CHAPTER 4

Unified Communications Deployment Models for IPv6

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This chapter describes the deployment models you can use with IPv6 in Cisco Unified Communications networks. Cisco Unified Communications Manager (Unified CM) 8.0 supports the following deployment models:

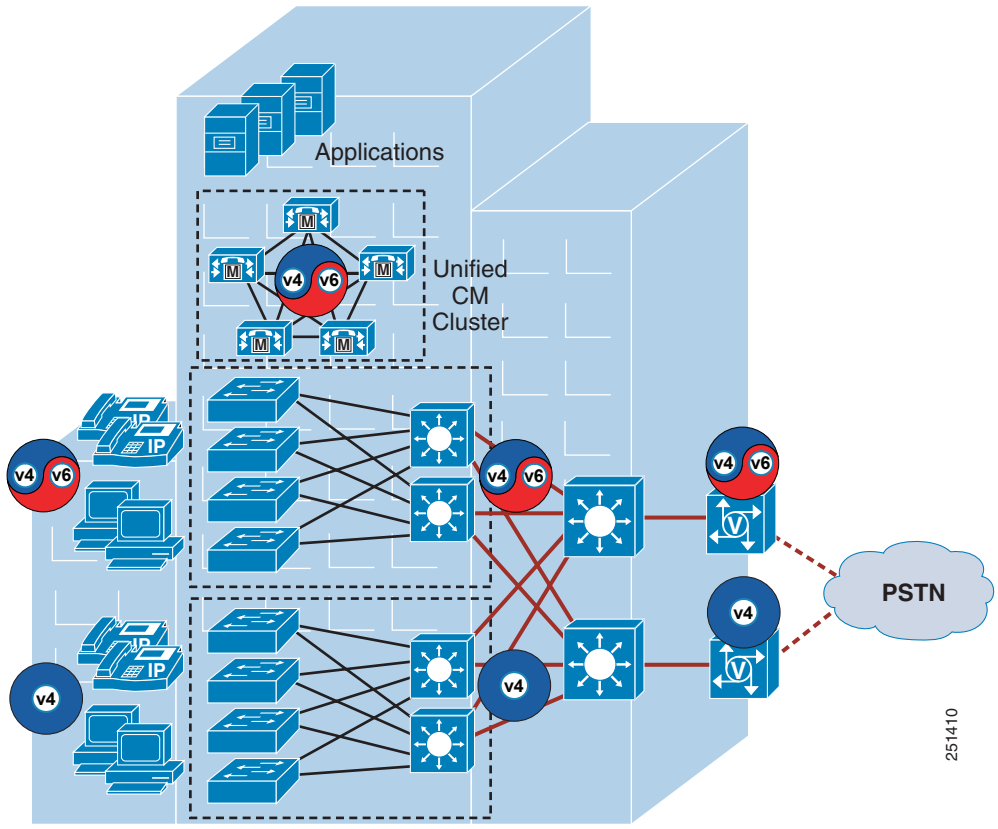
- Single-site deployments
- Multi-site WAN deployments with distributed call processing
- Multi-site deployments with centralized call processing and Survivable Remote Site Telephony (SRST)

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

Single-Site Deployments

The single-site model for Cisco Unified Communications consists of a call processing agent cluster located at a single site, or campus, with no telephony services provided over an IP WAN. An enterprise would typically deploy the single-site model over a LAN or metropolitan area network (MAN), which carries the voice traffic within the site. (See Figure 4-1.) In this model, calls beyond the LAN or MAN use the public switched telephone network (PSTN).

Figure 4-1 Single-Site IPv6 Deployment



The characteristics and benefits of the IPv6 single-site model are the same as those for IPv4 single-site deployments, as described in the *Cisco Unified Communications Solution Reference Network Design (SRND)*, available at http://www.cisco.com/go/ucsrnd. However, the IPv6 single-site model includes the additional IPv6 and dual-stack product capabilities and features discussed throughout this document.

Best Practices for IPv6 Single-Site Deployments

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

- IPv4 only
- IPv4 and IPv6 (Recommended)
- IPv6 only (Not recommended for production environments)

If IPv6 phones are configured as Dual Stack (IPv4 and IPv6), they should also be configured as follows:

- To use IPv6 for signaling to Unified CM
- To prefer IPv6 over IPv4 for media

One or more PSTN gateways can be deployed in a single-site deployment. If only one gateway is deployed, a Unified CM SIP trunk and Cisco IOS SIP gateway should be used. Both the Unified CM SIP trunk and Cisco IOS SIP gateway should be configured as follows:

- Dual stack (IPv4 and IPv6)
- With ANAT enabled
- To use IPv6 for signaling
- To prefer IPv6 over IPv4 for media

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

- SIP Early Offer (**MTP required** checked and used for every call.)
- SIP Delayed Offer (**MTP Required** unchecked, although MTPs may be inserted dynamically for some calls for conversions between IPv4 and IPv6 addresses.)

If a single dual-stack gateway is used and the cluster-wide preference for media is set to IPv6, an MTP will be used for all calls to IPv4-only devices to convert from IPv4 to IPv6. 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 a dual-stack SIP gateway using SIP Delayed Offer as described above, and the other as a standard IPv4-only gateway. Calling search spaces and partitions can then be used to direct PSTN calls from IPv4-only and dual-stack devices to their respective gateways.

For specific device configuration options and preferences, refer to the chapters on Trunks, page 7-1, and Unified Communications Endpoints, page 15-1.

The Campus LAN

If the campus LAN uses Layer 2 switching only, Multicast Listener Discovery (MLD) should be enabled, in the campus switches if it is supported. Enabling MLD is not mandatory, but it is preferred because it reduces unwanted multicast traffic in the LAN.

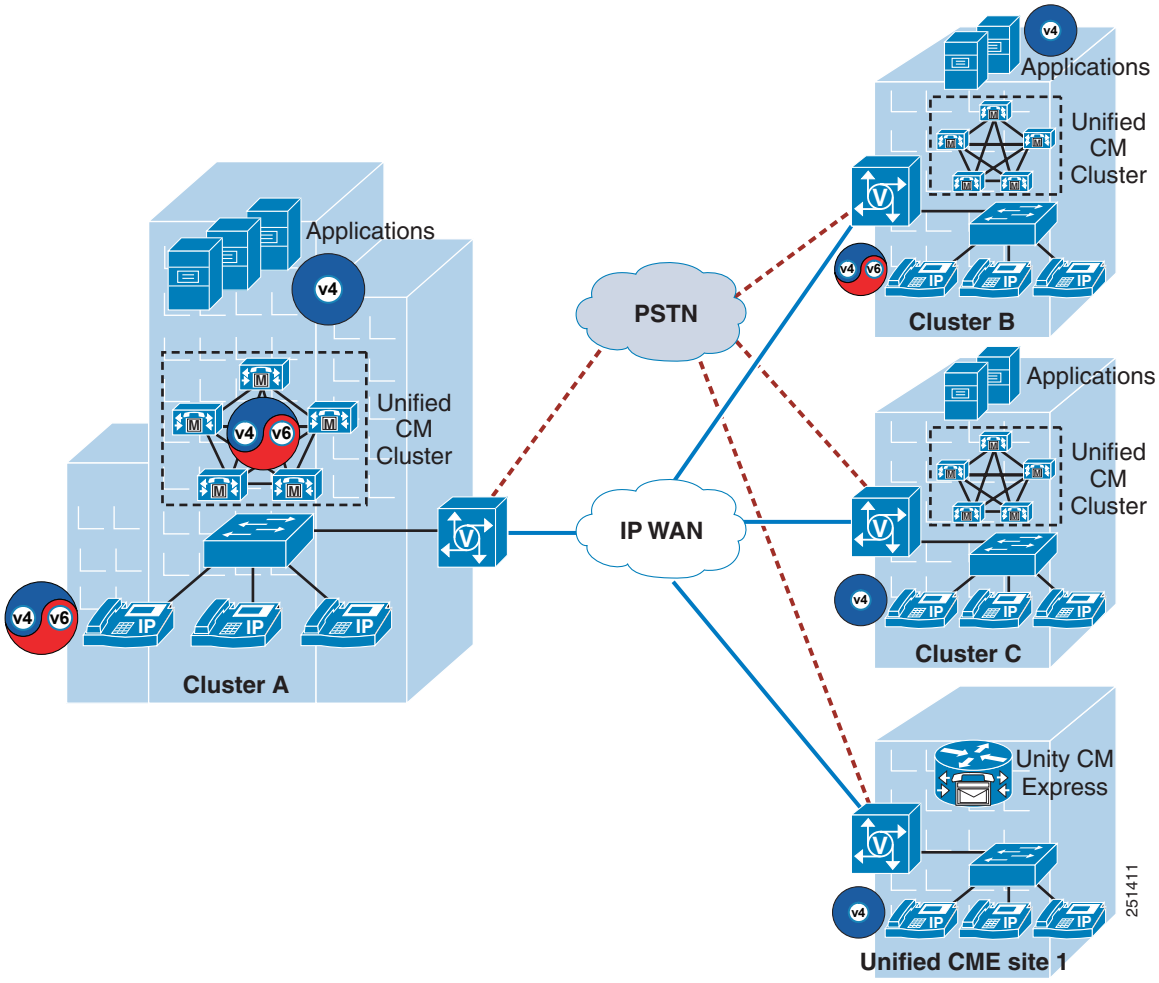
If the campus LAN also includes Layer 3 routing devices, these devices should be configured to support dual-stack (IPv4 and IPv6) routing.

Note If a single PSTN gateway (as described above) 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 dual-stack 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. (See Figure 4-2.)

Figure 4-2 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)
  - A standard (IPv4 only) Cisco Unified Communications Manager (Unified CM)
  - A 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 dual-stack (IPv4 and IPv6))

For dual-stack (IPv4 and IPv6) sites, IPv6 devices should be configured as dual stack, with a preference of IPv6 for signaling and media. This configuration maximizes the amount of IPv6 traffic and minimizes 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 Unified Communications Solution Reference Network Design (SRND)*, available at http://www.cisco.com/go/ucsrnd. However, the IPv6 multi-site model includes the additional IPv6 and dual-stack product capabilities and features discussed in this document.

Best Practices for IPv6 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-enabled SIP intercluster trunks to connect to other IPv6 Unified CM clusters. However, for intercluster trunk connections to IPv4-only Unified CM clusters, IPv6 intercluster trunks should not be used.

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

The Unified CM SIP intercluster trunks should be configured as follows:

- Dual stack (IPv4 and IPv6)
- With ANAT enabled
- To use IPv6 for signaling
- To prefer IPv6 over IPv4 for media (by configuring the cluster-wide addressing mode preference for media to IPv6)

The Unified CM SIP intercluster trunk can be configured to use either of the following:

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

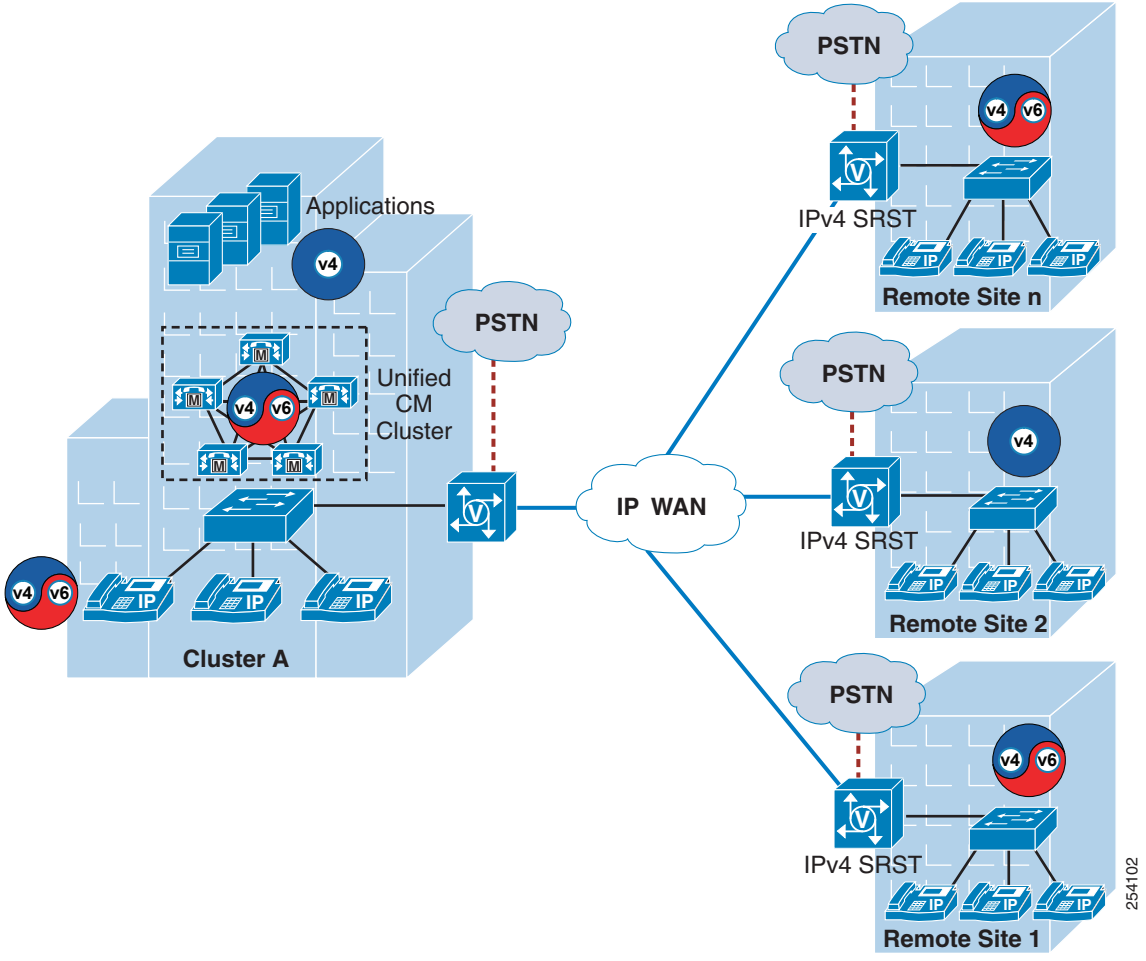
For specific device configuration options and preferences, refer to the chapters on Trunks, page 7-1, and Unified Communications Endpoints, page 15-1.

Multi-Site Deployments with Centralized Call Processing and Survivable Remote Site Telephony (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. Figure 4-3 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 4-3 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-stack or IPv4 only. For dual-stack (IPv4 and IPv6) sites, IPv6 devices should be configured as dual-stack with a preference of IPv6 for signaling and media. This configuration maximizes the amount of IPv6 traffic and minimizes the use of 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 those for IPv4 multi-site centralized call processing deployments, as described in the *Cisco Unified Communications Solution Reference Network Design (SRND)*, available at http://www.cisco.com/go/ucsrnd. However, the IPv6 multi-site centralized call processing deployment model includes the additional IPv6 and dual-stack product capabilities and features discussed in this document.

Best Practices for IPv6 Multi-Site Deployments with Centralized Call Processing and Survivable Remote Site Telephony (SRST)

IPv6 multi-site deployments with centralized call processing can contain sites with a mixture of 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.

IPv6-capable phones can be configured as:

- IPv4 only
- IPv4 and IPv6 (Recommended)
- IPv6 only (Not recommended for production environments)

If IPv6 phones are configured as Dual Stack (IPv4 and IPv6), they should also be configured as follows:

- To use IPv6 for signaling to Unified CM
- To prefer IPv6 over IPv4 for media

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

SRST routers at remote sites support IPv4 only in SRST mode. Dual-stack (IPv4 and IPv6) phones will revert to IPv4-only when in SRST mode and revert back to dual-stack mode when the connection to the Unified CM cluster is restored. IPv6-only phones do not support SRST and are not recommended for deployment at remote sites in production environments.

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 Unified Communications Solution Reference Network Design (SRND)*, available at http://www.cisco.com/go/ucsrnd.

Intra-Cluster Communications

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

Clustering Over the WAN

Clustering over the WAN with dual-stack Unified CM clusters has not been tested by Cisco Systems and is not currently supported.

Call Detail Records (CDR) and Call Management Records (CMR)

Call detail records and call management records, 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.