

## Cox SIP Trunking:

Cisco Unified Communications Manager  
11.0.1 with Cisco Unified Border Element  
(CUBE 11.5.0) on ISR4321 [IOS-XE 3.17]  
using SIP

April 2017



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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 11.5.0) on ISR 4321 with IOS-XE 3.17 can be used. The Cisco Unified Border Element 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 11.5.0) on ISR 4321 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)

## Network Topology

### Basic Call Setup

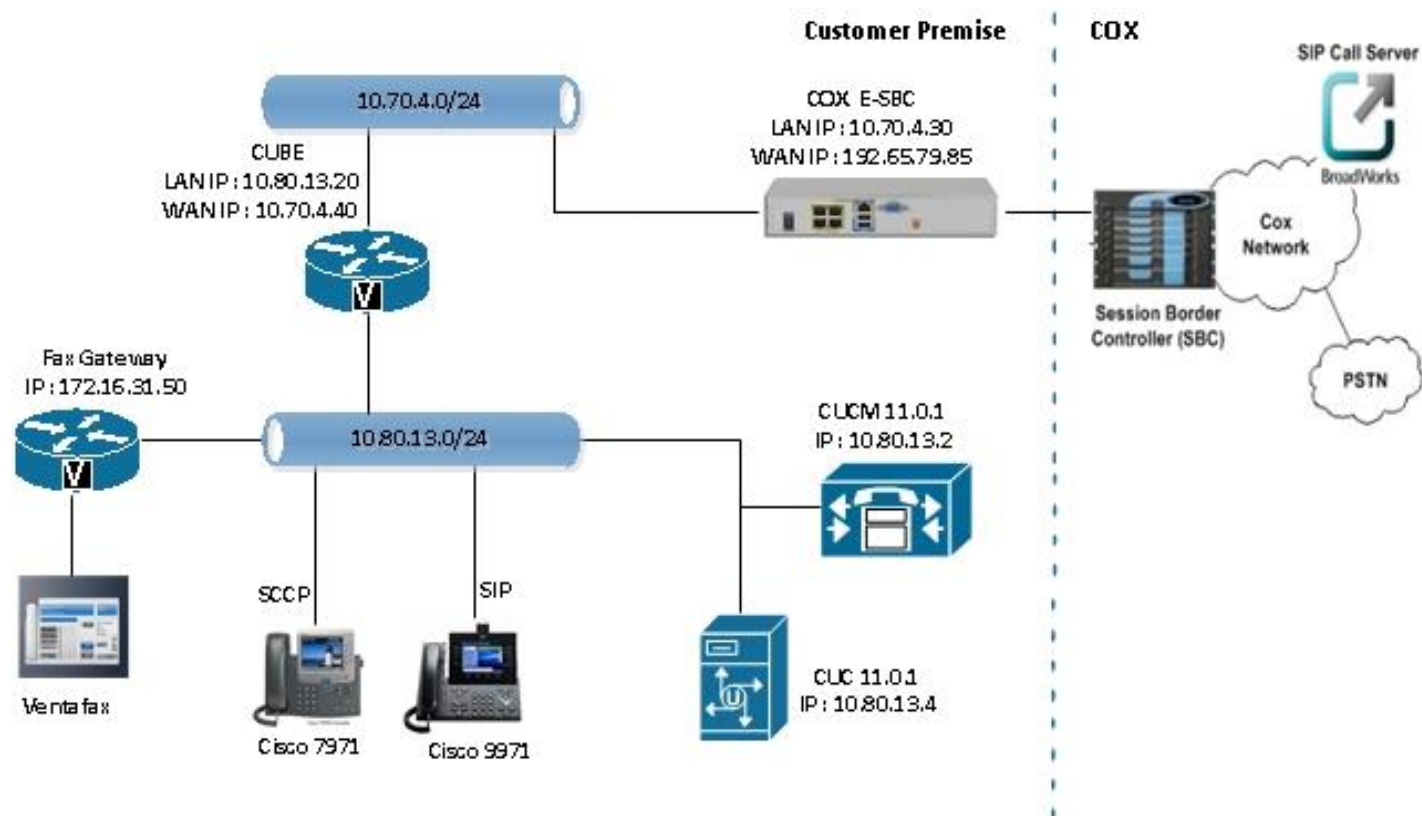


Figure 1: Network Topology



## **System Components**

### ***Hardware Components***

- Cisco UCM and Unity Connection run on VMware
- ISR 2851 router as Fax Gateway
- ISR 4321 router as CUBE
- IP phones 9971(SIP) and 7971(SCCP)

### ***Software Requirements***

- Cisco Unified Communications Manager 11.0.1
- IOS-15.6(1)S for Cisco Unified Border Element on ISR4321
- IOS 15.1(4)M5 for Fax Gateways on ISR2851
- 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***

- 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 – CSCuv04539



## 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 CUBE LAN
ip address 10.80.13.20 255.255.255.0
negotiation auto
```

!

```
interface GigabitEthernet0/0/1
description COX CUBE WAN
ip address 10.70.4.40 255.255.255.0
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 20 voip
```

```
description " Incoming to IP-PBX - IP-PBX facing side "
```

```
destination-pattern 402614....
```

```
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 rate disable
```

```
fax protocol pass-through g711ulaw
```

```
no vad
```

```
!
```

```
dial-peer voice 21 voip
```

```
description " Incoming IP-PBX - EdgeMarc facing side "
```

```
no modem passthrough
```

```
session protocol sipv2
```

```
incoming called-number 402614....
```

```
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
!
dial-peer voice 15 voip
description "Outgoing To EdgeMarc" -EdgeMarc facing side
destination-pattern [0-9]T
no modem passthrough
session protocol sipv2
session target ipv4:10.70.4.30:5060
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 profiles 1
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 16 voip
description "Outgoing To EdgeMarc" -IP PBX facing side"
no modem passthrough
session protocol sipv2
incoming called-number [0-9]T
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 profiles 1
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
```



## ISR Configuration

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

```
COX_CUBE#sh running-config
```

```
Building configuration...
```

```
!
```

```
version 15.6
```

```
service timestamps debug datetime msec
```

```
service timestamps log datetime msec
```

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

```
!
```

```
hostname COX_CUBE
```

```
!
```

```
boot-start-marker
```

```
boot system flash:isr4300-universalk9.03.17.00.S.156-1.S-std.SPA.bin
```

```
boot system bootflash:isr4300-universalk9.03.17.00.S.156-1.S-std.SPA.bin
```

```
boot-end-marker
```

```
!
```

```
!
```

```
vrf definition Mgmt-intf
```

```
!
```

```
address-family ipv4
```

```
exit-address-family
```

```
!
```

```
address-family ipv6
```

```
exit-address-family
```

```
!
```

```
no aaa new-model
```

```
no ip domain lookup
```

```
ip domain name tekvizion.com
```



```
!  
subscriber templating  
multilink bundle-name authenticated  
!  
voice service voip  
no ip address trusted authenticate  
address-hiding  
mode border-element license capacity 50  
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 sip-profiles 1  
request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:402614\1@\2>"  
!  
license udi pid ISR4321/K9 sn FDO17860MW3  
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 CUBE LAN  
ip address 10.80.13.20 255.255.255.0  
negotiation auto  
!  
interface GigabitEthernet0/0/1  
description COX CUBE WAN  
ip address 10.70.4.40 255.255.255.0  
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 10.64.0.0 255.255.0.0 10.80.13.1
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 20 voip
description " Incoming to IP-PBX - IP-PBX facing side "
destination-pattern 402614....
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 rate disable
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 21 voip
description " Incoming IP-PBX - EdgeMarc facing side "
session protocol sipv2
incoming called-number 402614....
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
!
dial-peer voice 15 voip
description "Outgoing To EdgeMarc" -EdgeMarc facing side
destination-pattern [0-9]T
session protocol sipv2
session target ipv4:10.70.4.30:5060
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 profiles 1
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 16 voip
description "Outgoing To EdgeMarc" -IP PBX facing side"
session protocol sipv2
incoming called-number [0-9]T
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 profiles 1
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
!
sip-ua
credentials username 4023159990 password 7 15465B5E577B7E7D716A65 realm 10.70.4.30
```



```
authentication username 4023159990 password 7 01475656085A535678151E
registrar ipv4:10.70.4.30 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 UCM Version

**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 ▾

### Cisco Unified CM Administration

**System version: 11.0.1.21900-11**

VMware Installation: 1 vCPU Intel(R) Xeon(R) CPU E5-2680 0 @ 2.70GHz, disk 1: 80Gbytes, 4096Mbytes RAM, Partitions aligned  
**WARNING: DNS unreachable**

**No successful backup has been taken after the Upgrade**

User administrator last logged in to this cluster on Monday, January 11, 2016 12:31:07 AM CST, to node 10.80.13.2, from 172.16.31.203 using HTTPS

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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 our [Export Compliance Product Report](#) web site.

For information about Cisco Unified Communications Manager please visit our [Unified Communications System Documentation](#) web site.

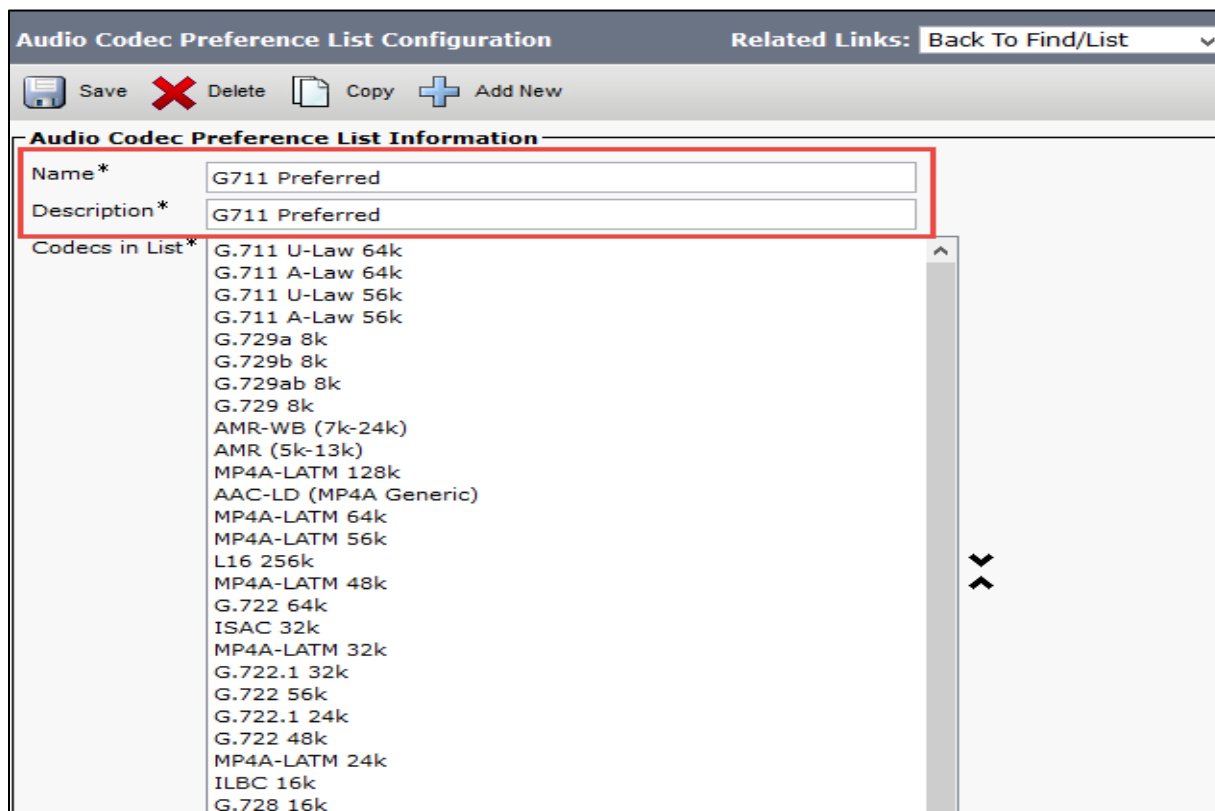
For Cisco Technical Support please visit our [Technical Support](#) web site.

Figure 2: Cisco UCM Version

## Cisco UCM Audio Codec Preference List

**Navigation Path:** System → Region Information → Audio codec preference list

Cisco UCM 11.0.1 has a feature called Audio Codec Preference List. This feature allows to configure the order of audio codec preference both for Inter and Intra Region calls. Audio Codec Preference list is assigned to the Region used by the Device Pool for Phones and by Conference Bridges. Based on user requirement, different codec regions can be assigned as their first choice codec with this configuration for inbound calls as well as conferences initiated by Cisco IP phones. Audio codec preference for outbound calls is determined by Cisco UBE (via configuration of voice-class codec)



**Audio Codec Preference List Configuration** Related Links: [Back To Find/List](#)

Save Delete Copy Add New

**Audio Codec Preference List Information**

Name\* G711 Preferred

Description\* G711 Preferred

Codecs in List\*

- G.711 U-Law 64k
- G.711 A-Law 64k
- G.711 U-Law 56k
- G.711 A-Law 56k
- G.729a 8k
- G.729b 8k
- G.729ab 8k
- G.729 8k
- AMR-WB (7k-24k)
- AMR (5k-13k)
- MP4A-LATM 128k
- AAC-LD (MP4A Generic)
- MP4A-LATM 64k
- MP4A-LATM 56k
- L16 256k
- MP4A-LATM 48k
- G.722 64k
- ISAC 32k
- MP4A-LATM 32k
- G.722.1 32k
- G.722 56k
- G.722.1 24k
- G.722 48k
- MP4A-LATM 24k
- ILBC 16k
- G.728 16k

Figure 3: Audio Codec Preference List



## Cisco UCM Region Configuration

Navigation Path: System → Region Information → Region

Region Configuration

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

Save

Delete

Reset

Apply Config

Add New

Region Information

Name\*

G711 region

Region Relationships

Region	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls	Maximum Session Bit Rate for Immersive Video Calls
Default	G711 Preferred	64 kbps (G.722, G.711)	384 kbps	2147483647 kbps
G711 region	G711 Preferred	64 kbps (G.722, G.711)	384 kbps	2147483647 kbps
G729 Region	G729 Preferred	64 kbps (G.722, G.711)	384 kbps	2147483647 kbps
NOTE: Regions not displayed	Use System Default	Use System Default	Use System Default	Use System Default

Modify Relationship to other Regions

Regions	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls	Maximum Session Bit Rate for Immersive Video Calls
<div><div>Default</div><div>G711 region</div><div>G711 test</div><div>G729 Region</div><div>test</div></div>	<div>Keep Current Setting</div>	<div><div><div>Keep Current Setting</div><div>Use System Default</div><div>None</div></div><div>kbps</div></div>	<div><div><div>Keep Current Setting</div><div>Use System Default</div><div>None</div></div><div>kbps</div></div>	<div><div><div>Keep Current Setting</div><div>Use System Default</div><div>None</div></div><div>kbps</div></div>

Figure 4: Region Configuration

## Device Pool Configuration

Navigation Path: System → Device Pool

“G711 pool” Device Pool is configured for testing the interoperability. No special consideration needs to be taken when configuring the Device Pools. Optionally, a Media Resource Group List can be added to the Device Pools, if needed, to assign selected Media Resources (Conference Bridges, Transcoders, MoH servers, Annunciators) to devices.

<b>Device Pool Information</b>	
Device Pool: G711 pool (14 members**)	
<b>Device Pool Settings</b>	
Device Pool Name *	G711 pool
Cisco Unified Communications Manager Group *	Default
Calling Search Space for Auto-registration	< None >
Adjunct CSS	< None >
Reverted Call Focus Priority	Default
Intercompany Media Services Enrolled Group	< None >
<b>Roaming Sensitive Settings</b>	
Date/Time Group *	CMLocal
Region *	G711 region
Media Resource Group List	MRGL_MTP
Location	< None >
Network Locale	< None >
SRST Reference *	Disable
<b>Connection Monitor Duration***</b>	
Single Button Barge *	Default
Join Across Lines *	Default
Physical Location	< None >
Device Mobility Group	< None >
Wireless LAN Profile Group	< None > <a href="#">View Details</a>
<b>Local Route Group Settings</b>	
Standard Local Route Group	< None >
<b>Device Mobility Related Information****</b>	
Device Mobility Calling Search Space	< None >
AAR Calling Search Space	< None >

Figure 5: Device Pool Configuration

AAR Group	< None >
Calling Party Transformation CSS	< None >
Called Party Transformation CSS	< None >

---

**Geolocation Configuration**

Geolocation	< None >
Geolocation Filter	< None >

---

**Call Routing Information**

**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
National Number	Default		< None >
International Number	Default		< None >
Unknown Number	Default		< None >
Subscriber Number	Default		< None >

---

**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
National Number	Default	0	< None >
International Number	Default	0	< None >
Unknown Number	Default	0	< None >
Subscriber Number	Default	0	< None >

---

**Phone Settings**

**Caller ID For Calls From This Phone**

Calling Party Transformation CSS	< None >
----------------------------------	----------

---

**Connected Party Settings**

Connected Party Transformation CSS	< None >
------------------------------------	----------

---

**Redirecting Party Settings**

Redirecting Party Transformation CSS	< None >
--------------------------------------	----------

Figure 6: Device Pool Configuration – Cont.

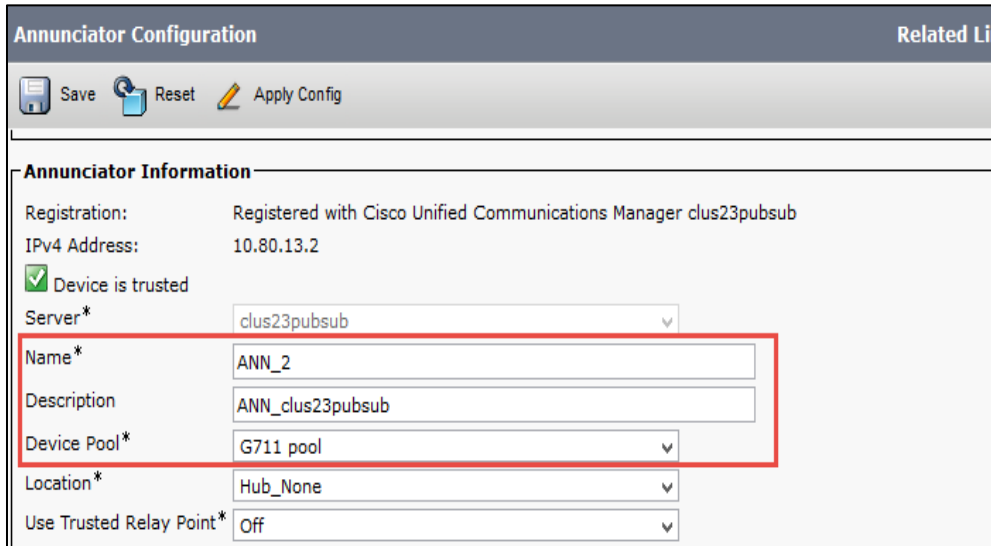
## Annunciator Configuration

**Navigation:** Media Resource → Annunciator

Set Name\* = ANN\_2

Set Description = ANN\_clus23pubsub. This is used for this example.

Set Device Pool\* = G711 pool



The screenshot shows the 'Annunciator Configuration' page in a web interface. At the top, there are tabs for 'Save', 'Reset', and 'Apply Config'. Below this is the 'Annunciator Information' section. It displays the following configuration details:

Registration:	Registered with Cisco Unified Communications Manager clus23pubsub
IPv4 Address:	10.80.13.2
<input checked="" type="checkbox"/> Device is trusted	
Server*	clus23pubsub
Name*	ANN_2
Description	ANN_clus23pubsub
Device Pool*	G711 pool
Location*	Hub_None
Use Trusted Relay Point*	Off

A red rectangular box highlights the 'Name\*', 'Description', and 'Device Pool\*' fields and their corresponding values.

Figure 7: Annunciator Configuration

## Conference Bridge Configuration

**Navigation:** Media Resources → Conference Bridge

Set Conference Bridge Type\* = Cisco Conference Bridge Software

Set Host Server = clus23pubsub. This is used for this example

Set Conference Bridge Name\* = CFB\_2

Set Description = CFB\_clus23pubsub. This is used in this example.

Set Device Pool\* = G711 pool

Conference Bridge Information	
Conference Bridge :	CFB_2 (CFB_clus23pubsub)
Registration:	Registered with Cisco Unified Communications Manager clus23pubsub
IPv4 Address:	10.80.13.2

Software Conference Bridge Info	
Conference Bridge Type*	Cisco Conference Bridge Software
Host Server	clus23pubsub
⚠ Device is not trusted	
Conference Bridge Name*	CFB_2
Description	CFB_clus23pubsub
Device Pool*	G711 pool
Common Device Configuration	< None >
Location*	Hub_None
Use Trusted Relay Point*	Default

Figure 8: Conference Bridge Configuration

## Media Termination Point Configuration

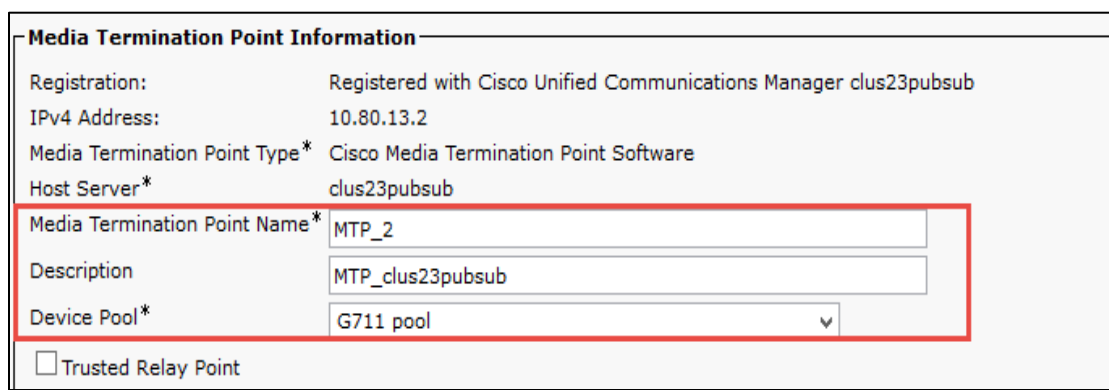
**NOTE\*\*\*** Cox certification is a SIP trunk certification between Cox Business eSBC, Cisco CUBE and Cisco CUCM only. In the event the customer requires DTMF between dissimilar endpoints an external MTP is required”

**Navigation:** Media Resource → Media Termination Point

Set Media Termination Point Name\* = MTP\_2

Set Description = MTP\_clus23pubsub is used for this example

Set Device pool\* = G711 pool



Media Termination Point Information	
Registration:	Registered with Cisco Unified Communications Manager clus23pubsub
IPv4 Address:	10.80.13.2
Media Termination Point Type*	Cisco Media Termination Point Software
Host Server*	clus23pubsub
Media Termination Point Name*	MTP_2
Description	MTP_clus23pubsub
Device Pool*	G711 pool
<input type="checkbox"/> Trusted Relay Point	

Figure 9: Media Termination Point Configuration

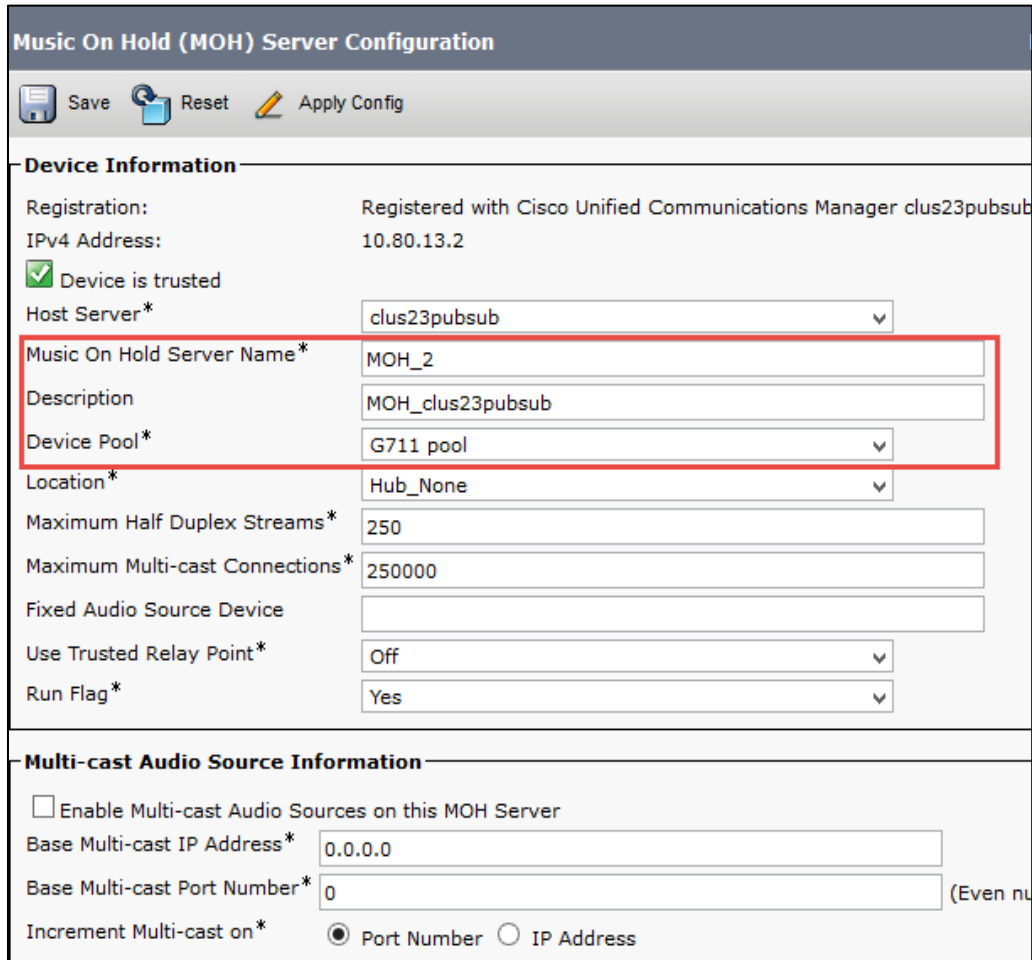
## Music on Hold Server Configuration

**Navigation:** Media Resources → Music on Hold Server

Set Music on Hold Server Name\* = MOH\_2

Set Description = MOH\_clus23pubsub. This is used for this example.

Set Device Pool\* = G711 pool



**Music On Hold (MOH) Server Configuration**

Save Reset Apply Config

**Device Information**

Registration: Registered with Cisco Unified Communications Manager clus23pubsub

IPv4 Address: 10.80.13.2

☒ Device is trusted

Host Server\*: clus23pubsub

**Music On Hold Server Name\***: MOH\_2

Description: MOH\_clus23pubsub

**Device Pool\***: G711 pool

Location\*: Hub\_None

Maximum Half Duplex Streams\*: 250

Maximum Multi-cast Connections\*: 250000

Fixed Audio Source Device:

Use Trusted Relay Point\*: Off

Run Flag\*: Yes

**Multi-cast Audio Source Information**

☐ Enable Multi-cast Audio Sources on this MOH Server

Base Multi-cast IP Address\*: 0.0.0.0

Base Multi-cast Port Number\*: 0 (Even nu

Increment Multi-cast on\*: ☒ Port Number ☐ IP Address

Figure 10: MOH Configuration

## Media Resource Group Configuration

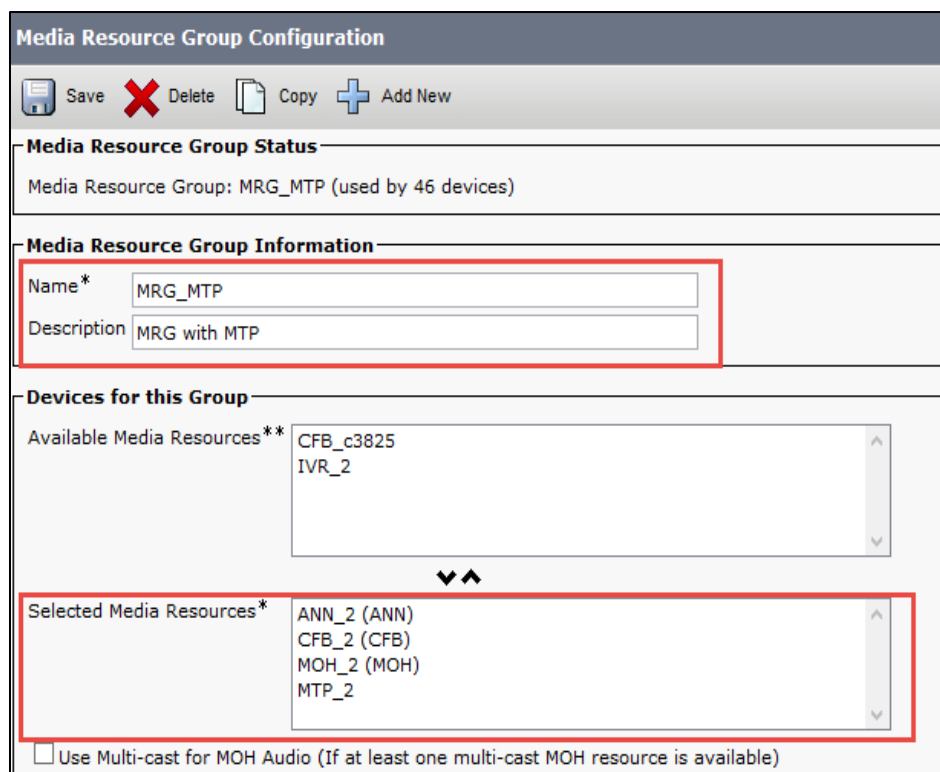
**Navigation Path:** Media Resources → Media Resources group

The Media Resource Group (MRG) contains media resources, such as Conference Bridge, Transcoder, MoH server and Annunciator. It will be assigned to a Media Resource Group List (MRGL) which is used to allocate media resources to groups of devices through Device Pools, or individually by configuring a valid MRGL at the device configuration page.




Set Name\*= MRG\_MTP. This is used for this example.

Set Description = MRG with MTP. This text is used to define this Media Resource Group List.

Set all Resources in the selected Media Resources Box



**Media Resource Group Configuration**

Save  Delete  Copy  Add New

**Media Resource Group Status**

Media Resource Group: MRG\_MTP (used by 46 devices)

**Media Resource Group Information**

Name\*

Description

**Devices for this Group**

Available Media Resources\*\*   
IVR\_2

Selected Media Resources\*   
CFB\_2 (CFB)  
MOH\_2 (MOH)  
MTP\_2

☐ Use Multi-cast for MOH Audio (If at least one multi-cast MOH resource is available)

Figure 11: Media Resource Group Configuration

## Media Resource Group List Configuration

**Navigation Path:** Media Resources → Media Resource Group List

Set Name = MRGL\_MTP

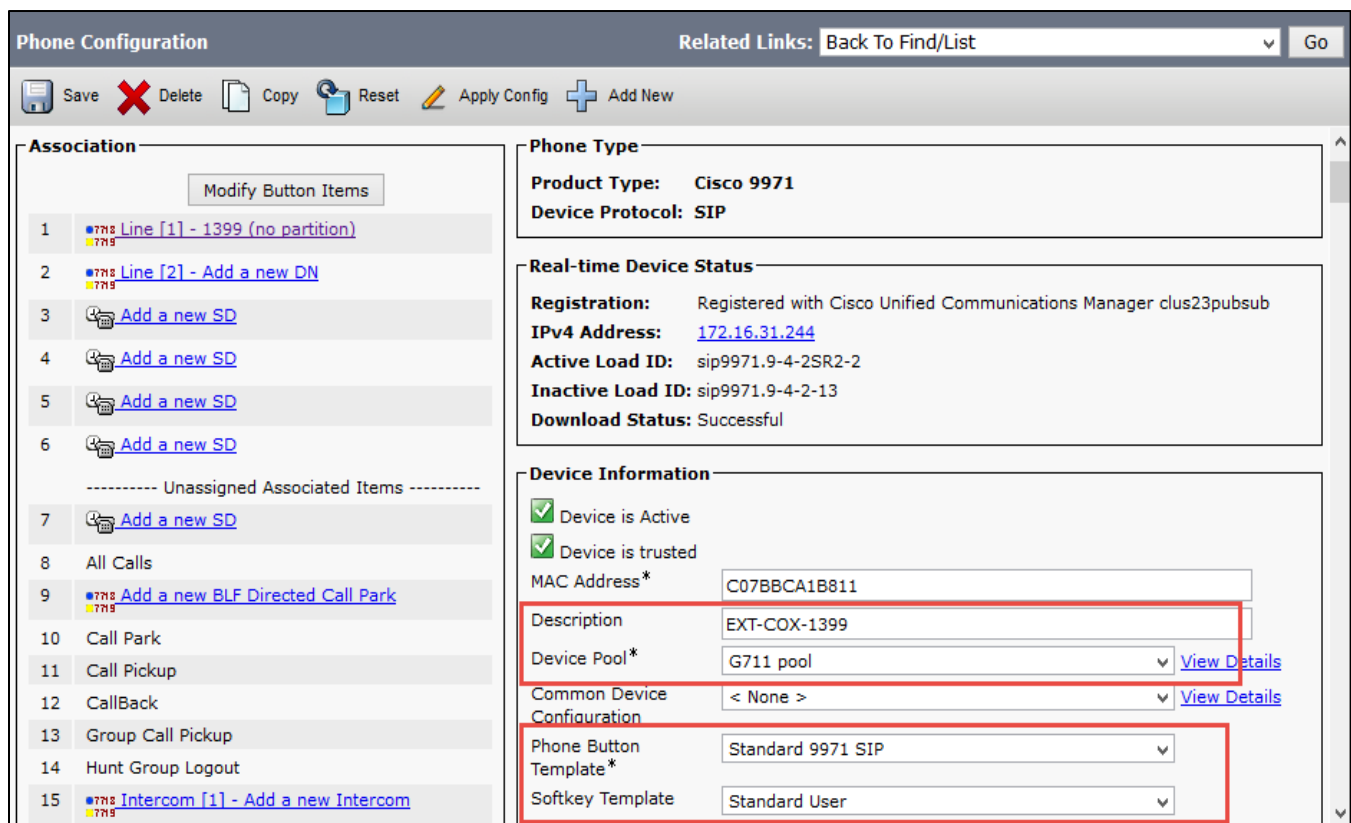
Set selected Media Resource Groups = MRG\_MTP

<b>Media Resource Group List Status</b>	
Media Resource Group List: MRGL_MTP (used by 46 devices)	
<b>Media Resource Group List Information</b>	
Name *	<input type="text" value="MRGL_MTP"/>
<b>Media Resource Groups for this List</b>	
Available Media Resource Groups	<div><div></div><div></div></div>
v v	
Selected Media Resource Groups	<div><div>MRG_MTP</div><div></div></div>

Figure 12: Media Resource Group List Configuration

## Cisco IP Phone 9971 SIP Configuration

Set MAC Address\* = the below mac is used in this example  
Set Description = EXT-COX-1399. This text is used to identify this Phone  
Set Device Pool\* = G711 pool. This is used in this example  
Set Phone Button Template\* = Standard 9971 SIP. This is used in this example  
Set Soft key Template = Standard User. This is used in this example  
Set Media Resource Group List = MRGL\_MTP. This is used in this example  
Set User Hold MOH Audio Source = 1-SampleAudioSource  
Set Network Hold MOH Audio Source = 1-SampleAudioSource



**Phone Configuration** Related Links: [Back To Find/List](#) [Go](#)

Save Delete Copy Reset Apply Config Add New

**Association**

Modify Button Items

- Line [1] - 1399 (no partition)
- Line [2] - Add a new DN
- Add a new SD
- Add a new SD
- Add a new SD
- Add a new SD
- Unassigned Associated Items -----
- Add a new SD
- All Calls
- Add a new BLF Directed Call Park
- Call Park
- Call Pickup
- CallBack
- Group Call Pickup
- Hunt Group Logout
- Intercom [1] - Add a new Intercom

**Phone Type**

Product Type: Cisco 9971  
Device Protocol: SIP

**Real-time Device Status**

Registration: Registered with Cisco Unified Communications Manager clus23pubsub  
IPv4 Address: 172.16.31.244  
Active Load ID: sip9971.9-4-2SR2-2  
Inactive Load ID: sip9971.9-4-2-13  
Download Status: Successful

**Device Information**

☒ Device is Active  
☒ Device is trusted  
MAC Address\*: C07B8CA1B811  
Description: EXT-COX-1399  
Device Pool\*: G711 pool [View Details](#)  
Common Device Configuration: < None > [View Details](#)  
Phone Button Template\*: Standard 9971 SIP  
Softkey Template: Standard User

Figure 13: Cisco IP Phone 9971 SIP Configuration

Common Phone Profile*	Standard Common Phone Profile	<a href="#">View Details</a>
Calling Search Space	< None >	
AAR Calling Search Space	< None >	
Media Resource Group List	MRGL_MTP	
User Hold MOH Audio Source	1-SampleAudioSource	
Network Hold MOH Audio Source	1-SampleAudioSource	
Location*	Hub_None	
AAR Group	< None >	
User Locale	< None >	
Network Locale	< None >	
Built In Bridge*	Default	
Privacy*	Default	
Device Mobility Mode*	Default	<a href="#">View Current Device Mobility Settings</a>
Owner	<input type="radio"/> User <input checked="" type="radio"/> Anonymous (Public/Shared Space)	
Owner User ID		
Phone Personalization*	Default	
Services Provisioning*	Default	
Phone Load Name		
Use Trusted Relay Point*	Default	
BLF Audible Alert Setting (Phone Idle)*	Default	
BLF Audible Alert Setting (Phone Busy)*	Default	
Always Use Prime Line*	Default	
Always Use Prime Line for Voice Message*	Default	
Geolocation	< None >	
Feature Control Policy	< None >	
<input type="checkbox"/> Ignore Presentation Indicators (internal calls only) <input checked="" type="checkbox"/> Allow Control of Device from CTI <input checked="" type="checkbox"/> Logged Into Hunt Group <input type="checkbox"/> Remote Device <input type="checkbox"/> Protected Device**** <input type="checkbox"/> Require off-premise location		

Figure 14: Cisco IP Phone 9971 SIP Configuration – Cont.



Expansion Module Information		
Module 1	< None >	
Module 1 Load Name		
Module 2	< None >	
Module 2 Load Name		
Module 3	< None >	
Module 3 Load Name		

External Data Locations Information (Leave blank to use default)		
Information		
Directory		
Messages		
Services		
Authentication Server		
Proxy Server		
Idle		
Idle Timer (seconds)	1	
Secure Authentication URL		
Secure Directory URL		
Secure Idle URL		
Secure Information URL		
Secure Messages URL		
Secure Services URL		

Extension Information		
<input type="checkbox"/> Enable Extension Mobility		
Log Out Profile	-- Use Current Device Settings --	
Log in Time	< None >	
Log out Time	< None >	

MLPP and Confidential Access Level Information		
MLPP Domain	< None >	
MLPP Indication*	Default	
MLPP Preemption*	Default	
Confidential Access Mode	< None >	
Confidential Access Level	< None >	

Do Not Disturb		
<input type="checkbox"/> Do Not Disturb		
DND Option*	Use Common Phone Profile Setting	
DND Incoming Call Alert	< None >	

Secure Shell Information		
Secure Shell User	administrator	
Secure Shell Password	.....	

Product Specific Configuration Layout		
	Parameter Value	Override Common Settings
<input type="checkbox"/> Disable Speakerphone		
<input type="checkbox"/> Disable Speakerphone and Headset		
PC Port *	Enabled	
Back USB Port*	Enabled	<input type="checkbox"/>
Side USB Port*	Enabled	<input type="checkbox"/>
Cisco Camera*	Disabled	<input type="checkbox"/>
Console Access*	Disabled	<input type="checkbox"/>

Figure 16: Cisco IP Phone 9971 SIP Configuration – Cont.

Video Capabilities*	Disabled	<input type="checkbox"/>
Enable/Disable USB Classes	Mass Storage	<input type="checkbox"/>
	Human Interface Device	
	Audio Class	
SDIO *	Disabled	<input type="checkbox"/>
Bluetooth *	Enabled	<input type="checkbox"/>
Wifi *	Enabled	<input type="checkbox"/>
Bluetooth Profiles*	Handsfree	<input type="checkbox"/>
	Human Interface Device	
Settings Access*	Enabled	<input type="checkbox"/>
Gratuitous ARP*	Disabled	
PC Voice VLAN Access*	Enabled	
Web Access*	Disabled	<input type="checkbox"/>
Show All Calls on Primary Line*	Disabled	
Days Display Not Active	Sunday	<input type="checkbox"/>
	Monday	
	Tuesday	
Display On Time	07:30	<input type="checkbox"/>
Display On Duration	10:30	<input type="checkbox"/>
Display rule Timeout	01:00	<input type="checkbox"/>
HTTPS Server*	http and https Enabled	<input type="checkbox"/>
Enable Power Save Plus	Sunday	<input type="checkbox"/>
	Monday	
	Tuesday	
Phone On Time	00:00	<input type="checkbox"/>
Phone Off Time	24:00	<input type="checkbox"/>
Phone Off Idle Timeout*	60	<input type="checkbox"/>
<input type="checkbox"/> Enable Audible Alert		<input type="checkbox"/>
EnergyWise Domain		<input type="checkbox"/>
EnergyWise Endpoint Security Secret		<input type="checkbox"/>
<input type="checkbox"/> Allow EnergyWise Overrides		<input type="checkbox"/>
Span to PC Port*	Disabled	
Logging Display*	Disabled	
Load Server		<input type="checkbox"/>
IPv6 Load Server		<input type="checkbox"/>
Recording Tone*	Disabled	
Recording Tone	100	

Figure 17: Cisco IP Phone 9971 SIP Configuration – Cont.

Recording Tone*	Disabled	
Recording Tone Local Volume*	100	
Recording Tone Remote Volume*	50	
Recording Tone Duration		
Display On When Incoming Call*	Enabled	<input type="checkbox"/>
RTCP*	Disabled	<input type="checkbox"/>
Log Server		<input type="checkbox"/>
IPv6 Log Server		<input type="checkbox"/>
Remote Log*	Disabled	<input type="checkbox"/>
Log Profile	Default Preset Telephony	<input type="checkbox"/>
Advertise G.722 and iSAC Codecs*	Use System Default	
Wideband Headset UI Control*	Enabled	
Wideband Headset*	Enabled	
Peer Firmware Sharing*	Enabled	<input type="checkbox"/>
Cisco Discovery	Enabled	<input type="checkbox"/>
Switch Port		
Cisco Discovery Protocol (CDP): PC Port*	Enabled	<input type="checkbox"/>
Link Layer Discovery Protocol - Media Endpoint Discover (LLDP-MED): Switch Port*	Enabled	<input type="checkbox"/>
Link Layer Discovery Protocol (LLDP): PC Port*	Enabled	<input type="checkbox"/>
LLDP Asset ID		
LLDP Power Priority*	Unknown	
802.1x Authentication*	User Controlled	<input type="checkbox"/>
FIPS Mode*	Disabled	<input type="checkbox"/>
Detect Unified CM Connection Failure*	Normal	<input type="checkbox"/>
Switch Port Remote Configuration*	Disabled	<input type="checkbox"/>
PC Port Remote Configuration*	Disabled	<input type="checkbox"/>
Automatic Port	Disabled	<input type="checkbox"/>

Figure 18: Cisco IP Phone 9971 SIP Configuration – Cont.

Automatic Port Synchronization *	Disabled	<input type="checkbox"/>
Power Negotiation *	Enabled	<input type="checkbox"/>
Restrict Data Rates *	Disabled	
SSH Access *	Disabled	<input type="checkbox"/>
Incoming Call Toast Timer *	5	<input type="checkbox"/>
Provide Dial Tone from Release Button *	Disabled	<input type="checkbox"/>
Hide Video By Default *	Disabled	<input type="checkbox"/>
Background Image		<input type="checkbox"/>
Simplified New Call UI *	Disabled	<input type="checkbox"/>
Enable VXC VPN for MAC		
VXC VPN Option *	Dual Tunnel	<input type="checkbox"/>
VXC Challenge *	Challenge	<input type="checkbox"/>
VXC-M Servers		<input type="checkbox"/>
Revert to All Calls *	Disabled	<input type="checkbox"/>
RTCP for Video *	Enabled	<input type="checkbox"/>
Record Call Log	Disabled	<input type="checkbox"/>
Record Call Log For Remote Private Calls *	Enabled	
Show Call History for Selected Line Only. *	Disabled	<input type="checkbox"/>
Actionable Incoming Call Alert *	Disabled	<input type="checkbox"/>
DF bit *	0	<input type="checkbox"/>
Default Line Filter		
Separate Audio and Video Mute *	Disabled	<input type="checkbox"/>
Softkey Control *	Feature Control Policy	<input type="checkbox"/>
Start Video Port		<input type="checkbox"/>
Stop Video Port		<input type="checkbox"/>
Lowest Alerting Line State Priority *	Disabled	<input type="checkbox"/>
TLS Resumption Timer *	3600	<input type="checkbox"/>
Audio EQ *	Default : Default	<input type="checkbox"/>

Figure 19: Cisco IP Phone 9971 SIP Configuration – Cont.



Set Directory Number\* = 1399. This is used in this example.  
Set Description = SIP-1399. This is used in this example.  
Set Alerting Name = SIP-1399. This is used in this example.  
Set ASCII Alerting Name = SIP-1399. This is used in this example.

<b>Directory Number Information</b>				
Directory Number*	1399	<input type="checkbox"/> Urgent Priority		
Route Partition	< None >			
Description	SIP-1399			
Alerting Name	SIP-1399			
ASCII Alerting Name	SIP-1399			
External Call Control Profile	< None >			
<input checked="" type="checkbox"/> Allow Control of Device from CTI				
Associated Devices	SEPC07BBCA1B811	<a href="#">Edit Device</a>		
		<a href="#">Edit Line Appearance</a>		
Dissociate Devices				
<b>Directory Number Settings</b>				
Voice Mail Profile	< None >	(Choose <None> to use system default)		
Calling Search Space	< None >			
BLF Presence Group*	Standard Presence group			
User Hold MOH Audio Source	1-SampleAudioSource			
Network Hold MOH Audio Source	1-SampleAudioSource			
Auto Answer*	Auto Answer Off			
<input type="checkbox"/> Reject Anonymous Calls				
<b>Enterprise Alternate Number</b>				
<a href="#">Add Enterprise Alternate Number</a>				
<b>+E.164 Alternate Number</b>				
<a href="#">Add +E.164 Alternate Number</a>				
<b>Directory URIs</b>				
Primary	URI	Partition	Advertise Globally via ILS	Remove
<input checked="" type="radio"/>		< None >	<input checked="" type="checkbox"/>	<a href="#">Remove</a>
<a href="#">Add Row</a>				
<b>PSTN Failover for Enterprise Alternate Number, +E.164 Alternate Number, and URI Dialing</b>				
Advertised Failover Number < None >				
<b>AAR Settings</b>				
Voice Mail	AAR Destination Mask	AAR Group		
AAR <input type="checkbox"/> or		< None >		
<input checked="" type="checkbox"/> Retain this destination in the call forwarding history				
<b>Call Forward and Call Pickup Settings</b>				
Voice Mail	Destination	Calling Search Space		
Calling Search Space Activation Policy		Use System Default		

Figure 20: Cisco IP Phone 9971 SIP Configuration – Cont.



Set Display (Caller ID) = SIP-1399. This is used in this example  
Set ASCII Display (Caller ID) = SIP-1399. This is used in this example  
Set Line Text Label = SIP-1399. This is used in this example  
Set External Phone Number Mask = 4026141399. This is used in this example

Forward All	<input type="checkbox"/> or	<input type="text"/>	< None >
Secondary Calling Search Space for Forward All			< None >
Forward Busy Internal	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward Busy External	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward No Answer Internal	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward No Answer External	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward No Coverage Internal	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward No Coverage External	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward on CTI Failure	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward Unregistered Internal	<input type="checkbox"/> or	<input type="text"/>	< None >
Forward Unregistered External	<input type="checkbox"/> or	<input type="text"/>	< None >
No Answer Ring Duration (seconds)		<input type="text"/>	
Call Pickup Group		< None >	
<b>Park Monitoring</b>			
	<b>Voice Mail</b>	<b>Destination</b>	<b>Calling Search Space</b>
Park Monitoring Forward No Retrieve Destination External	<input type="checkbox"/> or	<input type="text"/>	< None > A blank value means to call the parker's line.
Park Monitoring Forward No Retrieve Destination	<input type="checkbox"/> or	<input type="text"/>	< None > A blank value means to call the
Park Monitoring Reversion Timer		<input type="text"/>	A blank value will use value set in Park Monitoring Reversion Timer service parameter
<b>MLPP Alternate Party And Confidential Access Level Settings</b>			
Target (Destination)		<input type="text"/>	
MLPP Calling Search Space		< None >	
MLPP No Answer Ring Duration (seconds)		<input type="text"/>	
Confidential Access Mode		< None >	
Confidential Access Level		< None >	
Call Control Agent Profile		< None >	
<b>Line Settings for All Devices</b>			
Hold Reversion Ring Duration (seconds)		<input type="text"/>	Setting the Hold Reversion Ring Duration to zero will disable the feature
Hold Reversion Notification Interval (seconds)		<input type="text"/>	Setting the Hold Reversion Notification Interval to zero will disable the feature
Party Entrance Tone*		Default	
<b>Line 1 on Device SEPC07BBCA1B811</b>			
Display (Caller ID)		SIP-1399	Display text for a line appearance is intended for displaying text such as a name instead of a directory number for calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.
ASCII Display (Caller ID)		SIP-1399	
Line Text Label		SIP-1399	
External Phone Number Mask		4026141399	

Figure 21: Cisco IP Phone 9971 SIP Configuration – Cont.



Visual Message Waiting Indicator Policy*	Use System Policy	
Audible Message Waiting Indicator Policy*	Default	
Ring Setting (Phone Idle)*	Use System Default	
Ring Setting (Phone Active)	Use System Default	Applies to this line when any line on the phone has a call in progress.
Call Pickup Group Audio Alert Setting(Phone Idle)	Use System Default	
Call Pickup Group Audio Alert Setting(Phone Active)	Use System Default	
Recording Option*	Call Recording Disabled	
Recording Profile	< None >	
Recording Media Source*	Gateway Preferred	
Monitoring Calling Search Space	< None >	
<input checked="" type="checkbox"/> Log Missed Calls		

---

**Multiple Call/Call Waiting Settings on Device SEPC07BBCA1B811**

Note:The range to select the Max Number of calls is: 1-200

Maximum Number of Calls*	4	
Busy Trigger*	2	(Less than or equal to Max. Calls)

---

**Forwarded Call Information Display on Device SEPC07BBCA1B811**

<input type="checkbox"/> Caller Name
<input type="checkbox"/> Caller Number
<input type="checkbox"/> Redirected Number
<input type="checkbox"/> Dialed Number

Figure 22: Cisco IP Phone 9971 SIP Configuration – Cont.

## Cisco IP Phone 7971 SCCP Configuration

Set MAC Address\* = the below mac is used in this example  
Set Description = EXT-COX-1958. This text is used to identify this Phone  
Set Device Pool\* = G711 pool. This is used in this example  
Set Phone Button Template\* = Standard 7971 SCCP. This is used in this example  
Set Soft key Template = Standard User. This is used in this example  
Set Media Resource Group List = MRGL\_MTP. This is used in this example  
Set User Hold MOH Audio Source = 1-SampleAudioSource  
Set Network Hold MOH Audio Source = 1-SampleAudioSource

Association		Phone Type	
<div>Modify Button Items</div> <div> <div>1</div> <div>Line [1] - 1958 (no partition)</div> </div> <div> <div>2</div> <div>Line [2] - Add a new DN</div> </div> <div> <div>3</div> <div>Add a new SD</div> </div> <div> <div>4</div> <div>Add a new SD</div> </div> <div> <div>5</div> <div>Add a new SD</div> </div> <div> <div>6</div> <div>Add a new SD</div> </div> <div> <div>7</div> <div>Add a new SD</div> </div> <div> <div>8</div> <div>Add a new SD</div> </div> <div> <div>9</div> <div>Unassigned Associated Items -----</div> </div> <div> <div>10</div> <div>Add a new SD</div> </div> <div> <div>11</div> <div>Add a new SURF</div> </div> <div> <div>12</div> <div>Add a new BLF SD</div> </div> <div> <div>13</div> <div>Add a new BLF Directed Call Park</div> </div> <div> <div>14</div> <div>Call Back</div> </div>		<div> <div>Product Type:</div> <div>Cisco 7971</div> </div> <div> <div>Device Protocol:</div> <div>SCCP</div> </div>	
		<div>Real-time Device Status</div> <div> <div>Registration:</div> <div>Registered with Cisco Unified Communications Manager clus23pubsub</div> </div> <div> <div>IPv4 Address:</div> <div>172.16.31.119</div> </div> <div> <div>Active Load ID:</div> <div>SCCP70.9-4-2SR1-1S</div> </div> <div> <div>Download Status:</div> <div>None</div> </div>	
		<div>Device Information</div> <div> <div> <input checked="" type="checkbox"/> Device is Active </div> <div> <input checked="" type="checkbox"/> Device is trusted </div> <div> <div>MAC Address*</div> <div>001DA266A924</div> </div> <div> <div>Description</div> <div>EXT-COX-1958</div> </div> <div> <div>Device Pool*</div> <div>G711 pool</div> <div>View Details</div> </div> <div> <div>Common Device Configuration</div> <div>&lt; None &gt;</div> <div>View Details</div> </div> <div> <div>Phone Button Template*</div> <div>Standard 7971 SCCP</div> </div> <div> <div>Softkey Template</div> <div>Standard User</div> </div> <div> <div>Common Phone Profile*</div> <div>Standard Common Phone Profile</div> <div>View Details</div> </div> </div>	
<div>14</div> <div>Call Park</div>			

15

Call Pickup

16

Conference List

17

Conference

18

Do Not Disturb

19

End Call

20

Forward All

21

Group Call Pickup

22

Hold

23

Hunt Group Logout

24

Intercom [1] - Add a new Intercom

25

Malicious Call Identification

26

Meet Me Conference

27

Mobility

28

New Call

29

Other Pickup

30

Quality Reporting Tool

31

Redial

32

Remove Last Participant

33

Transfer

34

Video Mode

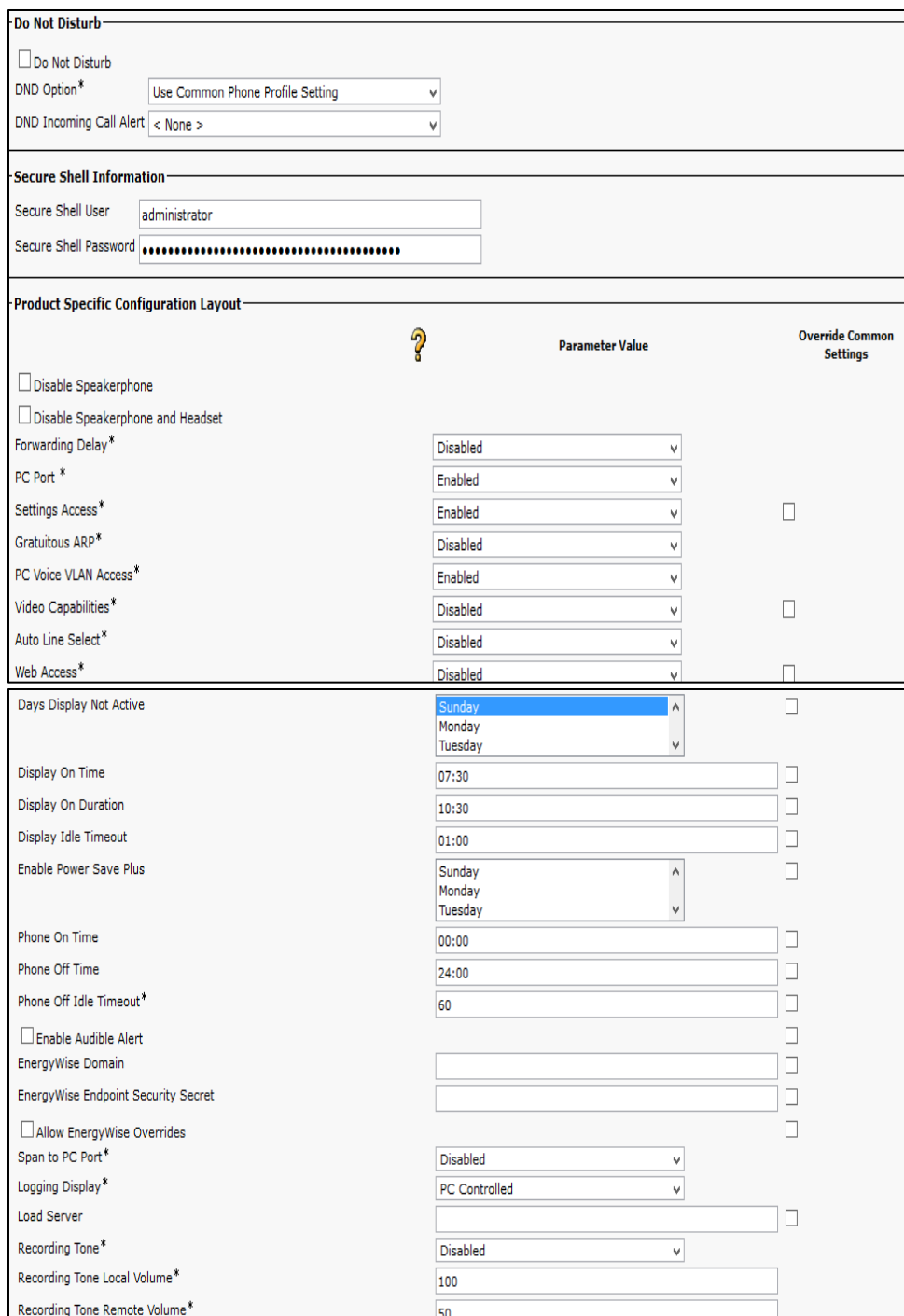
Figure 23: Cisco IP Phone 7971 SCCP Configuration

Use Trusted Relay Point*	Default
BLF Audible Alert Setting (Phone Idle)*	Default
BLF Audible Alert Setting (Phone Busy)*	Default
Always Use Prime Line*	Default
Always Use Prime Line for Voice Message*	Default
Geolocation	< None >
<input checked="" type="checkbox"/> Retry Video Call as Audio <input type="checkbox"/> Ignore Presentation Indicators (internal calls only) <input checked="" type="checkbox"/> Allow Control of Device from CTI <input checked="" type="checkbox"/> Logged Into Hunt Group <input type="checkbox"/> Remote Device <input type="checkbox"/> Protected Device***** <input type="checkbox"/> Hot line Device***** <input type="checkbox"/> Require off-premise location	
<b>Number Presentation Transformation</b>	
<b>Caller ID For Calls From This Phone</b>	
Calling Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS (Caller ID For Calls From This Phone)	
<b>Remote Number</b>	
Calling Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)	
<b>Protocol Specific Information</b>	
Packet Capture Mode*	None
Packet Capture Duration	0
BLF Presence Group*	Standard Presence group
Device Security Profile*	Cisco 7971 - Standard SCCP Non-Secure Profile
SUBSCRIBE Calling Search Space	< None >
<input type="checkbox"/> Unattended Port <input type="checkbox"/> Require DTMF Reception <input type="checkbox"/> RFC2833 Disabled	
<b>Certification Authority Proxy Function (CAPF) Information</b>	
Certificate Operation*	No Pending Operation
Authentication Mode*	By Null String
Authentication String	
<input type="button" value="Generate String"/>	
Key Order*	RSA Only
RSA Key Size (Bits)*	2048

Figure 24: Cisco IP Phone 7971 SCCP Configuration – Cont.

EC Key Size (Bits)	< None >
Operation Completes By	2016 1 22 12 (YYYY:MM:DD:HH)
Certificate Operation Status: None	
Note: Security Profile Contains Addition CAPF Settings.	
<b>Expansion Module Information</b>	
Module 1	< None >
Module 1 Load Name	
Module 2	< None >
Module 2 Load Name	
<b>External Data Locations Information (Leave blank to use default)</b>	
Information	
Directory	
Messages	
Services	
Authentication Server	
Proxy Server	
Idle	
Idle Timer (seconds)	
Secure Authentication URL	
Secure Directory URL	
Secure Idle URL	
Secure Information URL	
Secure Messages URL	
Secure Services URL	
<b>Extension Information</b>	
<input type="checkbox"/> Enable Extension Mobility	
Log Out Profile	-- Use Current Device Settings --
Log in Time	< None >
Log out Time	< None >
<b>MLPP and Confidential Access Level Information</b>	
MLPP Domain	< None >
MLPP Indication*	Default
MLPP Preemption*	Default
Confidential Access Mode	< None >
Confidential Access Level	< None >

Figure 25: Cisco IP Phone 7971 SCCP Configuration – Cont.



Recording Tone Duration	<input type="text"/>	
Display On When Incoming Call*	Disabled	<input type="checkbox"/>
RTCP*	Disabled	<input type="checkbox"/>
"more" Soft Key Timer	5	
Auto Call Select*	Enabled	
Log Server	<input type="text"/>	<input type="checkbox"/>
Advertise G.722 Codec*	Use System Default	
Wideband Headset UI Control*	Enabled	
Wideband Headset UI Control*	Enabled	
Wideband Headset*	Enabled	
Wideband Headset*	Use Phone Default	
Peer Firmware Sharing*	Enabled	<input type="checkbox"/>
Cisco Discovery Protocol (CDP): Switch Port*	Enabled	<input type="checkbox"/>
Cisco Discovery Protocol (CDP): PC Port*	Enabled	<input type="checkbox"/>
Link Layer Discovery Protocol - Media Endpoint Discover (LLDP-MED): Switch Port*	Enabled	<input type="checkbox"/>
Link Layer Discovery Protocol (LLDP): PC Port*	Enabled	<input type="checkbox"/>
LLDP Asset ID	<input type="text"/>	
LLDP Power Priority*	Unknown	
IPv6 Load Server	<input type="text"/>	<input type="checkbox"/>
IPv6 Log Server	<input type="text"/>	

802.1x Authentication*	User Controlled	<input type="checkbox"/>
Detect Unified CM Connection Failure*	Normal	<input type="checkbox"/>
Minimum Ring Volume*	0-Silent	
Headset Sidetone Level*	Use Phone Default	
HTTPS Server*	http and https Enabled	<input type="checkbox"/>
Enbloc Dialing*	Enabled	
Switch Port Remote Configuration*	Disabled	<input type="checkbox"/>
PC Port Remote Configuration*	Disabled	<input type="checkbox"/>
Automatic Port Synchronization*	Disabled	<input type="checkbox"/>
SSH Access*	Disabled	<input type="checkbox"/>
LOGIN Access*	Enabled	<input type="checkbox"/>
FIPS Mode*	Disabled	<input type="checkbox"/>
80-bit SRTP*	Disabled	<input type="checkbox"/>
Customer Support Use	<input type="text"/>	<input type="checkbox"/>

Figure 27: Cisco IP Phone 7971 SCCP Configuration – Cont.



Set Directory Number\* = 1958 is used in this example  
Set Description = SCCP-1958 is used in this example  
Set Alerting Name = SCCP-1958 is used in this example  
Set ASCII Alerting Name = SCCP-1958 is used in this example

Directory Number Information				
Directory Number*	1958		<input type="checkbox"/> Urgent Priority	
Route Partition	< None >			
Description	SCCP-1958			
Alerting Name	SCCP-1958			
ASCII Alerting Name	SCCP-1958			
External Call Control Profile	< None >			
<input checked="" type="checkbox"/> Allow Control of Device from CTI				
Associated Devices	SEP001DA266A924		<a href="#">Edit Device</a>	
			<a href="#">Edit Line Appearance</a>	
Dissociate Devices				

Directory Number Settings	
Voice Mail Profile	< None > (Choose <None> to use system default)
Calling Search Space	< None >

BLF Presence Group*	Standard Presence group
User Hold MOH Audio Source	< None >
Network Hold MOH Audio Source	< None >
Auto Answer*	Auto Answer Off
<input type="checkbox"/> Reject Anonymous Calls	

Enterprise Alternate Number	
<a href="#">Add Enterprise Alternate Number</a>	

+E.164 Alternate Number	
<a href="#">Add +E.164 Alternate Number</a>	

Directory URIs				
Primary	URI	Partition	Advertise Globally via ILS	Remove
<input checked="" type="radio"/>		< None >	<input checked="" type="checkbox"/>	<a href="#">-</a>
<a href="#">Add Row</a>				

PSTN Failover for Enterprise Alternate Number, +E.164 Alternate Number, and URI Dialing	
Advertised Failover Number	< None >

AAR Settings		
	Voice Mail	AAR Destination Mask
AAR	<input type="checkbox"/> or	< None >
<input checked="" type="checkbox"/> Retain this destination in the call forwarding history		

Figure 28: Cisco IP Phone 7971 SCCP Configuration – Cont.

Call Forward and Call Pickup Settings			
	Voice Mail	Destination	Calling Search Space
Calling Search Space Activation Policy			Use System Default
Forward All	<input type="checkbox"/> or		< None >
Secondary Calling Search Space for Forward All			< None >
Forward Busy Internal	<input type="checkbox"/> or		< None >
Forward Busy External	<input type="checkbox"/> or		< None >
Forward No Answer Internal	<input type="checkbox"/> or		< None >
Forward No Answer External	<input type="checkbox"/> or		< None >
Forward No Coverage Internal	<input type="checkbox"/> or		< None >
Forward No Coverage External	<input type="checkbox"/> or		< None >
Forward on CTI Failure	<input type="checkbox"/> or		< None >
Forward Unregistered Internal	<input type="checkbox"/> or		< None >
Forward Unregistered External	<input type="checkbox"/> or		< None >
No Answer Ring Duration (seconds)			
Call Pickup Group			< None >

Park Monitoring			
	Voice Mail	Destination	Calling Search Space
Park Monitoring Forward No Retrieve Destination External	<input type="checkbox"/> or		< None > A blank value means to call the parker's line.
Park Monitoring Forward No Retrieve Destination Internal	<input type="checkbox"/> or		< None > A blank value means to call the parker's line.
Park Monitoring Reversion Timer			A blank value will use value set in Park Monitoring Reversion Timer service parameter

MLPP Alternate Party And Confidential Access Level Settings	
Target (Destination)	
MLPP Calling Search Space	< None >
MLPP No Answer Ring Duration (seconds)	
Confidential Access Mode	< None >
Confidential Access Level	< None >
Call Control Agent Profile	< None >

Line Settings for All Devices	
Hold Reversion Ring Duration (seconds)	Setting the Hold Reversion Ring Duration to zero will disable the feature
Hold Reversion Notification Interval (seconds)	Setting the Hold Reversion Notification Interval to zero will disable the feature
Party Entrance Tone*	Default

Figure 29: Cisco IP Phone 7971 SCCP Configuration – Cont.



Set Display (Caller ID) = SCCP-1958 is used in this example  
Set ASCII Display (Caller ID) = SCCP-1958 is used in this example  
Set Line Text Label = SCCP-1958 is used in this example  
Set External Phone Number Mask = 4026141958 is used in this example

Line 1 on Device SEP001DA266A924	
Display (Caller ID)	SCCP-1958
Display text for a line appearance is intended for displaying text such as a name instead of a directory number for calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.	
ASCII Display (Caller ID)	SCCP-1958
Line Text Label	SCCP-1958
External Phone Number Mask	4026141958
Visual Message Waiting Indicator Policy*	Use System Policy
Audible Message Waiting Indicator Policy*	Default
Ring Setting (Phone Idle)*	Use System Default
Ring Setting (Phone Active)	Use System Default
Applies to this line when any line on the phone has a call in progress.	
Call Pickup Group Audio Alert Setting(Phone Idle)	Use System Default
Call Pickup Group Audio Alert Setting(Phone Active)	Use System Default
Recording Option*	Call Recording Disabled
Recording Profile	< None >
Recording Media Source*	Gateway Preferred
Monitoring Calling Search Space	< None >
<input checked="" type="checkbox"/> Log Missed Calls	
Multiple Call/Call Waiting Settings on Device SEP001DA266A924	
Note:The range to select the Max Number of calls is: 1-200	
Maximum Number of Calls*	4
Busy Trigger*	2
(Less than or equal to Max. Calls)	
Forwarded Call Information Display on Device SEP001DA266A924	
<input type="checkbox"/> Caller Name	
<input type="checkbox"/> Caller Number	
<input type="checkbox"/> Redirected Number	
<input type="checkbox"/> Dialed Number	

Figure 30: Cisco IP Phone 7971 SCCP Configuration – Cont.

## SIP Trunk Security Profile

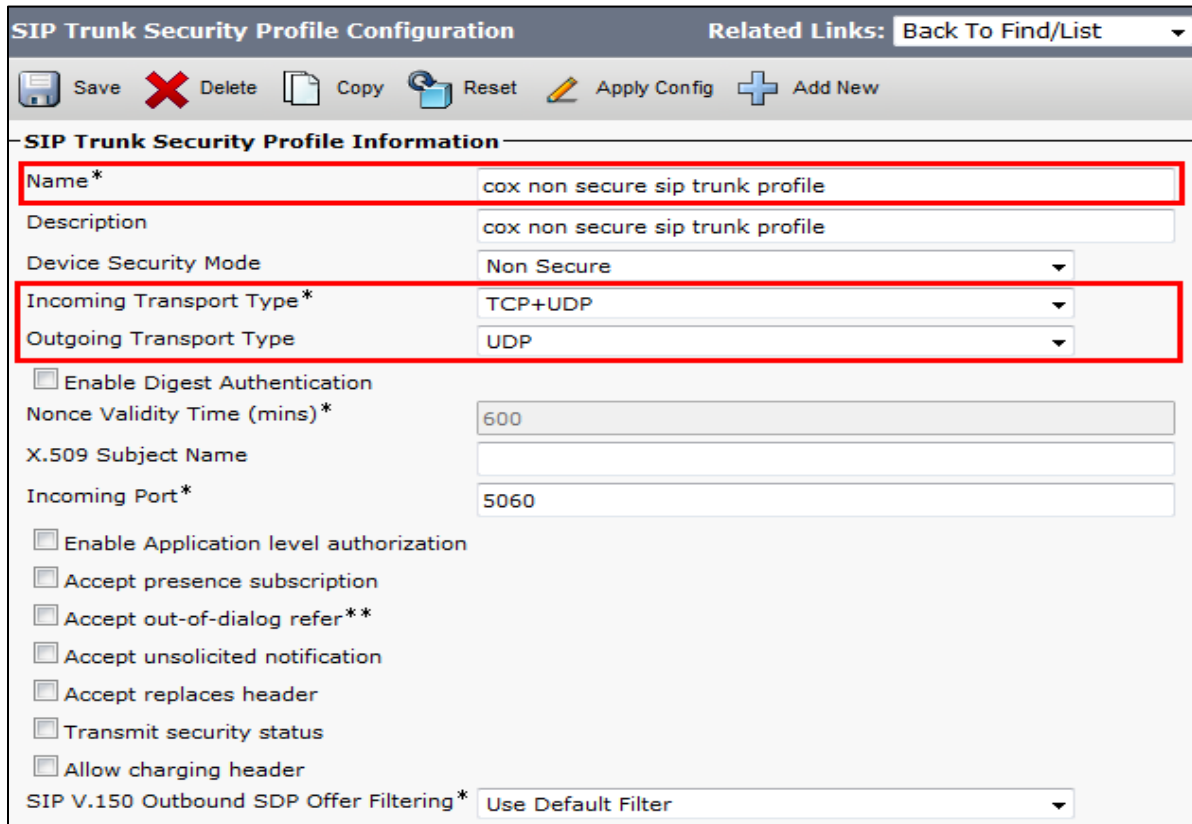
**Navigation:** System → Security → SIP Trunk Security Profile

Set Name\* =cox non secure sip trunk profile is used in this example

Set Description = Non Secure SIP Trunk Profile authenticated by null String is used in this example

Set Incoming Transport Type\* = TCP+UDP

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

Save Delete Copy Reset Apply Config Add New

**SIP Trunk Security Profile Information**

Name\* cox non secure sip trunk profile

Description cox non secure sip trunk profile

Device Security Mode Non Secure

Incoming Transport Type\* TCP+UDP

Outgoing Transport Type UDP

☐ Enable Digest Authentication

Nonce Validity Time (mins)\* 600

X.509 Subject Name

Incoming Port\* 5060

☐ Enable Application level authorization

☐ Accept presence subscription

☐ Accept out-of-dialog refer\*\*

☐ Accept unsolicited notification

☐ Accept replaces header

☐ Transmit security status

☐ Allow charging header

SIP V.150 Outbound SDP Offer Filtering\* Use Default Filter

Figure 31: SIP Trunk Security Profile

## SIP Profile

**Navigation:** Device → Device Settings → SIP Profile

Set SIP profile Name \* = cox\_sip\_profile. This is used for this example

Set SIP Rel1xx Options\* = Send PRACK if 1xx contains SDP

**NOTE\*** = Some PSTN network call prompters utilize early-media cut-through to offer menu options to the caller (DTMF select menu) before the call is connected. In order for Cisco UCM/Cisco UBE solution to achieve successful early-media cut-through, the Cisco UCM to Cisco UBE call leg must be enabled with SIP PRACK. To enable SIP PRACK on the Cisco UCM, the SIP Profile “SIP Rel1XX Options” setting must be set to “Send PRACK”.

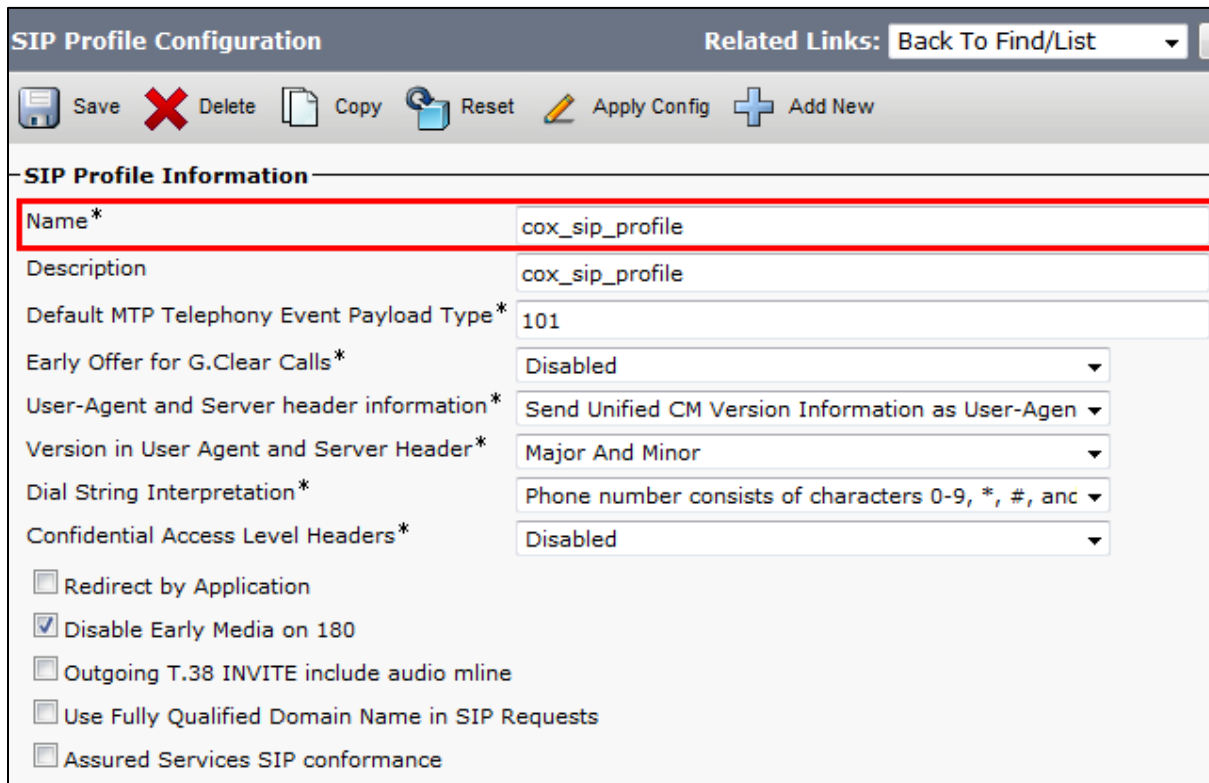


Figure 32: SIP Profile Configuration

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 33: SIP Profile Configuration – 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	

---

**Normalization Script**

Normalization Script < None >

☐ Enable Trace

	Parameter Name	Parameter Value
1		

---

**Incoming Requests FROM URI Settings**

Caller ID DN

Caller Name

---

**Trunk Specific Configuration**

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)

☐ Enable ANAT  
☐ Deliver Conference Bridge Identifier  
☐ Allow Passthrough of Configured Line Device Caller Information  
☐ Reject Anonymous Incoming Calls  
☐ Reject Anonymous Outgoing Calls  
☐ Send ILS Learned Destination Route String

Figure 34: SIP Profile Configuration – Cont.

SIP OPTIONS Ping	
<input checked="" type="checkbox"/> 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	
<input checked="" type="checkbox"/> Send send-receive SDP in mid-call INVITE	
<input type="checkbox"/> Allow Presentation Sharing using BFCP	
<input type="checkbox"/> Allow iX Application Media	
<input type="checkbox"/> Allow multiple codecs in answer SDP	

Figure 35: SIP Profile Configuration – Cont.



## SIP Trunk to Cisco UBE Configuration

**Navigation:** Device → Trunk

Set Device Name\* = cox\_trunk. This is used for this example

Set Description = cox\_trunk. This is used for this example

Set Media Resource Group List = MRGL\_MTP




Trunks (1 - 3 of 3)												Rows per Page 50	
Find Trunks where <div>Device Name</div> contains <div></div> <div>Find</div> <div>Clear Filter</div> <div>+</div> <div>-</div>													
Select item or enter search text													
<input type="checkbox"/>		Name ^	Description	Calling Search Space	Device Pool	Route Pattern	Partition	Route Group	Priority	Trunk Type	SIP Trunk Status	SIP Trunk Duration	SIP Trunk Security Profile
<input type="checkbox"/>		<a href="#">COX_UNITY</a>	COX_UNITY		<a href="#">G711 pool</a>	<a href="#">3000</a>				SIP Trunk	Full Service	Time In Full Service: 4 days 0 hour 34 minutes	<a href="#">cox_non_secure_sip_trunk_profile</a>
<input type="checkbox"/>		<a href="#">cox_trunk</a>	cox_trunk		<a href="#">G711 pool</a>	<a href="#">9.@</a>				SIP Trunk	None	None	<a href="#">cox_non_secure_sip_trunk_profile</a>
<input type="checkbox"/>		<a href="#">trunk to fax</a>			<a href="#">G711 pool</a>	<a href="#">3229</a>				SIP Trunk	Full Service	Time In Full Service: 2 days 2 hours 50 minutes	<a href="#">cox_non_secure_sip_trunk_profile</a>

Figure 36: SIP Trunk List

<b>SIP Trunk Status</b>	
<b>Service Status:</b> Full Service	
<b>Duration:</b> Time In Full Service: 0 day 0 hour 44 minutes	
<b>Device Information</b>	
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
<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	
<b>Intercompany Media Engine (IME)</b>	
E.164 Transformation Profile	< None >

Figure 37: SIP Trunk to CUBE

Set Significant Digits\* = 4. 4 digits Extension for all CPE phones

**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 38: SIP Trunk to CUBE Configuration – Cont.



Set Destination Address = 10.80.13.20 .IP address of the CUBE  
Set SIP Trunk Security Profile = Cox Non Secure SIP Trunk Profile. SIP Trunk Security Profile configured earlier  
Set SIP Profile = Cox SIP Profile. SIP Profile configured earlier  
DTMF Signaling Method = RFC 2833. RFC 2833 is supported for DTMF transport to/from Cox

Outbound Calls	
Called Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Called Party Transformation CSS	
Calling Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS	
Calling Party Selection*	Originator
Calling Line ID Presentation*	Default
Calling Name Presentation*	Default
Calling and Connected Party Info Format*	Deliver DN only in connected party
<input type="checkbox"/> Redirecting Diversion Header Delivery - Outbound	
Redirecting Party Transformation CSS	< None >
<input checked="" type="checkbox"/> Use Device Pool Redirecting Party Transformation CSS	
<b>Caller Information</b>	
Caller ID DN	
Caller Name	
<input type="checkbox"/> Maintain Original Caller ID DN and Caller Name in Identity Headers	

SIP Information	
<b>Destination</b>	
<input type="checkbox"/> Destination Address is an SRV	
<b>Destination Address</b>	<b>Destination Address IPv6</b>
1* 10.80.13.20	
MTP Preferred Originating Codec*	711ulaw
BLF Presence Group*	Standard Presence group
SIP Trunk Security Profile*	cox 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*	cox_sip_profile <a href="#">View Details</a>
DTMF Signaling Method*	RFC 2833

Figure 39: SIP Trunk to CUBE Configuration – Cont.

<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>Recording Information</b>							
<input checked="" type="radio"/> None <input type="radio"/> This trunk connects to a recording-enabled gateway <input type="radio"/> This trunk connects to other clusters with recording-enabled gateways							
<b>Geolocation Configuration</b>							
Geolocation	< None >						
Geolocation Filter	< None >						
<input type="checkbox"/> Send Geolocation Information							

Figure 40: SIP Trunk to CUBE Configuration – Cont.

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

## SIP Trunk to Fax Gateway Configuration

**Navigation:** Device → Trunk

Set Device Name\* = trunk\_to\_fax. This is used for this example

Set Media Resource Group List = MRGL\_MTP

SIP Trunk Status	
<b>Service Status:</b>	Full Service
<b>Duration:</b>	Time In Full Service: 2 days 3 hours 4 minutes
Device Information	
Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	trunk_to_fax
Description	trunk_to_fax
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
<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* <span>When using both sRTP and TLS</span> Route Class Signaling Enabled* <span>Default</span> Use Trusted Relay Point* <span>Default</span> <input checked="" type="checkbox"/> PSTN Access <input type="checkbox"/> Run On All Active Unified CM Nodes	
Intercompany Media Engine (IME)	
E.164 Transformation Profile	< None >

Figure 41: SIP trunk to FAX Gateway Configuration

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

Figure 42: SIP Trunk to FAX Gateway Configuration – Cont.



Set Destination Address = 172.16.31.50 .IP address of the FAX Gateway  
Set SIP Trunk Security Profile = Cox Non Secure SIP Trunk Profile. SIP Trunk Security Profile configured earlier  
Set SIP Profile = Cox SIP Profile. SIP Profile configured earlier

**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  
Calling and Connected Party Info Format\* Deliver DN only in connected party  
☐ Redirecting Diversion Header Delivery - Outbound  
Redirecting Party Transformation CSS < None >  
☒ 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 *	172.16.31.50		5060

MTP Preferred Originating Codec\* 711ulaw  
BLF Presence Group\* Standard Presence group  
SIP Trunk Security Profile\* cox 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\* cox\_sip\_profile [View Details](#)  
DTMF Signaling Method\* No Preference

**Normalization Script**  
Normalization Script < None >  
☐ Enable Trace  

	Parameter Name	Parameter Value		
1			+	-

Figure 43: SIP Trunk to FAX Gateway Configuration – Cont.

<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>Recording Information</b>							
<input checked="" type="radio"/> None <input type="radio"/> This trunk connects to a recording-enabled gateway <input type="radio"/> This trunk connects to other clusters with recording-enabled gateways							
<b>Geolocation Configuration</b>							
Geolocation	< None >						
Geolocation Filter	< None >						
<input type="checkbox"/> Send Geolocation Information							

Figure 44: SIP Trunk to Fax Gateway Configuration – Cont.

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

## SIP Trunk to CUC

**Navigation:** Device → Trunk

Set Device Name\* = COX\_UNITY. This is used for this example

Set Device Pool\* = G711 pool. This is used for this example

SIP Trunk Status	
<b>Service Status:</b> Full Service	
<b>Duration:</b>	Time In Full Service: 4 days 1 hour 44 minutes
Device Information	
Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	COX_UNITY
Description	COX_UNITY
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
<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 checked="" 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* <span>When using both sRTP and TLS</span> Route Class Signaling Enabled* <span>Default</span> Use Trusted Relay Point* <span>Default</span> <input checked="" type="checkbox"/> PSTN Access <input type="checkbox"/> Run On All Active Unified CM Nodes	
Intercompany Media Engine (IME)	
E.164 Transformation Profile	< None >

Figure 45: SIP Trunk to Unity

**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\* 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"/>

**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 46: SIP Trunk to Unity – Cont.



Set Destination Address = 10.80.13.4 .IP address of the Unity  
Set SIP Trunk Security Profile = Cox Non Secure SIP Trunk Profile. SIP Trunk Security Profile configured earlier  
Set SIP Profile = Cox SIP Profile. SIP Profile configured earlier

Outbound Calls		
Called Party Transformation CSS	< None >	
<input checked="" type="checkbox"/> Use Device Pool Called Party Transformation CSS		
Calling Party Transformation CSS	< None >	
<input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS		
Calling Party Selection*	Originator	
Calling Line ID Presentation*	Default	
Calling Name Presentation*	Default	
Calling and Connected Party Info Format*	Deliver DN only in connected party	
<input type="checkbox"/> Redirecting Diversion Header Delivery - Outbound		
Redirecting Party Transformation CSS	< None >	
<input checked="" type="checkbox"/> Use Device Pool Redirecting Party Transformation CSS		

Caller Information	
Caller ID DN	
Caller Name	
<input type="checkbox"/> Maintain Original Caller ID DN and Caller Name in Identity Headers	

SIP Information		
<input type="checkbox"/> Destination Address is an SRV		
Destination Address	Destination Address IPv6	Destination Port
1 * 10.80.13.4		5060

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

Figure 47: SIP Trunk to Unity – Cont.

<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>Recording Information</b>							
<input checked="" type="radio"/> None <input type="radio"/> This trunk connects to a recording-enabled gateway <input type="radio"/> This trunk connects to other clusters with recording-enabled gateways							
<b>Geolocation Configuration</b>							
Geolocation	< None >						
Geolocation Filter	< None >						
<input type="checkbox"/> Send Geolocation Information							

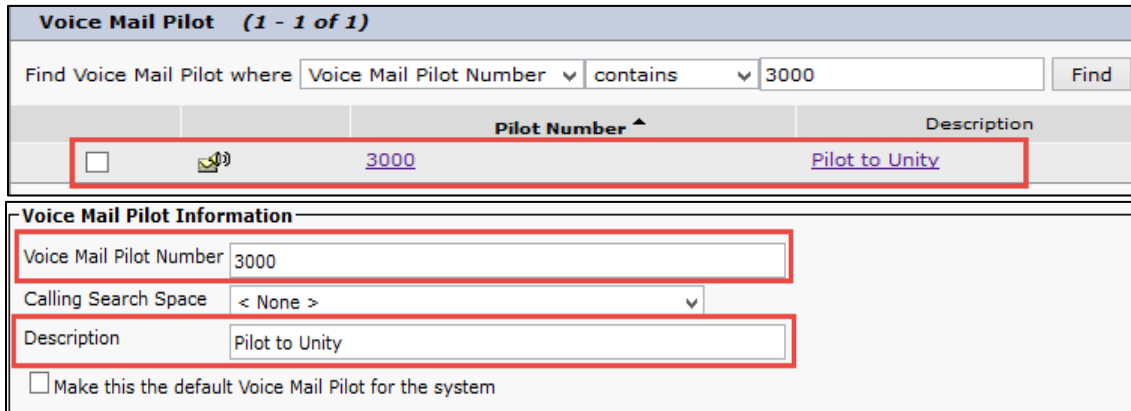
Figure 48: SIP Trunk to Unity – Cont.

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

## Voicemail Pilot Configuration

**Navigation:** Advanced Features → Voice Mail → Voice Mail Pilot

Set Voice Mail Pilot Number = 3000. This is used for this example  
Set Description = Pilot to Unity. This is used for this example.



Voice Mail Pilot (1 - 1 of 1)	
Find Voice Mail Pilot where Voice Mail Pilot Number contains 3000 Find	
Pilot Number	Description
3000	Pilot to Unity

**Voice Mail Pilot Information**

Voice Mail Pilot Number 3000

Calling Search Space < None >

Description Pilot to Unity

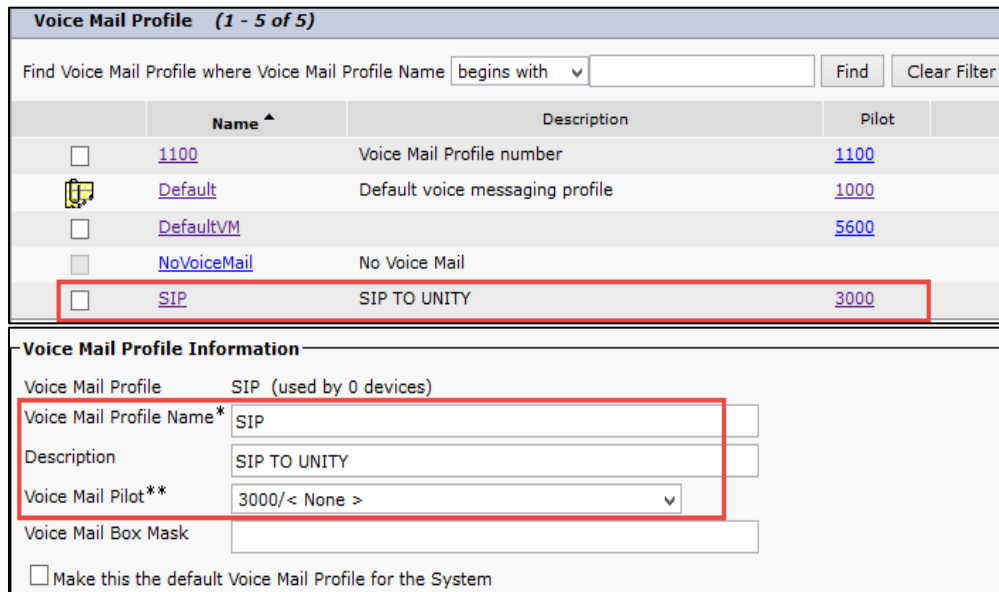
☐ Make this the default Voice Mail Pilot for the system

Figure 49: Voicemail Pilot Configuration

## Voicemail Profile Configuration

**Navigation:** Advanced Features → Voice Mail → Voice Mail Profiles

Set Voice Mail Profile Name\* = SIP. This is used for this example  
Set Description = SIP TO UNITY. This is used for this example  
Set Voice Mail Pilot\*\* = 3000. This is used for this example



Voice Mail Profile (1 - 5 of 5)			
Find Voice Mail Profile where Voice Mail Profile Name begins with Find Clear Filter			
Name	Description	Pilot	
1100	Voice Mail Profile number	1100	<input type="checkbox"/>
Default	Default voice messaging profile	1000	<input checked="" type="checkbox"/>
DefaultVM		5600	<input type="checkbox"/>
NoVoiceMail	No Voice Mail		<input type="checkbox"/>
SIP	SIP TO UNITY	3000	<input type="checkbox"/>

**Voice Mail Profile Information**

Voice Mail Profile SIP (used by 0 devices)

Voice Mail Profile Name\* SIP

Description SIP TO UNITY

Voice Mail Pilot\*\* 3000/< None >

Voice Mail Box Mask

☐ Make this the default Voice Mail Profile for the System

Figure 50: Voicemail Profile Configuration

## Route Pattern Configuration

**Navigation:** Call Routing → Route/Hunt → Route Pattern

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 3229 will send to fax gateway
- 3000 is the Pilot Number for voicemail to Unity Connection

Route Patterns (1 - 3 of 3)					
Find Route Patterns where		Description	contains	cox	Find Clear Filter
<input type="checkbox"/>	Pattern ^	Description	Partition	Route Filter	Associated De
<input type="checkbox"/>	3229	COX-FAX test			<a href="#">trunk to fax</a>
<input type="checkbox"/>	3000	COX_UNITY			<a href="#">COX_UNITY</a>
<input type="checkbox"/>	9.@	To Cox			<a href="#">cox_trunk</a>

Figure 51: Route Patterns

Set Route Pattern\* =9.@ for Voice call ,3000 for Unity and 3229 for the Fax call. Specify appropriate Route Pattern

Set Gateway/Route List\* = cox\_trunk. SIP Trunk name configured earlier

Discard Digits = PreDot. Specifies how to modify digit before they are sending to Cox ESBC

Pattern Definition	
Route Pattern *	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
Gateway/Route List *	cox_trunk (Edit)
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	

Figure 52: Route Patterns for Voice

Authorization Level*	0
<input type="checkbox"/> Require Client Matter Code	
<b>Calling Party Transformations</b>	
<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
<b>Connected Party Transformations</b>	
Connected Line ID Presentation*	Default
Connected Name Presentation*	Default
<b>Called Party Transformations</b>	
Discard Digits	PreDot
Called Party Transform Mask	
Prefix Digits (Outgoing Calls)	
Called Party Number Type*	Cisco CallManager
Called Party Numbering Plan*	Cisco CallManager
<b>ISDN Network-Specific Facilities Information Element</b>	
Network Service Protocol	-- Not Selected --
Carrier Identification Code	
Network Service	Service Parameter Name
-- Not Selected --	< Not Exist >

Figure 53: Route Patterns for Voice – Cont.

Pattern Definition	
Route Pattern*	3000
Route Partition	< None >
Description	COX_UNITY
Numbering Plan	-- Not Selected --
Route Filter	< None >
MLPP Precedence*	Default
<input type="checkbox"/> Apply Call Blocking Percentage	
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Gateway/Route List*	COX_UNITY
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	

[\(Edit\)](#)

Figure 54: 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"/> <input type="button" value="v"/> Calling Name Presentation* <input type="text" value="Default"/> <input type="button" value="v"/> Calling Party Number Type* <input type="text" value="Cisco CallManager"/> <input type="button" value="v"/> Calling Party Numbering Plan* <input type="text" value="Cisco CallManager"/> <input type="button" value="v"/>					
<b>Connected Party Transformations</b>					
Connected Line ID Presentation* <input type="text" value="Default"/> <input type="button" value="v"/> Connected Name Presentation* <input type="text" value="Default"/> <input type="button" value="v"/>					
<b>Called Party Transformations</b>					
Discard Digits <input type="text" value="&lt; None &gt;"/> <input type="button" value="v"/> Called Party Transform Mask <input type="text"/> Prefix Digits (Outgoing Calls) <input type="text"/> Called Party Number Type* <input type="text" value="Cisco CallManager"/> <input type="button" value="v"/> Called Party Numbering Plan* <input type="text" value="Cisco CallManager"/> <input type="button" value="v"/>					
<b>ISDN Network-Specific Facilities Information Element</b>					
Network Service Protocol <input type="text" value="-- Not Selected --"/> <input type="button" value="v"/> Carrier Identification Code <input type="text"/> <table border="0"> <tr> <td>Network Service</td> <td>Service Parameter Name</td> </tr> <tr> <td><input type="text" value="-- Not Selected --"/> <input type="button" value="v"/></td> <td><input type="text" value="&lt; Not Exist &gt;"/></td> </tr> </table>		Network Service	Service Parameter Name	<input type="text" value="-- Not Selected --"/> <input type="button" value="v"/>	<input type="text" value="&lt; Not Exist &gt;"/>
Network Service	Service Parameter Name				
<input type="text" value="-- Not Selected --"/> <input type="button" value="v"/>	<input type="text" value="&lt; Not Exist &gt;"/>				

Figure 55: Route Patterns for Unity – Cont.

Pattern Definition			
Route Pattern*	3229		
Route Partition	< None >		
Description	COX-FAX test		
Numbering Plan	-- Not Selected --		
Route Filter	< None >		
MLPP Precedence*	Default		
<input type="checkbox"/> Apply Call Blocking Percentage			
Resource Priority Namespace Network Domain	< None >		
Route Class*	Default		
Gateway/Route List*	trunk_to_fax		
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 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		
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	Service Parameter Name	Service Parameter Value	
-- Not Selected --	< Not Exist >		

Figure 56: Route Patterns for Fax



## FAX Gateway Configuration

```
cme.in.tekvizion.com#sh running config
!
version 15.1
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname cme.in.tekvizion.com
!
boot-start-marker
boot-end-marker
!
aaa new-model
!
!
aaa authentication login local_auth local
!
aaa session-id common
clock timezone IST 5 30
network-clock-participate wic 2
network-clock-participate wic 3
!
dot11 syslog
ip source-route
!
!
ip cef
```



```
!  
ip host Clus1-862-Pub 172.16.26.2  
no ipv6 cef  
multilink bundle-name authenticated  
!  
isdn switch-type primary-qsig  
voice rtp send-recv  
!  
voice service pots  
!  
voice service voip  
no ip address trusted authenticate  
allow-connections sip to sip  
no supplementary-service sip handle-replaces  
redirect ip2ip  
fax protocol pass-through g711ulaw  
sip  
g729 annexb-all  
!  
voice class codec 3  
codec preference 1 g711ulaw  
!  
voice-card 0  
!  
crypto pki token default removal timeout 0  
!  
!  
!  
!
```



```
license udi pid CISCO2851 sn FHK7867F4LY
```

```
username cisco password 0
```

```
!
```

```
interface GigabitEthernet0/0
```

```
ip address 172.16.31.50 255.255.255.0
```

```
duplex auto
```

```
speed auto
```

```
!
```

```
interface GigabitEthernet0/1
```

```
no ip address
```

```
ip nat outside
```

```
ip virtual-reassembly in
```

```
shutdown
```

```
duplex auto
```

```
speed auto
```

```
ip forward-protocol nd
```

```
!
```

```
ip http server
```

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

```
!
```

```
ip route 0.0.0.0 0.0.0.0 172.16.31.1
```

```
!
```

```
access-list 1 permit 172.16.31.0 0.0.0.255
```

```
!
```

```
snmp-server community public RO
```

```
snmp-server location Chennai
```

```
!
```

```
control-plane
```



!

voice-port 0/1/1

cptone IN

station-id number

caller-id enable

!

dial-peer voice 777 pots

huntstop

destination-pattern 3229

port 0/1/1

forward-digits all

!

dial-peer voice 9224 voip

description CUCM to Gateway

session protocol sipv2

session transport udp

incoming called-number 3229

voice-class codec 3

dtmf-relay rtp-nte

fax-relay ecm disable

fax rate disable

fax protocol pass-through g711ulaw

no vad

!

dial-peer voice 92240 voip

description Gateway to CUCM

destination-pattern 9T

session protocol sipv2

session target ipv4:10.80.13.2



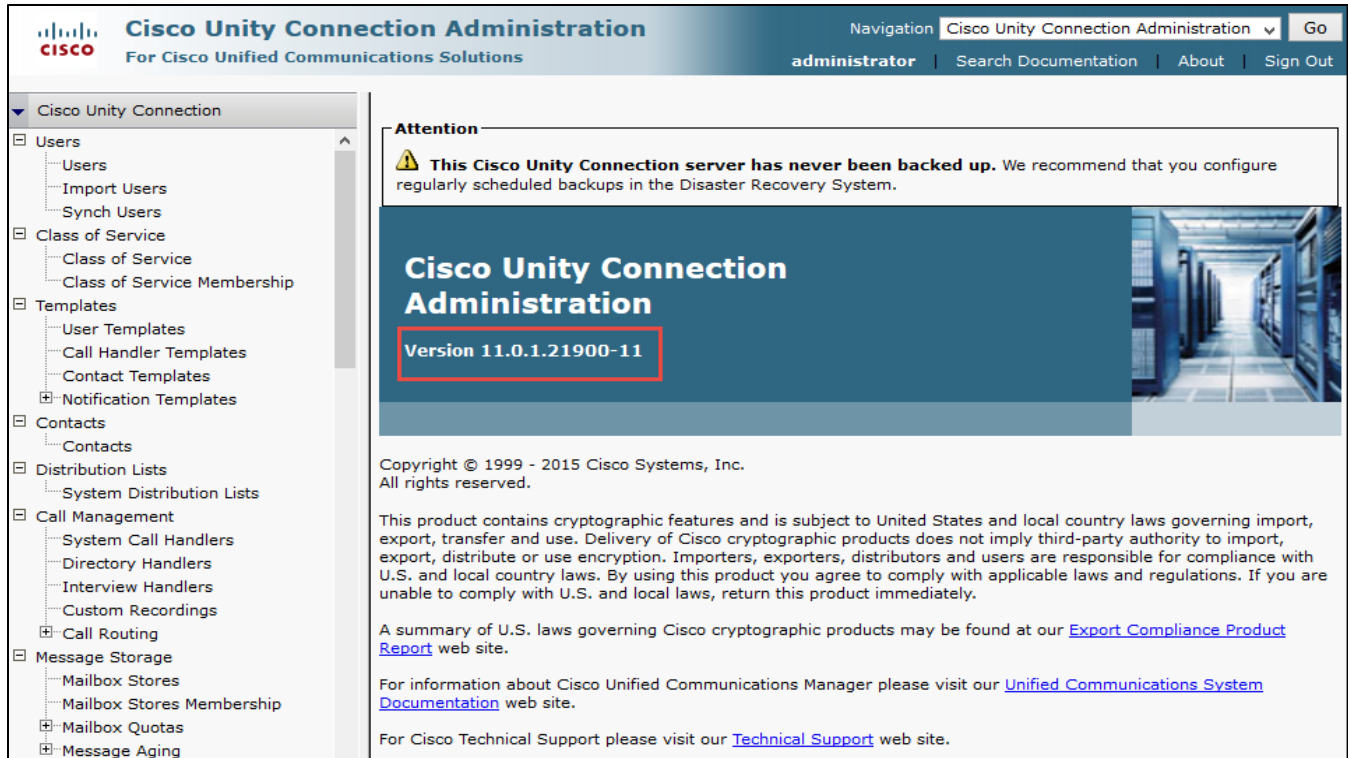
```
session transport udp
voice-class codec 3
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax protocol pass-through g711ulaw
no vad
!
!
gateway
timer receive-rtp 1200
!
sip-ua
retry register 5
timers connection aging 30
timers update 1000
no timers hold
timers register 1000
!
!
line con 0
line aux 0
line 66
no activation-character
no exec
transport preferred none
transport input all
transport output pad telnet rlogin lapb-ta mop udptn v120 ssh
line 194
```



```
no activation-character
no exec
transport preferred none
transport input all
transport output all
line vty 0 4
session-timeout 180
exec-timeout 0 0
password
login authentication local_auth
transport input all
!
scheduler allocate 20000 1000
ntp server
end
```

## Cisco UCM SIP Integration with Cisco Unity Connection (CUC)

### CUC Version



**Cisco Unity Connection Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unity Connection Administration Go

administrator | Search Documentation | About | Sign Out

**Attention**

**⚠ This Cisco Unity Connection server has never been backed up.** We recommend that you configure regularly scheduled backups in the Disaster Recovery System.

**Cisco Unity Connection Administration**

**Version 11.0.1.21900-11**

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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 our [Export Compliance Product Report](#) web site.

For information about Cisco Unified Communications Manager please visit our [Unified Communications System Documentation](#) web site.

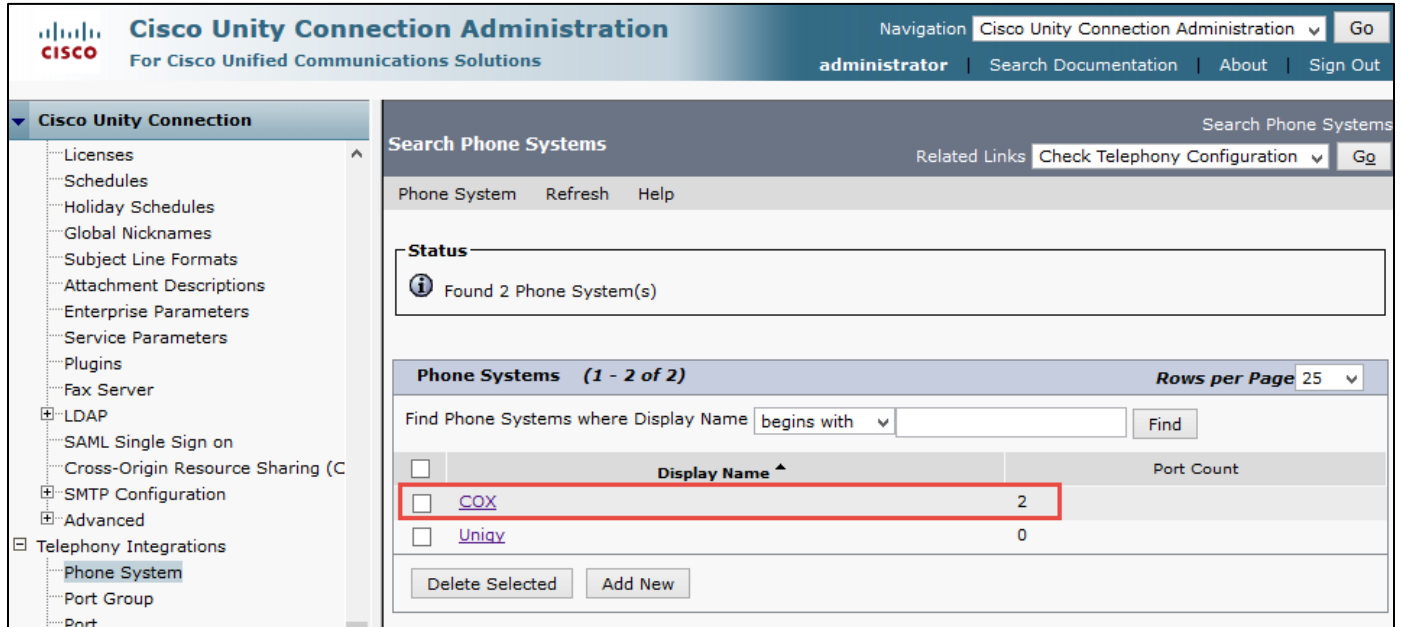
For Cisco Technical Support please visit our [Technical Support](#) web site.

Figure 57: CUC Version

## CUC Telephony Integration with Cisco UCM

**Navigation:** Telephony Integrations → Phone system

Set Phone System Name\* = COX. This is used for this example



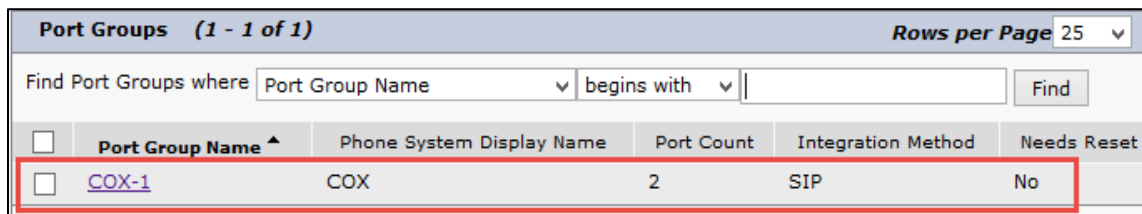
The screenshot shows the Cisco Unity Connection Administration interface. The left sidebar contains a navigation tree with 'Cisco Unity Connection' expanded, showing various configuration options. The main content area is titled 'Search Phone Systems' and displays a status message: 'Found 2 Phone System(s)'. Below this, a table lists the phone systems. The table has columns for 'Display Name' and 'Port Count'. The first row is 'COX' with a port count of 2, and the second row is 'Univ' with a port count of 0. The 'COX' row is highlighted with a red border. Below the table are buttons for 'Delete Selected' and 'Add New'.

Display Name	Port Count
COX	2
Univ	0

Figure 58: CUC Phone System

## CUC Port Group

**Navigation:** Telephony Integration → Port Group



The screenshot shows the Cisco Unity Connection Administration interface for Port Groups. The main content area is titled 'Port Groups (1 - 1 of 1)' and displays a table with columns: 'Port Group Name', 'Phone System Display Name', 'Port Count', 'Integration Method', and 'Needs Reset'. The first row is 'COX-1' with a phone system display name of 'COX', a port count of 2, integration method of 'SIP', and 'Needs Reset' set to 'No'. The entire row is highlighted with a red border.

Port Group Name	Phone System Display Name	Port Count	Integration Method	Needs Reset
COX-1	COX	2	SIP	No

Figure 59: CUC Port Group



Set Display Name\* = COX-1. This is used in this example  
Set Integration Method = SIP  
Check "Enable Message Waiting Indicator Settings" box

**Port Group**  
Display Name\*   
Integration Method   
Reset Status

**Session Initiation Protocol (SIP) Settings**  
☒ Register with SIP Server  
☒ Authenticate with SIP Server  
Authentication Username   
Authentication Password   
Contact Line Name   
SIP Security Profile   
SIP Transport Protocol

**Advertised Codec Settings**  


Display Name	Packet Size
G.711 mu-law	<input type="text" value="20"/>
G.729	<input type="text" value="20"/>

**Message Waiting Indicator Settings**  
☒ Enable Message Waiting Indicators  
Delay between Requests  milliseconds  
Maximum Concurrent Requests   
Retries After Successful Attempt   
Retry Interval After Successful Attempt  milliseconds

Figure 60: CUC Port Group – Cont.

## CUC Port Settings

Navigation: Telephony Integration → Port


**Cisco Unity Connection Administration**  
For Cisco Unified Communications Solutions

Navigation **Cisco Unity Connection Administration** **Go**  
**administrator** | [Search Documentation](#) | [About](#) | [Sign Out](#)

**Cisco Unity Connection**

- Licenses
- Schedules
- Holiday Schedules
- Global Nicknames
- Subject Line Formats
- Attachment Descriptions
- Enterprise Parameters
- Service Parameters
- Plugins
- Fax Server
- LDAP
- SAML Single Sign on
- Cross-Origin Resource Shar
- SMTP Configuration
- Advanced
- Telephony Integrations
  - Phone System
  - Port Group
  - Port**
  - Speech Connect Port

**Search Ports**
Search Ports

Related Links **Check Telephony Configuration** **Go**

Port Refresh Help

**Status**  
Found 2 Port(s)

**Port (1 - 2 of 2)** **Rows per Page** 25

Find Port where Display Name begins with Find

	Display Name	Phone System Display Name	Extension	Server	Enabled	Answer Calls	Message Notification	Dialout MWI	TRAP Connection	Security Mode
<input type="checkbox"/>	<a href="#">COX-1-001</a>	COX		clus23-unity	X	X	X	X	X	NA
<input type="checkbox"/>	<a href="#">COX-1-002</a>	COX		clus23-unity	X	X	X	X	X	NA

Delete Selected Add New

**Port Basics (COX-1-001)**
Search Ports Port Basics (COX-1-001)

Related Links **Check Telephony Configuration** **Go**

Port Refresh Help

Save Delete Previous Next

**Phone System Port**

- ☒ Enabled
- Port Name  **Restart**
- Phone System
- Port Group
- Server

**Port Behavior**

- ☒ Answer Calls
- ☒ Perform Message Notification
- ☒ Send MWI Requests (may also be disabled by the port group)
- ☒ Allow TRAP Connections

Figure 61: CUC Port Settings

## CUC Sample User Basic Settings

**Navigation:** Cisco Unity Connection → Users → Users

Set Alias = 1399 is one of the extensions used for this testing

Set Extension = 1399 is used for this example

Set Partition = clus23-unity Partition is used for this example

Select Search Space = clus23-unity Search Space

Set Phone System = COX

Name	
Alias*	1399
First Name	
Last Name	
Display Name	1399
SMTP Address	1399 @clus23-unity.lab.tekvizion.com
Initials	
Title	
Employee ID	
LDAP Integration Status	
<input type="radio"/> Integrate with LDAP Directory <input checked="" type="radio"/> Do Not Integrate with LDAP Directory	
Phone	
Extension*	1399
Cross-Server Transfer Extension or URI	
Outgoing Fax Number	
Outgoing Fax Server	--- Not Selected ---
Partition	clus23-unity Partition
Search Scope	clus23-unity Search Space
Phone System	COX
Class of Service	Voice Mail User COS

Figure 62: CUC Sample User Basic Settings

Active Schedule
Weekdays
View

☐ Set for Self-enrollment at Next Sign-In
☒ List in Directory
☒ Send Non-Delivery Receipts on Failed Message Delivery
☐ Skip PIN When Calling From a Known Extension
☐ Use Short Calendar Caching Poll Interval

Recorded Name
Play/Record

**Location**

Address
Building
City
State
Postal Code
Country
United States
☒ Use System Default Time Zone
Time Zone
(GMT-06:00) America/Chicago
Language
☒ Use System Default Language
☐ English(United States)
Department
Manager
Billing ID

Corporate Email Address
☐ Generate SMTP Proxy Address From Corporate Email Address
Directory URI
Corporate Phone Number

Figure 63: CUC Sample User Basic Settings – Cont.

## Acronyms

Acronym	Definitions
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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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