IPv6 Deployment Guide

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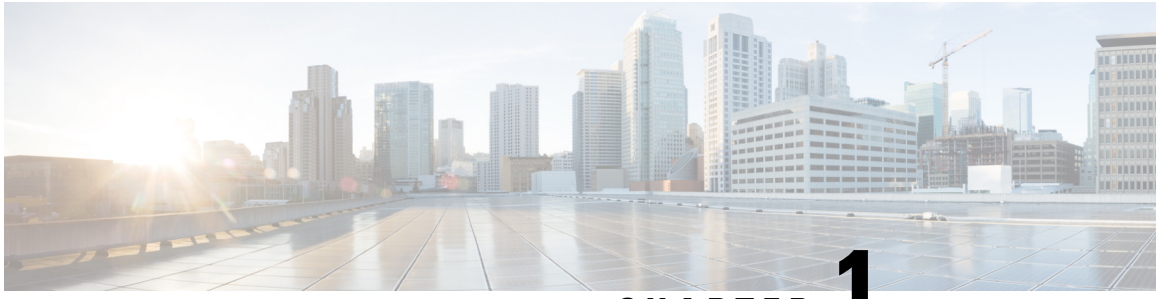


CHAPTER 1

IPv6 Introduction

Documentation Changes

Table 1: Documentation Changes

Date	Change	Link to Topic
November 2023		-

Date	Change	Link to Topic
	The entire guide is updated. The following are the updates to the guide. A few to mention - • Table 8 is updated. • Deleted “for DOD Network only” • Replaced “Recommended” to “Supported” • Removed “CSR 12” and “CSR 14” references • Removed release numbers across the guide to keep it generic • Removed occurrences of “Solution Tested” • Deleted all Deployment Gaps and created a new list • Jabber on IPv6, Expressway-E external on IPv6, Expressway-E internal interface on IPv4, UCM, IM&P, CUC on IPv4 are supported deployments • **Dual-stack:** Used for application servers, such as Cisco Unified Communications Manager, Cisco Unity Connection, Cisco Emergency Responder, and Cisco Unified Survivable Remote Site Telephony (Unified SRST) • Table 8 and Table 10: Titles “Supported” instead of “Recommended”, remove “Solution Tested" column • Note: IPv6 NAT-based solution is not supported • DHCP v6 modify the text accordingly • MRA: Jabber on IPv6, Expressway-E external on IPv6, Expressway-E internal interface on IPv4, UCM, IM&P, CUC on IPv4 are supported deployments • The SIP Trunk configuration settings discussed in this section are applied through the Common Device Configuration profile that is created and assigned to the SIP trunk (IP Addressing	

Date	Change	Link to Topic

	Mode and IP Addressing Mode Preference for Signaling), and through the SIP Profile configuration assigned to the SIP trunk (Enable ANAT should be disabled). Change “Enable ANAT should be disabled”. This is confusing. Instead, mentioned "ANAT is not enabled."
	• Removed the following from ANAT section:
	  • Media Selection for Outbound Early Offer Calls with ANAT
	  • Media Selection for Inbound Early Offer Calls with ANAT
	  • Media Selection for Delayed Offer Calls Over Unified CM SIP Trunks with ANAT
	  • Media Selection for Outbound Delayed Offer Calls with ANAT
	  • Inbound Delayed Offer Calls with ANAT
	  • Inbound Delayed Offer Calls with ANAT and Supported: sdp anat
	  • Inbound Delayed Offer Calls with ANAT and Require: sdp-anat
	• Included this: The SIP trunk uses either the Unified CM IPv4 address or the Unified CM IPv6 address for signaling, and an MTP. We recommend IPv6 Only for production environments.
	Removed the media part. Instead mentioned, MTP or other trunk address for media.
	Modified to: Outbound SIP best effort early offer (without MTP/TRP) in SIP trunk as we don’t support in IPv6 mode.
	• Removed this Note: In CSR 12.0, SIP Early Offer does not support IPv6. CUBE or Unified SRST gateways will not support ANAT.	

Date	Change	Link to Topic
	• The following changes are made to the section "IPv6-Only Phones": • Existing content is removed. Replaced with the following: IPv6 only Supported Features: Mobility – MVA/IVR, CAPFv3, TVS and TFTP, Extension Mobility, EMCC, SRST, SIP GW, SIP Route Pattern, Barge, Intercom, Phone NTP Reference, Device Mobility, Self Care Portal, Self Provisioning using with and without IVR, Phone services - Corporate Directory, Missed Calls, Placed Calls, Intercom Calls, Personal Directory, Voicemail, Received Calls, Web Dialer, Quality Report Tool (QRT), SNMPv6 MIB, Phone Config. download, CTI, IPVMS (Unicast MoH, software MTP, ANN, Conference Bridge, IVR), CDR IPv6-Only Not Supported Features • Best Effort Early Offer over SIP Trunk • External MoH, Multicast MoH, Video On Hold, Voicemail, Recording (BIB Recording, GW Recording), Enhanced E911 • SSH, Cluster Manager, LDAP, Syslog Server/Alarm Library, Serviceability Applications - IPMA, RTMT • DRS, IPSec • Media Resources (IOS Xcoder/ hardware MTP/TRP)/Conference Bridge), Video conference resource (MCU/Conductor), FAX/ATA • MRA (Jabber in IPv6 and Expressway-E external interface in v6 and enterprise	

Date	Change	Link to Topic
	network in v4 is supported) • Silent Monitoring, Whisper Coaching, Agent Greeting, VG as analog line interface using SCCP (pending test) • Removed ISR 4000 references • IPv6 endpoints should be configured as IPv6 with preference • Dual Stack with preference IPv6 • Removed "with no telephony services provided over an IP WAN." • Remove CSR 12.0 • Addressed CDET "CSCwh69667" • Removed SRST 12.0 • Table 8: Added row for "Cisco Room Series, Board Series, and Desk Series". All "Yes" • Change 7832 - "Yes" • Removed row Wi-Fi enabled devices: Cisco IP Phone 8861, 8865 • The first row "Dual- Stacks: IPv4 and IPv6" is "Yes" for all 78xx and 88xx. Added the following phones to this row: 7811, 7821, 7841, 7861 • For 7832 and 8832: YES to Dual-Stack • For 7832 - "Yes" in 8861 all columns • Removed the entire row that says "Cisco Unified IP Phone 6901, 7945G, 7965G, 7975G …" It is End of Support • Removed "Cisco DX70 and DX80" • Removed row "Cisco TelePresence MX Series, SX Series" • Removed row "Cisco TelePresence System EX Series" • Removed row "Cisco Webex Share" • Remove row "Cisco IP Communicator"	

Date	Change	Link to Topic
	• Remove row Wi-Fi enabled devices: Cisco IP Phone 8861, 8865 • Document direction • Removed "We recommend Stateful DHCP host configuration for both IPv4 or IPv6 IP phones." • Remove dual-stack IP Phones that require ANAT, which is not supported in Unified SRST call agents • Removed the "Security and IPv6 Traffic" section • Removed ISR 4000 G3 • Include row for "Cisco Room Series, Board Series, and Desk Series". All "Yes" • Removed "Cisco ATA 190 Series Analog Telephone Adapters"	
June 08, 2022	Updated the table "Table 9: Supported Addressing Modes Communication Gateways".	Supported IPv6 Addressing Modes
April 01, 2021	Removed redundant topics from *Call Processing and Call Admission Control* chapter.	-
April 10, 2019	Changed book title to include Cisco Collaboration System Release.	-

Date	Change	Link to Topic
April 30, 2018	Updated overview based on Cisco Collaboration Systems Release.	IPv6 Deployment Overview
	Updated deployment gaps.	Deployment Recommendations for Enterprise Networks
	Added a note about IPv6 multicast application.	IPv6 Multicast Addresses, on page 19
	• Updated the supported IPv6 addressing modes for endpoints. • Added Cisco Meeting Server and Cisco TelePresence Management Suite.	IP Addressing Modes
	Updated the note.	IPv6 Bandwidth Provisioning
	• Added Cisco Meeting Server and Cisco TelePresence Management Suite in Video Conferencing section. • Added Multistream video section. • Updated Unified CM IM and Presence Service section to include dual-stack.	Applications Overview
	Added: • Cisco TelePresence EX Series • Cisco Meeting Server • Cisco TelePresence Management Suite	Product Configuration Resources for IPv6

IPv6 Deployment Overview

This document describes our recommendations on how to transition your Cisco Collaboration network design to use IPv6 in a dual-stack (IPv4 and IPv6) environment.

This document does not describe how to implement IPv6 in the campus and WAN, but does refer you to other documents for those details. It is assumed that data IPv6 desktop services are deployed first, for example internet access when Collaboration IPv6 upgrade is initiated in a dual-stack mode.

Application Server Addressing Mode

All Collaborations applications servers are deployed in IPv4 and IPv6 stacks to support IPv4 or IPv6 devices and components.

Branch Office Addressing Mode

In Cisco Collaboration Systems Release, a branch office can be in one of the following IP addressing modes. A cluster can support the following types of IP addressing modes and all can co-exist.

- Traditional IPv4-only configuration where the LAN is IPv4-only or dual-stack.
- IPv6-only configuration where the LAN is dual-stack.
- Dual-stack sites that support IPv4 stack or IPv6 stack devices, where the LAN is dual-stack.

Provides IPv6-only site or branch offices that can have the following components:

- Cisco Integrated Services Routers (ISR) Edge Routers
  - PSTN GW
  - SRST
  - Audio Conference
  - MTP
- Endpoints
  - Video TP CE (Cisco DX Series, Cisco TelePresence MX, SX, and EX and Series)
  - Cisco IP Phone 7800 and 8800 Series
  - Conference IP Phone (8832)
  - On-premise Cisco Jabber (Jabber)
  - ATA FAX

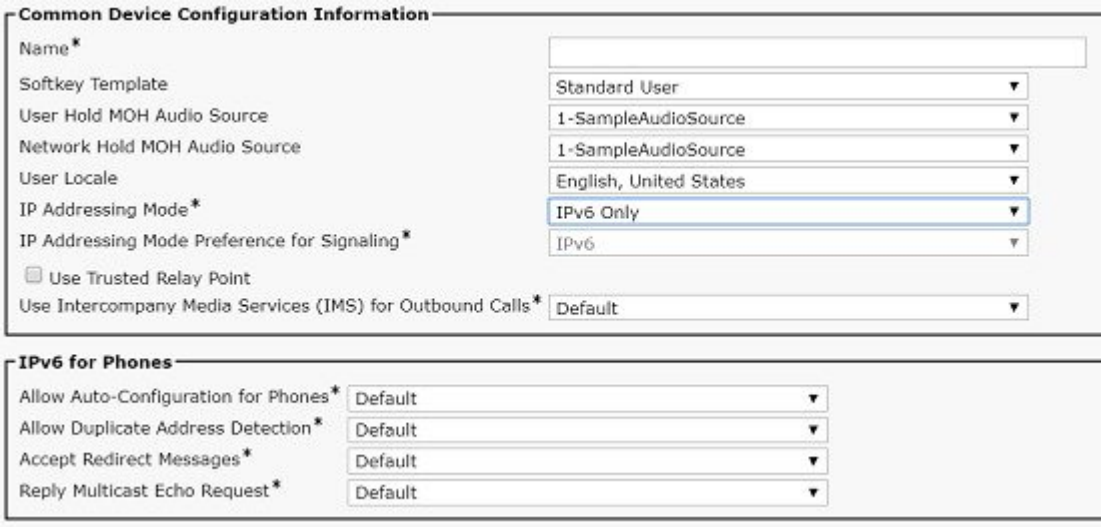
We strongly recommend that you use this document with following guides that provide in-depth guidance on Collaboration deployments using IPv4:

- *Cisco Collaboration System Solution Reference Network Design (SRND)*, available at: Link .
- *Cisco Preferred Architecture for Enterprise Collaboration 11.6 Design Overview*, available at: Link

Cisco Collaboration IPv6 transition follows the Cisco Preferred Architecture (PA) recommended deployment of IPv4 products:

- Cisco Unified Communications Manager (Unified CM) is the call control server.
- Cisco IP Phones, Jabber clients, and Cisco TelePresence video endpoints use SIP to register directly to Unified CM.
- Unified CM cluster's failover mechanism provides endpoint registration redundancy. If a WAN failure occurs, and endpoints at remote locations cannot register to Unified CM, they use SRST functionality for local and PSTN calls. However, some services such as Jabber presence might not be available.

Note As of Cisco Collaboration Systems Release, some default settings have not been adjusted for IPv6-only as a default, so follow the directions in this guide to ensure that you use the recommended settings for IPv6 deployment. Also some labels in the Admin Configuration UI have not been updated, but changes are noted in this document. For example, the Allow Auto-Configuration for Phones label has not been updated to Allow Stateless Auto-Configuration for Phones.



Move Toward IPv6-Only Network

In Cisco Collaboration Systems Release, the deployment of IPv6-only stack devices reduces dependency on IPv4 addresses. For all applications servers, emphasis is given to the deployment of dual-stack (IPv4 and IPv6) applications servers. Dual-stack applications servers offer a greater degree of functionality and interoperability with existing traditional IPv4-only devices that may not support IPv6-only stack because they are not developed.

The ultimate goal is for all collaboration endpoints, servers, and gateways to be configured as IPv6-only stack, meaning that they can provide collaboration functions without using any IPv4 addresses and they do not require IPv4 to IPv6 media interworking functions.

Table 2: Reduction in IPv4 Addresses

Scale of Deployment (Endpoints)	Servers and Gateways (Dual-stack or IPv4)	IPv4 Addresses Before Deploying IPv6-only Endpoints	IPv4 Addresses After Deploying IPv6-only Endpoints	Percent Reduction in IPv4 Addresses
500	6	506	6	98.8%
5,000	13	5,013	13	99.7%
10,000	25	10,025	25	99.8%

Note The scope of this document is limited to the solutions approved by Cisco. Supported IPv4-only stack applications are limited to those stated in this document. It is assumed that IP phones and gateways are configured with either an IPv4-only stack or IPv6-only stack using SIP signaling. Supported application servers support IPv4-only and IPv6-only stacks (dual-stack). Any applications that we did not develop or test are not supported by Cisco TAC. In IPv6 supported network configurations, Skinny Client Control Protocol (SCCP) for IP phones, voice gateways (Cisco VG Series Gateways), and Cisco Unity Connection are not configured except for the use of media termination points (MTP) by Cisco Unified Communications Manager.

Deployment Recommendations for Enterprise Networks

IPv4-only stack or IPv6-only stack devices support:

- Single-site call processing deployments.
- Multi-site distributed call processing deployments.
- Multi-site deployments with centralized call processing.

To move toward an IPv6 deployment and reduce dependency on IPv4 addresses, we recommend that you deploy:

- IPv6-only stack SIP phones, SIP gateways, and SIP trunks.
- Dual-stack Cisco Unified Communications Manager (Unified CM) clusters and other application servers.

Deployment Gaps

When you transition from a traditional IPv4 network to an IPv6 dual-stack network deployment, the following functionality is not supported.

- IPv6-only stack site or branch office (LAN)
- IPv6-only data center
- All IP phones with SCCP signaling
- Secure IP phones, CE endpoints, and gateways
- Applications based on ISR and ISR G2 Routers
- IP phone VPN
- IP phones with NTP
- Video conference deployment based on Cisco TelePresence Conductor, Cisco TelePresence MCU, and Cisco TelePresence Server
- Mobile Remote Access
- Off-premise Cisco Jabber
- Cisco Meeting Server

- Cisco TelePresence Management Suite
- SCCP signaling configuration IP phones
- RSVP call admission control
- Cisco Spark Hybrid Services
- Third-party API products
- Media Gateway Control Protocol (MGCP) gateways
- Unified CM supported H.323 gateways
- Products that are End-of-Support
- Cisco Jabber on IPv6
- Expressway-E external on IPv6
- Expressway-E internal interface on IPv4
- Unified CM , IM & Presence, CUC on IPv4

Comparison of IPv4 and IPv6

This section provides a brief description of the motivation behind deploying IPv6, and a summary comparison of IPv4 and IPv6.

Why Deploy IPv6?

The deployment of IPv6 is primarily driven by IPv4 address exhaustion. As the worldwide usage of IP networks increases, the number of applications, devices, and services requiring IP addresses is rapidly increasing. According to the Internet Assigned Numbers Authority (IANA) and Regional Internet Registries, their pool of unallocated IPv4 addresses is exhausted. For example, all area wireless networks in the American Registry for Internet Numbers (ARIN) are IPv6-only stack.

Because the current IPv4 address space is unable to satisfy the potential huge increase in the number of users and the geographical needs of the Internet expansion, many companies are either migrating to or planning their migration to IPv6, which offers a virtually unlimited supply of IP addresses.

Transforming the Internet from IPv4 to IPv6 is likely to take several years. During this period, IPv4 will co-exist with and then gradually be replaced by IPv6.

Advantages of IPv6 Over IPv4

As a new version of the Internet Protocol, IPv6 provides the following advantages over IPv4:

- Larger address space (Supported in Cisco Collaboration Systems Release)

  The main feature of IPv6 that is driving adoption today is the larger address space. Addresses in IPv6 are 128 bits long compared to 32 bits in IPv4. The larger address space avoids the potential exhaustion of the IPv4 address space without the need for network address translation (NAT) or other devices that break the end-to-end nature of Internet traffic. By avoiding the need for complex sub-netting schemes,

IPv6 addressing schemes are easier to understand, making administration of medium and large networks simpler.

- Address scopes (Supported)

  IPv6 introduces the concept of address scopes. An address scope defines the region, or span, where an address can be defined as a unique identifier of an interface. These spans are the link, the site network, and the global network, corresponding to link-local, site-local (or unique local unicast), and global addresses.

- Stateless Address Auto-Configuration (SLAAC) (Supported)

  IPv6 hosts can be configured automatically when connected to a routed IPv6 network using ICMPv6 router discovery messages. Address reconfiguration is also simplified. If IPv6 auto-configuration is not suitable, a host can be configured manually.

- Multicast (**Not** supported)

  Multicast is part of the base specifications in IPv6, unlike IPv4, where it was introduced later. Like IPv6 unicast addresses, the IPv6 multicast address range is much larger than that of IPv4. IPv6 does not have a link-local broadcast facility; the same effect can be achieved by multicasting to the all-hosts group address (FF02::1).

- Streamlined header format and flow identification (**Not** supported)

  The IPv6 header format reduces router processing overhead by using a fixed header length, performing fragmentation on hosts instead of routers, and using an improved header extension method and a new flow label to identify traffic flows requiring special treatment.

- Mobile IPv6 (**Not** supported)

  Mobile IPv6 allows a mobile node to change its locations and addresses, while maintaining a connection to a specific address that is always assigned to the mobile node and through which the mobile node is always reachable. Mobile IPv6 provides transport layer connection survivability when a node moves from one link to another by performing address maintenance for mobile nodes at the Internet layer.

Note Mobile IPv6 is not supported by Cisco IP Phones or other collaboration components.

- Network-layer security (**Not** supported)

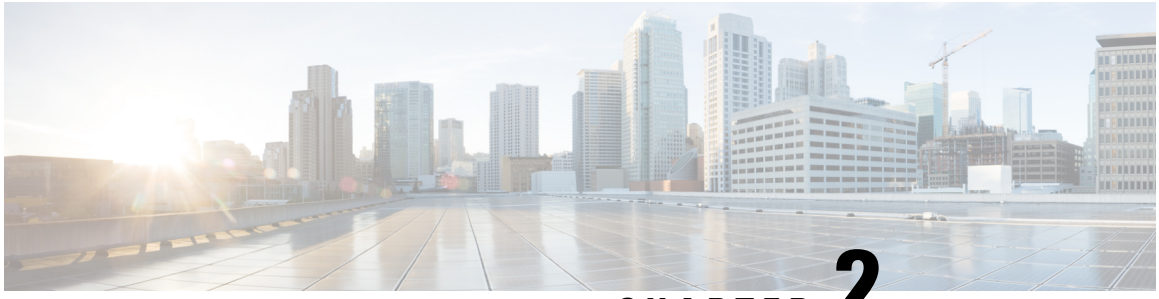
  IPsec, the protocol for IP network-layer encryption and authentication, is part of the base protocol suite in IPv6. This is unlike IPv4, where IPsec is optional. Because of its reduced payload and performance overhead, products use TLS and SRTP for authentication and encryption.

The following table summarizes the differences between IPv4 and IPv6 services.

Table 3: IPv4 and IPv6 Services

IP Service	IPv4 Feature	IPv6 Feature
Address range	32-bit, NAT	128-bit, multiple scopes
Auto-configuration	DHCP	Stateless, Easy Reconfiguration, DHCP

IP Service	IPv4 Feature	IPv6 Feature
Routing	RIP, OSPFv2, IS-IS, EIGRP, MP-BGP	RIPng, OSPFv3, IS-IS, EIGRP, MP-BGP
IP Security	IPsec	IPsec
Mobility	Mobile IP	Mobile IP with direct routing
Quality of Service (QoS)	Differentiated Service, Integrated Service	Differentiated Service, Integrated Service
IP multicast	IGMP, PIM, and Multicast BGP	MLD, PIM, and Multicast BGP; Scope Identifier



CHAPTER 2

IPv6 Basics

IPv6 Basics Overview

This chapter provides an introduction to the basics of IPv6 addressing, including the various address types, address assignment options, new DHCP features, and DNS. For further reading on IPv6, you can refer to the following documentation:

- *IPv6 Fundamentals: A Straightforward Approach to Understanding IPv6, 2nd Edition*, a Cisco Press publication available as a: book or video.
- *IPv6 Design and Deployment LiveLessons*, a Cisco Press publication available through Link
- Other Cisco online documentation at Link.

IPv6 Addressing

An IPv6 address consists of 8 sets of 16-bit hexadecimal values separated by colons (:), totaling 128 bits in length. For example:

```
2001:0db8:1234:5678:9abc:def0:1234:5678
```

You can omit leading zeros. For consecutive zeros in contiguous blocks, you can use a double colon (::). Double colons can appear only once in the address.

For example, here is the complete address:

```
2001:0db8:0000:130F:0000:0000:087C:140B
```

Here is the abbreviation:

```
2001:0db8:0:130F::87C:140B
```

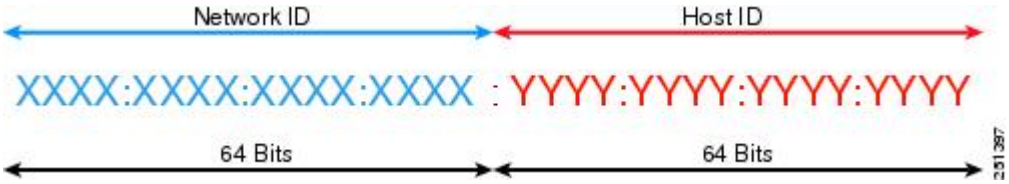
As with IPv4 addresses, you can represent an IPv6 address network prefix the same way:

```
2001:db8:12::/64
```

IPv6 Unicast Addresses: Network and Host IDs

IPv6 unicast addresses generally use 64 bits for the network ID and 64 bits for the host ID.

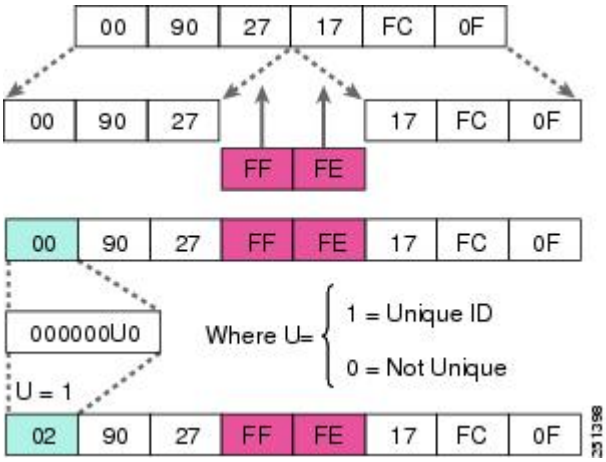
Figure 1: Format for IPv6 Unicast Network ID and Host ID



The network ID is administratively assigned. The host ID is configured manually or auto-configured by any of the following methods:

- Using a randomly generated number.
- Using DHCPv6.
- Using the Extended Unique Identifier (EUI-64) format. We commonly use the EUI-64 host ID format for devices such as Cisco IP Phones, gateways, and routers. This format, as illustrated, expands the device interface 48-bit MAC address to 64 bits by inserting FFFE into the middle 16 bits.

Figure 2: Conversion of EUI-64 MAC Address to IPv6 Host Address Format



Types of IPv6 Addresses

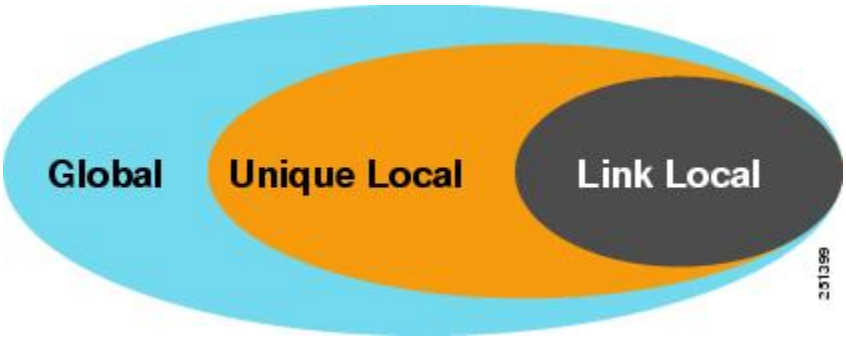
As with IPv4, IPv6 addresses are assigned to interfaces. However, unlike IPv4, an IPv6 interface is expected to have multiple addresses. There are several types:

- Unicast address—Identifies a single node or interface. Traffic destined for a unicast address is forwarded to a single interface.
- Multicast address—Identifies a group of nodes or interfaces. Traffic destined for a multicast address is forwarded to all the nodes in the group.
- Anycast address—Identifies a group of nodes or interfaces. Traffic destined to an anycast address is forwarded to the nearest node in the group. An anycast address is essentially a unicast address assigned to multiple devices with a host ID = 0000:0000:0000:0000. (Anycast addresses are not widely used today.)

With IPv6, broadcast addresses are no longer used. Broadcast addresses are too resource intensive, so IPv6 uses multicast addresses instead.

Address Scopes

An address scope defines the region where an address can be defined as a unique identifier of an interface. These scopes or regions are the link, the site network, and the global network, corresponding to link-local, unique local unicast, and global addresses.

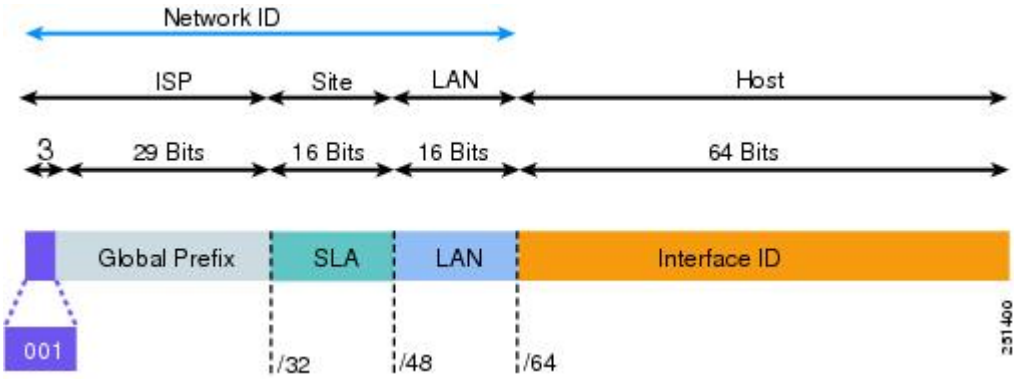


Global Unicast Addresses

Global unicast addresses are:

- Routable and reachable across the Internet.
- IPv6 addresses for widespread generic use.
- Structured as a hierarchy to allow address aggregation.
- Identified by their three high-level bits set to 001 (2000::/3).

Figure 3: Format for Global Unicast Addresses



The global routing prefix is assigned to a service provider by the Internet Assigned Numbers Authority (IANA). The site level aggregator (SLA), or subnet ID, is assigned to a customer by their service provider. The LAN ID represents individual networks within the customer site and is administered by the customer.

The Host or Interface ID has the same meaning for all unicast addresses. It is 64 bits long and is typically created by using the EUI-64 format.

Example:

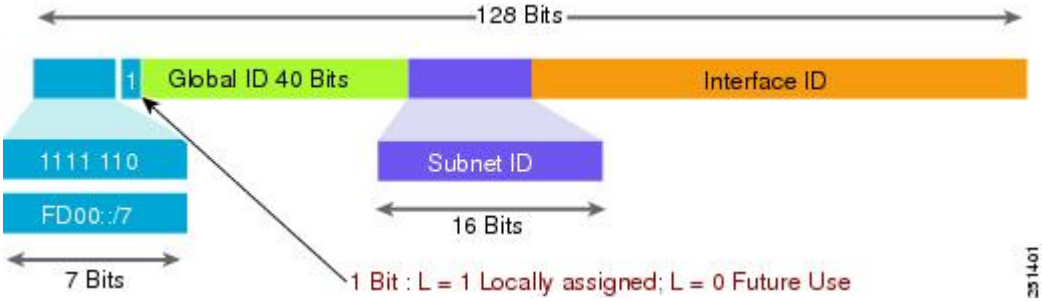
```
2001:0DB8:BBBB:CCCC:0987:65FF:FE01:2345
```

Unique Local Unicast Addresses

Unique local unicast addresses are:

- Analogous to private IPv4 addresses (for example, 10.1.1.254)
- Used for local communications, inter-site VPNs, and so forth
- Not routable on the Internet (routing would require IPv6 NAT)

Figure 4: Format of Local Unique Address



Global IDs do not have to be aggregated and are defined by the administrator of the local domain. Subnet IDs are also defined by the administrator of the local domain. Subnet IDs are typically defined using a hierarchical addressing plan to allow for route summarization.

The Host or Interface ID has the same meaning for all unicast addresses. It is 64 bits long and is typically created by using the EUI-64 format.

Example:

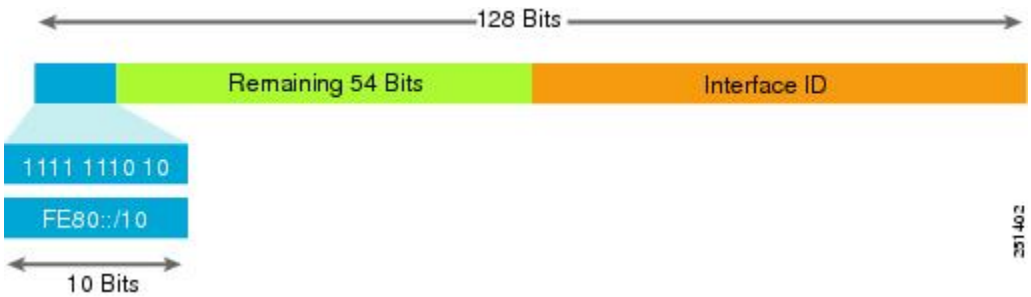
```
FD00:aaaa:bbbb:CCCC:0987:65FF:FE01:2345
```

Link Local Unicast Addresses

Link local unicast addresses are:

- Mandatory addresses that are used exclusively for communication between two IPv6 devices on the same link.
- Automatically assigned by device when IPv6 is enabled.
- Not routable addresses. Their scope is link-specific only.
- Identified by the first 10 bits (FE80).

Figure 5: Format of Link Local Unicast Address



The remaining 54 bits of the network ID could be zero or any manually configured value. The interface ID has the same meaning for all unicast addresses. It is 64 bits long and is typically created by using the EUI-64 format.

Example:

```
FE80:0000:0000:0000:0987:65FF:FE01:2345
```

This address would generally be represented in shorthand notation as:

```
FE80::987:65FF:FE01:2345
```

IPv6 Multicast Addresses

IPv6 multicast addresses have an 8-bit prefix, FF00::/8 (1111 1111). The second octet defines the lifetime and scope of the multicast address.

Figure 6: Multicast Address Format

Multicast addresses are always destination addresses. Multicast addresses are used for router solicitations (RS), router advertisements (RA), DHCPv6, multicast applications, and so forth.

Note IPv6 clients do not require a default gateway configuration because routers are discovered using RSs and RAs.

Table 4: Common Multicast Addresses

Address	Scope	Meaning
FF01::1	Node-local	Same node
FF02::1	Link-local	All nodes on a link
FF01::2	Node-local	Same router
FF02::2	Link-local	All routers on a link
FF05::2	Site-local	All routers on the Internet
FF02::1:FFxx:xxxx	Link-local	Solicited node

For more information on IPv6 multicast addresses, refer to the IANA documentation: Link.

Note IPv6 multicast application is not developed in Collaboration products, and is not supported industry wide through service provider networks.

Address Assignment for IPv6 Devices

Addresses can be assigned in the following ways:

- Manual Configuration, on page 20
- IPv6 Stateless Address Auto-Configuration (RFC2462), on page 20
- DHCP for IPv6, on page 21

  Stateless DHCP, on page 21

Manual Configuration

Because IPv6 addresses are so complex, you won't want to configure many of them manually. There's too much administrative overhead. But you might want static addresses for router interfaces and certain network resources.

IPv6 Stateless Address Auto-Configuration (RFC2462)

One of the easiest ways to assign IP addresses is to set up IPv6 Stateless address auto-configuration (SLAAC) on an IPv6 router.

The network administrator configures the router to send Router Advertisement (RA) announcements onto the link. Then the on-link connected IPv6 nodes configure themselves with an IPv6 address and routing parameters.

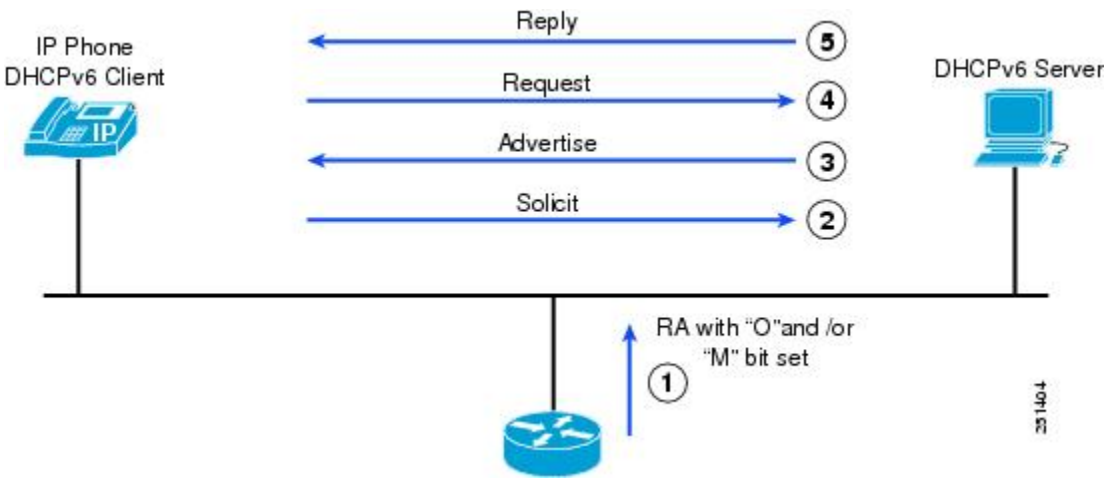
They get the IPv6 network prefix from the link-local router's RAs. They create the IPv6 host ID by using the device's MAC address and the EUI-64 format for host IDs.

DHCP for IPv6

IPv6 devices use multicast to acquire IP addresses and to find DHCPv6 servers. The basic DHCPv6 client-server concept is similar to DHCP for IPv4. If a client wants to receive configuration parameters, it sends out a request on the attached local network to detect available DHCPv6 servers.

- The DHCPv6 client requests parameters from an available server. This is done using Solicit and Advertise message and well-known DHCPv6 multicast addresses.
- The server responds with the requested information in a Reply message. As with DHCPv4, DHCPv6 uses an architectural concept of "options" to carry more parameters and information within DHCPv6 messages.

Figure 7: IPv6 DHCP Messages



The DHCPv6 client knows whether to use DHCPv6 based upon the instruction from a router on its link-local network. The default gateway has two configurable bits in its Router Advertisement (RA) available for this purpose:

- O bit—When this bit is set, the client can use DHCPv6 to retrieve other configuration parameters (for example, TFTP server address or DNS server address) but not the client's IP address.
- M bit—When this bit is set, the client can use DHCPv6 to retrieve a managed IPv6 address and other configuration parameters from a DHCPv6 server.

Stateless DHCP

Stateless DHCPv6, which is specified by RFC3736, is a combination of Stateless address auto-configuration (SLAAC) and Dynamic Host Configuration Protocol (DHCP) for IPv6.

When a router sends an RA with the O bit set but does not set the M bit, the client can use SLAAC to obtain its IPv6 address and use DHCPv6 to obtain additional information, such as TFTP server address or DNS

server address. This mechanism is known as Stateless DHCPv6 because the DHCPv6 server does not have to track the client address bindings.

IPv6 Address Assignment Table

To automatically configure IP phones with IPv6 address assignment, use the following configuration in Unified CM and in router Routing Advertisement (RA).

Table 5: Automatic IPv6 Address Assignment

Address Assignment Method	Auto_config parameter on Unified CM	M-bit	O-bit
IPv6 address Stateless prefix from RA and DHCP to pull TFTP and DNS address	On	Off	On
Invalid configuration	On	Off	Off
Invalid configuration	Off	Off	On
Invalid configuration	Off	Off	Off

DNS for IPv6

Cisco Unified Communications Manager (Unified CM) uses DNS name-to-address resolution in the following cases:

- If DNS names are used to define Unified CM servers (not supported)
- If SIP route patterns use DNS names to define destinations
- If SIP trunks use DNS names to define trunk destinations

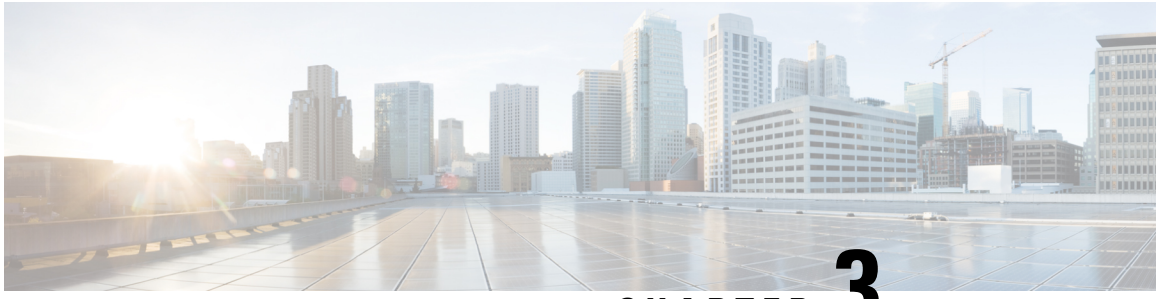
For IPv6, the principles of DNS are the same as for IPv4, with the following exceptions:

- The nomenclature is different (AAAA records are used instead of A records).
- DNS name-to-address queries can return multiple IPv6 addresses.

Table 6: Comparison of DNS Name and Address Resolution

Resolution of:	IPv4	IPv6
Hostname to IP address	A record: www.abc.test. A 192.168.30.1	AAAA record: www.abc.test AAAA 2001:db8:C18:1::2

Resolution of:	IPv4	IPv6
IP address to hostname	PTR record: 1.30.168.192.in-addr.arpa. PTR www.abc.test.	PTR record: 2.0.0.0.0.0.0.0.0.0.0.0.0.0.0.0.1.0.0.0.8.1.c.0. 8.b.d.0.1.0.0.2.ip6.arpa PTR www.abc.test.



CHAPTER 3

IP Addressing Modes for Cisco Collaboration Products

IP Addressing Modes

IP addressing modes specify the types of addresses that a device can communicate with and understand. The following IP addressing modes are used as part of the Cisco Collaboration Systems Release.

Table 7: IP Addressing Modes

	IP Addressing Mode	Description	Notes
	**IPv4-only stack**	Used for some endpoints and gateways.	
	**IPv6-only stack**	Used for some endpoints and gateways.	
	**Dual-stacks: IPv4-only and IPv6-only**	Used for applications servers, such as Cisco Unified Communications Manager, Cisco Unity Connection, Cisco Emergency Responder, and Cisco Unified Survivable Remote Site Telephony (Unified SRST).	The session initiation protocol (SIP) session description protocol (SDP) attribute should not be configured to transition to IPv6-only.

	IP Addressing Mode	Description	Notes
	IPv6 Aware	Used for applications servers that use IPv4 to transport IPv6 application information, such as the Cisco Prime Collaboration Provisioning module, Cisco Emergency Responder, and Unified CM.	IPv6 aware devices communicate with IPv4 addresses, but can receive and understand IPv6 addresses embedded in application protocol data units (PDUs).
	Dual-stack	Used for applications servers, such as Cisco Unified Communications Manager, Cisco Unity Connection, Cisco Emergency Responder, and Cisco Unified Survivable Remote Site Telephony (Unified SRST).	-

Supported IPv6 Addressing Modes

For Collaboration IPv6 implementations, we recommend that you configure:

- IPv6-only stack for endpoints and gateways where supported.
- IPv4-only and IPv6-only stack for application servers and interfaces where supported. Unified CM and other applications ensure interoperability with existing IPv4-only devices and applications.

The following tables illustrate the SIP IP addressing modes for Cisco Collaboration Systems Release (CSR) products. Any products not listed here should be configured in IPv4-only stack.

Note For a list of product configuration resources for IPv6, see Product Configuration Resources for IPv6, on page 117.

Table 8: Supported IPv6 Addressing Modes for Endpoints

Endpoint (SIP)	IPv4-only	IPv6-only	Dual- Stacks: IPv4 and IPv6	Comments
Cisco IP Phone 7811, 7821, 7841, 7861	Yes	Yes	Yes	

Endpoint (SIP)	IPv4-only	IPv6-only	Dual- Stacks: IPv4 and IPv6	Comments
Cisco IP Phone 8811, 8841, 8845, 8851, 8861, 8865, 8875	Yes	Yes	Yes	
Cisco IP Conference Phone 7832	Yes	Yes	Yes	
Cisco IP Conference Phone 8832	Yes	Yes	Yes	
Cisco Jabber	Yes	Yes	Yes	IPv6-only is for on-premise Jabber desktop deployment. No ANAT.
Cisco Webex Room Series, Board Series, Desk Series	Yes	Yes	Yes	

Table 9: Supported Addressing Modes for Communication Gateways

Gateway	Applications	IPv4-only	IPv6-only	Dual- Stacks: IPv4 and IPv6	Comments
	Note The "Yes" and "No" means whether the IP Addressing mode is supported or not.				
Cisco 4000 Series Integrated Services Routers (ISR)	IOS Gateways	Yes	No	Yes	For more information, see Cisco IOS Unified Communications Gateways with SIP
	Unified SRST	Yes	No	Yes	
	CUBE	Yes	No	Yes	Only ISR G2.
	MTP	Yes	No	Yes	

Gateway	Applications	IPv4-only	IPv6-only	Dual- Stacks: IPv4 and IPv6	Comments
Cisco 2900 and 3900 Series Integrated Services Routers (ISR)	IOS Gateways	Yes	No	No	EOS Notices for ISR 28xx/38xx/29xx/39xx
	Unified SRST	Yes	No	No	
	CUBE	Yes	No	Yes	
	MTP	Yes	No	Yes	
H.323 Gateways		Yes	No	No	EOS Notice for the H.323 Call Control features in Cisco IOS-XE Software EOS Notices for ISR 28xx/38xx/29xx/39xx
Media Gateway Control Protocol (MGCP) Gateways		Yes	No	No	EOS Notices for ISR 28xx/38xx/29xx/39xx

Table 10: Supported IPv6 Addressing Modes for Applications and Interfaces

Application or Interface	IPv4-only	IPv6-only	Dual- Stacks: IPv4 and IPv6	Comments
Cisco Unified Communications Manager (Unified CM)	Yes	No	Yes	Product required for IPv6 deployment.
Cisco Unified Communications Manager IM and Presence Service (IM and Presence Service)	Yes	No	Yes	Product required for IPv6 deployment.
Cisco Unified Communications Manager Express (Unified CME)	Yes	No	No	

Application or Interface	IPv4-only	IPv6-only	Dual- Stacks: IPv4 and IPv6	Comments
Cisco Unified Survivable Remote Site Telephony (Unified SRST)	Yes	No	Yes	Product required for IPv6 deployment for ISR G3.
Cisco Unified Contact Center Enterprise (Unified CCE)	Yes	No	Yes	For Agent IPv6-only stack, NAT64.
Cisco Unified Contact Center Express (Unified CCX)	Yes	No	Yes	For Agent IPv6-only stack, NAT64.
Cisco Emergency Responder (Emergency Responder)	Yes	No	Yes	
Cisco Unity Connection	Yes	No	Yes	
Cisco Meeting Server	Yes	No	No	
Cisco TelePresence Management Suite	Yes	No	No	
Cisco Unity Express	Yes	No	No	
End-of-Sale: Cisco Unified MeetingPlace	Yes	No	No	EOS Notice
Cisco Prime Collaboration	Yes	No	Yes	Applicable to IOS Gateways, Unified SRST, CUBE.
Cisco Smart Software Licensing	Yes	No	Yes	Applicable to Unified CM, IM and Presence Service, Cisco Unity Connection.
LAN/WAN	Yes	Yes	Yes	
Dynamic Host Configuration Protocol (DHCP)	Yes	Yes	Yes	

Application or Interface	IPv4-only	IPv6-only	Dual- Stacks: IPv4 and IPv6	Comments
Domain Name System (DNS)	Yes	Yes	Yes	
Directory LDAP	Yes	Yes	Yes	
NAT64	Yes	No	No	Except for Unified CCE and Unified CCX, which have internal NAT64.

IPv6 Addressing in Cisco Collaboration Products

As you design your network, you'll need information about the number of IPv6 addresses that the various products can support.

Cisco Unified Communications Manager and IPv6 Addresses

Each Cisco Media Convergence Server (MCS) can support the following addresses simultaneously:

- One IPv6 link local address (for example, FE80::987:65FF:FE01:2345)
- *Either* of the following:
  - One IPv6 unique local address (ULA), for example:

    ```
    FD00:AAAA:BBBB:CCCC:0987:65FF:FE01:2345
    ```

  - One IPv6 global address (GA), for example:

    ```
    2001:0DB8:BBBB:CCCC:0987:65FF:FE01:2345
    ```

- One IPv4 address

All IPv6 devices must have a link local address that is automatically created.

A unique local address is equivalent to a private address in IPv4 (for example, 10.10.10.1).

A global address is a globally unique public address.

Cisco IP Phones and IPv6 Addresses

A Cisco IP Phone can support a combination of the following addresses:

- One IPv6 link local address (for example, FE80::987:65FF:FE01:2345)
- Multiple IPv6 unique local addresses (for example, FD00:AAAA:BBBB:CCCC:0987:65FF:FE01:2345)
- Multiple IPv6 global addresses (for example, 2001:0DB8:BBBB:CCCC:0987:65FF:FE01:2345)

- One IPv4 address

Note IPv6 NAT-based solution is **not** supported.

Cisco IP Phones must support one link local address and can support a combination of up to 20 global or unique local addresses. The IP phone can use only IPv6 highest address scopes that can be received (RFC 4193) (global address or unique local IPv6 addresses) to register to Unified CM. After registration, this IPv6 address is used for signaling and media.

The following characteristics also apply to IPv6 addresses on IP phones:

- A link local address is never sent to Cisco Unified Communications Manager as a signaling and media address.
- If the IP phone has both unique local and global addresses, the highest scoped global addresses take precedence over unique local addresses.
- If the IP phone has multiple unique local addresses or multiple global addresses, the first address configured is the one used for signaling and media.

  The following priority order applies to IPv6 addresses configured on an IP phone:

  1. Use the IPv6 address configured manually through the phone's user interface (UI).
  2. If a manually configured address is not available, but stateless auto-configuration (SLAAC) is enabled for the phone, then the phone uses SLAAC to create an IPv6 address.

Note In Cisco Unified Communications Manager, SLAAC is On by default. With SLAAC, the phone uses the IPv6 network prefix advertised in the link local router's Router Advertisements (RAs). The phone creates the IPv6 host ID by using the phone's MAC address and the EUI-64 format for host IDs. If SLAAC is used, Stateless DHCP IPv6 provides IPv6 address for TFTP and DNS server.

Cisco IOS Devices and IPv6 Addresses

Each interface of a Cisco IOS device can support a combination of the following addresses:

- One IPv6 link local address, for example:

```
FE80::987:65FF:FE01:2345
```

- Multiple IPv6 unique local addresses, for example:

```
FD00:AAAA:BBBB:CCCC:0987:65FF:FE01:2345
```

- Multiple IPv6 global addresses, for example:

```
2001:0DB8:BBBB:CCCC:0987:65FF:FE01:2345
```

- Multiple IPv4 addresses

Cisco IOS media termination points (MTPs) are associated with the router's interface through the **sccp local**<*interface*> command. The MTP inherits the IPv4 and IPv6 addresses of the interface.

Configuration Parameters and Features for IPv6 in Unified CM

These configuration parameters and features support IPv6 in Cisco Unified Communications Manager (Unified CM):

- Common device configuration for phones and trunks
  - IP addressing mode
  - IP addressing mode preference for signaling
  - Allow Stateless auto-configuration for phones
- Role of the media termination point (MTP) in IPv6-enabled Unified CM clusters
- New enterprise parameters
- MTP selection

You can enable and configure IPv6 for an entire cluster and you can configure your IPv6 devices (as shown in Supported IPv6 Addressing Modes , on page 26) to use IPv6 for signaling and media.

Common Device Configuration

Rather than add IPv6 configuration parameters to specific IP phones and SIP trunks and phones, you can use Unified CM's common device configuration template. This template contains the IPv6-specific configuration parameters for IP phones and SIP trunks. You can create and associate multiple common device configuration profiles to IP phones and SIP trunks.

To find the template, choose **Device** > **Device Settings** > **Common Device Configuration**.

The profile contains this configuration information for IPv6:

- IP Addressing Mode
- IP Addressing Mode Preference for Signaling
- Allow (Stateless) Auto-Configuration for Phones

Default Common Device Configuration

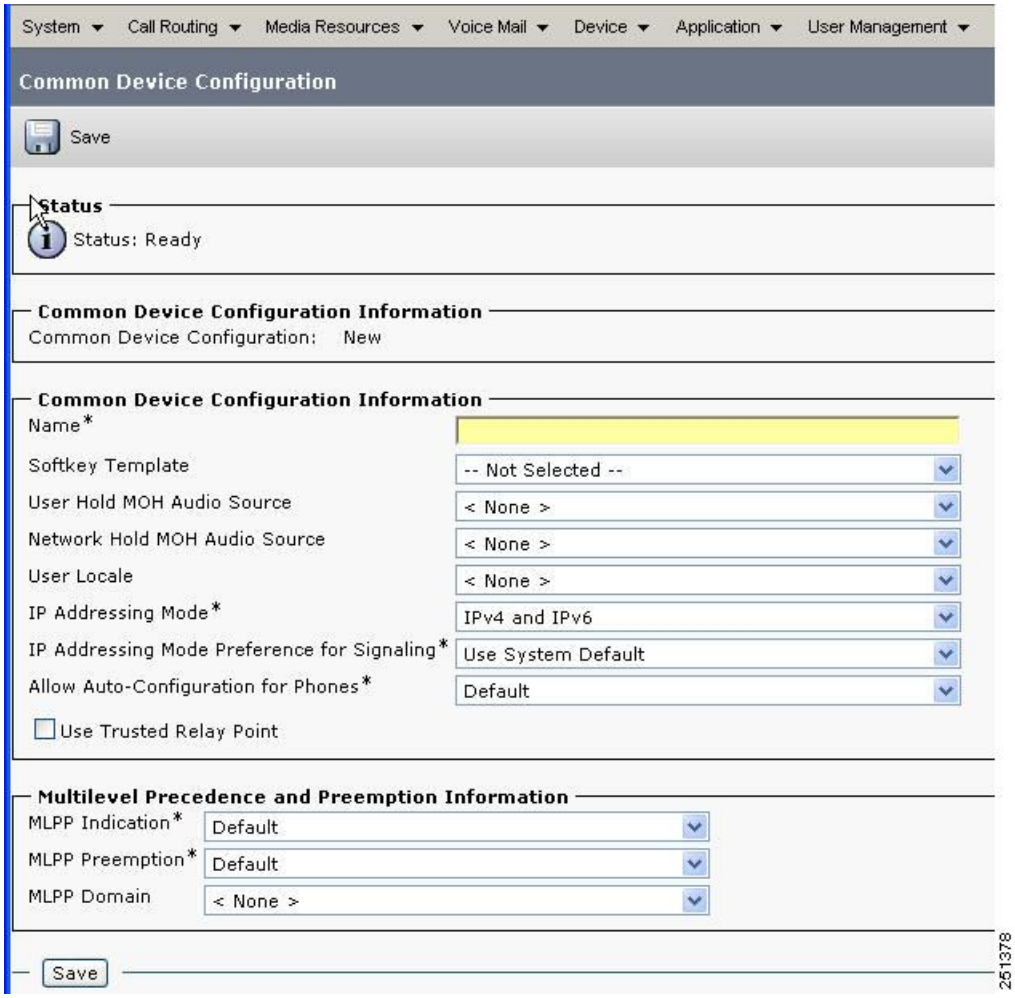
There is no default common device configuration profile. Devices are initially set to <None>. If you enable IPv6 in the Unified CM cluster in this <None> configuration, then your IPv6 devices adopt the following settings:

- IP Addressing Mode = IPv4 and IPv6

- IP Addressing Mode Preference for Signaling = Use System Default
- Allow (Stateless) Auto-Configuration for Phones = Default

Note For enterprise deployment models, you must configure IP Addressing Mode to IPv6.

Figure 8: Initial Common Device Configuration Settings



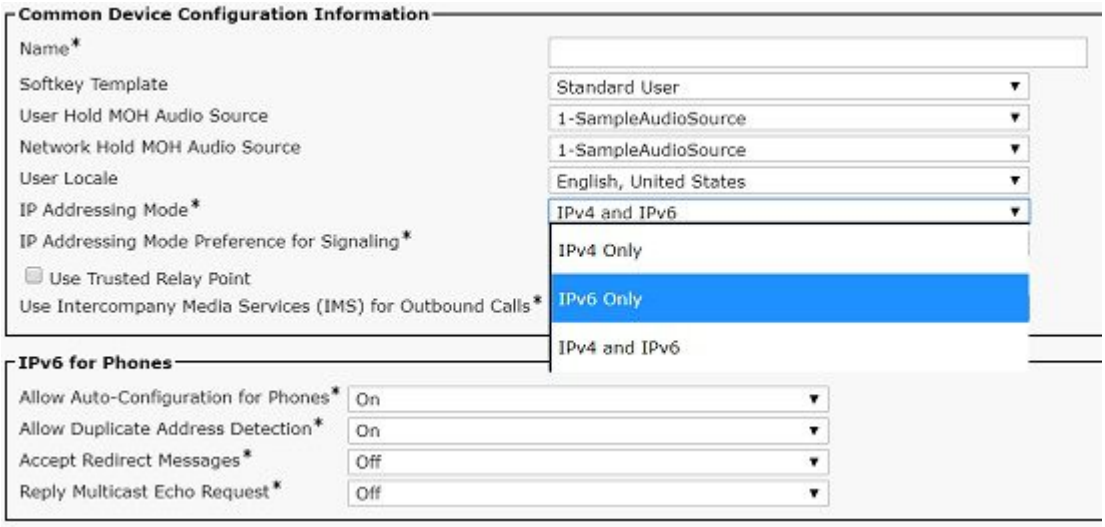
IP Addressing Mode for IPv6 Phones

After you configure the common device configuration profile and assign it to your phones, the specified IP addressing modes will be applied. The IP addressing modes are:

- **IPv4 Only**—The phone acquires and uses only one IPv4 address for all signaling and media. If the phone acquired an IPv6 address previously, it releases the IPv6 address.
- **IPv6 Only**—The phone acquires and uses only one IPv6 address for all signaling and media. If the phone acquired an IPv4 address previously, it releases the IPv4 address.

- **IPv4 and IPv6**—The phone acquires and uses one IPv4 address and one IPv6 address. It can use the appropriate address as required for media. It uses either the IPv4 address or the IPv6 address for call control signaling.

Figure 9: Setting the Phone IP Addressing Mode



If IPv6 is enabled in the Unified CM cluster, the default phone setting for IP addressing mode is IPv6. IP phones should not be configured for **IPv4 and IPv6**, it should be configured for IPv6-only phones.

IP Addressing Modes for Media Streams Between Devices

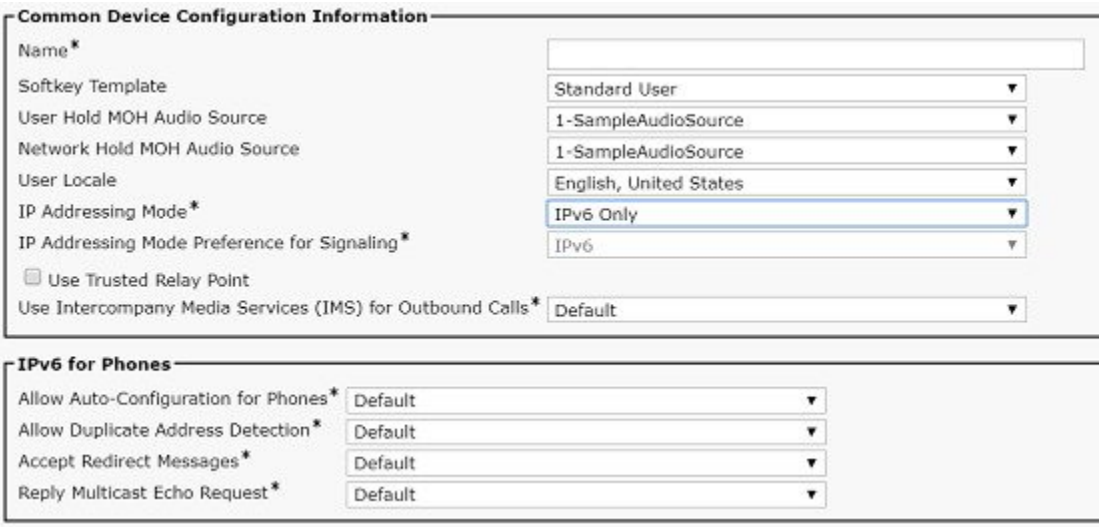
Depending on the devices that you have chosen and the configuration profiles that you have set up, you can potentially have an assortment of devices using IPv4 addresses or IPv6 addresses. For two devices (such as phones) that support mismatched addressing modes, an IP addressing version incompatibility exists when a device with an IPv4 address wants to establish a RTP voice stream with a device with an IPv6 address. To resolve this IP addressing incompatibility for media, Unified CM dynamically inserts a media termination point (MTP) to convert the media stream from IPv4 to IPv6 or conversely. For more information on how and when MTPs are used for IPv6 calls, see Media Resources and Music on Hold Overview, on page 79.

IP Addressing Mode Preference for Signaling for Phones

The phone IP Addressing Mode Preference for Signaling has three settings:

- **IPv4**—The phone uses its IPv4 address for call control signaling to Unified CM.
- **IPv6**—The phone uses its IPv6 address for call control signaling to Unified CM.
- **Use System Default**—The phone uses the cluster-wide setting for IP Addressing Mode for Signaling if it has an address of that type. At first glance, you might worry that a cluster-wide setting could lead to incompatibility issues: What if a phone is using an IPv4 address and it tries to set up a voice stream with a phone that is using an IPv6 address? Unified CM handles this situation dynamically. It inserts an MTP that converts the media stream from IPv4 to IPv6 and back again.

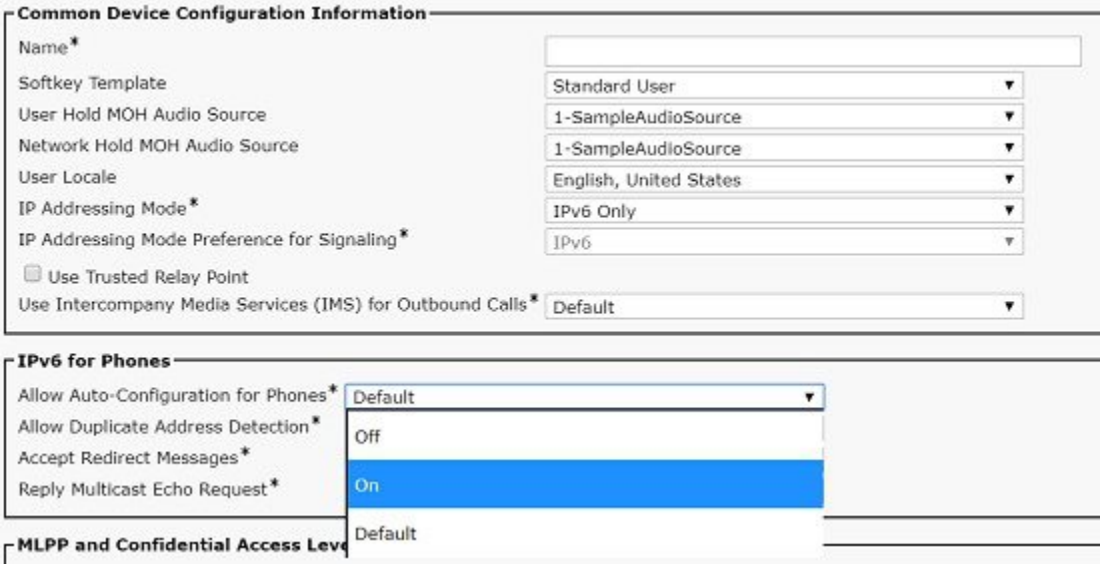
Figure 10: Common Device Configuration Information: IP Addressing Mode Preference for Signaling



Allow (Stateless) Auto-Configuration for Phones

You can allow phones to receive an IP address and other information automatically. Your options are partly dependent on how you configure the link local router.

Figure 11: Common Device Configuration Information: Allow (Stateless) Auto-Configuration for Phones



You have these options:

- **On**—The phone can use Stateless Address Auto-Configuration (SLAAC), if supported by the link local router's configuration. It depends on the O and M bits in the Router Advertisements (RAs).
  - If the O-bit is set—The router allows the phone to use SLAAC to acquire its IP address and to use the DHCP server to acquire other information (such as the TFTP server address and DNS server address). This is known as stateless DHCP IPv6.

- If the M-bit is set—The router does not allow the phone to use SLAAC but allows it to use the DHCP server to acquire its IP address and other information.
- If neither bit is set—The router allows the phone to use SLAAC to acquire an IP address but does not allow it to use DHCP for other information. You need to configure a TFTP server address through the phone's user interface (UI). The phone uses this TFTP server to download its configuration file and register to Unified CM. We do not recommend this for a production environment.

- **Off**—The phone does not use SLAAC to acquire an IPv6 address.
- **Default**—The phone uses the cluster-wide enterprise parameter configuration value for Allow (Stateless) Auto-Configuration for Phones. If IPv6 is enabled in the Unified CM cluster, the phone's default setting for Allow (Stateless) Auto-Configuration for Phones setting is **Default**. If the IP phone supports IPv6-only, it adopts the cluster-wide setting for Allow (Stateless) Auto-Configuration for Phones, but all IPv4 phones ignore this setting.

Common Device Profile Configuration for SIP Trunks

You can apply SIP trunk configuration settings through the Common Device Configuration profile that you create and assign to the IPv6-Only SIP trunk with ANAT disabled.

With IPv6 enabled and with IPv4 addresses defined on the Unified CM server, you can configure the SIP trunk to use either of these addresses as its source IP address for SIP signaling. The SIP trunk also listens for incoming SIP signaling on the configured incoming port number of the server's IPv4 or IPv6 address.

IP Addressing Mode for SIP Trunks

The SIP trunk IP addressing mode has three settings:

- **IPv4 Only**—The SIP trunk uses the Unified CM IPv4 address for signaling and either an MTP or phone IPv4 address for media.
- **IPv6 Only**—The SIP trunk uses the IPv6 address for signaling and either an MTP or phone IPv6 address for media.
- **IPv4 and IPv6**—For signaling, the SIP trunk will use either the Unified CM IPv4 address or the Unified CM IPv6 address. For media, the SIP trunk will use either an MTP IPv4 and/or IPv6 address or the phone IPv4 and/or IPv6 address.

If IPv6 is enabled in the Unified CM cluster, the default SIP trunk setting for the IP Addressing mode is IPv4 and IPv6. All IPv4 trunks (H.323 and MGCP) will ignore this setting.

For more information on these SIP trunk IP addressing modes, see SIP Trunks Using Delayed Offer, on page 73.

IP Addressing Mode Preference for Signaling for SIP Trunks

The SIP trunk IP Addressing Mode Preference for Signaling is used only for outbound calls. Unified CM listens for incoming SIP signaling on the configured incoming port number of the server's address.

The SIP trunk IP Addressing Mode Preference for Signaling has three settings:

- **IPv4**—The SIP trunk uses the Unified CM IPv4 address as its source address for SIP signaling.
- **IPv6**—The SIP trunk uses the Unified CM IPv6 address as its source address for SIP signaling.

Note If IPv6 is enabled in the Unified CM cluster, this is the default SIP trunk setting for IP Addressing Mode Preference for Signaling. All IPv4 trunks ignore this setting.

- **Use System Default**—The SIP trunk uses the cluster-wide enterprise parameter configuration value for its IP addressing mode for signaling.

If IPv6 is enabled in the Unified CM cluster, the default SIP trunk setting for IP Addressing Mode Preference for Signaling is **Use System Default**. With this setting the SIP trunk will adopt the cluster-wide setting for IP Addressing Mode Preference for Signaling. All IPv4 trunks will ignore this setting.

The SIP trunk IP Addressing Mode Preference for Signaling is used only for outbound calls. Unified CM will listen for incoming SIP signaling on the configured incoming port number of the server's IPv4 and IPv6 address.

Allow Auto-Configuration for Phones

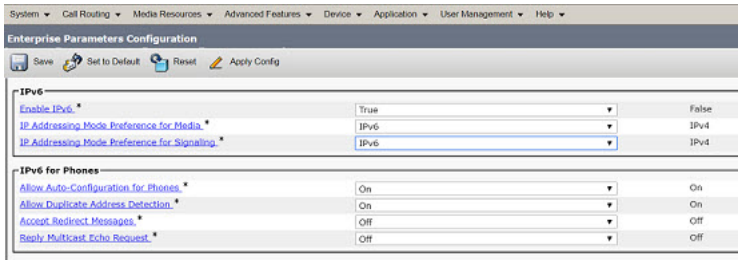
The parameter to Allow Auto-Configuration for Phones is not used by SIP trunks.

Cluster-Wide Configuration (Enterprise Parameters)

Before configuring the cluster-wide parameters in Unified CM, configure each server with an IPv6 address. For details on Unified CM IPv6 address configuration, see Configuring IPv6 in Cisco Unified CM, page A-1. In the Unified CM Administration interface, select **Enterprise Parameters** > **IPv6 Configuration Modes** to configure the following cluster-wide IPv6 settings for each Unified CM server.

- Enable IPv6
- IP Addressing Mode Preference for Media
- IP Addressing Mode Preference for Signaling
- Allow Auto-Configuration for Phones

Figure 12: Cluster-Wide IPv6 Configuration Modes



Enable IPv6

Set this parameter to **True** to enable IPv6. False is the default setting.

IP Addressing Mode Preference for Media

IP Addressing Mode Preference for Media has two setting options:

- IPv4 (Default setting)
- IPv6 setting

The cluster-wide IP Addressing Mode Preference for Media is different than the device-level IP addressing mode. The cluster-wide IP Addressing Mode Preference for Media:

- Selects which IP addressing version will be used for media when a call is made between two dual-stack devices.
- Is used when there is a mismatch in supported IP addressing versions between two devices. For example, if an IPv6-only device calls an IPv4-only device, an MTP must be inserted into the media path to convert from IPv4 to IPv6, and conversely.

Typically, both devices have MTP media resources available to them in their media resource group (MRG). The IP Addressing Mode Preference for Media is used to select which device's MTP is used to convert from IPv4 to IPv6 (and conversely) for the call, as follows:

- If the IP Addressing Mode Preference for Media is set to IPv4, the MTP associated with the IPv6-only device is selected, so that the longest call leg between the device and the MTP uses IPv4.
- If the IP Addressing Mode Preference for Media is set to IPv6, the MTP associated with the IPv4-only device is selected, so that the longest call leg between the device and the MTP uses IPv6.
- If the preferred device’s MTP is not available, the other device's MTP is used.
- If no MTPs are available, the call fails.

MTP resource allocation is discussed in detail in Media Resources and Music on Hold, on page 79.

IP Addressing Mode Preference for Signaling

The cluster-wide setting for the IP Addressing Mode Preference for Signaling is used by devices whose IP Addressing Mode Preference for Signaling is set to **Use System Default**. The IP Addressing Mode Preference for Signaling has two setting options:

- IPv4 (Default setting)
- IPv6 setting

Allow (Stateless) Auto-Configuration for Phones

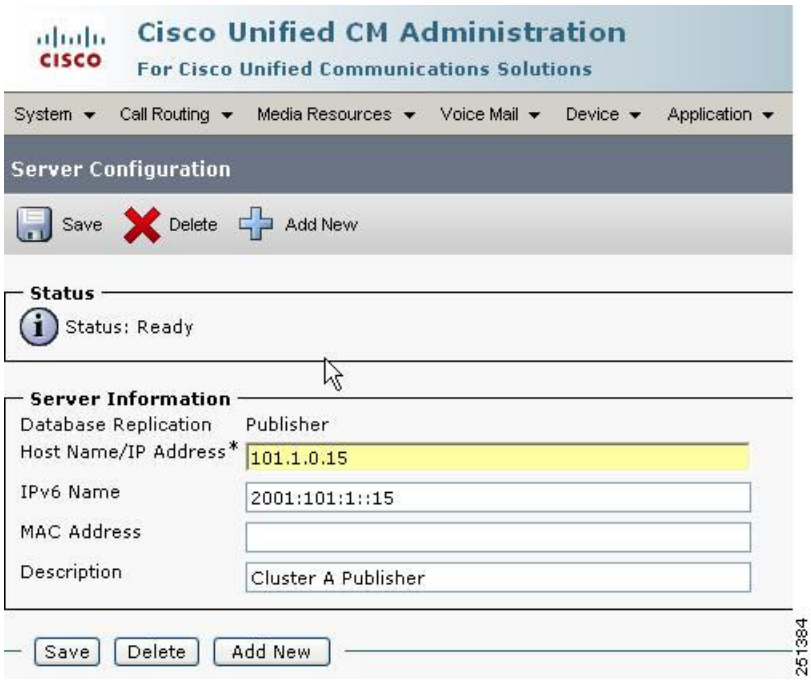
The cluster-wide setting to Allow (Stateless) Auto-Configuration for Phones is used by phones whose Allow (Stateless) Auto-Configuration for Phones parameter is set to Default. The parameter to Allow (Stateless) Auto-Configuration for Phones has two setting options:

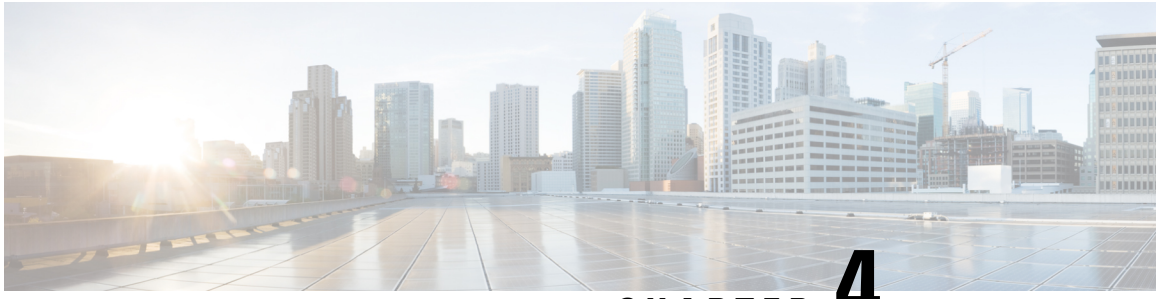
- On (Default setting)
- Off

IPv6 Address Configuration for Unified CM

After you configure an IPv6 address for the Unified CM server, you must also configure this address in the Unified CM Administration graphical user interface. This IPv6 address is used in the device configuration files stored on the cluster's TFTP servers. IPv6 devices can use this address to register with Unified CM. A server name can also be used, but an IPv6 DNS server is required to resolve this name to an IPv6 address.

Figure 13: IPv6 Address Configuration for Unified CM





CHAPTER 4

Collaboration Deployment Models for IPv6

Collaboration Deployment Models for IPv6 Overview

This chapter describes the deployment models you can use with IPv6 in Cisco Collaboration networks. Cisco Unified Communications Manager (Unified CM) supports the following traditional IPv4 deployment model examples:

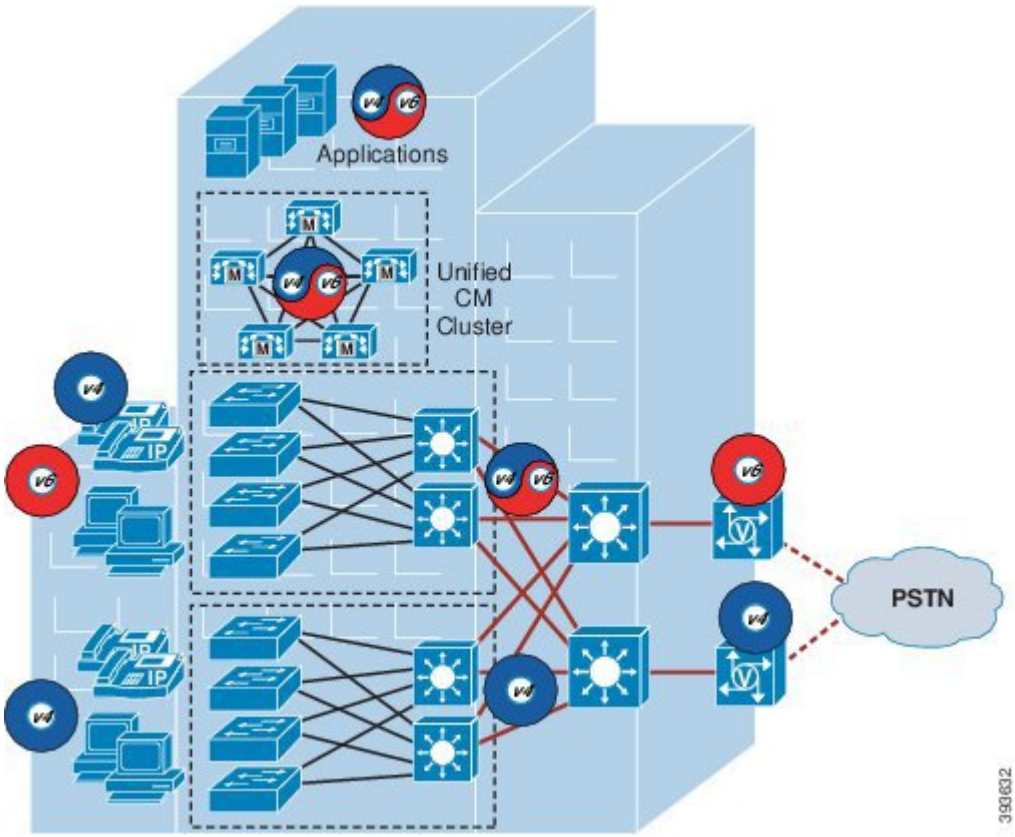
- Single-site deployments
- Multi-site WAN deployments with distributed call processing
- Multi-site deployments with centralized call processing and Cisco Unified Survivable Remote Site Telephony (Unified SRST)

With all of these deployment models, IPv6 endpoints should be configured as IPv6 with a preference of IPv6 for signaling and media. This configuration maximizes the amount of IPv6 traffic with the use of media termination points (MTPs) for conversions between IPv4 and IPv6.

Single-Site Deployments

An enterprise would typically deploy the single-site model over a LAN or metropolitan area network (MAN), which carries the voice, video, and IM traffic within the site. In this model, calls beyond the LAN or MAN use the public switched telephone network (PSTN).

Figure 14: Single-Site IPv6 Deployment



The characteristics and benefits of the IPv6 single-site model are the same as for IPv4 single-site deployments, as described in the Cisco Collaboration Solution Reference Network Design (SRND), available at Link. The IPv6 single-site model includes the additional IPv6-only endpoints and dual stack application servers' product capabilities and features discussed throughout this document. As shown in Supported IPv6 Addressing Modes , on page 26,

Best Practices for IPv6 Single-Site Deployments

Single-site IPv6 deployments can contain a mixture of IPv4 and IPv6 devices. Phones can be configured as:

- IPv4-only
- IPv6 only

One or more ISDN PSTN gateways can be deployed in a single-site deployment.

Both the Unified CM SIP trunk and Cisco IOS SIP gateway should be configured as follows:

- IPv6-only stack
- ANAT disabled
- To use IPv6 for signaling and media is to maximize the amount of IPv6 traffic.

The Unified CM SIP trunk and the SIP gateway can be configured to use:

- SIP Delayed Offer (With **MTP Required** unchecked, although MTPs may be inserted dynamically for some calls for conversions between IPv4 and IPv6 addresses.)

Note SIP Early Offer IPv6-only trunk is not supported.

If a single IPv6 gateway is used and the cluster-wide preference for media is set to IPv6, an MTP is used for all calls to IPv4-only devices to convert from IPv4 to IPv6 protocol. If the widespread use of MTPs is not acceptable in the single-site deployment, configure two PSTN gateways instead of just one. Configure one as IPv4-only SIP gateway using SIP Delayed Offer as described above, and the other as a standard IPv6-only gateway. Calling search spaces and partitions can then be used to direct PSTN calls from IPv4-only and IPv6-only devices to their respective gateways.

For specific device configuration options and preferences, refer to the chapters on Trunks, on page 65, and Collaboration Endpoints, on page 109.

The Campus LAN

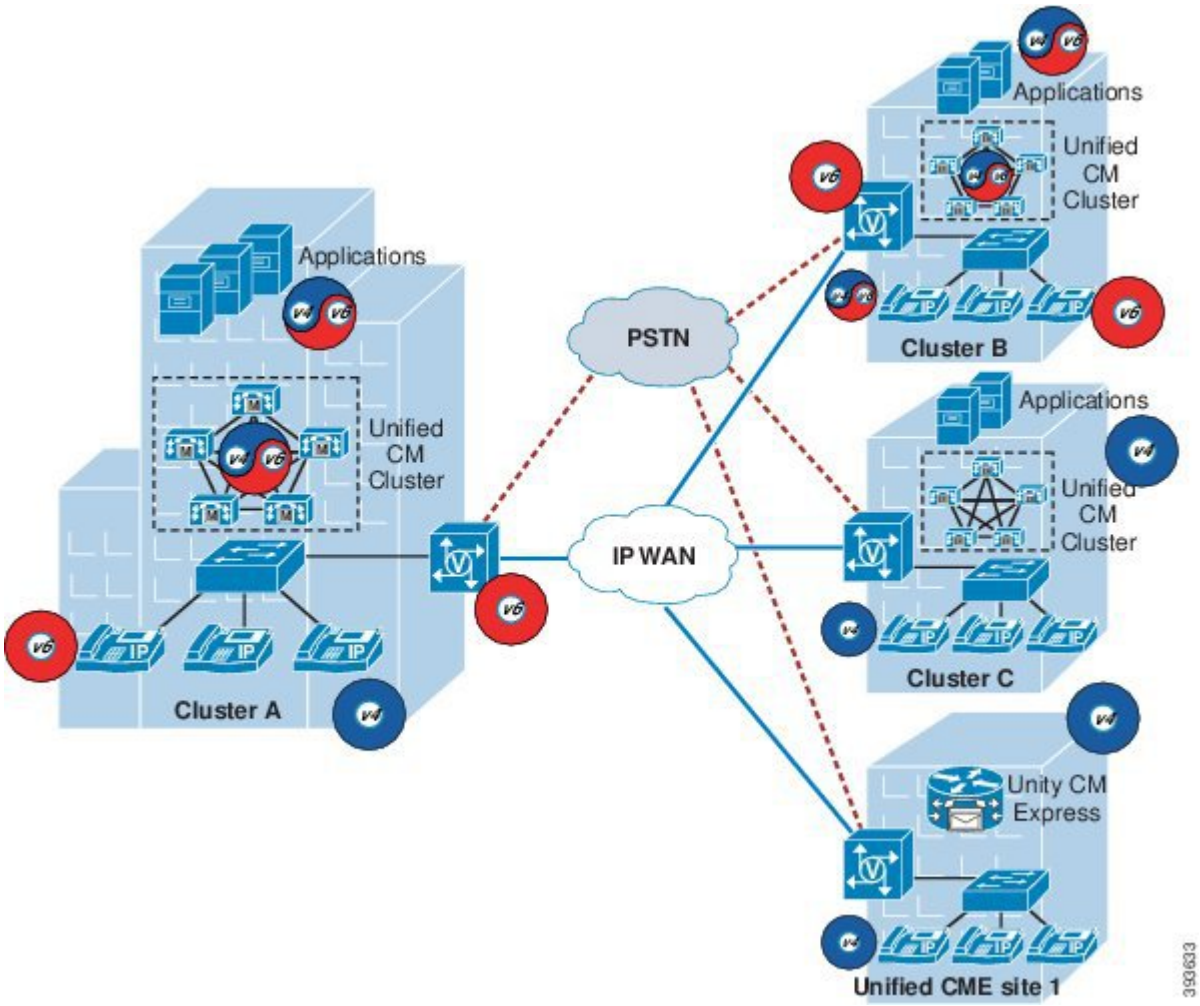
If the campus LAN also includes Layer 3 routing devices, configure these devices to support dual-stack (IPv4 and IPv6) routing. IPv6-only stack is **not** supported.

Note If a single PSTN gateway (as previously described) is used in this deployment model, then all Layer 3 LAN routing devices must be configured as dual-stack. If two gateways are used (one IPv6-only and one IPv4-only), then the portions of the network that contain IPv4-only devices do not have to be configured for dual-stack routing.

Multi-Site WAN Deployments with Distributed Call Processing

The model for a multi-site WAN deployment with distributed call processing consists of multiple independent sites, each with its own call processing cluster connected to an IP WAN that carries voice traffic between the distributed sites.

Figure 15: Multi-Site Deployment with Distributed Call Processing



Each site in the distributed call processing model can be one of the following:

- A single site with its own call processing agent, which can be either a:
  - Dual-stack (IPv4 and IPv6) Cisco Unified Communications Manager (Unified CM)
  - Standard (IPv4-only) Cisco Unified Communications Manager (Unified CM)
  - Standard (IPv4-only) Cisco Unified Communications Manager Express (Unified CME)
- Other IP PBX:
  - A standard (IPv4-only) centralized call processing site and all of its associated remote sites
  - A legacy PBX with Voice over IP (VoIP) gateway (IPv4-only or IPv6-only)

For dual-stack (IPv4 and IPv6) sites, the devices can be configured as either IPv4-only or IPv6-only. This configuration maximizes the amount of IPv6 traffic with the use of MTPs for conversions between IPv4 and IPv6 addresses.

The characteristics and benefits of an IPv6 multi-site WAN deployment with distributed call processing are the same as those for IPv4 multi-site WAN deployments with distributed call processing, as described in the Cisco Collaboration System Solution Reference Network Design (SRND), available at Link. The IPv6 multi-site model includes the additional IPv6-only and IPv6-only product capabilities and features discussed in this document.

Best Practices: Multi-Site WAN Deployments with Distributed Call Processing

A multi-site WAN deployment with distributed call processing has many of the same requirements as a single site. Follow the best practices from the single site model in addition to the ones listed here for the distributed call processing model.

IPv6 Unified CM clusters in multi-site WAN deployments with distributed call processing can use IPv6-only enabled SIP Delayed Offer intercluster trunks to connect to other IPv6 Unified CM clusters. However, for intercluster trunk connections to IPv4-only Unified CM clusters, we recommend using IPv4 SIP trunk delayed offer, not IPv6 intercluster trunks.

Note If IPv6-enabled SIP intercluster trunks are used, the WAN must support dual-stack (IPv4 and IPv6) routing.

Configure the Unified CM SIP intercluster trunks as follows:

- IPv6-only or IPv4-only.
- With ANAT disabled.
- To use IPv6 for signaling and media is to maximize the amount of IPv6 traffic.

Configure the Unified CM SIP intercluster trunk to use:

- SIP Delayed Offer with **MTP Required** unchecked. Although MTPs may be inserted dynamically for some calls for conversions between IPv4 and IPv6 addresses.

Note SIP IPv6-only trunk Early Offer with **MTP Required** checked and used for every call is not supported.

For specific device configuration options and preferences, refer to Trunks, on page 65, and Collaboration Endpoints, on page 109.

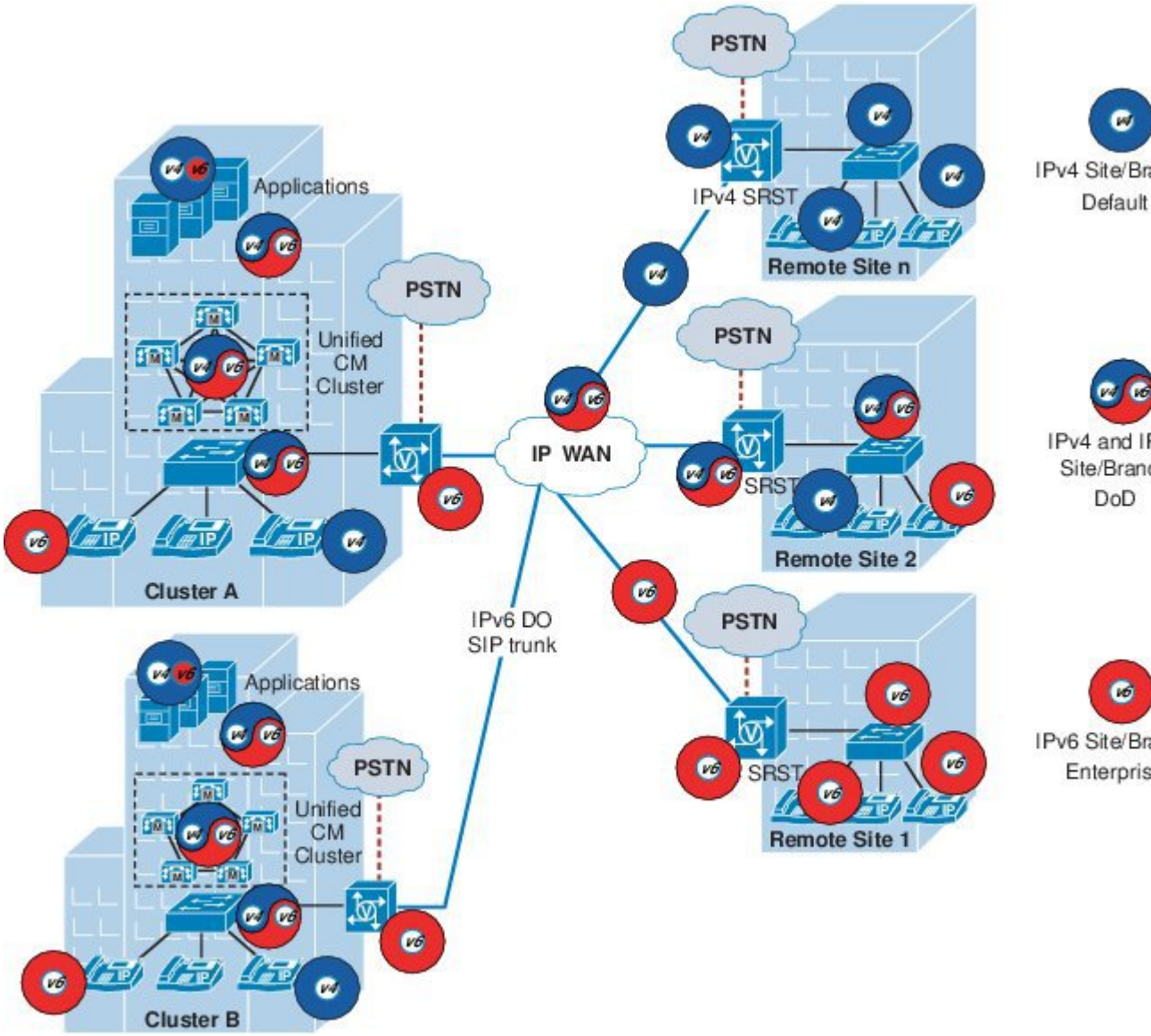
Multi-Site Deployments with Centralized Call Processing and Unified SRST

In this call processing deployment model, endpoints can be located remotely from the call processing service (Unified CM cluster), across a QoS-enabled Wide Area Network (WAN). Due to the limited quantity of bandwidth available across the WAN, call admission control is required to manage the number of calls admitted on any given WAN link, to keep the load within the limits of the available bandwidth. On-net communication between the endpoints traverses either a LAN/MAN (when endpoints are located in the same site) or a WAN

(when endpoints are located in different sites). Communication outside the enterprise goes over an external network such as the PSTN, through a gateway that is typically co-located with the endpoint.

The IP WAN also carries call control signaling between the central site and the remote sites. The following figure illustrates a typical centralized call processing deployment, with a Unified CM cluster as the call processing agent at the central site and an IP WAN to connect all the sites.

Figure 16: Multi-Site Deployment with Centralized Call Processing and SRST



For IPv6-enabled multi-site centralized call processing deployments, the centralized Unified CM cluster is enabled for IPv4 and IPv6. Each site may be configured as dual-stacks or IPv4-only. For dual-stack (IPv4 and IPv6) sites, IPv6 devices should be configured as IPv6-only with a preference of IPv6 for signaling and media. This configuration maximizes the amount of IPv6 traffic using MTPs for conversions between IPv4 and IPv6 addresses.

The characteristics and benefits of an IPv6 multi-site centralized call processing deployment are the same as for IPv4 multi-site centralized call processing deployments, as described in the *Cisco Collaboration System Solution Reference Network Design (SRND)*, available at Link. However, the IPv6 multi-site centralized call processing deployment model includes the additional IPv6-only product capabilities and features discussed in this document.

Best Practices: Multi-Site Deployments with Centralized Call Processing

IPv6 multi-site deployments with centralized call processing can contain sites with a mixture of traditional IPv4 and IPv6 devices. In each IPv6-enabled site, follow the best practices from the single-site model in addition to the ones listed here for the centralized call processing model.

You can configure phones as:

- IPv4-only
- IPv6-only

The IP WAN in IPv6 multi-site deployments with centralized call processing must support dual-stack (IPv4 and IPv6) routing.

Unified SRST routers at remote sites support IPv6-only or IPv4-only IP Phones in SRST mode. These SRST routers revert to original mode when the Unified CM cluster is restored. They operate in IPv6-only mode.

Call Admission Control

For multi-site deployments with distributed or centralized call processing, use locations-based call admission control and a WAN based on either Multiprotocol Label Switching (MPLS) or a hub-and-spoke topology. For more information on call admission control, refer to the Cisco Collaboration System Solution Reference Network Design (SRND), available at Link.

Intra-Cluster Communications

All intra-cluster server-to-server communications, such as Intra-Cluster Communication Signaling (ICCS) traffic, database traffic, and firewall management real-time traffic, use IPv4-only.

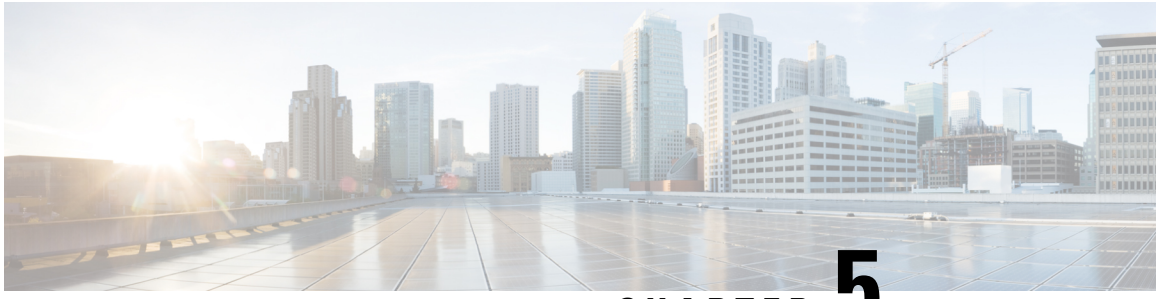
Clustering Over the WAN

We have clustering over the WAN with dual-stack Unified Communications Manager clusters, where IPv4 is used for intra-cluster communication such as ICCS and IPv6 is used for inter-cluster communication over SIP trunks.

Call Detail Records and Call Management Records

Call detail records (CDR) and call management records (CMR), when enabled, are collected by each subscriber server and are uploaded periodically to the publisher server, which stores the records in the CDR Analysis

and Reporting (CAR) database. CDR and CMR collect and store both IPv4 and IPv6 addresses. These servers interface with IPv4 addresses that are IPv6-aware.



CHAPTER 5

Network Infrastructure

Network Infrastructure Overview

The requirements of the network infrastructure needed to build an IPv6 Collaboration system in an enterprise environment are very similar to those for an IPv4 Collaboration system. Unified Communications places strict requirements on IP packet loss, packet delay, and delay variation (or jitter). Therefore, you must enable most of the Quality of Service (QoS) mechanisms available on Cisco switches and routers throughout the network. For the same reasons, redundant devices and network links that provide quick convergence after network failures or topology changes are also important to ensure a highly available infrastructure. The Cisco Catalyst 6000 Series and Catalyst 4000 Series Switches use the same QoS architecture (DSCP) for IPv6 as they used for IPv4. With the exception of the Cisco Catalyst 3560 Series and 3750 Series Switches (which support QoS trust features only for IPv6), the same QoS mechanisms (such as classification, policing, queuing, and so forth) used for IPv4 Unified Communications traffic in Cisco switches and routers can also be applied to IPv6 Unified Communications traffic. Likewise, the redundant design and availability mechanisms for IPv4 networks are generally available in Cisco switches and routers for IPv6.

This chapter discusses recommendations specific to IPv6 for Unified Communications network infrastructures. For other guidance on standard network infrastructure features required in IPv4 Unified Communications networks, refer to the Cisco Collaboration System Solution Reference Network Design (SRND), available at: Link.

The following list summarizes the key network infrastructure recommendations for IPv6 Unified Communications networks:

- For Layer 2 switched networks, enable Multicast Listener Discovery (MLD) snooping, if possible, so that multicast traffic can be forwarded selectively to the ports that you want to receive the data.
- Layer 3 routed networks require a mechanism to transport IPv6 traffic. Native Dual-stacks (IPv4 and IPv6) routing is supported, although various other IPv6 tunneling mechanisms may also be used.

- Use Hot Standby Router Protocol (HRSP) or Gateway Load Balancing Protocol (GLBP) if those protocols are supported by your Layer 3 campus devices. Otherwise, use IPv6 Neighbor Unreachability Detection.
- IPv6 traffic uses larger headers, which you must factor into the bandwidth requirements for IPv6 traffic, especially in the WAN where bandwidth can be limited.
- For intercluster IPv6 traffic over IPv6-only or IPv4-only SIP intercluster trunks, use call admission control that is based on topology-unaware locations. (RSVP is not supported for IPv6). Topology-unaware call admission control requires a hub-and-spoke topology for the WAN, or a spokeless hub in the case of a Multiprotocol Label Switching (MPLS) virtual private network (VPN).

LAN Infrastructure

Campus LAN infrastructure design is extremely important for proper IP telephony operation on a converged network. Proper LAN infrastructure design requires following basic configuration and design best practices for deploying a highly available network. Furthermore, proper LAN infrastructure design requires deploying end-to-end QoS on the network. This section discusses specific IPv6 design guidance for campus networks. For general guidance on designing collaboration campus networks, refer to the Cisco Collaboration System Solution Reference Network Design (SRND), available at Link.

General IPv6 LAN Design Guidance

The following sources provide general guidance for designing IPv6 networks:

- Introduction to IPv6 in Cisco products

  Reference: Link
- Deploying IPv6 in Campus Networks

  Reference: Link
- Deploying IPv6 in Branch Networks

  Reference: Link

IPv6 Design Guidance for Collaboration Campus Networks

The following sections provide design guidance for deploying IPv6 in collaboration campus networks.

MLD and MLD Snooping in Switched Layer 2 IPv6 Campus Networks

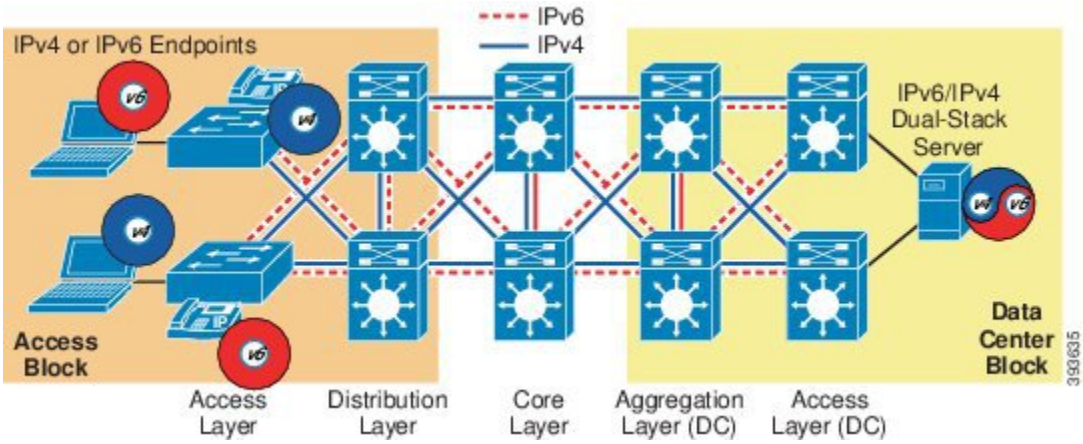
IPv6 multicast routers use Multicast Listener Discovery (MLD) protocol to discover the presence of multicast listeners (nodes wishing to receive IPv6 multicast packets) on its directly attached links and to discover which multicast packets are of interest to neighboring nodes.

MLD snooping is similar to Internet Group Management Protocol (IGMP) snooping for IPv4. With MLD snooping, IPv6 multicast data is selectively forwarded to a list of ports that want to receive the data, instead of being flooded to all ports in a VLAN. This list is constructed by snooping IPv6 multicast control packets. If possible, enable Multicast Listener Discovery (MLD) snooping in your IPv6 LAN to reduce unwanted multicast traffic.

Layer 3 Campus Networks

We recommend using dual-stack (IPv4 and IPv6) routing in Layer 3 campus IPv6 networks. The Cisco Catalyst Series Switches support Static, Routing Information Protocol next generation (RIPng), Enhanced Interior Gateway Routing Protocol (EIGRP), and Open Shortest Path First version 3 (OSPFv3) routing for IPv6.

Figure 17: Dual-Stack Routing in a Campus Network



First-Hop Redundancy Protocols

In the campus model, where the distribution switches are the L2/L3 boundary, they also act as the default gateway for the entire Layer 2 domain that they support. Some form of redundancy is required because this environment can be large and a considerable outage could occur if the device acting as the default gateway fails.

Hot Standby Router Protocol (HSRP) and Gateway Load Balancing Protocol (GLBP) first-hop redundancy protocols should be your first choice for high availability at the L2/L3 boundary and they also have useful features such as interface tracking, router prioritization, and preemption.

For IPv6 campus networks, the following Cisco IOS routing platforms support HSRP and GLBP first-hop redundancy protocols for IPv6:

- HSRP for IPv6 is supported on Cisco Catalyst 6500 Series Switches
- GLBP is supported on Cisco Catalyst 6500 Series Switches

Note If your design does not permit the use of HSRP or GLBP, you can use Neighbor Unreachability Detection (NUD). However, due to its limitations in comparison to first-hop redundancy protocols, NUD is not supported for collaboration IPv6 designs

Neighbor Discovery for IPv6 (RFC 2461) implements the use of NUD. NUD is a mechanism that enables a host to determine whether a router (neighbor) in the host's default gateway list is unreachable. Hosts receive the NUD value (which is known as the "reachable time") from the routers on the local link through regularly sent router advertisements (RAs). The default reachable time is 30 seconds and is configurable. Neighbor Unreachability Detection can be used where first-hop redundancy protocols are not available; however, due to its limitations in comparison to first-hop redundancy protocols, Neighbor Unreachability Detection is not supported for Unified Communications IPv6 designs.

For additional information on configuring first-hop redundancy protocols, refer to the Cisco IOS IPv6 Configuration Guide, available at: Link.

Network Services

As with IPv4 collaboration systems, the deployment of an IPv6 collaboration system requires the coordinated design of a well-structured, highly available, and resilient network infrastructure. It also requires an integrated set of IPv6 network services including Domain Name System (DNS), Dynamic Host Configuration Protocol (DHCP), and Trivial File Transfer Protocol (TFTP). The deployment guidelines for these network services are generally the same as for IPv4 collaboration systems, but IPv6 network services are configured differently to support their IPv6 functionality. This section discusses the product and configuration details for IPv6 network services.

IPv6 Domain Name System (DNS)

As with IPv4, IPv6 DNS enables the mapping of hostnames and network services to IPv6 addresses within a network or networks. DNS servers deployed within a network provide a database that maps network services to hostnames and, in turn, hostnames to IPv6 addresses. Devices on the network can query the DNS server and receive IPv6 addresses for other devices in the network, facilitating communications between network devices. Complete reliance on a single network service such as DNS can introduce an element of risk when a critical collaboration system is deployed. If the DNS server becomes unavailable and a network device is relying on that server to provide a hostname-to-IP-address mapping, communications will fail. For this reason, in networks requiring high availability, we recommend that you do not rely on DNS name resolution for any communication between Cisco Unified Communications Manager (Unified CM) and the endpoints.

Unified CM can use DNS name-to-address resolution in the following situations:

- DNS names are used to define Unified CM servers and all other applications servers.
- SIP route patterns use DNS names to define destinations.
- SIP trunks use DNS names to define trunk destinations.

We recommend that you use Cisco Network Registrar (CNR) or Microsoft Windows server as an IPv4 and IPv6 DNS server in your collaboration network.

Dynamic Host Configuration Protocol for IPv6 (DHCPv6)

IP phones can use DHCPv6 to obtain all of the initial configuration information that they need to register with Unified CM: an IPv6 address and an IPv6 TFTP server address.

In both IPv4 and IPv6 networks, DHCP eases the administrative burden of manually configuring each host with an IP address and other configuration information. DHCP also provides automatic reconfiguration of network addresses when devices are moved between subnets.

Note Unlike IPv4, which can use DHCP to inform a host of its default router, an IPv6 host uses Neighbor Discovery to find its local routers.

IPv6 devices can use the DHCPv6 server in two ways:

Stateless DHCP—Where the device uses stateless address auto-configuration (SLAAC) to obtain an IP address and uses the DHCP server to retrieve other information that it requires, such as TFTP server address.

DHCP and IPv4-only or IPv6-only Phones

When the power is cycled on a dual-stack (IPv4 or IPv6) phone, it requests both IPv4 and IPv6 addresses and TFTP server information from its DHCP server(s). The phone then requests its configuration file from the TFTP server, which contains information about its IP Addressing Mode setting. If the IP Addressing Mode is set to IPv4 only, the IP phone releases its IPv6 address; and if the IP Addressing Mode is set to IPv6-only, the IP phone releases its IPv4 address. If the IP Addressing Mode is set to **IPv4 and IPv6**, the IP Phone retains both addresses and uses the setting of the IP Addressing Mode Preference for Signaling (IPv4 or IPv6) in the configuration file to select which address to use to register with and signal to its Unified CM server(s).

DHCP Server Recommendations

We recommend that you use either a Cisco Network Registrar IPv4 and IPv6 DHCP server or a Cisco IOS IPv4 and IPv6 DHCP server in your collaboration network. Other DHCP server products from Microsoft Windows may be used.

DHCP Relay Agent

A DHCP relay agent is used to relay messages between the client and server. DHCP relay agent operation is transparent to the client. A client locates a DHCP server using a reserved, link-scoped multicast address, which typically requires the DHCP client and the server to be attached to the same link. However, in some situations it is desirable to allow a DHCP client to send a message to a DHCP server that is not connected to the same link. Use the **dhcp relay destination** command on your Cisco IOS router to forward DHCP client requests to a distant DHCP server.

The DHCP relay command is configured at the interface level, as follows:

- **dhcp relay destination** *ipv6-address <interface-type interface-number>*

Cisco IOS DHCPv6 Server

Current Cisco IOS releases support Cisco IOS DHCPv6 server functionality, but not all platforms support IPv6 vendor-specific options. Cisco Catalyst platforms support the IPv6 DHCP server with vendor-specific options in the latest Cisco IOS Release. Cisco IOS Router platforms support IPv6 DHCP server functionality with support for vendor-specific options in the latest Cisco IOS Release.

Example Configuration for a Cisco IOS IPv6 DHCP Server

The following example shows the configuration for a Cisco IOS IPv6 DHCP server.

```
! Activate DHCP Service on the IOS Device service dhcp

!
! Specify the name of this specific IPv6 DHCP pool, the address prefix and lifetime, the
link address and
! vendor-specific option and sub option with TFTP server address(es) ipv6 dhcp pool
v6-CLUSTER-B
address prefix 2001:101:2:1::/64 lifetime 172800 86400 link-address 2001:101:2:1::/64
vendor-specific 9
suboption 1 address 2001:101:2::10 2001:101:2::11
```

Usage Guidelines

The **ipv6 dhcp pool** command enables the DHCPv6 pool configuration mode. The following configuration commands are available in this mode:

- **address prefix** *IPv6-prefix*

  This command sets an address prefix for address assignment. This address must be in hexadecimal form, using 16-bit values between colons.

- **lifetime** *t1 t2*

  This command sets a valid (*t1*) and a preferred (*t2*) time interval (in seconds) for the IPv6 address. The range is 5 to 4294967295 seconds. The valid default is 2 days, and the preferred default is 1 day. The valid lifetime must be greater than or equal to the preferred lifetime. Specify *infinite* for no time interval.

- **link-address** *IPv6-prefix*

  This command sets a link-address IPv6 prefix. When an address on the incoming interface or a link-address in the packet matches the specified IPv6-prefix, the server uses the configuration information pool. This address must be in hexadecimal form, using 16-bit values between colons.

- **vendor-specific**

  This command enables the DHCPv6 vendor-specific configuration mode. The following configuration command options are available in this mode:

  - *vendor-id*

    Enter a vendor-specific identification number. This number is the vendor IANA Private Enterprise Number. The range is 1 to 4294967295. Cisco's Enterprise Number (vendor ID) is 9.

  - **suboption** *number*

    This command sets the vendor-specific suboption number. The range is 1 to 65535. Enter an IPv6 address, ASCII text, or a hexadecimal string, as defined by the suboption parameters.

    - **TFTP Server Addresses option**

      Use **suboption 1** for the TFTP Server Addresses option, and define the IPv6 addresses of the TFTP servers from which the client obtains its configuration file. List the TFTP server addresses in order of preference, and the client will attempt to obtain its configuration file from the TFTP servers in the order in which the addresses are listed.

    - **TFTP Service Name option**

      Use **suboption 2** for the TFTP Service option that contains the name for the locally assigned TFTP Service. If no TFTP Server Addresses are provided in the DHCP response, this name is resolved though a DNS service query. The name resolution may result in several addresses returned by the DNS server. This list contains the addresses of the TFTP servers from which the client obtains its configuration file. The TFTP server addresses are returned with an order of preference, and the client attempts to contact the target server with the lowest-numbered priority.

After you create the DHCPv6 configuration information pool, use the **ipv6 dhcp server** interface configuration command to associate the pool with a server on an interface. However, if you do not configure an information pool, you still need to use the **ipv6 dhcp server** interface configuration command to enable the DHCPv6 server function on an interface.

When you associate a DHCPv6 pool with an interface, only that pool services requests on the associated interface. The pool can also service other interfaces. If you do not associate a DHCPv6 pool with an interface, that pool can service requests on any interface.

Not using any IPv6 address prefix means that the pool returns only configured options.

The **link-address** keyword allows matching a link address without necessarily allocating an address. You can match the pool from multiple relays by using multiple **link-address** configuration commands inside a pool.

Because a longest match is performed on either the address pool information or the link information, you can configure one pool to allocate addresses and another pool on a subprefix that returns only configured options.

Trivial File Transfer Protocol (TFTP)

Within any Cisco Unified Communications Manager (Unified CM) system, endpoints such as IP phones rely on a TFTP-based process to acquire configuration files, software images, and other endpoint-specific information. The Cisco TFTP service is a file serving system that can run on one or more Unified CM servers. It builds configuration files and serves firmware files, ringer files, device configuration files, and so forth, to endpoints. To allow the TFTP server to serve files to devices using IPv6 signaling, the TFTP server inherits the IPv6 server address (the address configured through the server OS command line interface or the Cisco Unified Operating System Administration graphical user interface).

Note Peer-to-peer image file distribution is not supported with IPv6. However, a local IPv6 load server can be configured on IPv6 phones.

Network Time Protocol (NTP)

Network Time Protocol (NTP) enables network devices to synchronize their clocks to a network time server or network-capable clock. NTP is critical for ensuring that all devices in a network have the same time. When troubleshooting or managing a collaboration network, it is crucial to synchronize the time stamps within all error and security logs, traces, and system reports on devices throughout the network. This synchronization enables administrators to recreate network activities and behaviors based on a common timeline. Billing records and call detail records (CDRs) also require accurate synchronized time.

Unified CM NTP Time Synchronization

Unified CM supports NTP for IPv6. You can configure NTP v6 addresses through Unified CM Admin UI, BAT, and AXL. NTP v6 addresses are sent to a SIP Phone through phone configuration. IP phones do not use IPv6 addresses configured for NTP server under Phone NTP References.

IP Phone NTP Time Synchronization

IP Phone NTP Time Synchronization for IPv6 is not supported.

Cisco IOS and CatOS NTP Time Synchronization

Cisco IOS and Cisco Catalyst OS (CatOS) do **not** support NTP for IPv6. If Cisco IOS NTP is used, IPv4 NTP should be used for clock synchronization.

WAN Infrastructure

Proper WAN infrastructure design is important for normal collaboration operation on a converged network. Proper infrastructure design requires following basic configuration and design best practices for deploying a WAN that is as highly available as possible and that provides guaranteed throughput. Furthermore, proper WAN infrastructure design requires deploying end-to-end QoS on all WAN links. This section discusses specific IPv6 design guidance for WAN infrastructures in collaboration networks. For general guidance on designing WAN infrastructures for collaboration deployments, refer to the Cisco Collaboration System Solution Reference Network Design (SRND), available at Link.

General IPv6 WAN Design Guidance

The following sources provide general guidance for designing IPv6 WAN infrastructures:

- Introduction to IPv6 in Cisco products

  Reference: Link

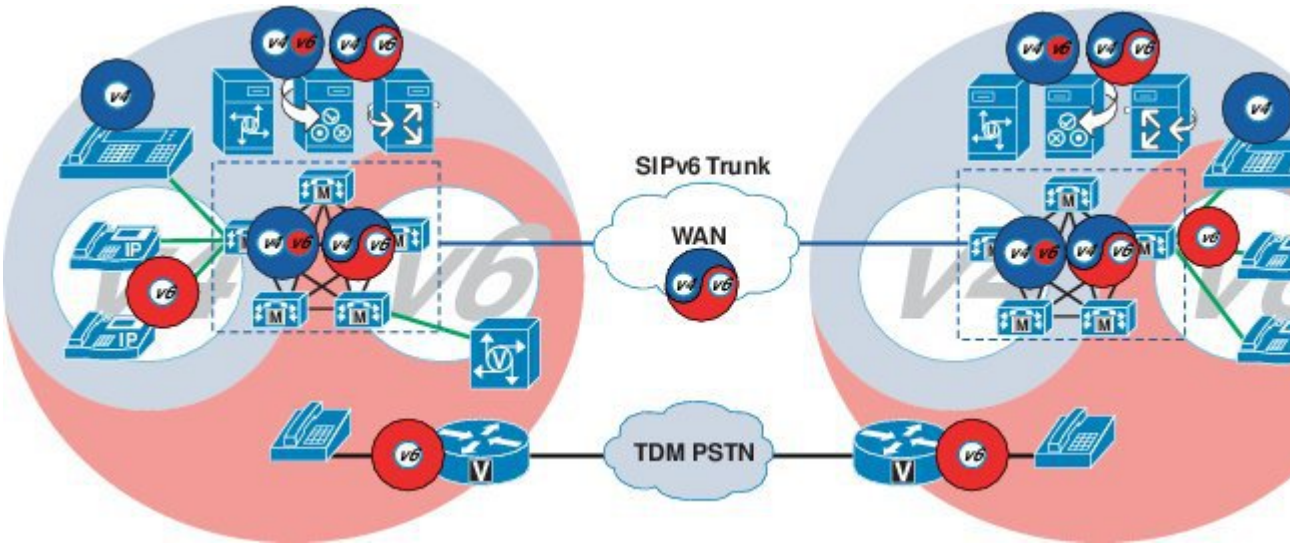
- *Deploying IPv6 in Branch Networks*

  Reference: Link

IPv6 Design Guidance for Unified Communications WAN Infrastructures

You may choose to run IPv6 Unified Communications traffic within your campus network only, in which case you can use standard IPv4 intercluster trunks between Unified CM clusters. If you wish to send IPv6 Unified Communications traffic between Unified CM clusters, then you must use IPv6 SIP intercluster trunks and an IPv6 WAN. Cisco recommends the deployment of both IPv4 and IPv6 routing protocols (a dual-stack WAN) for transporting IPv6 traffic over your WAN infrastructure.

Figure 18: Dual-Stack WAN Infrastructure



The following deployment options are available for deploying IPv6 in a branch campus and across the WAN.

- Run dual-stack (IPv4 and IPv6) routing protocols (Supported).
- Deploy tunneling of IPv6 over IPv4 using:
  - Manually configured GRE tunnels
  - Manually configured IPv6 over IPv4 tunnels
  - Automatically configured IPv6-to-IPv4 (6 to 4) tunnels (RFC 3056)
- IPSEC can also be used to send IPv6 traffic securely in IPv4 tunnels for VPNs.

Call Admission Control

Unified CM supports only locations-based topology-unaware call admission control for IPv6. Resource Reservation Protocol (RSVP) cannot be used as a call admission control technique within the cluster or between clusters. Likewise, Unified CM IPv4 and IPv6 SIP trunks support only locations-based call admission control.

Topology-unaware call admission control requires a hub-and-spoke topology for the WAN, or a spokeless hub in the case of a Multiprotocol Label Switching (MPLS) virtual private network (VPN). This topology ensures that call admission control, provided by Unified CM's locations mechanism, works properly to track the bandwidth available between any two sites in the WAN.

Because using IPv6 requires 20 more bytes of data in its header than IPv4, an IPv6 call requires more bandwidth than a similar IPv4 call that uses the same type of codec and media payload.

To reserve and adjust the location-based bandwidth for a call that uses IPv6, Unified CM calculates the IP bandwidth that is needed for an IPv6 call using any supported codec. After the device contacts Unified CM for bandwidth reservation during the call setup, Unified CM identifies the IP version. If the call uses IPv6, Unified CM reserves the bandwidth for IPv6; and if the call uses IPv4, Unified CM reserves the bandwidth for IPv4. If both IP versions are supported by the devices, Unified CM initially reserves the IPv6 bandwidth and, if required, adjusts the bandwidth after media negotiation occurs.

Locations-Based Call-Counting Call Admission Control

Unified CM also supports a type of locations-based, topology-unaware call admission control know as call counting. Less sophisticated than standard Unified CM locations-based call admission control, call counting uses a fixed bandwidth value for each voice and video call irrespective of the codec or actual bandwidth used.

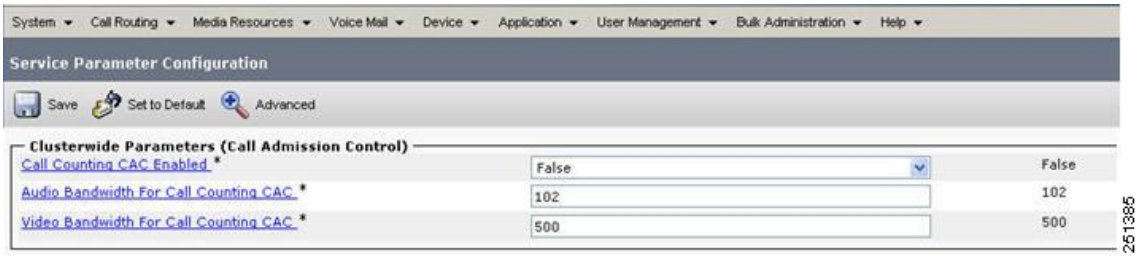
For call-counting call admission control, the following default values are used for Layer 3 voice and video bandwidth when calculating the amount of available bandwidth at a location:

- A voice call = 102 kbps
- A video call = 500 kbps

Although call counting provides a simplified form of call admission control (CAC), it also has the disadvantage that bandwidth reserved for voice and video in the WAN might not be used efficiently.

To enable call counting in Unified CM Administration, select **Service Parameters** > **Clusterwide Parameters (Call Admission Control)**. The default setting for **Call Counting CAC Enabled** is **False**. The voice and video bandwidth values for call counting are configurable.

Figure 19: Configuring Call Counting for Call Admission Control



IPv6 Bandwidth Provisioning

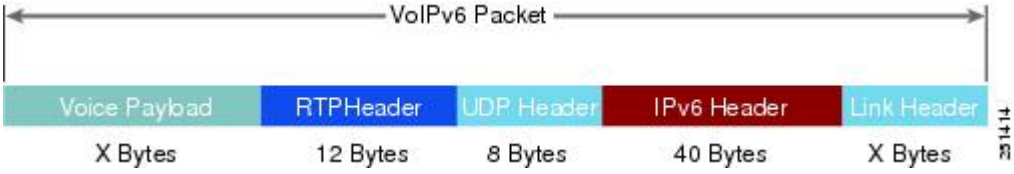
For general recommendations on bandwidth provisioning for collaboration traffic, refer to the bandwidth provisioning information in the Cisco Collaboration System Solution Reference Network Design (SRND), available at Link. However, when provisioning for IPv6 voice bearer traffic, you must consider the additional 20-byte overhead of the IPv6 header.

Note To determine the bandwidth requirements for video flows, refer to the bandwidth provisioning information in the Cisco Collaboration System Solution Reference Network Design (SRND), available at Link.

IPv6 Voice Bearer Traffic

As illustrated in the following figure, a Voice-over-IPv6 (VoIPv6) packet consists of the voice payload, Real-Time Transport Protocol (RTP) header, User Datagram Protocol (UDP) header, IPv6 header and Layer 2 Link header. When Secure Real-Time Transport Protocol (SRTP) encryption is used, the voice payload for each packet increases by 4 bytes. The link header varies in size according to the Layer 2 media used.

Figure 20: Typical VoIPv6 Packet



IPv6 Bandwidth Calculations

To calculate the bandwidth consumed by VoIPv6 streams, add the packet payload and all headers (in bits), then multiplying by the packet rate per second, as follows:

- Layer 2 bandwidth in kbps = [(Packets per second) * (X bytes for voice payload + 60 bytes for RTP/UDP/IP headers + Y bytes for Layer 2 overhead) * 8 bits] / 1000

- Layer 3 bandwidth in kbps = [(Packets per second) * (X bytes for voice payload + 60 bytes for RTP/UDP/IP headers) * 8 bits] / 1000
- Packets per second = [1/(sampling rate in msec)] * 1000
- Voice payload in bytes = [(codec bit rate in kbps) * (sampling rate in msec)] / 8

The following table details the Layer 3 bandwidth per VoIPv6 flow. It lists the bandwidth consumed by the voice payload and IPv6 header only, at a default packet rate of 50 packets per second (pps) and at a rate of 33.3 pps for both non-encrypted and encrypted payloads. It does not include Layer 2 header overhead. The codec sampling rate can be adjusted through the Unified CM Service Parameters menu.

Table 11: Layer 3 Bandwidth per VoIPv6 Flow

CODEC	Sampling Rate	Voice Payload in Bytes	Packets Per Second	Bandwidth Per Conversation
G.711 and G.722-64k	20ms	160	50.0	88.0 kbps
G.711 and G.722-64k (SRTP)	20ms	164	50.0	89.6 kbps
G.711 and G.722-64k	30ms	240	33.3	79.2 kbps
G.711 and G.722-64k (SRTP)	30ms	244	33.3	81.0 kbps
iLBC	20ms	38	50.0	39.2 kbps
iLBC (SRTP)	20ms	42	50.0	40.8 kbps
iLBC	30ms	50	33.3	29.3 kbps
iLBC (SRTP)	30ms	54	33.3	30.4 kbps
G.729A	20ms	20	50.0	32.0 kbps
G.729A (SRTP)	20ms	24	50.0	33.6 kbps
G.729A	30ms	30	33.3	24.0 kbps
G.729A (SRTP)	30ms	34	33.3	25.0 kbps

Compressed RTP (cRTP)

Cisco IOS does not currently support Compressed RTP for IPv6.

A more accurate method of bandwidth provisioning is to include the Layer 2 headers in the bandwidth calculations. The following table lists the amount of bandwidth consumed by IPv6 voice traffic when the Layer 2 headers are included in the calculations.

Table 12: Bandwidth Consumption with Layer 2 Headers Included

CODEC type and packet rate (packets per second)	Header Type and Size						
	Ethernet 14	PPP 6 Bytes	ATM 53-Byte Cells with a 48-Byte Payload	Frame Relay 4 Bytes	MLPPP 10 Bytes	MPLS 4 Bytes	WLAN 24 Bytes
G.711 and G.722-64k at 50.0 pps	93.6 kbps	90.4 kbps	109.2 kbps	89.6 kbps	92 kbps	89.6 kbps	97.6 kbps
G.711 and G.722-64k (SRTP) at 50.0 pps	95.2 kbps	92 kbps	110.8 kbps	91.2 kbps	93.6 kbps	91.2 kbps	99.2 kbps
G.711 and G.722-64k at 33.3 pps	83.7 kbps	81.5 kbps	94.0 kbps	80.1 kbps	82.6 kbps	80.1 kbps	86.3 kbps
G.711 and G.722-64k (SRTP) at 33.3 pps	84.7 kbps	82.6 kbps	95.1 kbps	82.1 kbps	83.7 kbps	82.1 kbps	87.4 kbps
iLBC at 50.0 pps	44.8 kbps	41.6 kbps	60.4 kbps	40.8 kbps	43.2 kbps	40.8 kbps	48.8 kbps
iLBC (SRTP) at 50.0 pps	46.4 kbps	43.2 kbps	62.0 kbps	42.4 kbps	44.8 kbps	42.4 kbps	50.4 kbps
iLBC at 33.3 pps	33.0 kbps	30.9 kbps	43.5 kbps	30.4 kbps	32.0 kbps	30.4 kbps	35.7 kbps
iLBC (SRTP) at 33.3 pps	34.1 kbps	32.0 kbps	44.5 kbps	31.5 kbps	33.1 kbps	31.5 kbps	36.8 kbps
G.729A at 50.0 pps	37.6 kbps	33.4 kbps	53.2 kbps	33.6 kbps	36.0 kbps	33.6 kbps	41.6 kbps
G.729A (SRTP) at 50.0 pps	39.2 kbps	36.0 kbps	54.8 kbps	35.2 kbps	37.6 kbps	35.2 kbps	43.2 kbps
G.729A at 33.3 pps	27.7 kbps	25.6 kbps	38.1 kbps	25.1 kbps	26.7 kbps	25.1 kbps	30.4 kbps

CODEC type and packet rate (packets per second)	Header Type and Size						
	Ethernet 14	PPP 6 Bytes	ATM 53-Byte Cells with a 48-Byte Payload	Frame Relay 4 Bytes	MLPPP 10 Bytes	MPLS 4 Bytes	WLAN 24 Bytes
G729A (SRTP) at 33.3 pps	28.8 kbps	26.7 kbps	39.2 kbps	26.1 kbps	27.8 kbps	26.1 kbps	31.5 kbps

Call Control Traffic Provisioning

Provisioning for call control traffic should not be a concern in a single-site Unified CM campus deployment. For multi-site WAN deployments with centralized or distributed call processing, you also need to consider bandwidth provisioning for inter-site signaling or intercluster trunk signaling traffic. For information on bandwidth provisioning for call control traffic over IPv4 trunks, refer to the Cisco Collaboration System Solution Reference Network Design (SRND), available at Link. For IPv6 signaling, add 10% to the bandwidth value calculated for call control traffic over IPv4.

RSVP

Resource Reservation Protocol (RSVP) call admission control is not supported for IPv6 calls. RSVP is not supported over Unified CM SIP trunks. Instead, use locations-based call admission control for intercluster trunks. We do **not** recommend the deployment of IPv6 in networks that use RSVP for call admission control.

WLAN

IPv6 is not supported by any Cisco wireless device. These devices support IPv4 only.

Network Management

This section discusses Cisco Prime Collaboration, a network management product.

Cisco Prime Collaboration

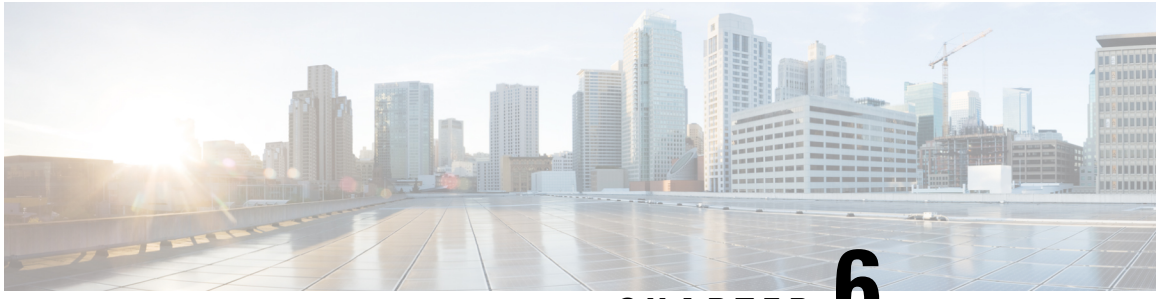
IPv6-aware Cisco Prime Collaboration provides simplified, unified management across voice and video collaboration networks. It offers automated provisioning, real-time monitoring, and proactive troubleshooting, plus long term trending and analytics—in one integrated product.

Cisco Prime Collaboration Advanced includes three separate modules:

- Provisioning (IPv6-aware and supports dual-stack and IPv6-only devices)
- Assurance (IPv4-only)

- Analytics (IPv4-only)

Cisco Prime Collaboration Standard includes a subset of the features available in the Provisioning and Assurance modules. The Analytics module and Cisco Prime Collaboration Contact Center Assurance are available as part of the Cisco Prime Collaboration Advanced offer only (IPv4-only).



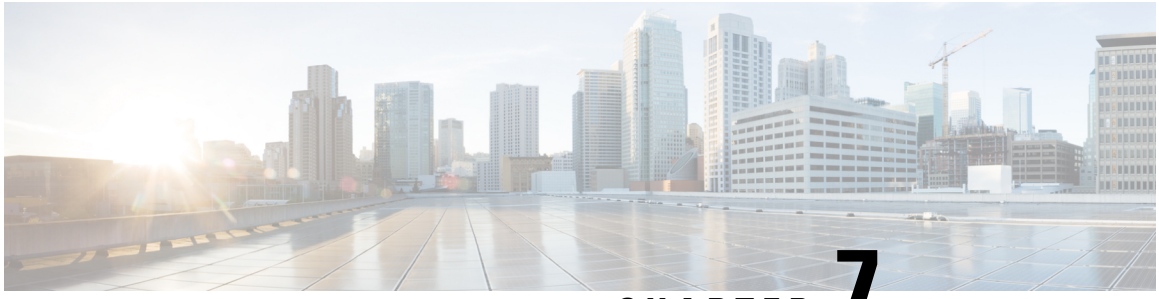
CHAPTER 6

Gateways

Gateways Overview

Gateways provide a number of methods for connecting an IP telephony network to the public switched telephone network (PSTN), legacy PBX systems, key systems, or analogue devices. Gateways range from specialized, entry-level and standalone voice gateways to high-end, feature-rich integrated routers and Cisco Catalyst gateways. For general guidance on gateway selection and features, refer to the Cisco Collaboration System Solution Reference Network Design (SRND), available at Link.

- IPv4-only stack supported gateways with SIP protocol as shown in: Supported IPv6 Addressing Modes , on page 26.
- Software and hardware media termination points (MTPs) for conversion between IPv4 and IPv6 use SCCP IPv4 signaling to Unified CM.
- All other gateway connections that use SCCP, H.323, and MGCP signaling protocols between Unified CM and the gateway have been deprecated.
- You can combine ISDN gateways, Unified SRST analog port, and MTP functionality on a single Cisco Integrated Services Router (ISR) platform. Unified SRST and CUBE cannot co-exist in the same router.
- Cisco 2800 and 3800 Series Integrated Services Routers and Cisco VG Series Gateways must be configured as IPv4-only stack gateway.



CHAPTER 7

Trunks

Trunks Overview

Cisco Unified Communications Manager (Unified CM) supports several different types of IP trunks for connectivity with external devices:

- H.225 (H.323)
- SIP
- Intercluster trunks

Only SIP trunks and SIP intercluster trunks can support IPv6. This chapter describes the new IPv6 features and capabilities of these trunks. For information on the general capabilities and functions of Unified CM trunks, refer to the Cisco Collaboration System Solution Reference Network Design (SRND), available at Link .

There are several possible configurations for Unified CM SIP trunks:

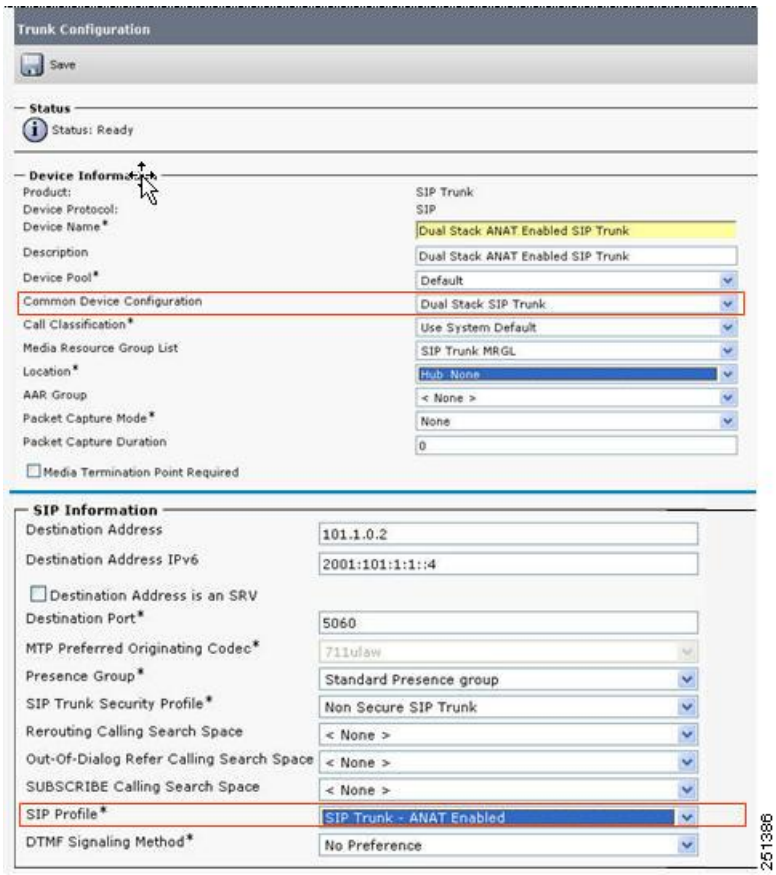
- Inbound and outbound SIP Early Offer trunk calls
- Inbound and outbound SIP Delayed Offer trunk calls (This chapter focuses on this supported option)

IPv6 SIP Trunks Configuration

To configure SIP trunks to gateways and Unified CM SIP intercluster trunks, select **Devices > Trunks > SIP Trunk in Unified CM Administration**.

The SIP trunk configuration settings discussed in this section are applied through the Common Device Configuration profile that is created and assigned to the SIP trunk (**IP Addressing Mode** and **IP Addressing Mode Preference for Signaling**), and through the SIP Profile configuration assigned to the SIP trunk (**ANAT is not enabled**). The IPv4 to IPv6 media interworking is supported by Unified CM inserting MTPs.

Figure 21: Trunk Configuration in Unified CM Administration



Common Device Configuration Settings for SIP Trunks

This section describes the configuration settings for SIP trunks.

SIP Trunk IP Addressing Mode

You can configure the IP Addressing Mode to one of the following settings:

- IPv4

  In this mode, the SIP trunk uses the Unified CM IPv4 address for signaling and either an MTP or phone IPv4 address for media.

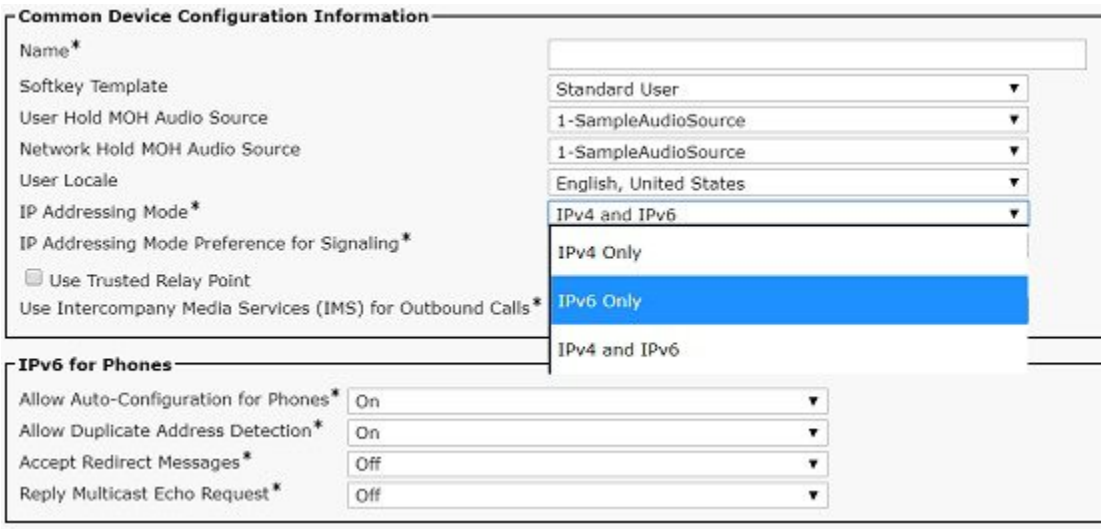
- IPv6

  In this mode, the SIP trunk uses the Unified CM IPv6 address for signaling and either an MTP or phone IPv6 address for media.

- IPv4 and IPv6

  The SIP trunk uses either the Unified CM IPv4 address or the Unified CM IPv6 address for signaling, and an MTP or other trunk address for media.

Figure 22: IP Addressing Mode



SIP Trunk IP Addressing Mode Preference for Signaling

You can configure the IP Addressing Mode Preference for Signaling to one of the following settings:

- IPv4

  In this mode, the SIP trunk uses the Unified CM IPv4 server address as its source address for SIP signaling.

- IPv6

  In this mode, the SIP trunk uses the Unified CM IPv6 server address as its source address for SIP signaling.

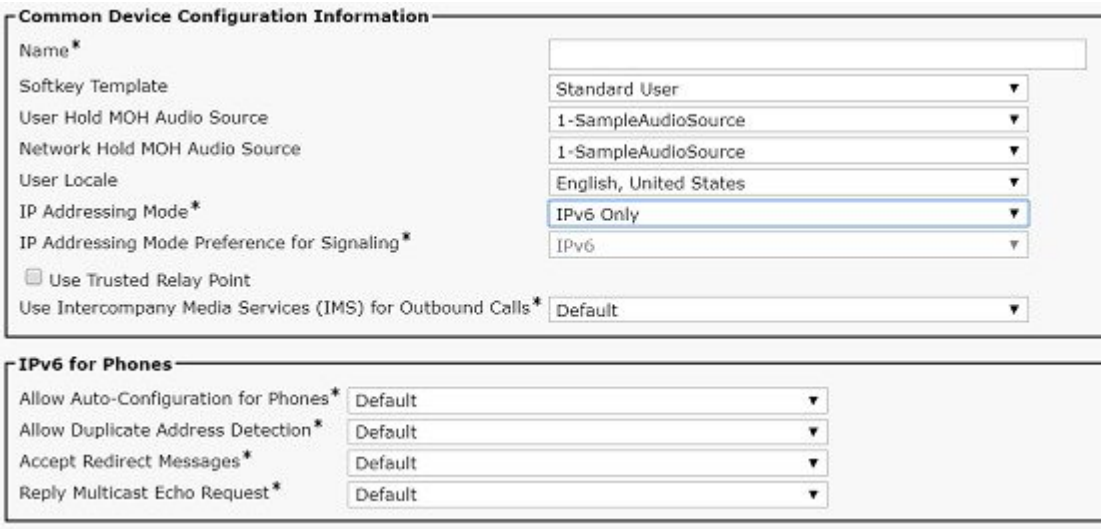
- Use System Default

  In this mode, the SIP trunk uses the cluster-wide Enterprise Parameter configuration value for its IP addressing mode for signaling.

If IPv6 is enabled in the Unified CM cluster, the default SIP trunk setting for the IP Addressing Mode for Signaling is **Use System Default**. With this setting, the SIP trunk adopts the cluster-wide setting for its IP addressing mode for signaling, if the trunk is configured with a destination address of that type. All IPv4 trunks ignore this setting.

The SIP trunk's IP Addressing Mode Preference for Signaling is used only for outbound calls. Unified CM listens for incoming SIP signaling on both the IPv4 and IPv6 address.

Figure 23: P Addressing Mode Preference for Signaling



Allow Auto-Configuration for Phones

The setting of Allow Stateful Auto-Configuration for Phones is not used by SIP trunks.

Supported IPv6 SIP Trunk Configurations and Associated Call Flows

How you configure your Unified CM IPv6 SIP trunk will, to some extent, depend upon the capabilities of the far-end SIP trunk device. Usually this far-end SIP trunk device is another Unified CM cluster, IPv6 SIP gateway, or third-party IPv6 SIP call agent.

Some general guidance on IPv6 SIP trunk configuration:

- IPv6 SIP trunks should be configured with an IP addressing mode of IPv4 and IPv6.
- SIP Early Offer and SIP Delayed Offer are supported, both in symmetric and asymmetric configurations, as follows:
  - Outbound and inbound SIP Early Offer (**Supported** option)
  - Outbound and inbound SIP Delayed Offer
  - Outbound SIP Early Offer and inbound SIP Delayed Offer
  - Outbound SIP Delayed Offer and inbound SIP Early Offer

Early Offer and SIP Trunk Calls

For all Unified CM SIP trunks, you must check the **MTP required** check box on the trunk configuration page to enable SIP Early Offer in IPv4 (IPv6 does not support Early Offer). When **MTP required** is checked, a

media termination point (MTP) is used in the media path for all inbound and outbound calls. This statically assigned MTP affects all calls in the following ways:

- Because the MTP is placed in the media path for all calls, rather than having one call leg from the calling phone to the called phone, the insertion of the MTP creates two legs: one from the calling phone to the MTP, and the other from the MTP to the called phone. For signaling purposes, this can be considered to be two calls. The calling phone and MTP negotiate media capabilities (such as codec, IP addresses, and UDP port numbers to be used), as do the MTP and the called phone at the far end of the SIP trunk.
- The statically assigned MTP (**MTP required** checked) must be configured to use one codec type (G711 or G729). Assigning a single voice codec to this statically assigned MTP disables the use of the pass-through codec. This, in turn, prevents the negotiation of the pass-through codec that is required for video calls or encrypted calls. (T.38 fax calls are supported with statically assigned MTPs.) Therefore, if support for video or encryption is required over the SIP trunk, SIP Delayed Offer (no statically assigned MTP) must be used.

Note The pass-through codec should be configured on all dynamically inserted MTPs. To enable the use of the pass-through codec, configure the MTP with both a standard codec and the pass-through codec.

If SIP Early Offer is required for dual-stack SIP Unified CM trunks, then you must configure the Cisco IOS MTP to use both an IPv6 and IPv4 address. For details, see Media Resources and Music on Hold Overview, on page 79.

Delayed Offer and SIP Trunks

Delayed Offer trunks do not have a statically assigned MTP and therefore MTP resources are not used for every call. For Delayed Offer calls, Unified CM attempts to set up the call using a single call leg between the calling and called device, and in doing so must consider the IP addressing mode configuration of both the Unified CM trunk and the IP phone registered with Unified CM. In certain calls where there are IP addressing mode mismatches between the Unified CM trunk and the registered phone, Unified CM dynamically inserts an MTP to resolve this mismatch. The pass-through codec is supported by this dynamically inserted MTP, and video calls and encrypted calls can be established with this MTP in the call path. The pass-through codec should be configured on all dynamically inserted MTPs. To enable the use of the pass-through codec, configure the MTP with both a standard codec and the pass-through codec.

Unified CM SIP Trunk Signaling

The following factors affect which IP addressing version is used for signaling on Unified CM SIP trunks:

- Call direction
- IP addressing mode of the trunk
- Configured destination addresses of the trunk
- Trunk's IP addressing mode preference for signaling
- Cluster-wide IP addressing mode preference for signaling

IP Addressing Version Used for SIP Signaling for Outbound

The IP addressing version for signaling is determined by the following factors, in the order listed here:

1. The IP Addressing Mode of the SIP trunk (IPv4 or IPv6)
2. The configured destination addresses of the SIP trunk (IPv4 or IPv6)
   - If only one destination address is configured (IPv4 or IPv6), the IP addressing version must match the IP Addressing Mode of the trunk. If these two values do not match, the SIP trunk connection is not established.
   - If two trunk destination addresses are configured (IPv4 and IPv6), then the IP addressing version is determined by the SIP trunk's IP Addressing Mode Preference for Signaling (IPv4, IPv6, or Use System Default). If the **Use System Default** setting is used, then the IP addressing version is determined by the cluster-wide IP Addressing Mode Preference for Signaling (IPv4 or IPv6).

IP Addressing Version Used for SIP Signaling for Inbound

For inbound calls, the IP addressing version used for signaling is based on the trunk destination addresses and port numbers configured in Unified CM. If the signaling source address and port number received from the calling device match a configured destination address and port number on the SIP trunk, then the signaling connection is established.

Unified CM provides the following configuration setting options for the SIP trunk destination address:

- One IPv4 address configured
- One IPv6 address configured
- One IPv4 and one IPv6 address configured

If IPv6 is enabled in the cluster, Unified CM servers listen for incoming SIP trunk calls destined to their configured IPv4 and IPv6 addresses and source port number.

Media Address Selection for Calls over Dual-Stack SIP Trunks

Many configuration options are possible for SIP trunks. Trunks may be single or dual stack and use SIP Early Offer or SIP Delayed Offer. This chapter, while not exhaustive, discusses the significant configuration options and their outcomes in terms of the addresses that are exchanged and used for media. Early Offer call scenarios are considered first, followed by Delayed Offer call scenarios.

Depending on the call scenario, media address selection for calls over dual-stack SIP trunks can be based on:

- Call direction
- Whether Delayed Offer or Early Offer is used
- The IP Addressing Mode of the trunk
- The cluster-wide IP Addressing Mode Preference for Media
- The IP Addressing Mode of the phone

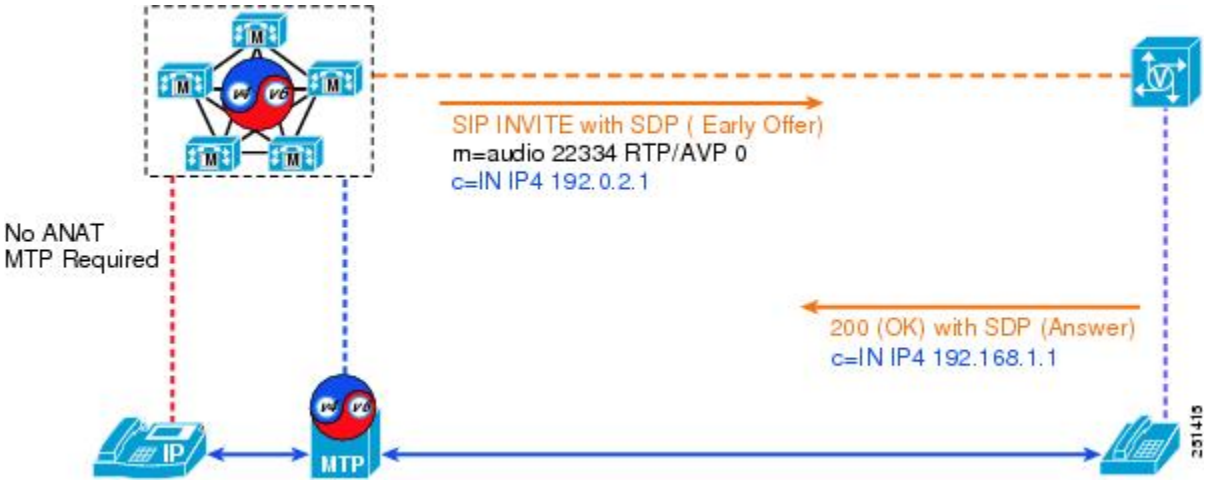
The remaining sections of this chapter review media selection for the following Unified CM call flows:

- SIP Early Offer calls
  - Outbound Early Offer calls without ANAT
  - Inbound Early Offer calls without ANAT
- SIP Delayed Offer calls
  - Outbound Delayed Offer calls without ANAT
  - Inbound Delayed Offer calls without ANAT

Media Selection for Outbound IPv6 Early Offer Calls Without ANAT

SIP Early Offer calls involve two call legs: one from the phone to trunk MTP, and the other from the trunk MTP to the SIP voice gateway. The Cisco IOS MTP is configured to support both IPv4 and IPv6 addresses. ANAT has not been enabled on the SIP trunk in the following figure, so as with a standard SIP trunk, only a single IP addressing version is exchanged.

Figure 24: Media Selection on Unified CM SIP Trunks for Outbound Early Offer Calls Without ANAT



Call Leg from Phone to Trunk MTP: Standard Unified CM In-Cluster Negotiation

The MTP is dual-stacked and can match the media addressing type of the phone if it is set to IPv4-only or IPv6-only. If the phone is also dual-stacked, the cluster-wide IP Addressing Mode Preference for Media (IPv4 or IPv6) determines which IP addressing version is used for media.

Call Leg from MTP Trunk to Voice Gateway: ANAT Not Enabled, and One Media Address Is Sent in SDP (IPv4 or IPv6)

For outbound Early Offer calls where ANAT is not enabled, the IP Addressing Mode of the SIP trunk determines what is sent in the SDP body of the SIP Offer, as follows:

- IP Addressing Mode = IPv4 only—The IPv4 address of the MTP is sent in the SDP body.

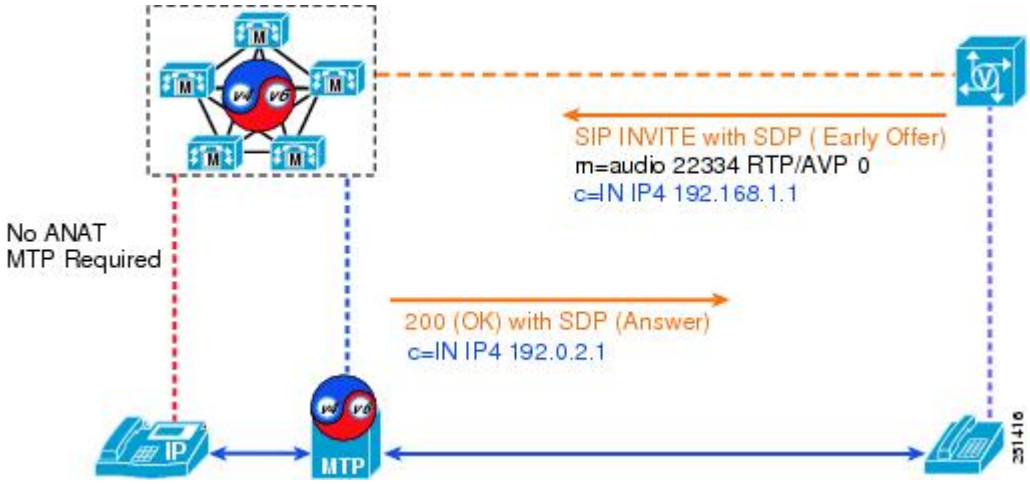
- IP Addressing Mode = IPv6 only—The IPv6 address of the MTP is sent in the SDP body.
- IP Addressing Mode = IPv4 and IPv6—The cluster-wide IP Addressing Mode Preference for Media (IPv4 or IPv6) is used to determine which MTP address is sent in the SDP body.

Media Selection for Inbound Early Offer Calls Without ANAT (IPv6 Not Supported)

Note IPv6 is **not** supported.

SIP Early Offer calls involve two call legs: one from the phone to the trunk MTP, and the other from the trunk MTP to the SIP voice gateway. The Cisco IOS MTP is configured to support both IPv4 and IPv6 addresses. ANAT has not been enabled on the SIP trunk in Figure 7-6, so as with a standard SIP trunk, only a single IP addressing version is exchanged in the SIP Offer and Answer.

Figure 25: Media Selection on Unified CM SIP Trunks for Inbound Early Offer Calls Without ANAT



Call Leg from Trunk MTP to Phone: Standard Unified CM In-Cluster Negotiation

The MTP is dual-stacked and can match the media addressing type of the phone if it is set to IPv4 only or IPv6 only. If the phone is also dual-stacked, the cluster-wide IP Addressing Mode Preference for Media (IPv4 or IPv6) determines which IP addressing version is used for media.

Call Leg from Voice Gateway to Trunk MTP: ANAT Not Enabled, and One Media Address Is Received in SDP

For inbound Early Offer calls where ANAT is not enabled, the IP Addressing Mode of the SIP trunk determines whether the address received in the SDP body of the SIP Offer is accepted or rejected, as follows:

- IP Addressing Mode = IPv4 only:
  - If an IPv4 address is received in the SDP body, proceed with the call.
  - If an IPv6 address is received in the SDP body, reject the call.

- IP Addressing Mode = IPv6 only:
  - If an IPv6 address is received in the SDP body, proceed with the call.
  - If an IPv4 address is received in the SDP body, reject the call.

Note For these trunk calls, Unified CM does not insert an MTP to resolve a media addressing version mismatch between the two voice devices.

- IP Addressing Mode = IPv4 and IPv6:
  - If an IPv4 address is received in the SDP body, proceed with the call.
  - If an IPv6 address is received in the SDP body, proceed with the call.

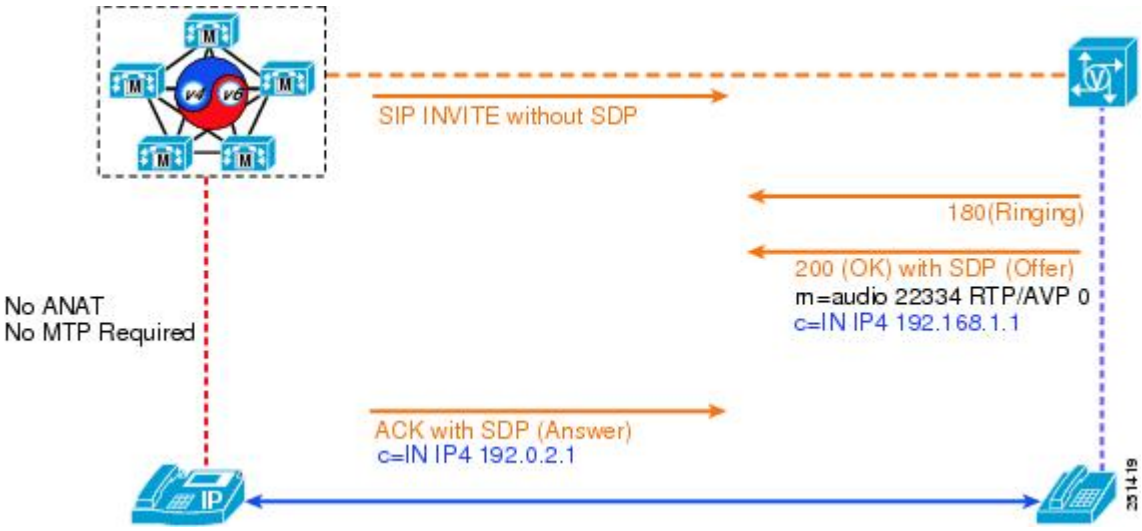
SIP Trunks Using Delayed Offer

With Delayed Offer, SIP trunks do not use a statically assigned MTP, and typically only one call leg is created between the calling phone and called phone or device. From the perspective of Unified CM, this makes the selection of which IP addressing version to use a little more involved because in this case both the trunk's settings and the phone's settings must be taken into account.

Media Selection for Outbound Delayed Offer Calls Over Unified CM SIP Trunks Without ANAT

As shown in the following figure, SIP Delayed Offer calls typically involve a single call leg from the phone to the SIP voice gateway. ANAT has not been enabled on this SIP trunk, so as with a standard SIP trunk, only a single IP addressing version is exchanged in the SIP Offer and Answer.

Figure 26: Media Selection on Unified CM SIP Trunks for Outbound Delayed Offer Calls Without ANAT



For outbound Delayed Offer calls, the IP Addressing Mode settings of both the trunk and the phone influence the call setup in the following ways:

- The IP Addressing Mode setting of the trunk determines whether the received SIP Offer is accepted or rejected.
- The IP Addressing Mode setting of the phone determines which address (phone or MTP) is returned in the SIP Answer from Unified CM.

In this scenario, Unified CM can dynamically insert an MTP, if needed, into the call to convert the IP addressing version of the voice media stream between the calling and called devices. As mentioned previously, dynamically inserted MTPs support the pass-through codec, allowing video calls and encrypted calls to be established.

IP Addressing Mode of the Trunk

- IP Addressing Mode = IPv4 only:
  - If an IPv4 address is received in the SDP body, proceed with the call.
  - If an IPv6 address is received in the SDP body, reject the call.
- IP Addressing Mode = IPv6 only:
  - If an IPv6 address is received in the SDP body, proceed with the call.
  - If an IPv4 address is received in the SDP body, reject the call.

Note For trunk call signaling, Unified CM does not insert an MTP to resolve a media addressing version mismatch.

- IP Addressing Mode = IPv4 and IPv6 (Supported configuration):
  - If an IPv4 address is received in the SDP body, proceed with the call.

- If an IPv6 address is received in the SDP body, proceed with the call.

For SIP trunks using Delayed Offer and not using ANAT, the supported trunk IP Addressing Mode setting is **IPv4 and IPv6** because both IPv6 calls and IPv4 calls are accepted by the trunk.

IP Addressing Mode of the Phone

- IP Addressing Mode = IPv4 only:
  - If an IPv4 address is received in the SDP body, proceed with the call and return the IPv4 address of the phone in the SDP body of the SIP answer.
  - If an IPv6 address is received in the SDP body, dynamically insert an MTP into the media path to convert IP addressing versions, then proceed with the call. Return the IPv6 address of the MTP in the SDP body of the SIP answer.
- IP Addressing Mode = IPv6 only:
  - If an IPv6 address is received in the SDP body, proceed with the call and return the IPv6 address of the phone in the SDP body of the SIP answer.
  - If an IPv4 address is received in the SDP body, dynamically insert an MTP into the media path to convert IP addressing versions, then proceed with the call. Return the IPv4 address of the MTP in the SDP body of the SIP answer.
- IP Addressing Mode = IPv4 and IPv6:
  - If an IPv4 address is received in the SDP body, proceed with the call and return the IPv4 address of the phone in the SDP body of the SIP answer.
  - If an IPv6 address is received in the SDP body, proceed with the call and return the IPv6 address of the phone in the SDP body of the SIP answer.

When an MTP Is Required, Will the MTP of the Phone or the Trunk Be Used?

The cluster-wide IP Addressing Mode Preference for Media determines whether the MTP of the phone or of the trunk is used to convert the voice media stream between IPv4 and IPv6. This preference is used to select an MTP so that the longest Real-Time Transport Protocol (RTP) call leg in the cluster matches the cluster-wide preference.

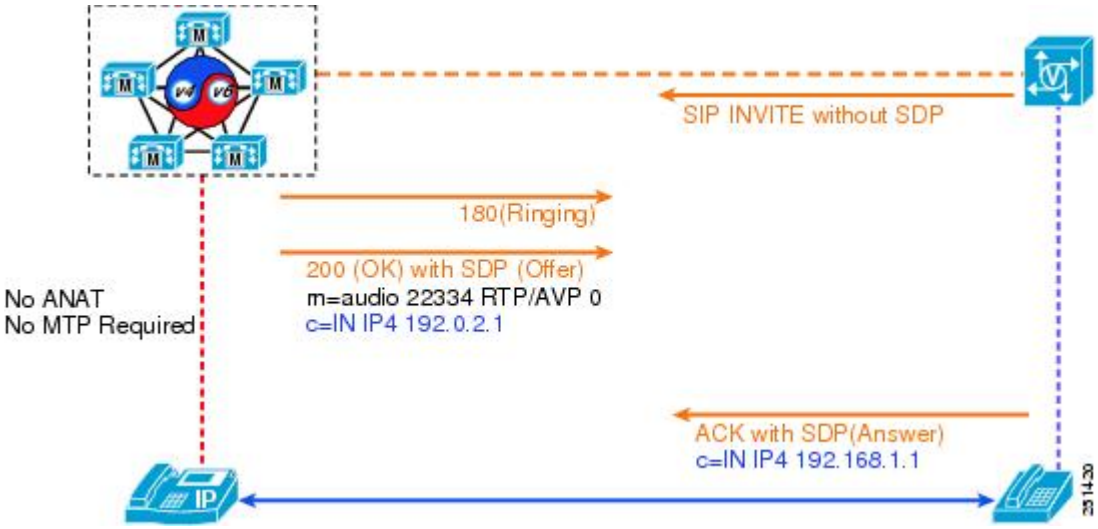
Deployment Considerations for Delayed Offer Calls over Trunks without ANAT

If a call from an IPv4-only phone receives a SIP Offer that contains an IPv6 address, or if a call from an IPv6-only phone receives a SIP Offer that contains an IPv4 address, Unified CM dynamically inserts an MTP to convert between IPv4 and IPv6. In deployments with large numbers of IPv4-only phones, any SIP trunk call to or from an IPv6-only device requires an MTP for conversion between IPv4 and IPv6. Therefore, we recommend that you provide MTP resources for IPv4-only and IPv6-only devices in the Unified CM cluster.

Media Selection for Inbound Delayed Offer Calls Over Unified CM SIP Trunks Without ANAT

As shown in the following figure, SIP Delayed Offer calls typically involve a single call leg from the phone to the SIP voice gateway. ANAT has not been enabled on this SIP trunk, so as with a standard SIP trunk, only a single IP addressing version is exchanged in the SIP Offer and Answer.

Figure 27: Media Selection on Unified CM SIP Trunks for Inbound Delayed Offer Calls Without ANAT



For inbound Delayed Offer calls, the combined settings of the IP Addressing Mode of both the trunk and the phone determine which IP addressing version and which device's IP address is sent in the SDP body of the SIP Offer.

For inbound Delayed Offer calls, if a mismatch exists between the IP addressing modes of the phone and the trunk, Unified CM can dynamically insert an MTP into the call path to convert the IP addressing version of the voice media stream from the IP phone, so that it matches that configured on the trunk. In this case, the address of the MTP is sent in the SDP body of Unified CM's SIP Offer.

For SIP trunks using Delayed Offer and not using ANAT, the supported IP Addressing Mode setting for the trunk is **IPv4 and IPv6**. With this setting, Unified CM does not need to insert MTPs for inbound SIP Delayed Offer calls.

When an MTP Is Required, Will the MTP of the Phone or the Trunk Be Used?

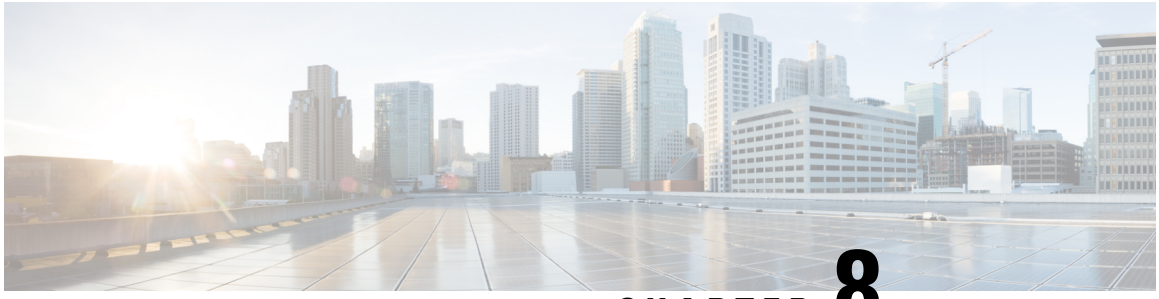
The cluster-wide IP Addressing Mode Preference for Media determines whether the MTP of the phone or of the trunk is used to convert the voice media stream between IPv4 and IPv6. See the following table for details. This preference is used to select an MTP so that the longest Real-Time Transport Protocol (RTP) call leg in the cluster matches the cluster-wide preference.

As mentioned previously, dynamically inserted MTPs do support the pass-through codec, allowing video calls and encrypted calls to be established.

Table 13: IP Addressing Mode Preference for Media

IP Addressing Mode of Phone	IP Addressing Mode of Trunk	Address Sent in SIP Offer by Unified CM
IPv4 Only	IPv4 Only	IPv4 address of phone
IPv4 Only	IPv4 and IPv6	IPv4 address of phone
IPv6 Only	IPv6 Only	IPv6 address of phone
IPv6 Only	IPv4 and IPv6	IPv6 address of phone
IPv4 Only	IPv6 Only	Insert an MTP and use its IPv6 address.
IPv6 Only	IPv4 Only	Insert an MTP and use its IPv4 address.
IPv4 and IPv6	IPv4 Only	IPv4 address of phone
IPv4 and IPv6	IPv6 Only	IPv6 address of phone
IPv4 and IPv6	IPv4 and IPv6 Only	IPv4 or IPv6 address of phone[1]

[1] The cluster-wide IP Addressing Mode Preference for Media determines which phone address (IPv4 or IPv6) Unified CM sends in the SDP body of the SIP Offer.



CHAPTER 8

Media Resources and Music on Hold

Media Resources and Music on Hold Overview

A media resource is a software-based or hardware-based entity that performs media processing functions on the data streams to which it is connected. Media processing functions include mixing multiple streams to create one output stream (conferencing), passing the stream from one connection to another (media termination point), converting the data stream from one compression type to another (transcoding), echo cancellation, signaling, termination of a voice stream from a TDM circuit (coding and decoding), packetization of a stream, streaming audio (Annunciator and Music on Hold), and so forth.

This chapter focuses on new media termination point (MTP) functionality introduced to support IPv6 Collaboration deployments: namely, the capability of Cisco IOS MTPs to convert a voice media stream from IPv4 to IPv6 and conversely. Other media resources such as conferencing and transcoding are also discussed in context with this new MTP functionality. For all other media resource and Music on Hold (MoH) design guidance, refer to the Cisco Collaboration System Solution Reference Network Design (SRND), available at Link.

Media Termination Point (MTP)

The following Cisco Integrated Services Router (ISR) MTPs support media conversion between IPv4 and IPv6 for devices with mismatched media IP address versions:

- Cisco IOS hardware MTPs—Digital signal processors (DSPs) on the Cisco ISR Motherboard and NM-HDV2 with PVDM2 DSPs
- Cisco IOS software MTPs

Cisco IOS Release IOS XE Everest 16.6.1 and above supports these MTPs.

Address Conversion Between IPv4 and IPv6

When a mismatch exists between the IP addressing versions supported by two devices, Unified CM dynamically inserts an MTP resource to convert the voice media stream from IPv4 to IPv6 and conversely. Dynamically inserted MTPs support the pass-through codec, which allows the MTP to support not only voice calls but also video calls and encrypted calls. The pass-through codec should be configured on all dynamically inserted MTPs. To enable the use of the pass-through codec, configure the MTP with both a standard codec and the pass-through codec.

Unified CM trunks can also be configured with a statically assigned MTP (**MTP Required** check box checked). The trunk's statically defined MTP does not support the pass-through codec and supports only a single configured codec (G.711 or G.729), which limits all calls that use this trunk to either G.711 or G.729 voice calls and T.38 Fax calls only. This statically assigned MTP also has the capability to convert the voice media stream from IPv4 to IPv6 and conversely.

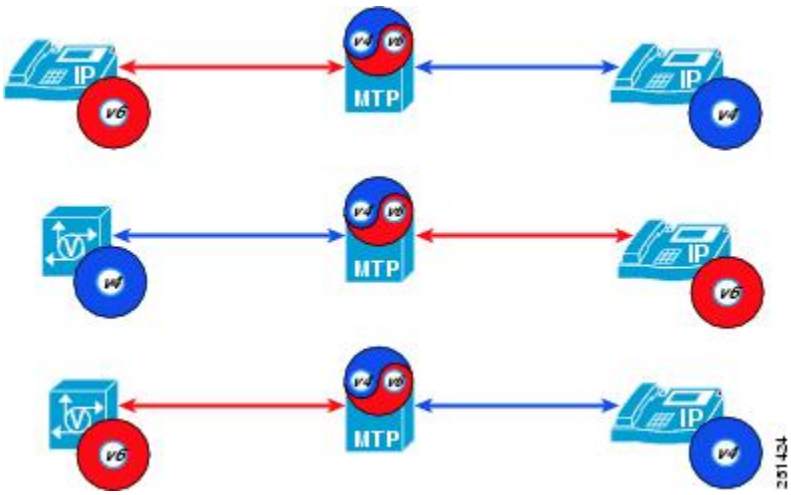
To be dynamically or statically inserted into a call path, the Cisco IOS MTPs must be associated with the media resource group (MRG) for each device (phone or trunk).

The following Cisco IOS configuration is an example of a software MTP. The **sccp local GigabitEthernet0/0** command associates the IPv4 and IPv6 addresses on this interface with the MTP for both SCCP signaling and media addresses.

```
interface GigabitEthernet0/0
ip address 192.168.1.5 255.255.255.0
! MTP's IPv4 address
ipv6 address 2001:0db8:101:1:1::5/64
! MTP's IPv6 address
!
sccp local GigabitEthernet0/0
sccp ccm 192.168.0.15 identifier 1 version 7.0
! Unified CM's IPv4 address
sccp ccm 2001: 0db8:101:1::15 identifier 2 version 7.0
! Unified CM's IPv6 address
sccp
!
sccp ccm group 1
associate ccm 1 priority 1
associate ccm 2 priority 2
associate profile 1 register MTP-1
!
dspfarm profile 1 mtp
codec g711ulaw
codec pass-through
maximum sessions software 100
associate application SCCP
!
```

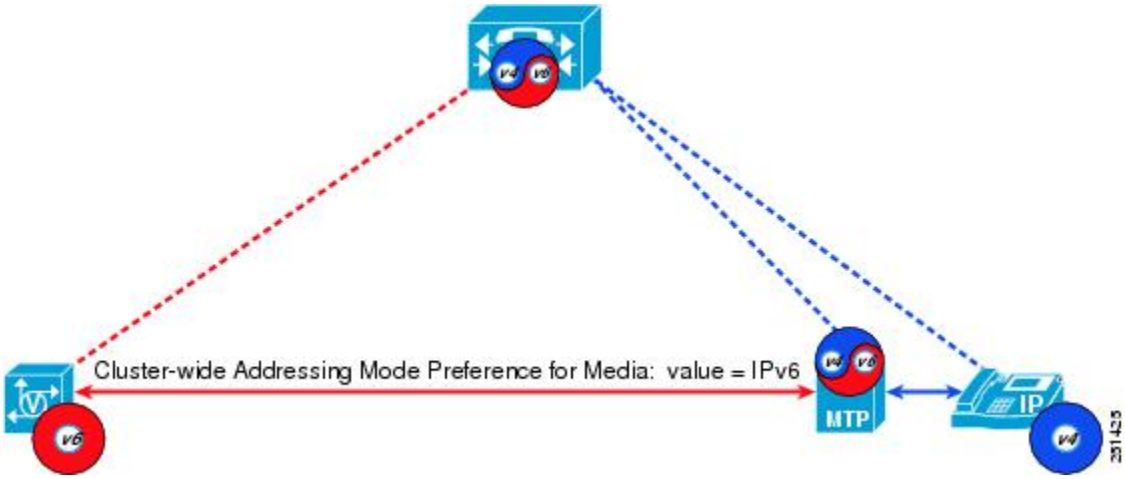
The following figure shows when an MTP is inserted between two devices to convert from IPv4 to IPv6, and conversely. However, both devices can have MTP resources available in their media resource groups, and Unified CM must decide which MTP to use. Unified CM uses the cluster-wide setting of IP Addressing Mode Preference for Media to determine which MTP to use for conversions between IPv4 and IPv6.

Figure 28: MTP Insertion for Address Conversion Between IPv4 and IPv6



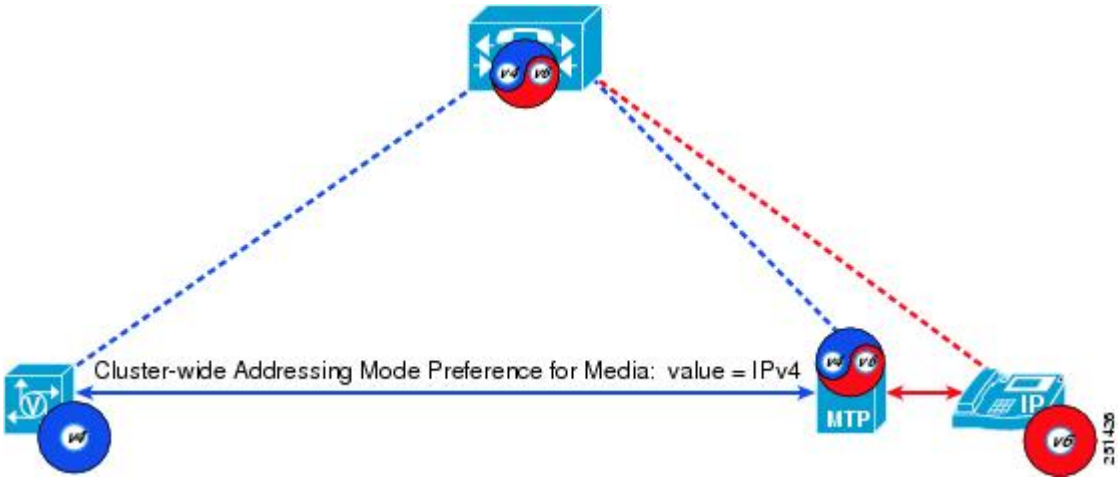
If the cluster-wide IP Addressing Mode Preference for Media is set to IPv6, Unified CM chooses the MTP associated with the IPv4 device.

Figure 29: Unified CM Chooses MTP of the IPv4 Device



Conversely, if the cluster-wide IP Addressing Mode Preference for Media is set to IPv4, Unified CM chooses the MTP associated with the IPv6 device.

Figure 30: Unified CM Chooses MTP of the IPv6 Device



By choosing the device whose addressing mode does not match the cluster-wide IP Addressing Mode Preference for Media, Unified CM selects the cluster-wide media preference value that matches the longest call leg between the two devices in the cluster.

Note If the preferred device's MTP is not available, the other device's MTP is used as a last resort. If no MTPs are available, the call fails.

Transcoder DSP resources can also be used as hardware MTPs. If both transcoding resources and software MTPs exist in the same media resource group, Unified CM uses these media resources in a round-robin fashion for conversions between IPv4 and IPv6. To prioritize transcoding DSP resources for transcoding purposes only, place the software MTP and hardware MTPs in a separate media resource group (MRG) and give this MRG precedence (a higher order) over the transcoder MRG in the media resource group list (MRGL).

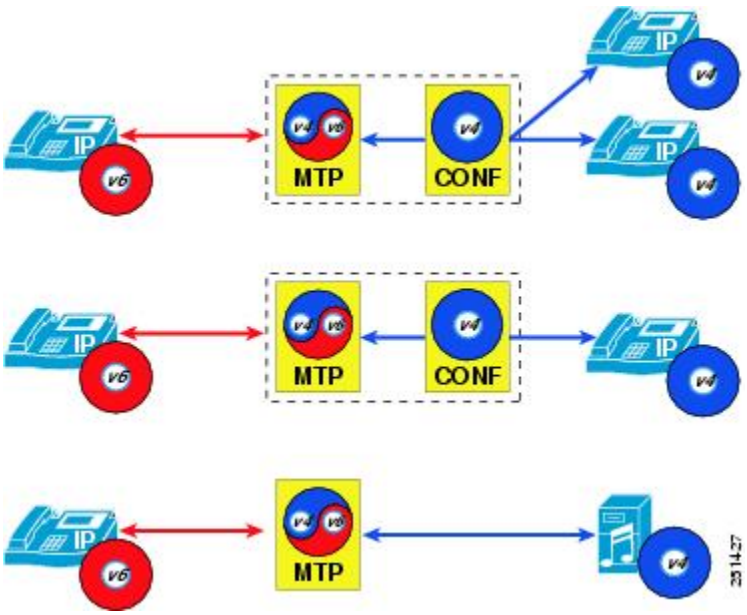
The number of MTPs required is based on the traffic that requires media header addressing conversation from IPv4 to IPv6. In the worst case, the number of MTPs required is the same as the number of IPv6-only devices that are in service. The MTPs should be located in a site where IPv4-only phones are deployed. For IPv6-only sites, the MTPs should be located where servers are deployed, for example, in a data center. MTPs should be located where IPv4 and IPv6 addresses are available. Quicker transition to IPv6- only devices is supported to phase out MTPs from a network.

IPv6 and Other Media Resources

The following media resources do not support IPv6 for voice. If these resources are invoked, an MTP resource is required to convert the voice stream between IPv6 and IPv4, as shown in the following figure.

- Video conferencing
- Video Transcoding
- IPv6 Music on Hold (MoH)

Figure 31: MTP Inserted for Conversions Between IPv4 and IPv6



Voice conferencing resources and hardware MTP resources can reside on the same DSP. Similarly, voice transcoding resources and hardware MTP resources can reside on the same DSP.

IPv6-only devices do not support multicast music on hold.

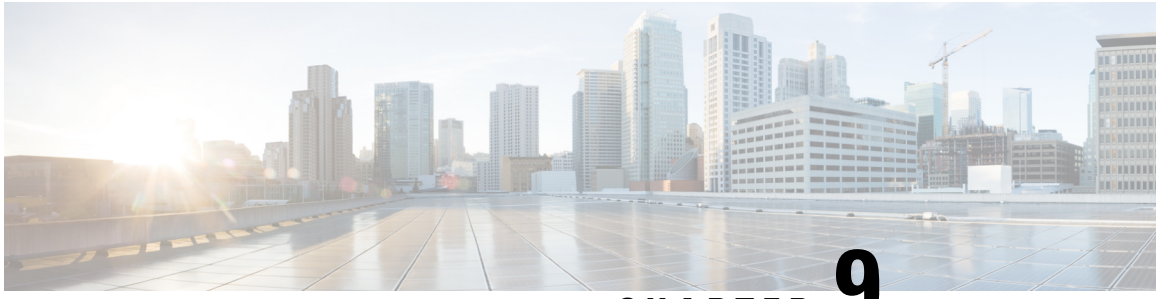
Cisco IOS RSVP Agent and IPv6

The Cisco IOS Resource Reservation Protocol (RSVP) agent does not support IPv6, and RSVPv4 cannot be used within the cluster for call admission control. For SIP trunks and intercluster trunks, locations-based call admission control must be used.

IPv4 to IPv6 Conversion When Multiple MTPs Exist Within a Media Resource Group (MRG)

Unified CM server-based MTPs do not support the pass-through codec. If Cisco IOS MTPs (which support the pass-through codec) and Unified CM MTPs (which do not support the pass-through codec) are listed in the same media resource group, Unified CM requests an MTP with pass-through codec support for conversions between IPv4 and IPv6.

If an MTP is statically assigned to the SIP trunk (Early Offer), then conversions between IPv4 an dIPv6 can occur only for the codec specified on the MTP (G.711 or G.729).



CHAPTER 9

Call Processing and Call Admission Control

Call Processing and Call Admission Control Overview

This chapter discusses aspects of call processing and call admission control that apply specifically to IPv6.

Call Processing

This section describes a few changes to call processing operation for IPv6, along with some IPv6 configuration information. For information on designing scalable and resilient call processing systems with Cisco Unified Communications Manager (Unified CM), refer to the Cisco Collaboration System Solution Reference Network Design (SRND), available at Link.

Enable Call Processing for IPv6

To enable call processing for IPv6, you must first enable IPv6 throughout the Cisco Unified Communications Manager (Unified CM) cluster, as outlined in the following steps:

1. Configure IPv6 on each server in the Unified CM cluster, either through the server operating system (OS) command line interface (CLI) or through the Cisco Unified Operating System Administration graphical user interface (GUI).
2. Configure IPv6 in Cisco Unified CM Administration.

Configure IPv6 on Each Server in Cluster Using CLI

Before you enable IPv6 in Cisco Unified CM Administration, you must configure IPv6 on each server in the cluster by using the following server operating system (OS) CLI commands:

Procedure

Step 1 Enable IPv6 by using the **set network ipv6 service enable** command.

Step 2 Set a static IPv6 address for your server by using the **set network ipv6 static_address** *<addr>* *<mask>* command.

Step 3 To view the platform's IPv6 address settings, use the **show network ipv6 settings** command. Example output from this command is as follows:

```
IPv6 : enabled
DHCPv6 : disabled
IPv6 addresses:
Address:2001:db8:c18:1:21c:c4ff:feef:ca0 Mask:64
Scope:Global Duplicate:no
Address:fe80::21c:c4ff:feef:ca0 Mask:64
Scope:Link
```

Configure Unified CM Server IPv6 Address Using GUI

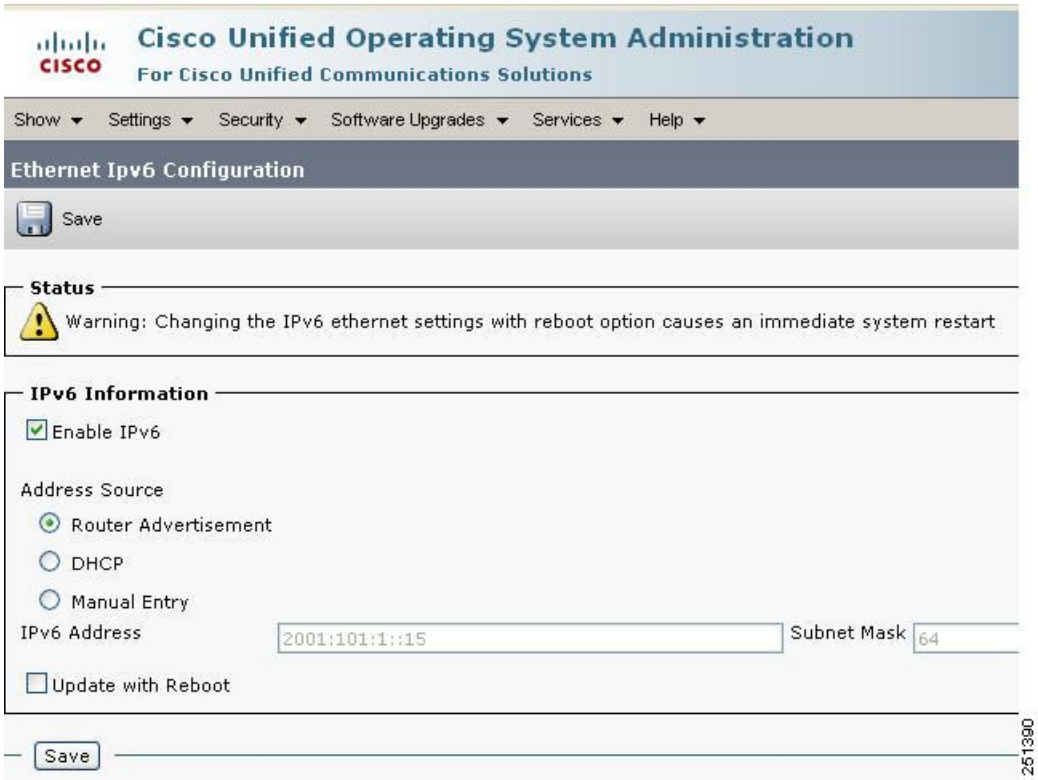
You can configure the IPv6 address of the server platform by using the Cisco Unified Operating System Administration GUI.

Procedure

To set the address, select **Settings > IP > Ethernet IPv6**.

The following figure shows how to configure the IPv6 address of the server platform by using the Cisco Unified Operating System Administration GUI.

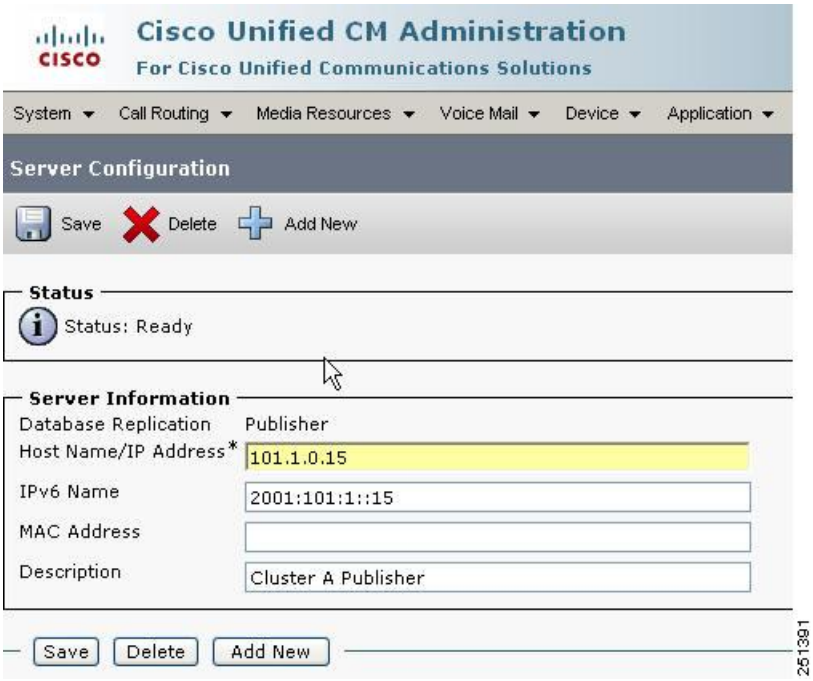
Figure 32: Configuring the Server Platform IPv6 Address in Cisco Unified Operating System Administration



Define Unified CM Server IPv6 Addresses

After you configure the server platform IPv6 address, define the IPv6 address for each Unified CM server by using Cisco Unified CM Administration. Select **System > Server**, and enter the IPv6 address in the **IPv6 Name** field. This IPv6 address allows IP phones to retrieve the IPv6 address of this Unified CM from the configuration file downloaded from the TFTP server.

Figure 33: Configuring the Unified CM Server IPv6 Address in Unified CM Administration

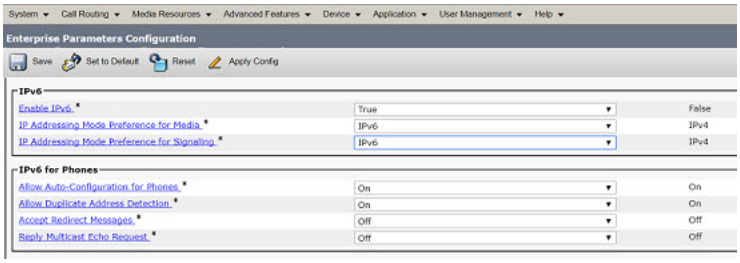


Cluster-Wide IPv6 Configuration

You can configure the following cluster-wide IPv6 settings for each Unified CM server through the Enterprise Parameters page for **IPv6** and **IPv6 for Phones** in Unified CM Administration:

- Enable IPv6 (Set to **True**)
- IP Addressing Mode Preference for Media (Set to **IPv6**)
- IP Addressing Mode Preference for Signaling (Set to **IPv6**)
- Allow Auto-Configuration for Phones (Set to **On**)

Figure 34: Cluster-Wide IPv6 Configuration Modes



Enable IPv6

Set this parameter to **True** to enable IPv6. The default setting is False.

IP Addressing Mode Preference for Media

This parameter has two setting options:

- IPv4 (Default)
- IPv6 (Set to **IPv6**)

This cluster-wide IP Addressing Mode Preference for Media is different than the device-level IP addressing mode, and it serves two purposes:

- The cluster-wide Addressing Mode Preference for Media is used to select which IP addressing version to use for media when a call is made between two dual-stack devices.
- The cluster-wide Addressing Mode Preference for Media is also used when there is a mismatch in supported IP addressing versions between two devices. If an IPv6-only device calls an IPv4-only device, an MTP must be inserted into the media path to convert between IPv4 and IPv6. Typically both devices will have MTP media resources available to them in their media resource group (MRG). The cluster-wide Addressing Mode Preference for Media determines which device's MTP is used to convert between IPv4 and IPv6 for the call.

MTP resource allocation is discussed in detail in Media Resources and Music on Hold Overview, on page 79.

IP Addressing Mode Preference for Signaling

The cluster-wide IP Addressing Mode Preference for Signaling setting is used by devices whose IP Addressing Mode Preference for Signaling is set to **Use System Default**. The cluster-wide IP Addressing Mode Preference for Signaling has two setting options:

- IPv4 (Default)
- IPv6 (Set to **IPv6**)

Allow Auto-Configuration for Phones

The cluster-wide setting of Allow Auto-Configuration for Phones is used by phones whose Allow Auto-Configuration for Phones parameter is set to **Default**. Allow Auto-Configuration for Phones has two settings:

- On (Default)
- Off

Intra-Cluster Communications

There are two primary types of intra-cluster communications, database replication and Intra-Cluster Communication Signaling (ICCS), both of which support IPv4-only.

TFTP Server

Within any Unified CM system, endpoints such as IP phones rely on a TFTP process to acquire configuration files, software images, and other endpoint-specific information. The Cisco TFTP service is a file serving

system that can run on one or more Unified CM servers. It builds configuration files and serves firmware files, ringer files, device configuration files, and so forth, to endpoints.

When IPv6 is enabled in the Unified CM cluster, the TFTP server inherits its IPv6 server address from the configured server address. This allows the TFTP server to serve files to devices using IPv6 signaling.

Unified CM CTI

Computer Telephony Integration (CTI) provides IP address information through the JTAPI and TAPI interfaces, which can support IPv4 and IPv6 addresses. To support IPv6, applications need to use a JTAPI/TAPI client interface version that supports IPv6.

Unified CM AXL/SOAP

Unified CM's Administrative XML (AXL) Simple Object Access Protocol (SOAP) interface is IPv6-aware. The AXL/SOAP interface communicates with IPv4 addresses, but it can receive and understand IPv6 addresses embedded in application protocol data units (PDUs).

SNMP

Simple Network Management Protocol (SNMP) for Unified CM includes:

- SNMPv6 MIB support.
- Ability to accept SNMP requests from v6-only hosts.
- Ability to configure v6-only SNMP Notification Destination.
- IPv6 support for SNMP V1/V2C/V3 protocols.

Cisco Collaboration Applications

Cisco Collaboration applications servers that are documented in this document are dual-stack, but Cisco Prime Collaboration is IPv6-aware. All the third-party applications remain in IPv4-only and are IPv6 aware where applicable.

Unified CM Platform Capacity Planning

IPv6 addresses require more Unified CM server memory when compared with IPv4 addresses. Therefore, in a Unified CM deployment with many IPv6 devices, the busy hour call completion (BHCC) capacity is approximately 3% to 5% less than the capacity of IPv4-only deployments.

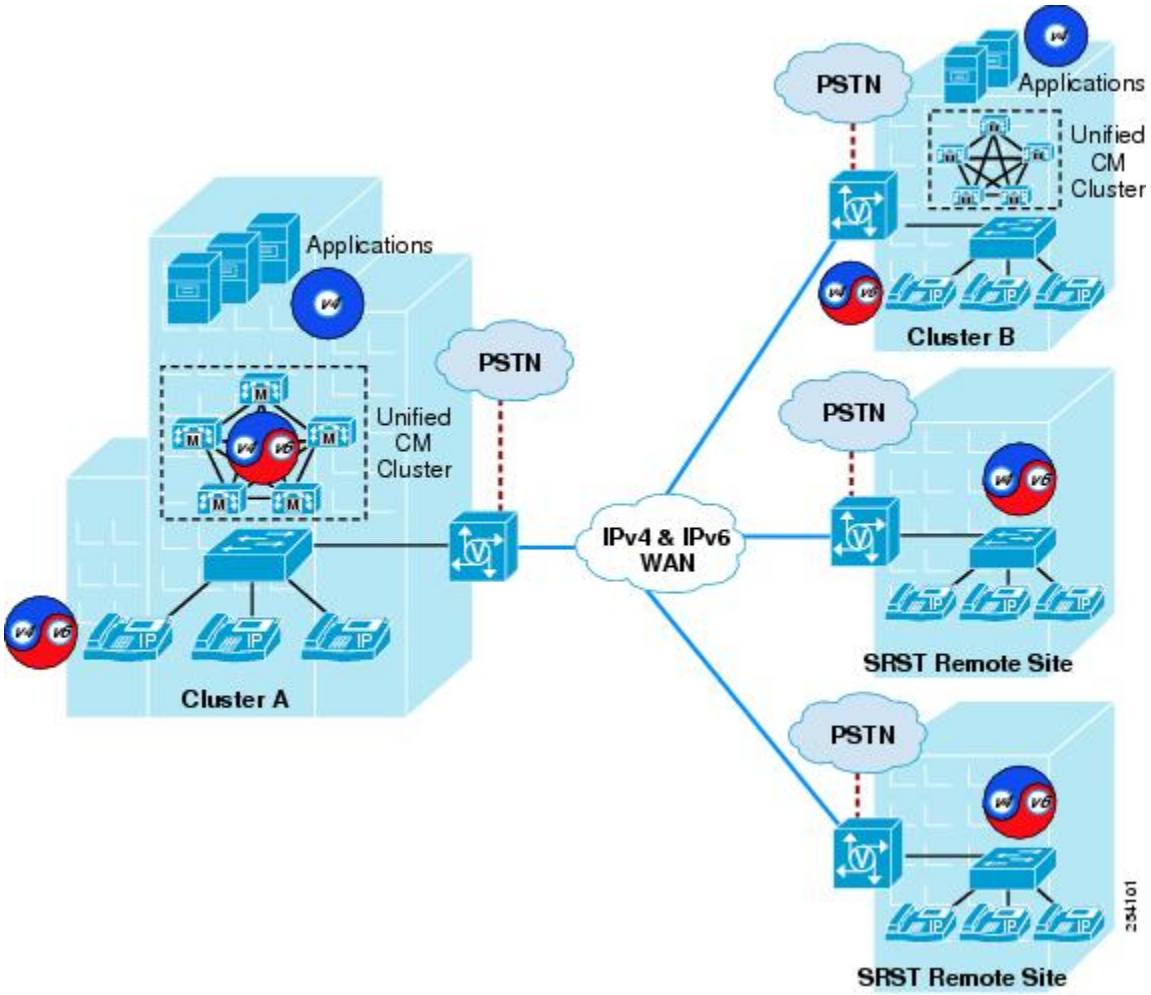
Interoperability of Unified CM and Unified CM Express

Cisco Unified Communications Manager Express (Unified CME) supports IPv4-only. If you are deploying Unified CME with Unified CM, the interface is IPv4-only. Follow the design guidance in the *Cisco Collaboration System Solution Reference Network Design (SRND)*, available at Link.

Call Admission Control (CAC)

IPv6-enabled Unified CM supports single-site deployments, multi-site WAN deployments with distributed call processing, and multi site deployments with centralized call processing. Call admission control is required where calls are made over a WAN between remote sites on the same cluster or between Unified CM clusters.

Figure 35: Multi-Site Collaboration Deployment



Call Admission Control with Unified Communications IPv6 Deployments

Collaboration IPv6 deployments with Unified CM support locations-based topology-unaware call admission control only for calls between remote sites in the same cluster and over intercluster trunks. Topology-unaware call admission control requires the WAN to be hub-and-spoke, or a spokeless hub in the case of a Multiprotocol Label Switching (MPLS) virtual private network (VPN). This topology ensures that call admission control, provided by the locations configuration mechanism in Unified CM, works properly in tracking the bandwidth available between any two sites in the WAN. For general guidance on topology-unaware call admission

control, refer to the *Cisco Collaboration System Solution Reference Network Design (SRND)*, available at Link.

Topology-aware Resource Reservation Protocol (RSVP) cannot be used within the cluster or between clusters as a call admission control technique. For IPv6-enabled Unified CM clusters:

- Locations-based call admission control must be used between sites controlled by the same Unified CM cluster.
- Unified CM SIP trunks support only locations-based call admission control (IPv4 and/or IPv6).
- Unified CM MGCP trunks support only locations-based call admission control (IPv4-only).
- Unified CM H.323 trunks support locations-based call admission control and gatekeeper-controlled call admission (IPv4-only).

For IPv6 traffic, Unified CM uses the values shown in Table 11: Layer 3 Bandwidth per VoIPv6 Flow, on page 59 in its locations-based call admission control algorithm.

Locations-Based Call Counting Call Admission Control

Unified CM also supports a type of locations-based, topology-unaware call admission control known as call counting. Less sophisticated than standard Unified CM locations-based call admission control, call counting uses a fixed bandwidth value for each voice and video call, irrespective of the codec or actual bandwidth used.

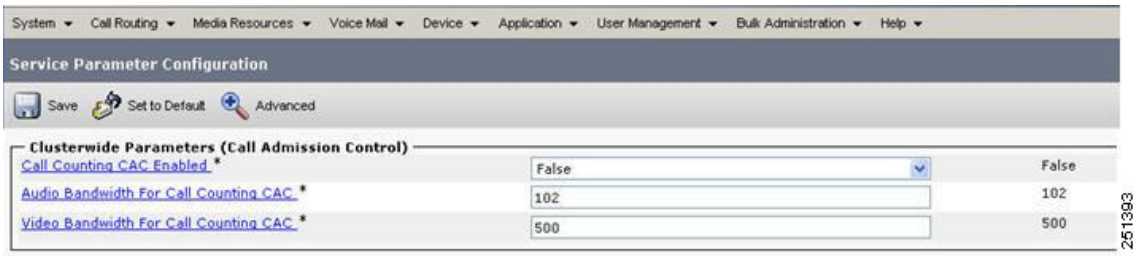
For call counting, the following default values are used for Layer 3 voice and video bandwidth when calculating the amount of available bandwidth at a location:

- Voice calls = 102 kbps
- Video calls = 500 kbps

Although call counting provides a simplified form of call admission control, it also has the disadvantage that bandwidth reserved for voice and video in the WAN might not be used efficiently.

To enable call counting in Unified CM Administration, select **Service Parameters > Clusterwide Parameters (Call Admission Control)**. The default setting for **Call Counting CAC Enabled** is False. The voice and video bandwidth values for call counting are configurable.

Figure 36: Configuring Call Counting



CHAPTER 10

Dial Plan

Dial Plan Overview

The dial plan is one of the key elements of a Unified Communications system, and an integral part of all call processing agents. Generally, the dial plan is responsible for instructing the call processing agent on how to route calls. Specifically, the dial plan performs the following main functions:

- Endpoint addressing

  Reachability of internal destinations is provided by assigning directory numbers (DNs) to all endpoints.

- Path selection

  Depending on the calling device, different paths can be selected to reach the same destination.

- Calling privileges

  Different groups of devices can be assigned to different classes of service, by granting or denying access to certain destinations.

- Digit manipulation

  Sometimes, it is necessary to manipulate the dialed string before routing the call.

- Call coverage

  Special groups of devices can be created to handle incoming calls for a certain service according to different rules (top-down, circular hunt, longest idle, or broadcast).

For general dial plan guidance and design considerations, refer to the Cisco Collaboration Solution Reference Network Design (SRND), available at Link.

IPv6 and Unified CM Dial Plans

The deployment of IPv6 with Cisco Unified Communications Manager (Unified CM) affects two areas of dial plan functionality:

- IPv6 addressing for SIP route patterns
- Path selection considerations for IPv6 calls over IPv6-capable networks

SIP IPv6 Route Patterns

Unified CM can use SIP route patterns to route or block both internal and external calls to SIP endpoints. SIP route patterns can use the destination domain name, an IPv4 address, or an IPv6 address to provide a match for call routing.

A SIP request to call a device can take either of the following forms:

- Using an address:

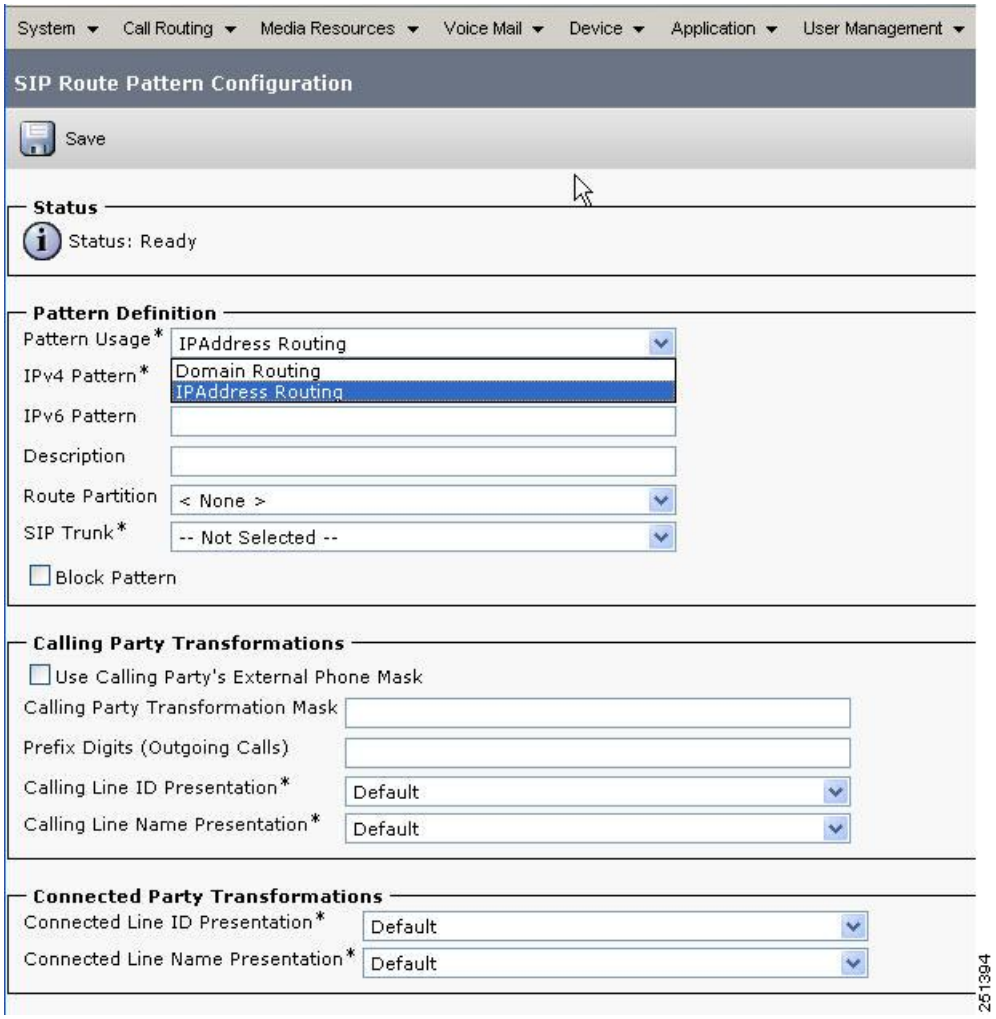
  INVITE sip:5001@2001:0db8:2::1 5060 SIP/2.0

- Using a domain name:

  INVITE sip:5001@example.com 5060 SIP/2.0

To process the SIP request, the Unified CM administrator can add domains, IP addresses, and IP network addresses, and associate them to SIP trunks (only), as shown in the following figure. This method allows requests that are destined for these domains to be routed through particular SIP trunk interfaces.

Figure 37: SIP Route Pattern Configuration in Unified CM



The following guidelines and examples apply to SIP route patterns:

- Domain name examples:
  - example.com
  - my-pc.example.com
  - *.com
  - rtp-ccm[1-5].example.com
- Valid characters for domain names:

  [, -, ., 0-9, A-Z, a-z, *, and]
- If domains names are used, then DNS must be configured in the Unified CM cluster.
- IPv4 address examples:
  - 192.168.201.119 (explicit IP host address)

    - 192.168.0.0/16 (IP network)

- IPv6 address examples:
    - 2001:0db8:2::1 (explicit IPv6 host address)
    - 2001::/16 (IPv6 network)

- Valid characters for IPv6 addresses:

  0-9, A-F, :, and /

Path Selection Considerations for IPv6 Calls

If you create an IPv6 route pattern, then that route pattern must be associated with an IPv6-capable SIP trunk. Likewise, the campus network or WAN that the IPv6 call traverses must be IPv6-capable.

Emergency Services

Cisco Emergency Responder (Emergency Responder) can track the IPv6-only phones through the switch port-based tracking, or access point-based tracking, or as manually configured phones. For the switch port-based tracking, Emergency Responder can talk to Cisco switches configured as the IPv4 address or IPv6 address through the SNMP protocol. Access point-based tracking requires the IPv6-only phone to communicate its upstream infrastructure information to Unified CM. For more information on Access Point Configuration and Discovery, refer to the Cisco Emergency Responder Administration Guide.

Emergency Responder tracks registered endpoints in Unified CM and provides Emergency Call Treatment for reaching correct PSAP and with correct Location information.

Emergency Responder interfaces with the following components:

- A Unified CM cluster by the following methods:
    - JTAPI, SNMP V3/V2, and AXL using IPv4 interface, to collect IPv6 information about its configured IP Phones. Emergency Responder and Unified CM are IPv6-aware.
    - JTAPI, to allow for the call processing associated with redirection of the call to the proper PSAP gateway.

- The access switches (through SNMP IPv4 or IPv6) where the phones associated with Unified CM are connected. Unified CM is IPv6-aware to communicate with Emergency Responder using IPv4 transport. The Emergency Responder to Switch interface is IPv6, supporting SNMP V3/V2.

Emergency Responder Limitations for IPv6-Only Endpoints

Some limitations for IPv6-only endpoints include:

- A call coming to Emergency Responder from an analog phone must be connected to an IPv4 gateway.
- Teleworker and off-premises features work only with IPv4.

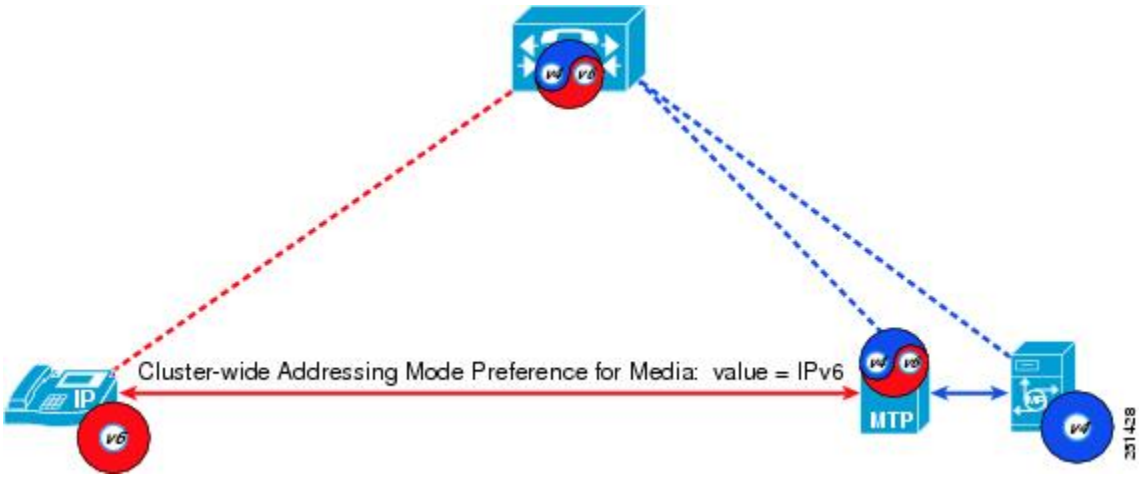
CHAPTER 11

Applications

Applications Overview

All Cisco applications that can terminate voice media support IPv4-only. For these applications, if a call is extended from an IPv6-only device to the IPv4-only application, Cisco Unified Communications Manager (Unified CM) inserts a media termination point (MTP) to convert the voice media from IPv6 to IPv4, as shown in the following figure.

Figure 38: MTP Inserted for Conversion Between IPv6 and IPv4



Voicemail

Cisco Unity Connection supports IPv6 for all interfaces, and Cisco Unity Express supports IPv4 only.

LDAP Directory Integration

Unified CM supports IPv4 with Lightweight Directory Access Protocol (LDAP) for directory sync. IP Phones 8800/7800 exercise UDS (http). IP Phones use CCMPD and CCMCIP services for Personal and Corporate Directory. These services are IPv6 compliant. Back-end interfaces from Unified CM to AD are not validated for IPv6, and should be IPv4-only.

Native Unified CM Applications

Cluster-wide Extension Mobility, IP Phones Services, Cisco Unified Communications Manager Assistant, Cisco Unified Communications Attendant Console, MRA, and Web Dialer support IPv4-only.

Video Conferencing

Cisco Meeting Server and Cisco TelePresence Management Suite (TMS) for video conference support is in a traditional IPv4 stack. IPv4 or IPv6 endpoints are supported in a cluster. Any IPv6-only endpoint can join Cisco Meeting Server video conference service. Additional MTP is required in Unified CM to provide media inter-working IPv4 to IPv6.

Multistream Video

Multistream video that allows an IPv4 or IPv6 endpoint to send multiple resolution video streams is supported. The bridge passes the most appropriate streams to the far-end video units that can be IPv4 or IPv6 stack. The far end video unit receives a full resolution stream of the active speakers.

Device Mobility

Device Mobility is supported only for IPv4-only, IPv6-only, and dual-stack devices.

Unified CM IM and Presence Service

Cisco Unified Communications Manager IM and Presence Service (IM and Presence Service), Cisco IM and Presence Service Client, and Cisco IP Phone Messenger support dual-stack or IPv6-only.

Cisco Mobility Applications

Mobility applications support IPv6-only.

Cisco Jabber Applications

Cisco Jabber applications support IPv6-only for desktop on-premise application deployments only. For MRA, Jabber should be configured for dual-stack (with IPv4 preferred) because of the IPv6-only support limitation in Cisco Expressway. In this dual-stack, ANAT as media is not applicable; it is for applications server interface only based on RFC 6555. Jabber applications in Android and iOS 3G/4G mobile networks will support IPv6 because of a NAT64 solution developed by mobile network service providers. In an off-premise deployment, we recommend Jabber dual-stack due to users roaming in various service provider networks.

Cisco Unified Survivable Remote Site Telephony

Cisco Unified Survivable Remote Site Telephony (Unified SRST) functions as a dual-stack application server that supports the following devices, which have IP phone line side feature such as: Hold/Resume, Forwarding, 3WC, Transfer, MOH, ATA Fax in IPv4 mode:

- Unified SRST supports the following Cisco IP Phones for voice only services in IPv6-only addressing mode using SIP signaling based on the Routers:
    - Support for Cisco IP Phone 7811, 7821, 7841 and 7861
    - Support for Cisco IP Phones 8811, 8841, 8845, 8851, 8861, and 8865
    - Not supported endpoints: Jabber and SCCP IP Phones

- Unified SRST will not support SCCP signaling and SIP SDP ANAT dual-stack attributes to support media selection that can be IPv4 or IPv6.
  - All the legacy dual-stack phones default to IPv4-only SIP mode in Unified CM.
  - PSTN gateways are configurable to SIP IPv4-only or IPv6-only. This gateway supports incoming, outgoing, and emergency calls to service provider PSTN.
- Media Inter-working: Unified SRST provides IPv4 to IPv6 RTP media inter-working for voice and video endpoints. This media inter-working is required when using IPv6-only endpoints, or when gateways are configured to IPv4-only mode.

For more information, refer to the Cisco Unified SCCP and SIP SRST System Administrator Guide.

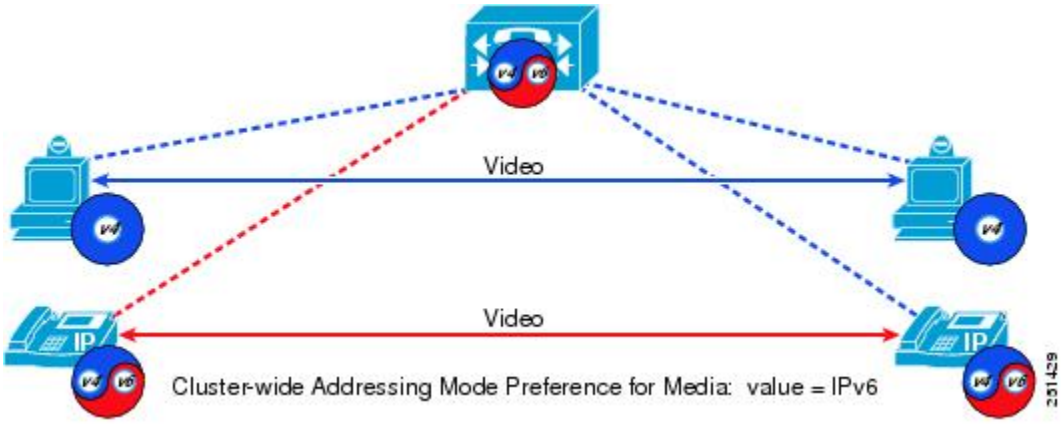
CHAPTER 12

IP Video Telephony

IP Video Telephony Overview

IPv6 transition will follow Cisco Preferred Architecture deployment recommendations of IPv4. Cisco Unified Communications Manager (Unified CM) is the call control server for the Cisco Preferred Architecture for Enterprise Collaboration deployment. Cisco IP Phones, Cisco Jabber clients, and Cisco TelePresence video endpoints use SIP to register directly to Unified CM. The Unified CM cluster's failover mechanism provides endpoint registration redundancy. If a WAN failure occurs and endpoints at remote locations cannot register to Unified CM, they use SRST functionality for local and PSTN calls, but some services such as voicemail and presence might not be available.

Figure 39: Addressing Modes for Voice and Video Media Streams



CHAPTER 13

IP Telephony Migration Options

IP Telephony Migration Options Overview

For first-time installations of IPv6 with Cisco Unified Communications Manager (Unified CM) or an upgrade to a traditional deployment, we recommend the following guidelines:

Deployment Models

For Cisco Unified CM IPv6 is supported for single-site deployments, multi-site deployments with distributed call processing, and multi-site deployments with centralized call processing. Minimum IPv6 deployment requires the following products:

- Dual-stack IPv6 Unified CM server
- MTP for media interworking
- IPv6-only IP Phones

All other components and interfaces can remain in IPv4. Unified CM will insert MTPs to provide media interworking for various use cases.

Campus Network and WAN

Before you enable IPv6 in the Unified CM cluster, make sure that both the campus network and the WAN support both IPv4 and IPv6 traffic. Dual-stack (IPv4 and IPv6) routing is supported for Layer 3 campuses and WANs.

Cluster-Wide IPv6 Configuration Settings

To maximize the amount of IPv6 traffic on your Unified Communications network, use the following settings for the cluster-wide IPv6 Enterprise Parameters:

- **Enable IPv6**: True
- **IP Addressing Mode Preference for Media**: IPv6
- **IP Addressing Mode Preference for Signaling**: IPv6

Common Device Configuration Profiles

Use multiple Common Device Configuration Profiles so that individual phones and trunks or groups of phones and SIP trunks can be selectively configured to support IPv6.

The Common Device Configuration Profile in Unified CM Administration (**Device > Device Settings > Common Device Configuration**) contains the following IPv6 configuration information:

- IP Addressing Mode
- IP Addressing Mode Preference for Signaling
- Allow Stateful Auto-Configuration for Phones

DHCP

Stateless DHCP can be used with Stateless Address Auto-Configuration (SLAAC) for phones.

MTPs

For multi-site distributed call processing deployments, you will most likely need to use Cisco IOS media termination points (MTPs) for conversions between IPv4 and IPv6. These MTPs should be associated with both phone and SIP trunk media resource groups (MRGs).

IPv6 SIP Trunks

IPv6 SIP trunks should be configured as Delayed Offer without Alternative Network Address Types (ANAT) enabled. However, SIP trunks configured for Early Offer (MTP Required) using an MTP resource for every call are do not support video calls or encrypted calls.

IP Phones

IPv6-only phone support is a major feature as it helps to mitigate IPv4 exhaustion. We recommend that you configure all IP Phones as IPv6-only where applicable. All other IP Phones must be configured as IPv4.

ISDN PSTN Gateway and SRST

To transition to an IPv6-only network and to reduce the MTP requirements, configure ISDN PSTN gateways and SRST with Delay Offered SIP IPv6-only trunk.

CHAPTER 14

Security

Security Overview

When comparing IPv4 and IPv6 in terms of how secure each protocol is, IPv6 has some advantages and some disadvantages, but overall it is no more or less secure than IPv4. One inherent benefit of IPv6 is the enormous size of IPv6 subnets and networks, which offer improvements in protection against automated scanning and worm propagation. Typical security drawbacks are the addressing complexity of IPv6 and the likelihood that network administrators will not be familiar with the IPv6 protocol and IPv6 security tools.

In general, most of the legacy issues with IPv4 security remain in IPv6. For example, Address Resolution Protocol (ARP) security issues in IPv4 are replaced with neighbor discovery (ND) security issues in IPv6.

IPv6 security settings have the same functionality as IPv4, such as Firewall, sRTP and TLS.

Privacy and Encryption for IPv6 Voice Signaling and Media

The Internet Engineering Task Force (IETF) and RFCs 4301-4303 mandate authentication and encryption for IPv6 using IP Security (IPsec). However, to avoid interworking issues with legacy IPv4 Unified Communications endpoints, Cisco Unified Communications Manager (Unified CM) IPv4 and IPv6 deployments continue to use Transport Layer Security (TLS) and Secure Real-Time Transport Protocol (SRTP) for authentication and encryption between IP phones and between IP phones and SIP gateways and trunks.

IPsec can also be used for IPv4-based H.323 and Media Gateway Control Protocol (MGCP) gateway connections.

Cisco Unified CM provides the following secure transport protocols:

- Transport Layer Security (TLS)

  TSL provides secure and reliable data transfer between two systems or devices by using secure ports and certificate exchange. TLS secures and controls connections between Unified CM-controlled systems,

devices, and processes to prevent access to the voice domain. Unified CM uses TLS to secure Skinny Client Control Protocol (SCCP) calls to phones that are running SCCP, and to secure SIP calls to phones or trunks that are running SIP.

- IP Security (IPsec)

  IPsec provides secure and reliable data transfer between Unified CM and gateways. IPv4-based IPsec implements signaling authentication and encryption to Cisco IOS MGCP and H.323 gateways.

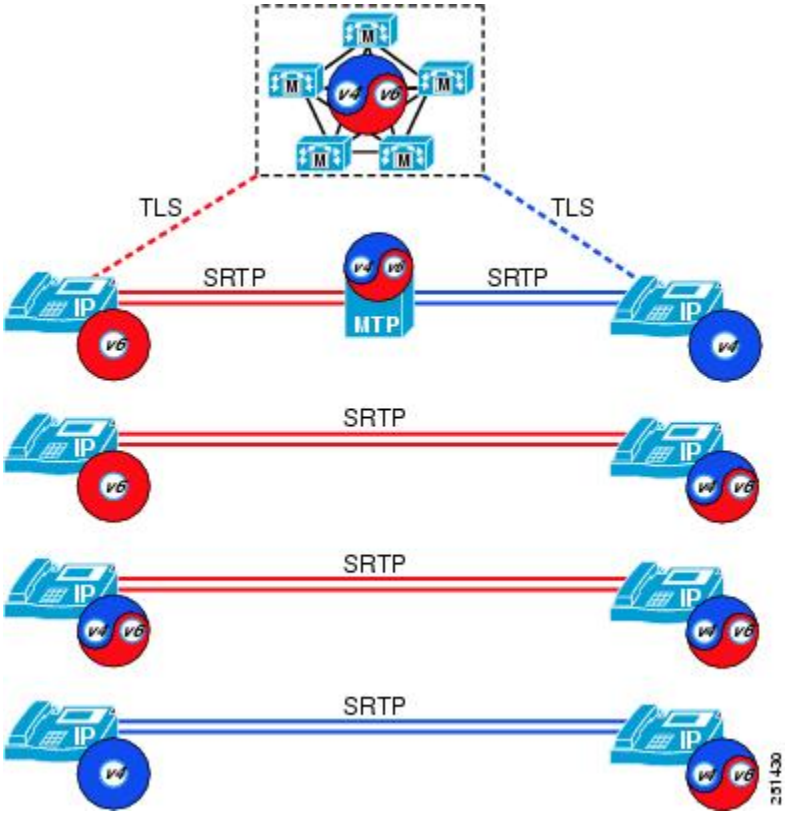
You can add Secure Real-Time Transport Protocol (SRTP) to TLS and IPsec transport services for the next level of security on devices that support SRTP. SRTP authenticates and encrypts the media stream to ensure that voice conversations originating or terminating on Cisco Unified IP Phones and either TDM or analog voice gateway ports, are protected from eavesdroppers who might have gained access to the voice domain. SRTP adds protection against replay attacks.

For more information on Unified CM security, refer to the *Cisco Unified Communications Manager Security Guide*, available at: Link.

Encrypted Media and MTPs Between IPv4 and IPv6

Unified CM supports encrypted calls between dual-stack (IPv4 and IPv6) and single-stack (IPv4 or IPv6) devices. If an IP addressing version mismatch exists between the called and calling device, Unified CM dynamically inserts an MTP to convert the IP header of the encrypted voice stream. This dynamically inserted MTP uses its pass-through codec for the encrypted media stream and changes only the IP headers from IPv4 to IPv6 and conversely.

Figure 40: Addressing Mode Resolution by Unified CM



CAPF and CTL

Certificate Authority Proxy Function (CAPF) supports both IPv4 and IPv6 addressing and uses TCP/IP to communicate with phones and to perform its standard security certificate functions. In an IPv6-enabled Unified CM cluster, CAPF has the following capabilities:

- Issuing and upgrading certificates to IPv4-only IP phones
- Issuing and upgrading certificates to IPv6-only IP phones
- Issuing and upgrading certificates to dual-stack (IPv4 and IPv6) IP phones

No new IPv6 functionality is needed for Certificate Trust List (CTL).

IPv6 Collaboration Traffic and Firewalls

The Cisco Adaptive Security Appliance (ASA) supports SCCP or SIP for IPv6, therefore can be used to open pinholes dynamically for IPv6 voice traffic. This product supports basic firewall and traffic filtering function for IPv6 traffic.

If you want to implement basic firewall capability for IPv6, refer to the following documents:

- Cisco ASA 5500-X Series Firewalls Configuration Guides

  Reference: Link
- Catalyst 6500 Series Switch Configuration Guides

  Reference: Link

Cisco Unified Border Element

The Cisco IOS-based Cisco Unified Border Element can:

- Terminate a SIP IPv6 call on one leg of a session, and generate a SIP IPv4 call on the other leg.
- Terminate a SIP IPv6 call on one leg of a session, and generate a SIP IPv6 call on the other leg.

This functionality allows for basic interconnection between enterprise IPv6 networks and service provider IPv4 networks.

Basic calls with both media and signaling processing are supported. Basic Supplementary Services over IPv6 are supported, but H.323 IPv6 calls are not supported.

CHAPTER 15

Collaboration Endpoints

IPv6 Capabilities of Collaboration Endpoints

This chapter describes a subset of the Cisco Collaboration endpoints and their IPv6 capabilities. The following table lists the endpoint types and summarizes their IPv6 capabilities. IPv6 support extends to only a subset of the Cisco analog gateways and IP phones. The following sections discuss these devices in more detail.

Table 14: IPv6 Capabilities of Collaboration Endpoints

Endpoint Type	IPv6 Capability
Analog gateways	Does not support Cisco VG Series Gateways for IPv6-only.
Cisco IP Phones	Cisco Unified IP Phones 78xx and 88xx Series support IPv6-only.
Software-based endpoints	Jabber software-based endpoints support IPv6.
Wireless endpoints	No wireless endpoints support IPv6.
Cisco Unified IP Conference Station	Conference stations support IPv6.
Cisco TelePresence video endpoints (CE software)	-
Third-party SIP IP phones	Third-party IPv6-capable phones are not supported.

IPv6 Support on Cisco IP Phones

For a list of endpoints that support IPv6-only, see Supported IPv6 Addressing Modes , on page 26. Any endpoints not listed there should be configured in IPv4-only stack.

When a phone with a single IPv4 address needs to communicate with a phone with a single IPv6 address, an IP addressing version incompatibility exists. To resolve this media addressing incompatibility, Unified CM

dynamically inserts a media termination point (MTP) to convert the media stream from IPv4 to IPv6 (and conversely).

IP Phone Deployment Considerations in an IPv6 Network

The **PC Voice LAN Access** setting on IP phones is typically used to allow the monitoring of IP phone traffic by monitoring and recording applications and by network monitoring software. This setting also allows multicast traffic to be propagated from the voice VLAN to the IP phone's PC port. By default, the PC Voice VLAN Access setting is enabled on IP phones.

In IPv6-enabled networks, this default PC Voice VLAN Access setting allows Router Advertisement (RA) multicast messages to be sent from the voice VLAN to the IP phone's PC port. Ordinarily, the PC should drop any packets that it receives with a voice VLAN tag because it is not configured for the voice VLAN and does not understand 802.1g tagging. However, if the PC does accept packets with a voice VLAN tag and uses an RA from the voice VLAN, it can cause an IPv6 address from the voice VLAN to be assigned to the PC.

If you encounter the issue of incorrect IPv6 address assignment for PC port devices, use either of the following techniques to resolve this issue:

- Set the prefix lifetime of RAs sent by routers in the voice VLAN to a shorter lifetime value than the RAs sent by routers in the data VLAN, and also ensure that routers in the data VLAN have higher priority than those in the voice VLAN. IPv6 devices in the data VLAN using the Address Selection Algorithm (RFC 3484) choose the Network Prefix included in RAs with the longest lifetime, and thus prefer routers in the data VLAN.
- For all IP phones with connected devices that are affected by the above issue, set the IP phone's PC Voice VLAN Access setting to disabled. For large numbers of phones, this configuration change can be bulk-provisioned through the Unified CM Bulk Administration Tool (BAT).

Common Device Configuration Profile

The process for configuring IPv6 on the phones is similar to that for SIP trunks. In Cisco Unified CM Administration, select **Device > Device Settings > Common Device Configuration** > to create and configure a Common Device Configuration Profile and associated it with one or more IP phones. The Common Device Configuration Profile contains the following IPv6 configuration information:

- IP Addressing Mode
- IP Addressing Mode Preference for Signaling
- Allow Auto-Configuration for Phones

IP Addressing Mode

The phone IP Addressing Mode has three settings:

- IPv4 Only

  In this mode, the phone acquires and uses only one IPv4 address for all signaling and media. If the phone has previously acquired an IPv6 address, it releases that address.

- IPv6 Only

  In this mode, the phone acquires and uses only one IPv6 address for all signaling and media. If the phone has previously acquired an IPv4 address, it releases that address.

- IPv4 and IPv6

  In this mode, the phone acquires and uses one IPv4 address and one IPv6 address. It can use the IPv4 and the IPv6 address as required for media. It uses either the IPv4 address or the IPv6 address for call control signaling to Unified CM.

If IPv6 is enabled in the Unified CM cluster, the default phone setting for IP Addressing Mode is IPv4 and IPv6. If the IP phone supports both **IPv4 and IPv6**, it will adopt this setting, but all IPv4-only phones will ignore this setting.

We recommend setting the phone IP Addressing Mode to **IPv4 and IPv6**.

IP Addressing Mode Preference for Signaling

The phone IP Addressing Mode Preference for Signaling has three settings:

- IPv4

  If the phone has an IPv4 address, it uses that IPv4 address for call control signaling to Unified CM.

- IPv6

  If the phone has an IPv6 address, it uses that IPv6 address for call control signaling to Unified CM.

- Use System Default

  In this case, the phone uses the cluster-wide Enterprise Parameter configuration value for its IP Addressing Mode for Signaling.

If IPv6 is enabled in the Unified CM cluster, the default phone setting for IP Addressing Mode for Signaling is **Use System Default**. If the IP phone supports either IPv6 or both IPv4 and IPv6, it will adopt the cluster-wide setting for IP Addressing Mode for Signaling, but all IPv4-only phones will ignore this setting.

Allow Auto-Configuration for Phones

Allow Auto-Configuration for Phones has three settings:

- On

  With this setting, the phone is allowed to use Stateless Auto Address Configuration (SLAAC) to acquire an IPv6 address. Whether or not the phone uses SLAAC depends on the link-local router's O and M bit configuration for router advertisements (RAs), as follows:

  - If the O bit is set in the router's RAs, the phone uses SLAAC to acquire an IPv6 address and uses the DHCP server to acquire other information such as the TFTP server address.
  - If the M bit is set in the router's RAs, the phone does not use SLAAC. Instead, it uses the DHCP server to acquire its IP address and other information.
  - If neither the M bit nor the O bit is set, the phone uses SLAAC to acquire an IP address and does not use DHCP for other information. The phone also requires a TFTP server address to download its configuration file and register to Unified CM. This TFTP server address can be configured manually through the phone’s user interface (UI).

- Off

With this setting, the phone does not use Stateless Auto Address Configuration (SLAAC) to acquire an IPv6 address. In this case, the phone can be configured manually to acquire an IPv6 address and TFTP server address.

- Default

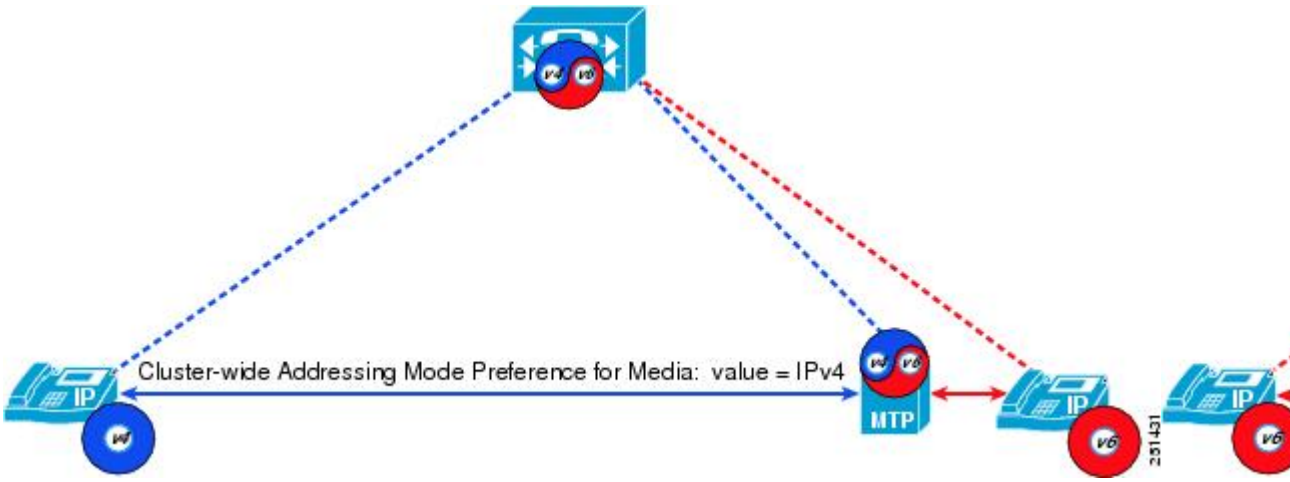
  With this setting, the phone uses the cluster-wide Enterprise Parameter configuration value for Allow Auto-Configuration for Phones.

Default Common Device Configuration Profile

By default, there are no Common Device Configuration profiles configured, and each device is associated with a <None> Common Device Configuration. If IPv6 is enabled in the Unified CM cluster with this default configuration, IPv6-capable devices adopt the following settings:

- IP Addressing Mode = IPv4 or IPv6
- IP Addressing Mode Preference for Signaling = Use System Default
- Allow Auto-Configuration for Phones = Default

Figure 41: Effect of IP Addressing Mode for Media on MTP Usage for IP Phones



Other IP Phone Functions

The following functions also affect the use of IPv6 on Cisco IP Phones.

TFTP

As previously described, TFTP servers support both IPv4 and IPv6 addresses. IPv4-only phones use their received IPv4 TFTP address to reach the TFTP server, and IPv6-only phones use their received IPv6 TFTP address to reach the TFTP server. Dual-stack phones can use either their IPv4 or IPv6 TFTP address to reach the TFTP server.

The following IPv6 information is sent to the phone in its configuration file from the TFTP server:

- Unified CM IPv4 and IPv6 addresses
- The setting for IP Addressing Mode
- The setting for IP Addressing Mode Preference for Signaling
- The setting for Allow Stateless Auto-Configuration for Phones

Unified CM Registration for Dual-Stack IP Phones

Dual-stack phones attempt to connect to Unified CM by using the preferred (IPv4 or IPv6) address specified in the IP Addressing Mode Preference for Signaling parameter found in the TFTP configuration file. If the first attempt at connection fails due to a TCP timeout, the phone then attempts to connect to Unified CM using its other address.

IP Phone HTTP Server Functionality

The HTTP server function of all IP phones uses IPv4-only or IPv6-only. The HTTP server provides several functions, including:

- Phone User Interface access to:
  - Directory search functions
  - Call history
  - All IP Phone Services
  - Extension Mobility
  - IPv4-only supported Quality Report Tool (QRT)
- Web access to the phone through the Unified CM phone administration graphical user interface

CDP and LLDP

IP phones support the sending and receiving of IPv4 and IPv6 addresses in data link layer Cisco Discovery Protocol (CDP) and Link Layer Discovery Protocol (LLDP) messages. Currently, Cisco applications do not use any of the IPv6 addresses sent in LLDP messages. CDP is applicable to Emergency Responder and discovery of a desk phone device.

IPv6-Only Supported and Not Supported Features

IPv6-Only Supported Features

The following are the IPv6-Only supported features:

- Mobility - MVA/IVR
- CAPFv3
- TVS and TFTP
- Extension Mobility
- EMCC

- SRST
- SIP GW
- SIP Route Pattern
- Barge
- Intercom
- Phone NTP Reference
- Device Mobility
- Self Care Portal
- Self Provisioning using with and without IVR
- Phone services - Corporate Directory
- Missed Calls
- Placed Calls
- Intercom Calls
- Personal Directory
- Voicemail
- Received Calls
- Web Dialer
- Quality Report Tool (QRT)
- SNMPv6 MIB
- Phone Config download
- CTI
- IPVMS (Unicast MoH, Software MTP, ANN, Conference Bridge, IVR)
- CDR

IPv6-Only Not Supported Features

The following are the IPv6-Only not supported features:

- Best Effort Early Offer over SIP Trunk
- External MoH, Multicast MoH , Video On Hold, Voicemail, Recording (BIB Recording, GW Recording), Enhanced E911
- SSH, Cluster Manager, LDAP, Syslog Server/Alarm Library, Serviceability Applications - IPMA, RTMT
- DRS, IPSec
- Media Resources (IOS Xcoder/ hardware MTP/TRP)/Conference Bridge), Video conference resource (MCU/Conductor), FAX/ATA

- MRA (Jabber in IPv6 and Exp-E external interface in v6 and enterprise network in v4 is supported)
- Silent Monitoring, Whisper Coaching, Agent Greeting, VG as analog line interface using SCCP (pending test)

CHAPTER 16

Appendix

Product Configuration Resources for IPv6

To configure products for IPv6, refer to the product configuration guides.

Table 15: Endpoints

Endpoint (SIP)	IPv6 Configuration Resources
Cisco IP Phone 7800 Series	Administration Guides
Cisco IP Phone 8800 Series	Administration Guides
Cisco Jabber	On-Premises Deployment Guide
Cisco DX70 and DX80 (CE endpoints)	Support
Cisco TelePresence MX Series (CE endpoints)	Support
Cisco TelePresence SX Series (CE endpoints)	Support
Cisco TelePresence EX Series (CE endpoints)	Support

Table 16: Communication Gateways

Gateway	IPv6 Configuration Resources
Cisco Integrated Services Routers (ISR)	Configuration Guides
Cisco 2900 and 3900 Series Integrated Services Routers (ISR) (For CUBE)	CUBE Configuration Guide

Table 17: Applications and Interfaces

Application or Interface	IPv6 Configuration Resources
Cisco Unified Communications Manager (Unified CM)	System Configuration Guide
Unified CM IM and Presence Service (IM and Presence Service)	Configuration and Administration Guide
Cisco Unified Survivable Remote Site Telephony (Unified SRST)	Administration Guides
Cisco Unified Contact Center Enterprise (Unified CCE)	Solution Design Guide
Cisco Unified Contact Center Express (Unified CCX)	Solution Design Guide
Cisco Emergency Responder	Administration Guide
Cisco Unity Connection	System Administration Guide Integration Guide
Cisco Meeting Server	Support
Cisco TelePresence Management Suite	Support
Cisco Prime Collaboration	Support

Document Direction

We recommend deploying IPv6-only device configuration where applicable, typically endpoints and PSTN gateways. All applications servers need IPv4 address and IPv6 address (dual-stack) to support all traditional IPv4 devices until all devices are IPv6-only capable including third-party products and applications.

To reduce dependency on IPv4 addresses, you can upgrade the following on-premise deployment modes to IPv6: single-site call processing deployments; multi-site distributed call-processing deployments; and multi-site deployments with centralized call processing. We recommend that you upgrade and configure:

- IPv6-only stack SIP phones, SIP gateways, and SIP trunks.
- IPv4-only stack and IPv6-only stack (dual-stack) Unified CM clusters and other application servers.
- All third-party products and applications in IPv4-only.

Deprecation of ANAT SIP SDP Attributes

Department of Defense (DoD) certification requirements drove the development of Alternative Network Address Types (ANAT) RFC 4091 and 4092. ANAT allows a set of network addresses to establish a media stream is useful in environments with both IPv4-only hosts and IPv6-only hosts. For this functionality, all the devices need IPv4 address and IPv6 address assignment. ANAT was also implemented in Cisco Unified

Communication System Release 7.1.2. However, since all device need both IPv4 and IPv6 addresses, ANAT does not help to reduce dependency on the now depleted number of IPv4 addresses. Based on large enterprise customer feedback, and their need for new IP addresses, ANAT SIP SDP attributes have since been deprecated by Internet Engineering Task Force (IETF).

We recommend deploying IPv6-only device configuration where applicable (typically endpoints). All servers, third-party products, and applications need IPv4 addresses and IPv6 addresses to support all traditional IPv4 devices until all devices are IPv6-only capable.