service	Time In Full Service: 0 day 0 hour 40 minutes
<input type="checkbox"/>		<a href="#">fax_19_2</a>	fax	<a href="#">Default</a>	<a href="#">9725980124</a>		SIP Trunk	Full Service	Time In Full Service: 20 days 3 hours 32 minutes
<input type="checkbox"/>		<a href="#">fax_19_2</a>	fax	<a href="#">Default</a>	<a href="#">4020</a>		SIP Trunk	Full Service	Time In Full Service: 20 days 3 hours 32 minutes
<input type="checkbox"/>		<a href="#">trunk_to_fax</a>		<a href="#">G711 pool</a>	<a href="#">5852</a>		SIP Trunk	Full Service	Time In Full Service: 0 day 0 hour 40 minutes
<input type="button" value="Add New"/> <input type="button" value="Select All"/> <input type="button" value="Clear All"/> <input type="button" value="Delete Selected"/> <input type="button" value="Reset Selected"/>									

Figure 26: SIP Trunk List


**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

**SIP Trunk Status**  
**Service Status:** Full Service  
**Duration:** Time In Full Service: 0 day 0 hour 44 minutes

**Device Information**

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	cox_trunk
Description	
Device Pool*	G711 pool
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	MRGL_MTP
Location*	Hub_None
AAR Group	< None >
Tunneled Protocol*	None
QSIG Variant*	No Changes
ASN.1 ROSE OID Encoding*	No Changes
Packet Capture Mode*	None

Packet Capture Duration 0

☐ Media Termination Point Required  
☒ 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

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

Figure 27: SIP Trunk to CUBE

**MLPP and Confidential Access Level Information**

MLPP Domain: < None >

Confidential Access Mode: < None >

Confidential Access Level: < None >

**Call Routing Information**

☒ Remote-Party-Id

☒ Asserted-Identity

Asserted-Type\*: Default

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:

☐ 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"/>

**Incoming Called 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"/>

Figure 28: SIP Trunk to CUBE - Cont.

**Outbound Calls**

Called Party Transformation CSS

☒ Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS

☒ Use Device Pool Calling Party Transformation CSS

Calling Party Selection\*

Calling Line ID Presentation\*

Calling Name Presentation\*

Calling and Connected Party Info Format\*

☐ Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS

☒ Use Device Pool Redirecting Party Transformation CSS

**Caller Information**

Caller ID DN

Caller Name

☐ Maintain Original Caller ID DN and Caller Name in Identity Headers

**SIP Information**

**Destination**

☐ Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1 *	<input type="text" value="10.80.13.20"/>	<input type="text"/>	<input type="text" value="5060"/>

MTP Preferred Originating Codec\*

BLF Presence Group\*

SIP Trunk Security Profile\*

Rerouting Calling Search Space

Out-Of-Dialog Refer Calling Search Space

SUBSCRIBE Calling Search Space

SIP Profile\*  [View Details](#)

DTMF Signaling Method\*

Figure 29: SIP Trunk to CUBE - Cont.



**Normalization Script**

Normalization Script < None >

☐ Enable Trace

	Parameter Name	Parameter Value		
1			+	-

**Recording Information**

☒ None

☐ This trunk connects to a recording-enabled gateway

☐ This trunk connects to other clusters with recording-enabled gateways

**Geolocation Configuration**

Geolocation < None >

Geolocation Filter < None >

☐ Send Geolocation Information

Save Delete Reset Add New

**Figure 30: SIP Trunk to CUBE - Cont.**

Parameter	Value	Description
Device Name	Cox_Trunk	Name for the trunk
Device Pool	G711 Pool	Default Device Pool is used for this trunk
Media Resource Group List	MRGL_MTP	MRG with resources: ANN, CFB, MOH and MTP
Significant Digits	4	4 digits Extension for all CPE phones
Destination Address	10.80.13.20	Virtual IP address of the CUBE
SIP Trunk Security Profile	Cox Non Secure SIP Trunk Profile	SIP Trunk Security Profile configured earlier
SIP Profile	Cox SIP Profile	SIP Profile configured earlier
DTMF Signaling Method	RFC 2833	RFC 2833 is supported for DTMF transport to/from Cox

Reset the trunk after the configuration is completed. Apply same procedure to create SIP trunks to Cisco Unity Connection.

## Dial Plan

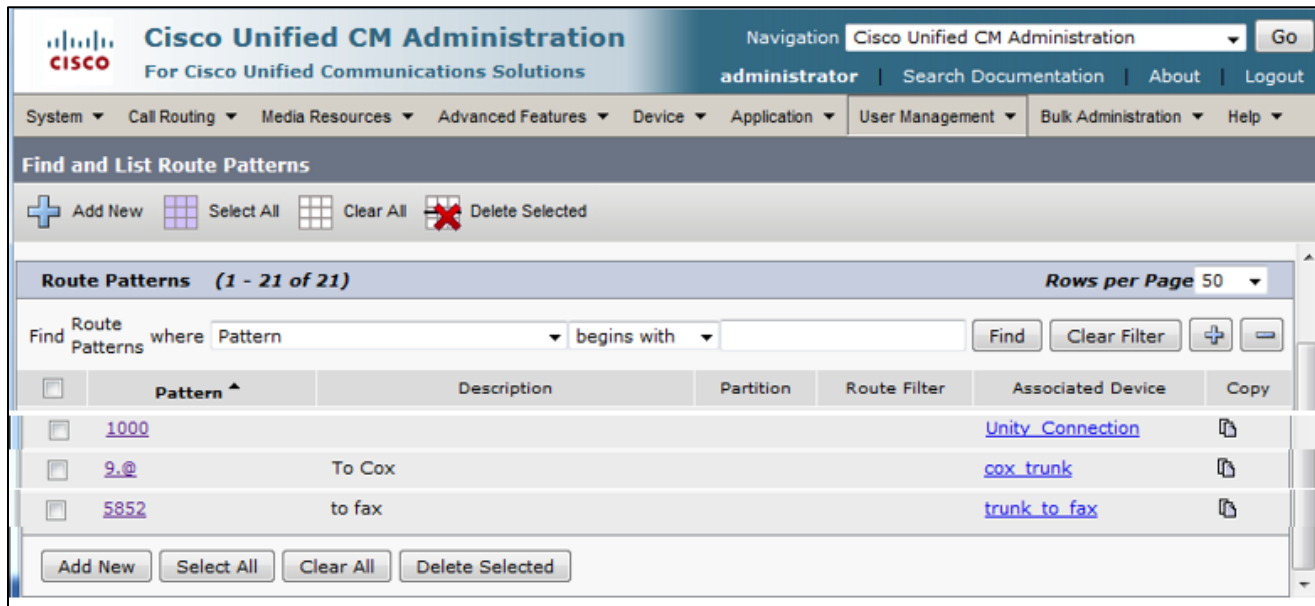
### Route Pattern Configuration

Route patterns are configured as below:

- Cisco IP phones dial 9+10 digits number to access PSTN via CUBE
- "9" is removed before send to CUBE
- For FAX call, Access Code 9 is used at fax gateway
- "9" is removed at UCM and 10 digits number is send to CUBE to Cox network
- Incoming fax call to 5852 will send to fax gateway
- 1000 is the Pilot Number for voicemail to Unity Connection

To Create Route Patterns

1. Navigate to **Call Routing > Route/Hunt > Route Pattern**
2. Click **Add New**




The screenshot shows the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and a navigation dropdown menu set to "Cisco Unified CM Administration". Below the navigation bar is a secondary bar with "administrator" and links for "Search Documentation", "About", and "Logout". A main navigation menu contains tabs for "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", "Bulk Administration", and "Help".

The "Call Routing" tab is selected, leading to the "Find and List Route Patterns" page. At the top of this page are buttons for "Add New", "Select All", "Clear All", and "Delete Selected". Below these is a table titled "Route Patterns (1 - 21 of 21)". The table has columns for "Pattern", "Description", "Partition", "Route Filter", "Associated Device", and "Copy". The first three rows of the table are visible:

Pattern	Description	Partition	Route Filter	Associated Device	Copy
1000				Unity Connection	
9.0	To Cox			cox_trunk	
5852	to fax			trunk to fax	

At the bottom of the table are buttons for "Add New", "Select All", "Clear All", and "Delete Selected".





Figure 31: Route Patterns


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

Navigation **Cisco Unified CM Administration**  
**administrator** | Search Documentation | Ab

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Ad

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

 Save
  Delete
  Copy
  Add New

**Pattern Definition**

<b>Route Pattern*</b>	9.@
Route Partition	< None >
Description	To Cox
Numbering Plan*	NANP
Route Filter	< None >
MLPP Precedence*	Default
<input type="checkbox"/> Apply Call Blocking Percentage	
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
<b>Gateway/Route List*</b>	cox_trunk
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error ▾
Call Classification*	OffNet
External Call Control Profile	< None >
<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*	0
<input type="checkbox"/> Require Client Matter Code	

**Calling Party Transformations**

<input checked="" type="checkbox"/> 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

**Figure 32: Route Patterns for Voice**

**Called Party Transformations**

Discard Digits PreDot

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 >

Save Delete Copy Add New

Figure 33: Route Patterns for Voice - Cont.

**Pattern Definition**

Route Pattern\* 1000

Route Partition < None >

Description

Numbering Plan -- Not Selected --

Route Filter < None >

MLPP Precedence\* Default

☐ Apply Call Blocking Percentage

Resource Priority Namespace Network Domain < None >

Route Class\* Default

Gateway/Route List\* Unity\_Connection [\(Edit\)](#)

Route Option ☒ Route this pattern ☐ Block this pattern No Error

Call Classification\* OnNet

External Call Control Profile < None >

☐ Allow Device Override ☐ Provide Outside Dial Tone ☐ Allow Overlap Sending ☐ Urgent Priority

Figure 34: Route Patterns for Unity

<input type="checkbox"/> Require Forced Authorization Code Authorization Level* <input type="text" value="0"/>	
<input type="checkbox"/> Require Client Matter Code	
<b>Calling Party Transformations</b>	
<input type="checkbox"/> Use Calling Party's External Phone Number Mask Calling Party Transform Mask <input type="text"/> Prefix Digits (Outgoing Calls) <input type="text"/> Calling Line ID Presentation* <input type="text" value="Default"/> Calling Name Presentation* <input type="text" value="Default"/> Calling Party Number Type* <input type="text" value="Cisco CallManager"/> Calling Party Numbering Plan* <input type="text" value="Cisco CallManager"/>	
<b>Connected Party Transformations</b>	
Connected Line ID Presentation* <input type="text" value="Default"/> Connected Name Presentation* <input type="text" value="Default"/>	
<b>Called Party Transformations</b>	
Discard Digits <input type="text" value=" &lt; None &gt;"/> Called Party Transform Mask <input type="text"/> Prefix Digits (Outgoing Calls) <input type="text"/> Called Party Number Type* <input type="text" value="Cisco CallManager"/> Called Party Numbering Plan* <input type="text" value="Cisco CallManager"/>	
<b>ISDN Network-Specific Facilities Information Element</b>	
Network Service Protocol <input type="text" value="-- Not Selected --"/> Carrier Identification Code <input type="text"/>	
Network Service <input type="text" value="-- Not Selected --"/>	Service Parameter Name <input type="text" value=" &lt; Not Exist &gt;"/>
<input type="button" value="Save"/> <input type="button" value="Delete"/> <input type="button" value="Copy"/> <input type="button" value="Add New"/>	

Figure 35: Route Patterns for Unity - Cont.

**Pattern Definition**

Route Pattern\*5852

Route Partition< None >

Descriptionto fax

Numbering Plan-- Not Selected --

Route Filter< None >

MLPP Precedence\*Default

☐ Apply Call Blocking Percentage

Resource Priority Namespace Network Domain< None >

Route Class\*Default

Gateway/Route List\*trunk\_to\_fax (Edit)

Route Option

☒ Route this pattern
☐ Block this pattern No Error

Call Classification\*OffNet

External Call Control Profile< None >

☐ 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

Connected Name Presentation\*Default

Figure 36: Route Patterns for Fax

**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 < Not Exist >
Service Parameter

Save Delete Copy Add New

**Figure 37: Route Patterns for Fax – Cont.**

Setting	Value	Description
Route Pattern	9.@ for Voice call and 8.@ for fax call	Specify appropriate Route Pattern
Gateway/Route List	COX	SIP Trunk name configured earlier
Require Forced Authorization Code	Checked when doing Authorization Code test	Specify if Authorization Code required when make call through this Route Pattern
Require Client Matter Code	Check when doing Account Code test	Specify if Account Code required when make call through this Route Pattern
Calling Party Transform mask	678238XXXX	Specify the Calling Line ID for outgoing call through this Route Pattern
Discard Digits	PreDot for RP 9.@	Specifies how to modify digit before they are sending to Cox ESBC



## Acronyms

Acronym	Definitions
CPE	Customer Premise Equipment
CUBE	Cisco Unified Border Element
CUCM	Cisco Unified Communications Manager
MTP	Media Termination Point
POP	Point of Presence
PSTN	Public Switched Telephone Network
ESBC	Enterprise Session Border Controller
SCCP	Skinny Client Control Protocol
SIP	Session Initiation Protocol





## **Important Information**

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## **Appendix A: Test Results**