



Connecting Cisco Unified Communication Manager [v11.5.1] to CableOne SIP Trunks via Cisco Unified Border Element v12.0.0 [IOS-XE 16.07.01]

May 16, 2018



Table of Contents

Introduction	4
Network Topology.....	5
System Components	6
Hardware Requirements.....	6
Software Requirements	6
Features	6
Features Supported	6
Features Not Supported.....	6
Caveats.....	7
Configuration	8
Configuring Cisco Unified Border Element	8
Network Interface.....	8
Global Cisco UBE settings.....	11
Codecs.....	11
Dial peer.....	12
Configuration example.....	14
Configuring Cisco UCM 11.5 Cluster	28
Cisco UCM Version	28
Cisco Call Manager Service Parameters.....	29
SIP Trunk Security Profile	30
SIP Profile	32
Trunk configuration.....	36
Routing configuration	42
Acronyms	47
Important Information.....	47



Table of Figures

Figure 1 Network Topology.....	5
Figure 2 High Availability topology	8
Figure 3: Cisco UCM Version	28
Figure 4: Service Parameters	29
Figure 5: Service Parameters (Cont.)	30
Figure 6: SIP Trunk Security Profile.....	31
Figure 7: SIP Profile	32
Figure 8: SIP Profile (Cont.)	33
Figure 9: SIP Profile (Cont.)	34
Figure 10: SIP Profile (Cont.)	35
Figure 11: Add New Trunk to Cisco UBE	36
Figure 12: Add SIP Trunk Type	37
Figure 13: SIP Trunk to Cisco UBE	38
Figure 14: SIP Trunk to Cisco UBE (Cont.)	39
Figure 15: SIP Trunk to Cisco UBE (Cont.)	40
Figure 16: SIP Trunk to Cisco UBE (Cont.)	41
Figure 17: Add New Route Pattern for Cisco UBE.....	42
Figure 18: Route Pattern Configuration for Cisco UBE-PSTN Access	43
Figure 19: Route Pattern Configuration for Cisco UBE-PSTN Access (Cont.).....	44
Figure 20: Route Pattern Configuration for Cisco UBE-PSTN Access - Anonymous Call.....	45
Figure 21: Route Pattern Configuration for Cisco UBE-PSTN Access - Anonymous Call (Cont.).....	46



Introduction

Service Providers today, such as CableOne, 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.

CableOne is a service provider offering 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 (Cisco UCM) and CableOne network, Cisco Unified Border Element (Cisco UBE) ISR 4331/K9 running IOS 16.07.01 can be used. The Cisco Unified Border Element 16.07.01 provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 11.5.1 connected to CableOne IP network.

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

- This application note describes how to configure Cisco UCM 11.5.1, and Cisco UBE on ISR 4331/K9 [IOS – 16.07.01] for connectivity to CableOne SIP Trunking service. The deployment model covered in this application note is CPE (Cisco UCM) to PSTN (CableOne) via Cisco UBE v12.0 [IOS-XE] 16.07.01.
- Testing was performed in accordance to CableOne generic SIP Trunking test methodology and among features verified were – basic calls, DTMF transport, Music on Hold (MOH), unattended and attended transfers, call forward, conferences and High Availability.
- The Cisco UBE configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between CableOne 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 CableOne SIP Trunking network.



Network Topology

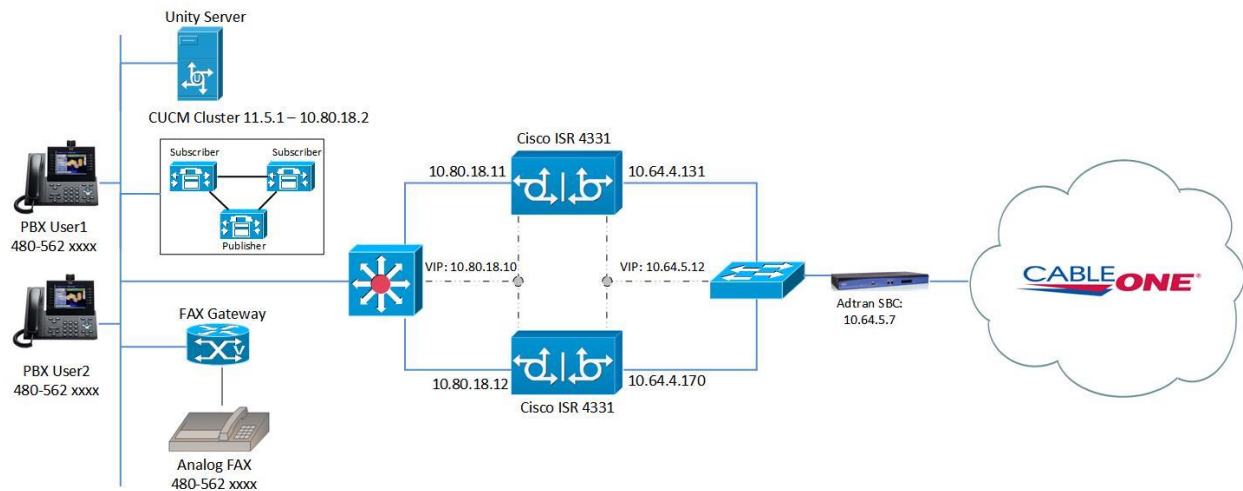


Figure 1 Network Topology

- The network topology includes the Cisco UCM Cluster, Unity Voicemail system, Cisco Fax gateway and 2 Cisco Endpoints. Cisco UCM has a trunk configured to Cisco UBE's Virtual IP Address. CableOne was used as the service provider with SIP trunk to the Cisco UBE using the WAN Virtual IP Address.
- 2 Cisco Unified Border Elements are used here for High Availability.
- SIP Trunk transport type used between Cisco UBE and Cisco UCM is TCP and to CableOne is UDP.

Cisco UCM and Cisco UBE Settings:

Setting	Value
Transport from Cisco UBE to Cisco UCM	TCP with RTP
Transport from Cisco UBE to CableOne	UDP with RTP
Voice Mail Support	YES
Session Refresh	YES



System Components

Hardware Requirements

- Cisco UBE on Cisco ISR 4331/K9 router
- CUCM cluster on UCS C240, 1 Publisher node and 2 Subscriber nodes
- Cisco 2851 with FXS ports and Analog Fax machine
- Generic Cisco IP-Phones
- Adtran NetVanta 3140

Software Requirements

- CUBE-Version: 12.0.0 running IOS-XE 16.07.01
- CUCM UCOS 11.5.1.12900-21 for 1 Publisher and 2 Subscriber
- Cisco IOS v15.1(4)M5 for the fax gateway
- ADTRAN, Inc. OS version R12.3.3.E

Features

Features Supported

- Incoming and outgoing calls using G711ulaw codec
- International Calls and digit manipulations
- Call Conference
- CPE Voice Mail support
- Call hold & Resume with and without MoH
- Unattended and Attended Call transfer
- Call forward (all, busy and no answer)
- Inband DTMF
- Fax (G711 Pass-through)
- IP-PBX Calling number privacy
- High Availability

Features Not Supported

- Cisco UCM does not support Blind Call transfer
- In HA Redundancy mode, the Primary Cisco UBE will not take over the Primary/Active role after a reboot/network outage
- CableOne does not support T.38 Fax
- G.729 Codec for voice calls is not supported by CableOne



Caveats

- Caller ID is not updated on attended and unattended transfer scenarios.
- Only one IP PBX used for the testing.
- The Cisco UBE HA tested is layer 2 box to box Cisco UBE redundancy.
- Transcoder has been enabled on Cisco UBE to support Inband DTMF.



Configuration

Configuring Cisco Unified Border Element

Network Interface

The IP address used are for illustration only, the actual IP address can vary. The Active/Standby pair share the same virtual IP address and continually exchange status messages.

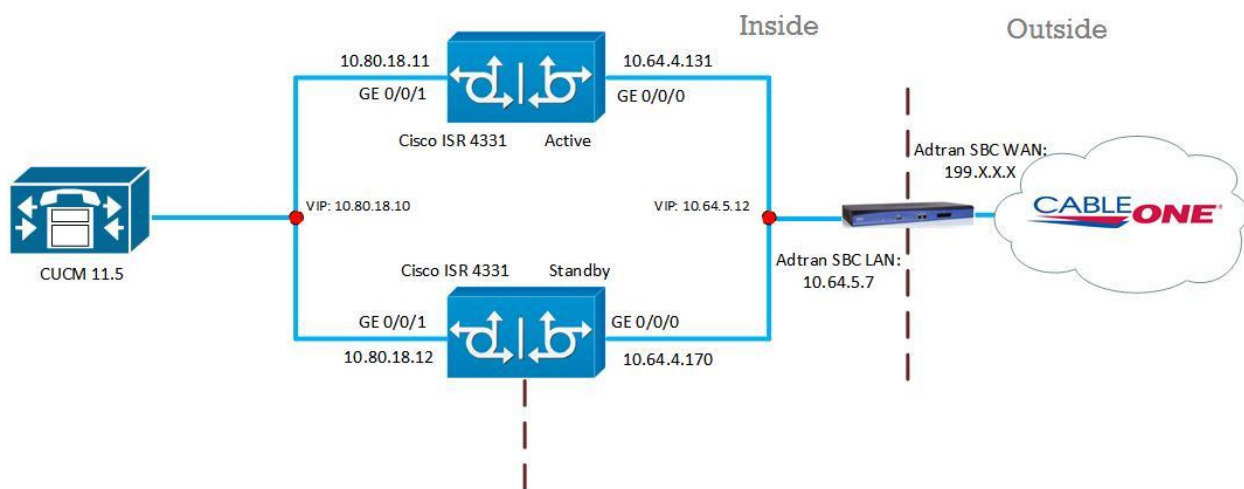


Figure 2 High Availability topology



Cisco UBE 1:

```
interface GigabitEthernet0/0/0
  description WAN
  ip address 10.64.4.131 255.255.0.0
  negotiation auto
  redundancy rii 1
  redundancy group 1 ip 10.64.5.12 exclusive
!
interface GigabitEthernet0/0/1
  description LAN
  ip address 10.80.18.11 255.255.255.0
  negotiation auto
  redundancy rii 2
  redundancy group 1 ip 10.80.18.10 exclusive
!
interface GigabitEthernet0/0/2
  description CUBE_HA
  ip address 10.89.20.12 255.255.255.0
  negotiation auto
!
interface Service-Engine0/4/0
!
interface GigabitEthernet0
  vrf forwarding Mgmt-intf
  no ip address
  shutdown
  negotiation auto
```

Cisco UBE 2:

```
interface GigabitEthernet0/0/0
  description WAN
  ip address 10.64.4.170 255.255.0.0
```



```
negotiation auto
redundancy rii 1
redundancy group 1 ip 10.64.5.12 exclusive
!
interface GigabitEthernet0/0/1
description LAN
ip address 10.80.18.12 255.255.255.0
negotiation auto
redundancy rii 2
redundancy group 1 ip 10.80.18.10 exclusive
!
interface GigabitEthernet0/0/2
description CUBE_HA
ip address 10.89.20.11 255.255.255.0
negotiation auto
!
interface Service-Engine0/4/0
!
interface GigabitEthernet0
vrf forwarding Mgmt-intf
no ip address
shutdown
negotiation auto
```



Global Cisco UBE settings

In order to enable Cisco UBE IP2IP SBC functionality, following command has to be entered:

```
voice service voip
  address-hiding
  mode border-element license capacity 20
  allow-connections sip to sip
  redundancy-group 1
  no supplementary-service sip handle-replaces
  fax protocol pass-through g711ulaw
  sip
    asserted-id pai
  midcall-signaling passthru
```

Explanation

Command	Description
allow-connections sip to sip	Allow IP2IP connections between two SIP call legs
redundancy-group 1	Enable High Availability for the VoIP service
fax protocol	Specifies the fax protocol
asserted-id	Specifies the privacy header in the outgoing SIP requests and response messages
midcall-signaling passthru	Pass-through media change method optimizes or consumes mid-call, media-related signaling within the call

Codecs

G711ulaw codec is used for this testing.

```
voice class codec 1
  codec preference 1 g711ulaw!
```



Dial peer

Outbound Dial-peer to CableOne:

```
description Incoming Call from CUCM International
session protocol sipv2
session transport tcp
incoming called-number .T
voice-class codec 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
no vad
!
dial-peer voice 2 voip
description Outgoing Call to CableOne
destination-pattern .T
session protocol sipv2
session target ipv4:10.64.5.7:5060
voice-class codec 1
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
no dtmf-interworking
no vad
```

Inbound Dial-peer from CableOne:

```
dial-peer voice 3 voip
description Incoming call from CableOne
session protocol sipv2
incoming called-number 480.....
voice-class codec 1
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/0
```



```
voice-class sip bind media source-interface GigabitEthernet0/0/0
no vad
!
dial-peer voice 4 voip
description Outgoing to CUCM
destination-pattern 480.....
session protocol sipv2
session target ipv4:10.80.18.2:5060
session transport tcp
voice-class codec 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
no vad
```



Configuration example

The following configuration snippet contains a sample configuration of Cisco UBE with all parameters mentioned previously.

Active Cisco UBE:

```
version 16.7
service config
service timestamps debug datetime msec
service timestamps log datetime msec
platform qfp utilization monitor load 80
no platform punt-keepalive disable-kernel-core
!
hostname CUBE10Cableone
!
boot-start-marker
boot-end-marker
!
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
!
no aaa new-model
!
!
```



```
subscriber templating
multilink bundle-name authenticated
!
crypto pki trustpoint TP-self-signed-2930804041
  enrollment selfsigned
  subject-name cn=IOS-Self-Signed-Certificate-2930804041
  revocation-check none
  rsakeypair TP-self-signed-2930804041
!
crypto pki trustpoint TP-self-signed-3616943619
  enrollment selfsigned
  subject-name cn=IOS-Self-Signed-Certificate-3616943619
  revocation-check none
  rsakeypair TP-self-signed-3616943619
!
!
crypto pki certificate chain TP-self-signed-2930804041
crypto pki certificate chain TP-self-signed-3616943619
!
voice service voip
  address-hiding
  mode border-element license capacity 20
  allow-connections sip to sip
  redundancy-group 1
  no supplementary-service sip handle-replaces
  fax protocol pass-through g711ulaw
  sip
    asserted-id pai
  midcall-signaling passthru
```



```
!  
voice class codec 1  
    codec preference 1 g711ulaw  
!  
voice class sip-profiles 1  
    request ANY sip-header Diversion modify "sip:(.*)@" "sip:480562\1@"  
!  
voice-card 0/4  
    dsp services dspfarm  
    no watchdog  
!  
license udi pid ISR4331/K9 sn FD021381F17  
diagnostic bootup level minimal  
spanning-tree extend system-id  
!  
redundancy  
    mode none  
    application redundancy  
        group 1  
            name voice-HA  
            priority 150 failover threshold 75  
            timers delay 30 reload 60  
            control GigabitEthernet0/0/2 protocol 1  
            data GigabitEthernet0/0/2  
            track 1 shutdown  
            track 2 shutdown  
!  
track 1 interface GigabitEthernet0/0/0 line-protocol  
track 2 interface GigabitEthernet0/0/1 line-protocol
```



```
interface GigabitEthernet0/0/0
  description WAN
  ip address 10.64.4.131 255.255.0.0
  negotiation auto
  redundancy rii 1
  redundancy group 1 ip 10.64.5.12 exclusive
!
interface GigabitEthernet0/0/1
  description LAN
  ip address 10.80.18.11 255.255.255.0
  negotiation auto
  redundancy rii 2
  redundancy group 1 ip 10.80.18.10 exclusive
!
interface GigabitEthernet0/0/2
  description CUBE_HA
  ip address 10.89.20.12 255.255.255.0
  negotiation auto
!
interface Service-Engine0/4/0
!
interface GigabitEthernet0
  vrf forwarding Mgmt-intf
  no ip address
  shutdown
  negotiation auto
!
ip forward-protocol nd
ip http server
```



```
ip http authentication local
ip http secure-server
ip http client source-interface GigabitEthernet0/0/0
ip tftp source-interface GigabitEthernet0/0/0
ip route 0.0.0.0 0.0.0.0 10.64.1.1
ip route 10.80.18.0 255.255.255.0 10.80.18.1
!
ip ssh server algorithm encryption aes128-ctr aes192-ctr aes256-ctr
ip ssh client algorithm encryption aes128-ctr aes192-ctr aes256-ctr
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
!
dspfarm profile 50 transcode
  codec g711ulaw
  maximum sessions 12
  associate application CUBE
!
dial-peer voice 1 voip
  description Incoming call from CUCM International
  session protocol sipv2
  session transport tcp
  incoming called-number .T
  voice-class codec 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
no vad
!
dial-peer voice 2 voip
description Outgoing call to CableOne
destination-pattern .T
session protocol sipv2
session target ipv4:10.64.5.7:5060
voice-class codec 1
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
no dtmf-interworking
no vad
!
dial-peer voice 3 voip
description Incoming call from CableOne
session protocol sipv2
incoming called-number 480.....
voice-class codec 1
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
no vad
!
dial-peer voice 4 voip
description Outgoing to CUCM
destination-pattern 480.....
```



```
session protocol sipv2
session target ipv4:10.80.18.2:5060
session transport tcp
voice-class codec 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
no vad
!
!
sip-ua
connection-reuse
!
!
line con 0
exec-timeout 0 0
password xxxxxxxx
login
transport input none
stopbits 1
line aux 0
stopbits 1
line vty 0 4
exec-timeout 0 0
password xxxxxxxxxx
login
transport input telnet
!
end
```



Standby Cisco UBE:

```
version 16.7
service timestamps debug datetime msec
service timestamps log datetime msec
platform qfp utilization monitor load 80
no platform punt-keepalive disable-kernel-core
!
hostname CUBE8Cableone
!
boot-start-marker
boot-end-marker
!
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
no aaa new-model
!
subscriber templating
multilink bundle-name authenticated
!
crypto pki trustpoint TP-self-signed-3616943619
enrollment selfsigned
subject-name cn=IOS-Self-Signed-Certificate-3616943619
```



```
revocation-check none
rsakeypair TP-self-signed-3616943619
!
crypto pki certificate chain TP-self-signed-3616943619
!
voice service voip
  no ip address trusted authenticate
  address-hiding
  mode border-element license capacity 20
  allow-connections sip to sip
  redundancy-group 1
  no supplementary-service sip handle-replaces
  fax protocol pass-through g711ulaw
  sip
    midcall-signaling passthru
  !
voice class codec 1
  codec preference 1 g711ulaw
  !
  !
voice class sip-profiles 1
  request ANY sip-header Diversion modify "sip:(.*)@" "sip:480562\1@"
  !
voice-card 0/4
  dsp services dspfarm
  no watchdog
  !
license udi pid ISR4331/K9 sn FD021381FEY
no license smart enable
```



```
diagnostic bootup level minimal
!
spanning-tree extend system-id
!
redundancy
mode none
application redundancy
group 1
    name voice-HA
    priority 150 failover threshold 75
    timers delay 30 reload 60
    control GigabitEthernet0/0/2 protocol 1
    data GigabitEthernet0/0/2
    track 1 shutdown
    track 2 shutdown
!
track 1 interface GigabitEthernet0/0/0 line-protocol
track 2 interface GigabitEthernet0/0/1 line-protocol
!
interface GigabitEthernet0/0/0
    description WAN
    ip address 10.64.4.170 255.255.0.0
    negotiation auto
    redundancy rii 1
    redundancy group 1 ip 10.64.5.12 exclusive
!
interface GigabitEthernet0/0/1
    description LAN
    ip address 10.80.18.12 255.255.255.0
```



```
negotiation auto
redundancy rii 2
redundancy group 1 ip 10.80.18.10 exclusive
!
interface GigabitEthernet0/0/2
description CUBE_HA
ip address 10.89.20.11 255.255.255.0
negotiation auto
!
interface Service-Engine0/4/0
!
interface GigabitEthernet0
vrf forwarding Mgmt-intf
no ip address
shutdown
negotiation auto
!
ip forward-protocol nd
ip http server
ip http authentication local
ip http secure-server
ip http client source-interface GigabitEthernet0/0/0
ip tftp source-interface GigabitEthernet0/0/0
ip route 0.0.0.0 0.0.0.0 10.64.1.1
ip route 10.80.18.0 255.255.255.0 10.80.18.1
!
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
!
dspfarm profile 50 transcode
  codec g711ulaw
  maximum sessions 12
  associate application CUBE
!
dial-peer voice 1 voip
  description Incoming Call from CUCM International
  session protocol sipv2
  session transport tcp
  incoming called-number .T
  voice-class codec 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
  no vad
!
dial-peer voice 2 voip
  description Outgoing Call to CableOne
  destination-pattern .T
  session protocol sipv2
  session target ipv4:10.64.5.7:5060
  voice-class codec 1
```



```
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
no dtmf-interworking
no vad
!
dial-peer voice 3 voip
description Incoming call from CableOne
session protocol sipv2
incoming called-number 480.....
voice-class codec 1
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
no vad
!
dial-peer voice 4 voip
description Outgoing to CUCM
destination-pattern 480.....
session protocol sipv2
session target ipv4:10.80.18.2:5060
session transport tcp
voice-class codec 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
no vad
!
!
```



```
sip-ua
  connection-reuse
!
!
line con 0
  exec-timeout 0 0
  password xxxxxxxxxxxx
  login
  transport input none
  stopbits 1
line aux 0
  stopbits 1
line vty 0 4
  exec-timeout 0 0
  password xxxxxxxxxxxx
  login
!
wsma agent exec
!
wsma agent config
!
wsma agent filesys
!
wsma agent notify
!
!
end
```

CUBE8Cableone#



Configuring Cisco UCM 11.5 Cluster

Cisco UCM Version

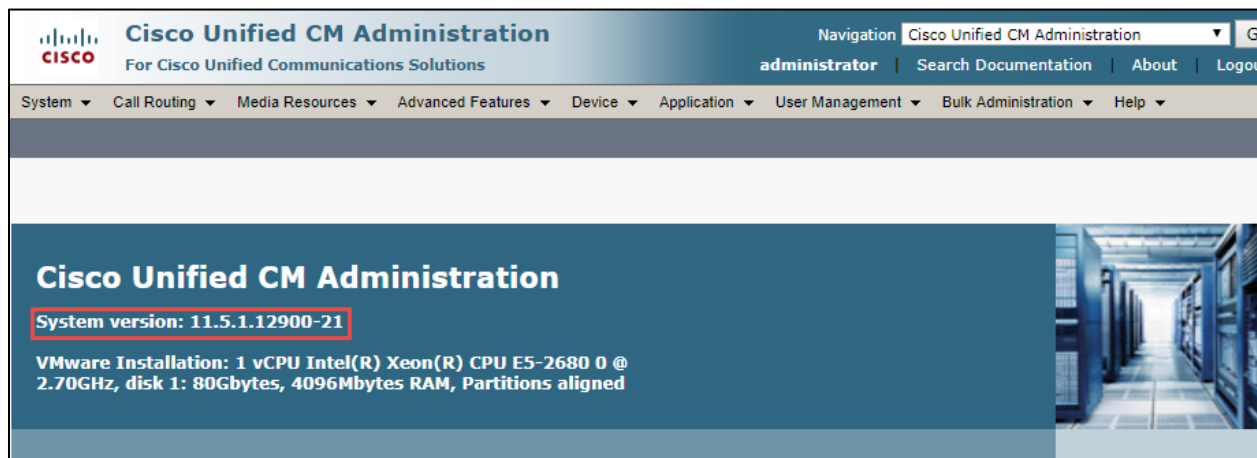


Figure 3: Cisco UCM Version



Cisco Call Manager Service Parameters

Navigation: System → Service Parameters

- Select Server* = Clus28pub--CUCM Voice/Video (Active)
- Select Service* = Cisco CallManager (Active)
- Duplex Streaming Enabled* = True
- All other fields are set to default values

Service Parameter Configuration

Navigation: Cisco Unified CM Administration Go

administrator | Search Documentation | About | Logout

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

Related Links: Parameters for All Servers Go

Save Set to Default Advanced

Status

Status: Ready

Select Server and Service

Server* Clus28pub--CUCM Voice/Video (Active)

Service* Cisco CallManager (Active)

All parameters apply only to the current server except parameters that are in the cluster-wide group(s).

Cisco CallManager (Active) Parameters on server Clus28pub--CUCM Voice/Video (Active)

Parameter Name	Parameter Value	Suggested Value
Call Throttling		
Code Yellow Entry Latency *	20	20
Code Yellow Exit Latency Calculation *	40	40
Code Yellow Duration *	5	5
Max Events Allowed *	2000	2000
System Throttle Sample Size *	10	10

Figure 4: Service Parameters



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

Figure 5: Service Parameters (Cont.)

SIP Trunk Security Profile

Navigation: System → Security → SIP Trunk Security Profile

- Name* = **Non Secure SIP Trunk Profile** is used as an example
- Description = **Non Secure SIP Trunk Profile authenticated by null String** is used as an example
- Device Security Mode = **Non Secure**
- Incoming Transport Type* = **TCP + UDP**
- Outgoing Transport Type = **TCP**



SIP Trunk Security Profile ConfigurationRelated Links: [Back To](#)

Save Delete Copy Reset Apply Config Add New

Status
 Status: Ready

SIP Trunk Security Profile Information
Name* Non Secure SIP Trunk Profile
Description Non Secure SIP Trunk Profile authenticated by null String
Device Security Mode Non Secure
Incoming Transport Type* TCP+UDP
Outgoing Transport Type TCP
☐ 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

Save Delete Copy Reset Apply Config Add New

Figure 6: SIP Trunk Security Profile



SIP Profile

Navigation: Device → Device Settings → SIP Profile

- Name*= **CableOne SIP Profile** is used as an example
- **SIP Rel1XX Options:** Send PRACK if 1XX contains SDP is selected for this example

SIP Profile ConfigurationRelated Links: [Back To Find/](#)

Save Delete Copy Reset Apply Config Add New

Status

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

SIP Profile Information

Name *	CableOne SIP Profile
Description	
Default MTP Telephony Event Payload Type *	101
Early Offer for G.Clear Calls *	Disabled ▼
User-Agent and Server header information *	Send Unified CM Version Information as User-Agent ▼
Version in User Agent and Server Header *	Major And Minor ▼
Dial String Interpretation *	Phone number consists of characters 0-9, *, #, and ▼
Confidential Access Level Headers *	Disabled ▼
☐ Redirect by Application	
☐ Disable Early Media on 180	
☐ Outgoing T.38 INVITE include audio mline	
☐ Offer valid IP and Send/Receive mode only for T.38 Fax Relay	
☐ Use Fully Qualified Domain Name in SIP Requests	
☐ Assured Services SIP conformance	
☐ Enable External QoS**	

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 ▼
☒ Require SDP Inactive Exchange for Mid-Call Media Change	
☐ Allow RR/RS bandwidth modifier (RFC 3556)	

Figure 7: SIP Profile



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	☒ Common Port Range for Audio and Video ☐ 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 ▼
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

Figure 8: SIP Profile (Cont.)



Call Forward URI*	x-cisco-serviceuri-cfwdall						
Speed Dial (Abbreviated Dial) URI*	x-cisco-serviceuri-abbrdial						
☒ Conference Join Enabled							
☐ RFC 2543 Hold							
☒ Semi Attended Transfer							
☐ Enable VAD							
☐ Stutter Message Waiting							
☐ MLPP User Authorization							
Normalization Script							
Normalization Script < None >							
☐ Enable Trace							
<table><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							
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* Disabled (Default value)							
☐ 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							
☐ Connect Inbound Call before Playing Queuing Announcement							

Figure 9: SIP Profile (Cont.)



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

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

Save

Delete

Copy

Reset

Apply Config

Add New

Figure 10: SIP Profile (Cont.)



Trunk configuration

Trunk configuration from Cisco UCM to the LAN side of the Cisco UBE:

Navigation: Device → Trunk → Add New

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

Find and List Trunks

+ Add New

Trunks

Find Trunks where Device Name ▾ begins with ▾ Find Clear Filter + -

Select item or enter search text ▾

No active query. Please enter your search criteria using the options above.

Add New

Figure 11: Add New Trunk to Cisco UBE

- Select 'Trunk Type' as SIP Trunk and 'Device Protocol' as SIP and select 'Next' as shown below.
- Set **Device Name**: CableOne is given for this example
- Set **Description**: CableOne SIP Trunk Certification is given for this example
- Select **Device Pool**: "Default" is selected for this example.
- Set Significant Digits: 4
- Under Destination:
- Set **Destination Address**: 10.80.18.12 is given for this example. This is the CUBE LAN side VIP IP address.
- Set **Destination Port**: 5060
- Set **BLF Presence Group**: Standard presence group is selected from dropdown menu
- Set **SIP Trunk security profile**: Non-Secured SIP trunk Profile is elected from drop down menu
- Set **SIP Profile**: CableOne SIP Profile is selected from drop down menu



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

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾

Trunk Configuration

 [Next](#)

Status

 Status: Ready

Trunk Information

Trunk Type*	SIP Trunk ▾
Device Protocol*	SIP ▾
Trunk Service Type*	None(Default) ▾

[Next](#)

 *- indicates required item.

Figure 12: Add SIP Trunk Type



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

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

Save Delete Reset Add New

-Status-

Status: Ready

-SIP Trunk Status-

Service Status: Full Service
Duration: Time In Full Service: 0 day 21 hours 46 minutes

-Device Information-

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	CableOne
Description	CableOne SIP Trunk Certification
Device Pool*	Default
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	MRGL_Default
Location*	Hub_None
AAR Group	< None >
Tunneled Protocol*	None
QSIG Variant*	No Changes
ASN.1 ROSE OID Encoding*	No Changes
Packet Capture Mode*	None
Packet Capture Duration	0

☐ Media Termination Point Required
☒ Retry Video Call as Audio
☐ Path Replacement Support
☐ Transmit UTF-8 for Calling Party Name
☐ Transmit UTF-8 Names in QSIG APDU
☐ Unattended Port
☐ SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.

Consider Traffic on This Trunk Secure*

Route Class Signaling Enabled*

Use Trusted Relay Point*

Figure 13: SIP Trunk to Cisco UBE



☐ PSTN Access

☐ Run On All Active Unified CM Nodes

Intercompany Media Engine (IME)

E.164 Transformation Profile < None >

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
Incoming Number	Default	0	< None >

Figure 14: SIP Trunk to Cisco UBE (Cont.)

- Configure the Virtual LAN IP address of the Cisco UBE and the Destination Port
- Configure the SIP Trunk Security Profile and SIP Profile as shown below
- The rest of the configuration is all default



- Click Save and Reset after completion

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 there is no prefix assigned.

Clear Prefix Settings **Default Prefix Settings**

Number Type	Prefix	Strip Digits	Calling Search Space
Incoming Number	Default	0	< None >

Connected Party Settings

Connected Party Transformation CSS

☒ Use Device Pool Connected Party Transformation CSS

Outbound Calls

Called Party Transformation CSS

☒ Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS

☒ Use Device Pool Calling Party Transformation CSS

Calling Party Selection*

Calling Line ID Presentation*

Calling Name Presentation*

Calling and Connected Party Info Format*

☒ Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS

☒ Use Device Pool Redirecting Party Transformation CSS

Caller Information

Caller ID DN

Caller Name

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

Figure 15: SIP Trunk to Cisco UBE (Cont.)



SIP Information

Destination

☐ Destination Address is an SRV

Destination Address

Destination Address IPv6

Destination

1* 10.80.18.10 5060

MTP Preferred Originating Codec* 711ulaw

BLF Presence Group* Standard Presence group

SIP Trunk Security Profile* 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* CableOne SIP Profile [View Details](#)

DTMF Signaling Method* RFC 2833

Normalization Script

Normalization Script < None >

☐ Enable Trace

Parameter Name

Parameter Value

1

Recording Information

☒ None

☐ This trunk connects to a recording-enabled gateway

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

Geolocation Configuration

Geolocation < None >

Geolocation Filter < None >

☐ Send Geolocation Information

Save

Delete

Reset

Add New

Figure 16: SIP Trunk to Cisco UBE (Cont.)



Routing configuration

Route Pattern for Cisco UBE:

Navigation: Call Routing → Route/Hunt → Route Pattern → Add New

Route patterns are configured as below:

Cisco IP phone dial “11” 10 digits number to access PSTN via Cisco UBE

- “1” is removed before sending to Cisco UBE
- The rest of the number is sent to Cisco UBE and to Cable One Network
- *6 is used to dial the outbound calls with caller ID blocking

The screenshot shows the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the title 'Cisco Unified CM Administration For Cisco Unified Communications Solutions', and a navigation dropdown menu set to 'Cisco Unified CM Administration' with a 'Go' button. Below this is a secondary navigation bar with links for 'administrator', 'Search Documentation', 'About', and 'Logout'. A main navigation bar contains various system categories like 'System', 'Call Routing', 'Media Resources', etc. The main content area is titled 'Find and List Route Patterns' and features an 'Add New' button with a plus icon. Below this is a 'Route Patterns' section with a search bar. The search bar has a dropdown menu set to 'Pattern' and a 'begins with' dropdown. To the right of the search bar are 'Find', 'Clear Filter', and '+'/'-' buttons. A message below the search bar states: 'No active query. Please enter your search criteria using the options above.' At the bottom of the 'Route Patterns' section, the 'Add New' button is highlighted with a red rectangular box.

Figure 17: Add New Route Pattern for Cisco UBE



Route Pattern ConfigurationRelated Links:

Save Delete Copy Add New

Status

Status: Ready

Pattern Definition

Route Pattern*	1.@
Route Partition	< None >
Description	To CableOne
Numbering Plan*	NANP
Route Filter	< None >
MLPP Precedence*	Default
☐ Apply Call Blocking Percentage	
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Gateway/Route List*	CableOne
Route Option	☒ Route this pattern ☐ Block this pattern No Error
Call Classification*	OffNet
External Call Control Profile	< None >
☐ Allow Device Override ☒ Provide Outside Dial Tone ☐ Allow Overlap Sending ☐ Urgent Priority	
☐ Require Forced Authorization Code	
Authorization Level*	0
☐ Require Client Matter Code	

[\(Edit\)](#)

Calling Party Transformations

☒ Use Calling Party's External Phone Number Mask

Calling Party Transform Mask	
Prefix Digits (Outgoing Calls)	
Calling Line ID Presentation*	Default
Calling Name Presentation*	Default
Calling Party Number Type*	Cisco CallManager
Calling Party Numbering Plan*	Cisco CallManager

Figure 18: Route Pattern Configuration for Cisco UBE-PSTN Access



Connected Party Transformations	
Connected Line ID Presentation*	Default ▼
Connected Name Presentation*	Default ▼
Called Party Transformations	
Discard Digits	PreDot ▼
Called Party Transform Mask	
Prefix Digits (Outgoing Calls)	
Called Party Number Type*	Cisco CallManager ▼
Called Party Numbering Plan*	Cisco CallManager ▼
ISDN Network-Specific Facilities Information Element	
Network Service Protocol	-- Not Selected -- ▼
Carrier Identification Code	
Network Service	Service Parameter Name
-- Not Selected -- ▼	< Not Exist >
Save Delete Copy Add New	

Figure 19: Route Pattern Configuration for Cisco UBE-PSTN Access (Cont.)



Route Pattern ConfigurationRelated Links: [Back](#)

Save Delete Copy Add New

Status

Status: Ready

Pattern Definition

Route Pattern*

Route Partition< None >

DescriptionCaller ID Restricted

Numbering Plan-- Not Selected --

Route Filter< None >

MLPP Precedence*Default

☐ Apply Call Blocking Percentage

Resource Priority Namespace Network Domain< None >

Route Class*Default

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

Route Option

☒ Route this pattern

☐ Block this pattern

Call Classification*

External Call Control Profile< None >

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

☐ Require Forced Authorization Code

Authorization Level*

☐ Require Client Matter Code

Calling Party Transformations

☒ Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation*

Calling Name Presentation*

Calling Party Number Type*

Calling Party Numbering Plan*

Figure 20: Route Pattern Configuration for Cisco UBE-PSTN Access - Anonymous Call



Connected Party Transformations	
Connected Line ID Presentation*	Default ▼
Connected Name Presentation*	Default ▼
Called Party Transformations	
Discard Digits	PreDot ▼
Called Party Transform Mask	
Prefix Digits (Outgoing Calls)	
Called Party Number Type*	Cisco CallManager ▼
Called Party Numbering Plan*	Cisco CallManager ▼
ISDN Network-Specific Facilities Information Element	
Network Service Protocol	-- Not Selected -- ▼
Carrier Identification Code	
Network Service	Service Parameter Name
-- Not Selected -- ▼	< Not Exist >
Save Delete Copy Add New	

Figure 21: Route Pattern Configuration for Cisco UBE-PSTN Access - Anonymous Call (Cont.)



Acronyms

Acronym	Definitions
CPE	Customer Premise Equipment
Cisco UBE	Cisco Unified Border Element
Cisco UCM	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

THE SPECIFICATIONS AND INFORMATION REGARDING THE PRODUCTS IN THIS MANUAL ARE SUBJECT TO CHANGE WITHOUT NOTICE. ALL STATEMENTS, INFORMATION, AND RECOMMENDATIONS IN THIS MANUAL ARE BELIEVED TO BE ACCURATE BUT ARE PRESENTED WITHOUT WARRANTY OF ANY KIND, EXPRESS OR IMPLIED. USERS MUST TAKE FULL RESPONSIBILITY FOR THEIR APPLICATION OF ANY PRODUCTS. IN NO EVENT SHALL CISCO OR ITS SUPPLIERS BE LIABLE FOR ANY INDIRECT, SPECIAL, CONSEQUENTIAL, OR INCIDENTAL DAMAGES, INCLUDING, WITHOUT LIMITATION, LOST PROFITS OR LOSS OR DAMAGE TO DATA ARISING OUT OF THE USE OR INABILITY TO USE THIS MANUAL, EVEN IF CISCO OR ITS SUPPLIERS HAVE BEEN ADVISED OF THE POSSIBILITY OF SUCH DAMAGES

**Corporate
Headquarters**

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

**European
Headquarters**

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

**Americas
Headquarters**

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

**AsiaPacific
Headquarters**

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

Cisco Systems has more than 200 offices in the following countries and regions. Addresses, phone numbers, and fax numbers are listed on the Cisco Web site at <http://www.cisco.com/go/offices>.

Argentina • Australia • Austria • Belgium • Brazil • Bulgaria • Canada • Chile • China PRC • Colombia • Costa Rica • Croatia • Czech Republic • Denmark • Dubai, UAE • Finland • France • Germany • Greece • Hong Kong SAR • Hungary • India • Indonesia • Ireland • Israel • Italy • Japan • Korea • Luxembourg • Malaysia • Mexico • The Netherlands • New Zealand • Norway • Peru • Philippines • Poland • Portugal • Puerto Rico • Romania • Russia • Saudi Arabia • Scotland • Singapore • Slovakia • Slovenia • South Africa • Spain • Sweden • Switzerland • Taiwan • Thailand • Turkey Ukraine • United Kingdom • United States • Venezuela • Vietnam • Zimbabwe

© 2016 Cisco Systems, Inc. All rights reserved.

CCENT, Cisco Lumin, Cisco Nexus, the Cisco logo and the Cisco Square Bridge logo are trademarks of Cisco Systems, Inc.; Changing the Way We Work, Live, Play, and Learn is a service mark of Cisco Systems, Inc.; and Access Registrar, Aironet, BPX, Catalyst, CCDA, CCDP, CCVP, CCIE, CCIP, CCNA, CCNP, CCSP, Cisco, the Cisco Certified Internetwork Expert logo, Cisco IOS, Cisco Press, Cisco Systems, Cisco Systems Capital, the Cisco Systems logo, Cisco Unity, EtherFast, EtherSwitch, Fast Step, Follow Me Browsing, FormShare, GigaDrive, HomeLink, Internet Quotient, IOS, iPhone, iQ Expertise, the iQ logo, iQ Net Readiness Scorecard, iQuick Study, LightStream, Linksys, Meeting Place, MGX, Networking Academy, Network Registrar, Packet, PIX, ProConnect, ScriptShare, SMARTnet, StackWise, The Fastest Way to Increase Your Internet Quotient, and TransPath are registered trademarks of Cisco Systems, Inc. and/or its affiliates in the United States and certain other countries.

All other trademarks mentioned in this document or Website are the property of their respective owners. The use of the word partner does not imply a partnership relationship between Cisco and any other company. (0705R)

Printed in the USA