

Cisco Unified Contact Center Express IPv6 Configuration

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Introduction

The purpose of this document is to describe the configuration of Cisco Unified Contact Center Express (Unified CCX), Cisco Unified Communications Manager (Unified Communications Manager), Cisco Unified Communications Manager IM and Presence Service (IM and Presence Service), Phones, Gateway, and Cisco Unified Border Element (CUBE) in a dual stack environment.

To test the various call flow combinations, change the common device configuration to V4 & V6, V6 only, and V4 only for phones. The trunk to Unified Communications Manager and IM and Presence Service is set as V4 & V6 for all combinations.

Design

For information about design considerations and guidelines to deploy Cisco Unified Contact Center Express, see:

- Cisco Unified Contact Center Express Design Guide, Release 10.5(1), IPv6 Support
- Cisco Unified Contact Center Express Design Guide, Release 10.5(1), Deployment Models chapter

Topologies

This section provides information about the Cisco Unified Contact Center Express deployment. In the test bed, various components were tested. The deployment adheres to principles and designs documented in the Cisco Unified Contact Center Express Design Guide, Release 10.5(1).

Component Deployment

Topologies

SIP Trunk **SIP Service** Provider 415 Unified CM PUB -----Unified IM and Presence Unified CUBE CCX A Service B and Pre 79xx, 69xx, 78x Nuance ASR/TTS CCX B Service A Unified CM SUB **Data Center B** 1 DE DM: **Data Center A** , by 町 99xx, 89xx, DX 650 MS Exchange 2010 Windows AD/DNS Unity Connection B Unity Cisco Agent Desktop, Connection Agent Cisco Agent Desktop, cialMine **Finesse Agents** 1 79xx, 69xx, 78xx 9 Media Sense XY ASR 1000 99xx, 89xx, DX 650 Jabber 29xx PSTN GW X **ASR 1000** - IP 29xx PSTN GW Internet MPLS WAN IP Phone with Delay Agent Home Route Phone VPN <u>i</u> isco Agent Desktop, 99xx, 89xx **Finesse Agent CVO VPN site** Site1 X 0 29xx PSTN GW 881 Route Jabbe 79xx, 69xx, 78xx Cisco Agent Desktop, **Finesse Agents** SIP PRI PSTN WAN BRI VPN

Cisco Unified Contact Center Express Deployment

Software Versions Used

The Unified CCX IPv6 deployment uses the following software versions for testing:

- Unified Communications Manager: 10.5.1.10000-7
- 2Unified CCX: 10.5.1.10000-20
- IM and Presence Service: 10.5.1.10000-9
- ☑Cisco IOS: 15.3(3)M2.9,15.3(3)M3

Call Flow Diagram

Customer Call > PRI Gateway > Voice Gateway or Unified Border Element > Unified Communications Manager > Unified CCX > Cisco Finesse Agent or Phone



Configuration

This section provides the high-level tasks and related information for configuring a Unified CCX deployment for IPv6:

The following table provides this information:

- Configure IOS
- Configure Unified CCX
- Configure Unified Communications Manager

Configure IOS

Step 1 Enable IPv6 on the gateway.

```
ip cef
ipv6 unicast-routing
ipv6 multicast rpf use-bgp
ipv6 cef
interface GigabitEthernet0/0
description "connected to PBL-DC1-c6506-gig3/14"
ip address 10.31.11.67 255.255.255.192
duplex auto
speed auto
ipv6 address 2001:10:31:11::67/122
ipv6 enable
ipv6 eigrp 23
```

no keepalive ! ipv6 router eigrp 23

Step 2 Enable dual stack on the gateway.

```
voice service voip
sip
call service stop
sip-ua
protocol mode dual-stack preference ipv6
voice service voip
sip
no call service stop
```

Step 3 Add Unified CM IPv6 and ANAT configuration in dial peer configuration.

```
dial-peer voice 32 voip
description automation connection of sip-trunk-to-uccx
translation-profile outgoing change-to-7-digit
preference 1
destination-pattern 0191660....
session protocol sipv2
session target ipv6:[2001:10:31:11::75]:5068 session transport udp/tcp
voice-class codec 4 [i.e. codec preference g711ulaw]
voice-class sip anat
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte h245-signal h245-alphanumeric
dtmf-interworking rtp-nte
no supplementary-service sip refer
!
```

Configure Unified CCX

For information on how to configure IPv6 for Unified CCX, see <u>Cisco Unified Communications Operating System Administration Guide for</u> <u>Cisco Unified CCX and Cisco Unified IP IVR, Release 10.5(1)</u>.

Configure Unified Communications Manager

Step 1 To enable IPv6 from Cisco Unified Operating System Administration, choose Settings > IP > Ethernet IPv6. Check the Enable IPv6 check box and enter IPv6 information. Click Save.

snow • Settings • :	Security ▼ Software Upgrades ▼ Services ▼ Help ▼
thernet Ipv6 Config	guration
Save	
Status	
🔥 Warning: Changir	ng the IPv6 ethernet settings with reboot option causes an immediate system restart
1Pv6 Information -	
🔽 Enable IPv6	
☑ Enable IPv6 Address Source	
☑ Enable IPv6 Address Source ○ Router Advertise	ment
 ✓ Enable IPv6 Address Source ○ Router Advertise ○ DHCP 	ment
 Enable IPv6 Address Source Router Advertise DHCP Manual Entry 	ment
 Enable IPv6 Address Source Router Advertise DHCP Manual Entry IPv6 Address 	ement 2001:10:31:11::75 Prefix Length 122
 Enable IPv6 Address Source Router Advertise DHCP Manual Entry IPv6 Address Default Gateway 	ement 2001:10:31:11::75 2001:10:31:11::65

Step 2 To add IPv6 to all servers in Unified Communications Manager cluster, from Cisco Unified CM Administrator, choose System > Server. Add IPv6 address to all Unified Communications Manager and IM and Presence Service nodes as shown in images.

System + Call Nothing + Media Nes	sources - Advanced Features - Device - Application -
Server Configuration	
🗐 Save 🗙 Delete 🛟 Add N	lew
Status	
1) Status: Ready	
- Server Information	CUCM Voice/Video
Database Replication	Publisher
같은 것	ccm-pub
Host Name/IP Address*	
Host Name/IP Address* IPv6 Address (for dual IPv4/IPv6)	2001:10:31:11::75
Host Name/IP Address* IPv6 Address (for dual IPv4/IPv6) MAC Address	2001:10:31:11::75
Host Name/IP Address* IPv6 Address (for dual IPv4/IPv6) MAC Address Description	2001:10:31:11::75
Host Name/IP Address* IPv6 Address (for dual IPv4/IPv6) MAC Address Description	2001:10:31:11::75
Host Name/IP Address* IPv6 Address (for dual IPv4/IPv6) MAC Address Description	2001:10:31:11::75

Cisco Unified Contact Center Express IPv6 Configuration

Configuration

Server Configuration	
- Status	
- Server Information	
Server Type	CUCM IM and Presence
Database Replication	Publisher
Fully Qualified Domain Name/IP Address*	uccx-cup-a
IPv6 Address (for dual IPv4/IPv6)	2001:10:31:11::78
Description	uccx-cup-a
- IM and Presence Server Information Presence Redundancy Group <u>DefaultCUPS</u> Assigned Users <u>13 users</u> <u>Presence Server Status</u>	ubcluster

Step 3 To set the media and signaling preferences, choose **System > Enterprise Parameters**. Complete **IPv6** information as shown in image.

ahaha Cisco Unif	ed CM Administration				Navig
CISCO For Cisco Unifie	d Communications Solutions				ccmadministra
System - Call Routing - Med	a Resources 👻 Advanced Features 👻	Device - Application -	User Management 👻	Bulk Administration 👻	Help 👻
Enterprise Parameters Con	figuration				
Save 🧬 Set to Default	🎦 Reset 🛛 🧷 Apply Config				
Cisco Support Use 1					-
Cisco Support Use 2					
Enable IPv6 *		True		,	✓ False
IP Addressing Mode Preferer	ce for Media_*	IPv6			▼ IPv4
IP Addressing Mode Preferen	ce for Signaling *	IPv6		-	✓ IPv4

Step 4 To create common device configuration, choose **Device > Device Settings > Common Device Configuration**. Complete **Common Device Configuration** as shown in image.

CISCO Cisco Unified CM Adminis For Cisco Unified Communications S	stration olutions	
System - Call Routing - Media Resources - Advance	ed Features 👻 Device 👻 Application 👻	User Management 👻 Bulk Administration 👻
Common Device Configuration		
🔚 Save 🗶 Delete 🗋 Copy 🎦 Reset 🧷 /	Apply Config 🔓 Add New	
- Status		
G Status: Ready		
.		
— Common Device Configuration Information —		
Common Device Configuration: v4andv6 (29 mem	bers**)	
- Common Device Configuration Information		
Name*	v4andv6	
Softkey Template	Not Selected	•
User Hold MOH Audio Source	< None >	
Network Hold MOH Audio Source	< None >	
User Locale	< None >	•
IP Addressing Mode*	IPv4 and IPv6	
IP Addressing Mode Preference for Signaling*	Use System Default	

Step 5 Associate **Common Device Configuration** to phones, CTI route point, trunk, CTI ports, and so on. Complete **Device Information** as shown in image.

cis	Cisco Unified CM Administ For Cisco Unified Communications So	tration lutions		Navigation Cisco L ccmadministrator Sear
Systen	n ▼ Call Routing ▼ Media Resources ▼ Advanced	Features 🕶 Device 👻 Application 👻 User	Management 👻 Bulk Administration 👻 Help	•
Phone	e Configuration			Related Links: Back To Find/List
:	Save 💥 Delete 🗋 Copy 🎦 Reset 🧷 Ap	ply Config 🕂 Add New		
7 8 9 10 11 12 13 14 15 16 17 18 19	All Calls Answer Oldest ************************************	Device Information	A0CF5B80E450 phone DP_A v6only v6only v4andv6 v4andv6 v4andv6 v4andv6 v4andv5 v6only Standord Common Phone Profile NormalUserCSS < None > MRGL_SideA	View Details View Details View Details View Details View Details View Details

Step 6 From the Cisco Unified IM and Presence Operating System Administration page, check the **Enable IPv6** check box. Then click **Manual Entry** and enter the **IPv6 Address** for IM and Presence Service as shown in image.

Note: If DHCP is configured, you can use it as the Address Source.

CISCO For Cisco	Unified IM and Pr Unified Communications	Solutions	iting System Ad	ministration
Show 👻 Settings 👻 S	ecurity 👻 Software Upgrades	▼ Services ▼ Help	•	
Ethernet Ipv6 Config	uration			
Save				
– Status – Warning: Changin	g the IPv6 ethernet settings	with reboot option cau	ises an immediate system	restart
– IPv6 Information –				
🔽 Enable IPv6				
Address Source				
🔘 Router Advertiser	nent			
O DHCP				
Manual Entry				
IPv6 Address	2001:10:31:11::78		Prefix Length	122
Default Gateway	2001:10:31:11::65			
🕅 Update with Reboot				
Save				

Step 7 To create and enable ANAT SIP profile for Unified Communications Manager, choose **Device > Device Settings > SIP Profile**. Complete **SIP Profile Information** as shown in image.

Save 🗙 Delete 🗋 Copy 資 Rese	t 🥒 Apply Config 🕂 /	Add New	
SIP Profile Information			
Name"	ANAT SIP Profile		
Description	Default SIP Profile		
Default MTP Telephony Event Payload Type*	101		
Early Offer for G.Clear Calls*	Disabled		
User-Agent and Server header information*	Send Unified CM Version	Information as User-Age	r 🔻
Version in User Agent and Server Header*	Major And Minor		
Dial String Interpretation*	Phone number consists of	f characters 0-9, *, #, an	ic 🔻
Confidential Access Level Headers*	Disabled		•
Redirect by Application			
Diaphle Early Media on 190			
Outgoing T.38 INVITE include audio mline			
 Disable Early Media on 180 Outgoing T.38 INVITE include audio mline Use Fully Qualified Domain Name in SIP R 	lequests		
 Disable Early Hedia on 180 Outgoing T.38 INVITE include audio mline Use Fully Qualified Domain Name in SIP R Assured Services SIP conformance 	lequests		
Outgoing T.38 INVITE include audio mline Use Fully Qualified Domain Name in SIP R Assured Services SIP conformance	lequests		
Disable Early Media on 180 Outgoing T.38 INVITE include audio mline Use Fully Qualified Domain Name in SIP R Assured Services SIP conformance SDP Information SDP Session-level Bandwidth Modifier for Ea	lequests arly Offer and Re-invites*	TIAS and AS	
Disable Early Media on 180 Outgoing T.38 INVITE include audio mline Use Fully Qualified Domain Name in SIP R Assured Services SIP conformance SDP Information SDP Session-level Bandwidth Modifier for Ea	equests arly Offer and Re-invites*	TIAS and AS Pass all unknown SDP at	tributes
Outgoing T.38 INVITE include audio mline Outgoing T.38 INVITE include audio mline Use Fully Qualified Domain Name in SIP R Assured Services SIP conformance SDP Information SDP Session-level Bandwidth Modifier for Ea SDP Transparency Profile Accept Audio Codec Preferences in Received	equests arly Offer and Re-invites*	TIAS and AS Pass all unknown SDP at Default	tributes
Outgoing T.38 INVITE include audio mline Outgoing T.38 INVITE include audio mline Use Fully Qualified Domain Name in SIP R Assured Services SIP conformance SDP Information SDP Session-level Bandwidth Modifier for Ea SDP Transparency Profile Accept Audio Codec Preferences in Received Require SDP Inactive Exchange for Mid.	arly Offer and Re-invites* d Offer*	TIAS and AS Pass all unknown SDP at Default	tributes

🚽 Save X Delete 🗋 Copy Reset 🍐	🖉 Apply Config 💾 Add New	
IP Rel1XX Options*	Disabled	-
ideo Call Traffic Class*	Mixed	
alling Line Identification Presentation*	Default	
ession Refresh Method*	Invite	
arly Offer support for voice and video calls st	Disabled (Default value)	-
Deliver Conference Bridge Identifier Allow Passthrough of Configured Line Device (Reject Anonymous Incoming Calls Reject Anonymous Outgoing Calls	Caller Information	
Deliver Conference Bridge Identifier Allow Passthrough of Configured Line Device (Reject Anonymous Incoming Calls Reject Anonymous Outgoing Calls Send ILS Learned Destination Route String	Caller Information	
 Deliver Conference Bridge Identifier Allow Passthrough of Configured Line Device (Reject Anonymous Incoming Calls Reject Anonymous Outgoing Calls Send ILS Learned Destination Route String SIP OPTIONS Ping Enable OPTIONS Ping to monitor destination 	Caller Information status for Trunks with Service Type	"None (Default)"
Deliver Conference Bridge Identifier Allow Passthrough of Configured Line Device (Reject Anonymous Incoming Calls Reject Anonymous Outgoing Calls Send ILS Learned Destination Route String SIP OPTIONS Ping Enable OPTIONS Ping to monitor destination Ping Interval for In-service and Partially In-serv	Caller Information status for Trunks with Service Type ice Trunks (seconds)* 60	"None (Default)"
Deliver Conference Bridge Identifier Allow Passthrough of Configured Line Device (Reject Anonymous Incoming Calls Reject Anonymous Outgoing Calls Send ILS Learned Destination Route String SIP OPTIONS Ping Enable OPTIONS Ping to monitor destination Ping Interval for In-service and Partially In-serv Ping Interval for Out-of-service Trunks (seconds	Caller Information status for Trunks with Service Type ice Trunks (seconds)* 60)*	"None (Default)"
Deliver Conference Bridge Identifier Allow Passthrough of Configured Line Device (Reject Anonymous Incoming Calls Reject Anonymous Outgoing Calls Send ILS Learned Destination Route String SIP OPTIONS Ping SIP OPTIONS Ping to monitor destination Ping Interval for In-service and Partially In-serv Ping Interval for Out-of-service Trunks (seconds Ping Retry Timer (milliseconds)*	Status for Trunks with Service Type ice Trunks (seconds)*)* 120 500	"None (Default)"

Step 8 To associate the ANAT SIP profile to the trunk, from Trunk Configuration page, add the V4 and V6 IPs of the gateway or Unified Border Element to the trunk in Unified Communications Manager as shown in image. Also, choose the **ANAT SIP Profile** from the **SIP Profile** drop-down list.

Related Documentation

🔜 Save 🗶 Delete 省 Reset 🛟 A	dd New					
SIP Information						
– Destination –						
Destination Address is an SRV						
Destination Address	i .	Destination Ad	ldress IPv6		Destination Port	Status Stat
1* 10.31.11.67		2001:10:31:11::67			5060	up
MTP Preferred Originating Codec*	711ulaw		v	1		
BLF Presence Group*	Standard Presence group		1			
SIP Trunk Security Profile*	Non Secure SIP Trunk Profile-5068		Ĩ			
Rerouting Calling Search Space	< None >					
Out-Of-Dialog Refer Calling Search Space	< None >		1			
SUBSCRIBE Calling Search Space	< None >		÷	1		
SIP Profile*	ANAT SIP Profile		÷	View Details		
DTMF Signaling Method *	No Preference		1			
Normalization Script						
Normalization Script < None >		•				
Enable Trace						
Parameter Name		Parameter	Value			
-						

Related Documentation

- <u>Cisco Unified Contact Center Express Design Guide, Release 10.5(1)</u>
- <u>Cisco Unified Communications Operating System Administration Guide for Cisco Unified CCX and Cisco Unified IP IVR, Release</u> 10.5(1)

Obtaining Documentation and Submitting a Service Request

For information on obtaining documentation, using the Cisco Bug Search Tool (BST), submitting a service request, and gathering additional information, see *What's New in Cisco Product Documentation* at: http://www.cisco.com/c/en/us/td/docs/general/whatsnew/whatsnew.html.

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