



CHAPTER 13

Cisco Unity

Last revised on: January 29, 2008

This chapter focuses on the design aspects of integrating Cisco Unity and Cisco Unity Connection with Cisco Unified Communications Manager (Unified CM). This chapter focuses on the following components:

- Cisco Unified CM 6.x
- Cisco Unity 5.x, 4.2 and later 4.x releases
- Cisco Unity Connection 2.x

For information on earlier releases of Cisco Unity, Unity Connection, and Unified CM, refer to the appropriate online product documentation available at

<http://www.cisco.com>

The design topics covered in this chapter apply to voicemail-only, unified messaging, and integrated messaging configurations. Additionally, this chapter discusses design issues for deploying Cisco Unity with Microsoft Exchange (2000, 2003, or 2007) or Lotus Notes Domino message stores and Microsoft Windows (2000 or 2003). This chapter does *not* cover deployments or upgrades from Microsoft NT 4.0 and/or Exchange 5.5. Cisco Unity Connection has no dependencies on an external message store.

For additional design information about Cisco Unity and Unity Connection, including integrations with other non-Cisco messaging systems, refer to the *Cisco Unity Design Guide*, available at

<http://www.cisco.com>

This chapter covers the following Cisco Unity and Unity Connection design topics:

- [Messaging Deployment Models, page 13-2](#)
- [Messaging System Infrastructure Components, page 13-4](#)
- [Integration with Unified CM, page 13-8](#)
- [Messaging and Unified CM Deployment Model Combinations, page 13-13](#)

Cisco Unified CM 6.x also provides functionality for Session Initiation Protocol (SIP) trunks, which support integration directly to Cisco Unity and Unity Connection without the need for a SIP proxy server.

Messaging Deployment Models

This section summarizes the various messaging deployment models for Cisco Unity and Unity Connection. For a complete discussion of the deployment models and design considerations specific to Cisco Unity, Unity Connection, and the various messaging components, refer to the *Cisco Unity Design Guide*, available at <http://www.cisco.com>.

Cisco Unity and Unity Connection support three primary messaging deployment models:

- [Single-Site Messaging, page 13-2](#)
- Multisite WAN deployment with [Centralized Messaging, page 13-2](#)
- Multisite WAN deployment with [Distributed Messaging, page 13-3](#)

Deployments involving both Cisco Unified CM and Cisco Unity or Unity Connection use one call processing model for Unified CM and one messaging model for Cisco Unity. The messaging deployment model is independent from the type of call processing model deployed.

In addition to the three messaging deployment models, Cisco Unity also supports messaging failover. (See [Messaging Failover, page 13-4](#).) Cisco Unity Connection does not currently support failover. Because Cisco Unity Connection uses only a single server without any networking between servers, it supports only the single-site and centralized messaging models.

All messaging deployment models support both voicemail and unified messaging installations. Cisco Unity Connection supports voicemail and integrated messaging. The following definitions apply to these configurations:

- *Voicemail-only* refers to a telephony voicemail integration where there is no access to the voicemail via any messaging client.
- *Integrated messaging* refers to voicemail with telephony access as well as voicemail-only access via a messaging client.
- *Unified messaging* refers to voicemail with telephony access as well as voicemail, email, and fax access via a messaging client.

Single-Site Messaging

In this model, the messaging systems and messaging infrastructure components are all located at the same site, on the same highly available LAN. The site can be either a single site or a campus site interconnected via high-speed metropolitan area networks (MANs). All clients of the messaging system are also located at the single (or campus) site. The key distinguishing feature of this model is that there are no remote clients.

Centralized Messaging

In this model, similar to the single-site model, all the messaging system and messaging infrastructure components are located at the same site. The site can be one physical site or a campus site interconnected via high-speed MANs. However, unlike the single-site model, centralized messaging clients can be located both locally and remotely.

Because messaging clients may be either local or remote from the messaging system, special design considerations apply to the following graphical user interface (GUI) clients: ViewMail for Outlook (VMO) with Microsoft Exchange and Domino Unified Communications Services (DUC) with Lotus Domino, and the use of the telephone record and playback (TRaP) and message streaming features.

Remote clients should not use TRaP and should be configured to download messages before playback. Because different features and operations for local and remote clients can cause user confusion, TRaP should be disabled on the voice ports and GUI clients should be configured to download messages and not use TRaP, regardless of whether the client is local or remote. Cisco Unity inbox accessed via Cisco Unity Personal Assistant (CPCA) should be allowed only for local clients. The Cisco Unity telephone user interface (TUI) operates the same way for both local and remote clients.

Distributed Messaging

A distributed messaging model consists of multiple single-site messaging systems distributed with a common messaging backbone. There can be multiple locations, each with its own messaging system and messaging infrastructure components. All client access is local to each messaging system, and the messaging systems share a messaging backbone that spans all locations. Message delivery from the distributed messaging systems occurs via the messaging backbone through a full mesh or hub-and-spoke type of message routing infrastructure. No messaging infrastructure components should be separated by a WAN from the messaging system they service. Distributed messaging is essentially multiple, single-site messaging models with a common messaging backbone.

The distributed messaging model has the same design criteria as centralized messaging with regard to local and remote GUI clients, TRaP, and message downloads.

Cisco Unity, Cisco Unity Connection, and Cisco Unity Express can all be networked together through Voice Profile for Internet Mail (VPIM) for a complete, virtualized enterprise and branch-office messaging solution.

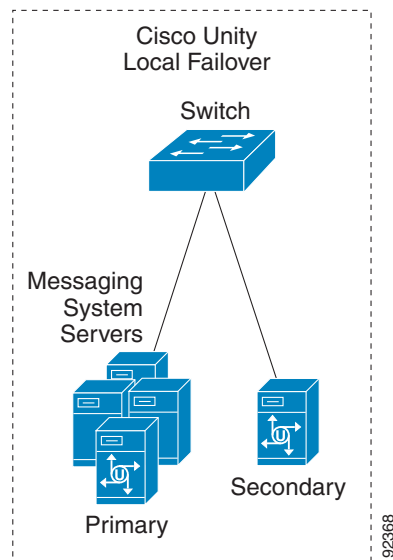
Smaller deployments can be networked directly with each other as a full mesh, or larger distributed voicemail networks can leverage the Cisco Unified Messaging Gateway to network VPIM v2-capable voicemail systems together in a hub-and-spoke or hierarchical structure.

The Cisco Unified Messaging Gateway allows for voicemail systems such as Cisco Unity, Cisco Unity Connection, Cisco Unity Express and other VPIM v2-capable voicemail systems to be fully networked without requiring the overhead of a fully meshed VPIM network. For a complete discussion of VPIM as a distributed messaging solution and for design guidelines on the Cisco Unified Messaging Gateway, refer to the Cisco Unified Messaging Gateway design guide available on <http://www.cisco.com>.

Messaging Failover

All three messaging deployment models support messaging failover. You can implement local messaging failover as illustrated in Figure 13-1. With local failover, both the primary and secondary Cisco Unity servers are located at the same site on the same highly available LAN. This design is only for Cisco Unity; Cisco Unity Connection does not currently support failover.

Figure 13-1 Local Failover of Cisco Unity Messaging



Messaging System Infrastructure Components

Cisco Unity interacts with various network resources, include Dynamic Domain Name Server (DDNS), directory servers, and message stores. (See Figure 13-2.) Cisco Unity supports both Microsoft Exchange and IBM Lotus Notes as message stores. Each of these message store types has different infrastructure components (refer to the appropriate deployment type in the *Cisco Unity Design Guide*, available at <http://www.cisco.com>).

Instead of viewing Cisco Unity as a single monolithic device, it is better to think of it as a system of interdependent components. Each messaging system infrastructure component in a Cisco Unity messaging system is required for normal operation, and it is critical that all of these components be on the same highly available LAN. (In most cases they will be physically co-located.) Any WAN links between these components can introduce delays that will impact the operation of Cisco Unity. These delays will manifest themselves as long delays and periods of silence during TUI operation. For more information, refer to the *Cisco Unity Design Guide*, available from <http://www.cisco.com>.

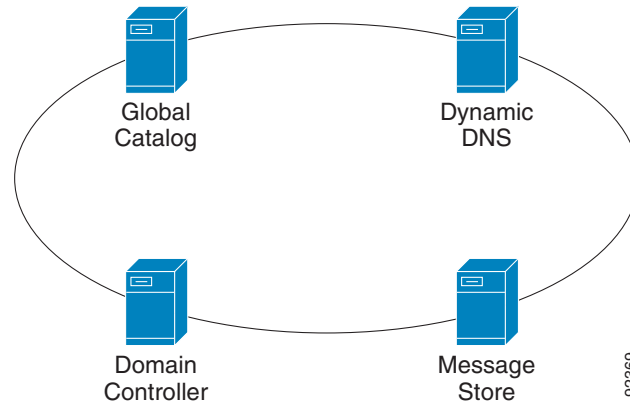
Figure 13-2 Cisco Unity Messaging System Infrastructure Components (Exchange Specific)

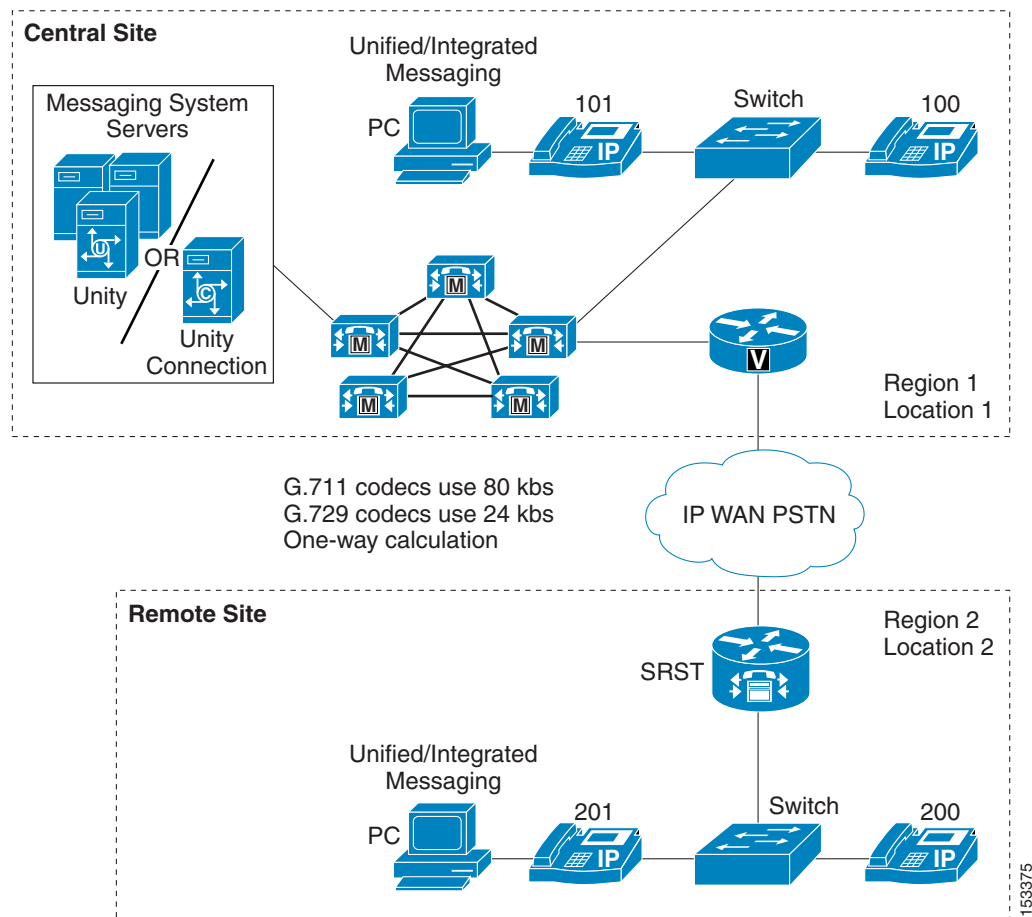
Figure 13-2 is a logical representation of the message system infrastructure components. Some of these components can be located on the same server. In the case of Domino Lotus Notes, the message store and directory (Names.nsf) are located on the same server. Microsoft Windows, the global catalog server, and domain controller may also be on the same server. You can also use multiple instances of each component, as in message store clustering, which Cisco Unity supports for Lotus Domino and Microsoft Exchange 2000, 2003, or 2007. All messaging system infrastructure components must be located on the same highly available LAN as the Cisco Unity server(s) that they service.

When deploying Cisco Unity Connection, it is possible to locate the automatic speech recognition (ASR) service on a separate server. In this deployment environment, the Cisco Unity Connection server and the ASR server should reside in the same location. Cisco Unity Connection does not have the same infrastructure requirements or dependencies as Cisco Unity.

Managing Bandwidth

Unified CM provides a variety of features for managing bandwidth. Through the use of regions, locations, and even gatekeepers, Unified CM can ensure that the number of voice calls going over a WAN link does not oversubscribe the existing bandwidth and cause poor voice quality. Cisco Unity and Unity Connection rely on Unified CM to manage bandwidth and to route calls. If you deploy Cisco Unity or Unity Connection in an environment where calls or voice ports might cross WAN links, these calls will be transparent to gatekeeper-based call admission control. This situation occurs any time the Cisco Unity or Unity Connection server is servicing either distributed clients (distributed messaging (Unity only) or distributed call processing) or when Unified CM is remotely located (distributed messaging (Unity only) or centralized call processing). Unified CM provides regions and locations for call admission control.

Figure 13-3 uses a small centralized messaging and centralized call processing site to illustrate how regions and locations work together to manage available bandwidth. For a more detailed discussion of regions and locations, refer to the chapter on [Call Admission Control](#), page 9-1.

Figure 13-3 **Locations and Regions**

In **Figure 13-3**, regions 1 and 2 are configured to use G.711 for intra-region calls and G.729 for inter-region calls. Locations 1 and 2 are both set to 24 kbps. Location bandwidth is budgeted only in the case of inter-location calls.

An intra-region (G.711) call would not be budgeted against the available bandwidth for the location. For example, when extension 100 calls extension 101, this call is not budgeted against the 24 kbps of available bandwidth for Location 1. However, an inter-region call using G.729 is budgeted against both bandwidth allocations of 24 kbps for Location 1 and Location 2. For example, when extension 100 calls extension 200, this call would be connected but any additional (simultaneous) inter-region calls would receive reorder (busy) tone.

Impact of Non-Delivery of RDNIS on Voicemail Calls Routed via AAR

Automated alternate routing (AAR), a Unified CM feature, can route calls over the PSTN when the WAN is oversubscribed. However, when calls are rerouted over the PSTN, Redirected Dialed Number Information Service (RDNIS) can be affected. Incorrect RDNIS information can impact voicemail calls that are rerouted over the PSTN by AAR when Cisco Unity or Unity Connection is remote from its messaging clients. If the RDNIS information is not correct, the call will not reach the voicemail box of the dialed user but will instead receive the auto-attendant prompt, and the caller might be asked to reenter the extension number of the party they wish to reach. This behavior is primarily an issue when the telephone carrier is unable to ensure RDNIS across the network. There are numerous reasons why the

carrier might not be able to ensure that RDNIS is properly sent. Check with your carrier to determine if they provide guaranteed RDNIS deliver end-to-end for your circuits. The alternative to using AAR for oversubscribed WANs is simply to let callers hear reorder tone in an oversubscribed condition.

Cisco Unity and Unity Connection Native Transcoding Operation

By default, the Cisco Unity or Unity Connection server performs transcoding automatically. Cisco Unity native transcoding does not use external hardware transcoders but instead uses the server's main CPU. For SCCP integrations only, the native transcoding must be disabled (turned off) on the Cisco Unity server via a registry setting in order for Unified CM to assign hardware transcoders to voice port calls. The registry setting is called *Audio - Enable G.729a codec support*, and the tool for setting it is the Advanced Settings Tool available at <http://www.CiscoUnityTools.com>. (To disable native transcoding on Cisco Unity Connection, the configuration of the messaging codec is done differently, as described at the end of this section.)

By default, there is no codec registry key, and therefore native transcoding is enabled (on). The Advanced Settings Tool adds a new registry key that enables you to select either one of the two available codecs. Cisco Unity will then "advertise" to Unified CM that it supports only one codec. If a call terminating or originating on the voice ports is using a different codec than the type configured for the Cisco Unity server, Unified CM will assign an external transcoding resource for the call. For detailed information on how to configure transcoding resources on Unified CM, refer to the chapter on [Media Resources](#), page 6-1.



Note

Currently, the Cisco Unity TAPI Service Provider (TSP) for Unified CM supports only G.729 and G.711 mu-law (a-law is not supported) for the Skinny Client Control Protocol (SCCP) voice ports. A software Media Termination Point (MTP), such as Unified CM itself, is required for mu-law to a-law conversion.

When the Advanced Settings Tool is used to enable only one codec, the Cisco Unity server system prompts must be the same as the codec used. By default, the system prompts are G.711. If the codec is set to G.711, the system prompts will work correctly. However, if G.729 is selected, you will have to change the system prompts. Refer to the *Cisco Unity System Administration Guide* on <http://www.cisco.com> for details on how to change the system prompts. If the system prompts are not the same as the codec selected by the registry, then the end users will hear unintelligible system prompts.

For information on disabling native transcoding on Unity when integrated via SIP, refer to the Cisco Unity product documentation online at <http://www.cisco.com>.

To change how Cisco Unity Connection advertises which codecs it supports, the configuration is done differently than for Cisco Unity. Instead of using the Cisco Unity Tools Depot, the configuration changes are made from the Cisco Unity Connection Administration page. Under the Telephony Integrations heading, select the Phone Systems link. On the Phone Systems page, you can change the configuration to advertise either G.711 only, G.729 only, or both, but "prefer" either G.711 or G.729. In the preferred setting, Cisco Unity Connection will advertise that it supports both protocols (native transcoding) but will prefer to use the one specified over the other. With native transcoding disabled, Unified CM will provide a transcoder resource when needed instead of using the CPU on the Unity or Unity Connection server.

Integration with Unified CM

Cisco Unified CM can integrate with Cisco Unity and Unity Connection via both SCCP or SIP. This section discusses some specifics of that integration regarding phones, SIP trunks, and voice ports.

Telephony Integrations

Cisco Unity supports multiple separate *telephony integrations*, which allow users to be associated with a particular telephony integration. The message waiting indication (MWI) ports are also associated with a particular integration; thus, MWI requests are made through the ports that are associated with that specific integration. This behavior is a marked difference from Cisco Unity 4.0.x and earlier versions, where MWI requests were broadcast out all ports of the system due to the lack of a user-to-integration association.

In Cisco Unity Connection this functionality is similar. Users are associated to a *phone system*, which contains one or more port groups. The port groups are associated with MWI ports; thus, the MWI requests are made through the ports associated to that specific port group.

Cisco Unity telephony integrations are configured with the Cisco Unity Telephony Integration Manager (UTIM), while Unity Connection phone systems and port groups are configured with the System Administrator.

Cisco Unity and Unity Connection now support an unlimited number of telephony integrations and are limited only by the number of ports per system. These features function the same way for both SCCP and SIP integrations. For details, refer to the appropriate Cisco Unity or Cisco Unity Connection administration guides available at <http://www.cisco.com>.

In addition to the option of adding multiple clusters by adding additional integrations for each new Unified CM cluster in Cisco Unity, Unified CM supports Annex M.1, Message Tunneling for QSIG, which gives administrators the ability to enable QSIG on intercluster trunks (ICTs) between Unified CM clusters. When QSIG is enabled on ICTs, Cisco Unity needs to integrate with only one Unified CM cluster and designate ports only in this one cluster for turning MWIs on and off, even when supporting multiple clusters. The Annex M.1 feature in Unified CM allows for propagation of the MWI requests across the ICTs to the proper Unified CM cluster and phone within that cluster. All calls originating in other clusters can be forwarded to the Unity server integrated to that one cluster. There is no need to designate MWI ports on the other clusters when Annex M.1 is enabled on the ICT.

For more information on Annex M.1, refer to the protocol descriptions in the *Cisco Unified Communications Manager System Guide*, available at <http://www.cisco.com>.

Configuring a Unified CM SIP Trunk for Voicemail Integration

Cisco Unified CM provides SIP support for line-side devices as well as trunk-side SIP. These features enable Unified CM to integrate directly to Cisco Unity and Unity Connection via SIP without the need for a SIP proxy server.

However, while Unified CM allows direct integration to voicemail via SIP trunks, it does not currently have the same Voicemail Port Wizard that Skinny Client Control Protocol (SCCP) ports do. Some manual configuration is required. The following steps are required to configure a SIP trunk for a basic integration scenario.

Step 1 Create a SIP Profile in Unified CM Administration by selecting **Device > Device Setting > SIP Profile**.

There is nothing specific to voicemail integrations here, but for easy and consistency of administration functions you will want to create a SIP profile specific for your voicemail integration(s) and name it accordingly.

Step 2 Create a SIP Trunk Security Profile in Unified CM Administration by selecting **System > Security Profile > SIP Trunk Security Profile**.

The **Incoming Port #** under the SIP Trunk Security Profile is specific to voicemail, and it is the number of the SIP port number that the Cisco Unity or Unity Connection server uses to connect to the Unified CM cluster. Additionally, check (enable) **Accept Unsolicited Notifications** so that Cisco Unity and Unity Connection can notify Unified CM of Message Waiting Indicator (MWI) events.



Note MWI functions are handled via Unsolicited Notifies. There is no need to configure the MWI DN's that are required for SCCP integrations.

Step 3 Create a SIP Trunk in Unified CM Administration by selecting **Device > Trunk [Add New]**. Under the trunk creation page, assign the previously configured SIP Profile and SIP Trunk Security Profile in the appropriate fields. Additionally, configure the destination address of the Unity or Unity Connection server and check (enable) **Unattended Port**. Also check (enable) **Redirecting Number IE Deliver - Outbound**.



Note One exception to the Unattended Port occurs when multiple Cisco Unity servers are attached to a single Unified CM cluster and live reply or cross-server logon are configured. In this scenario, you would want to leave Unattended Port unchecked (disabled) to allow calls transferred from voicemail ports on one server to be terminated on the other server. This exception currently applies to Unity only.

Step 4 Create a route pattern and specify the voicemail SIP trunk as the destination.

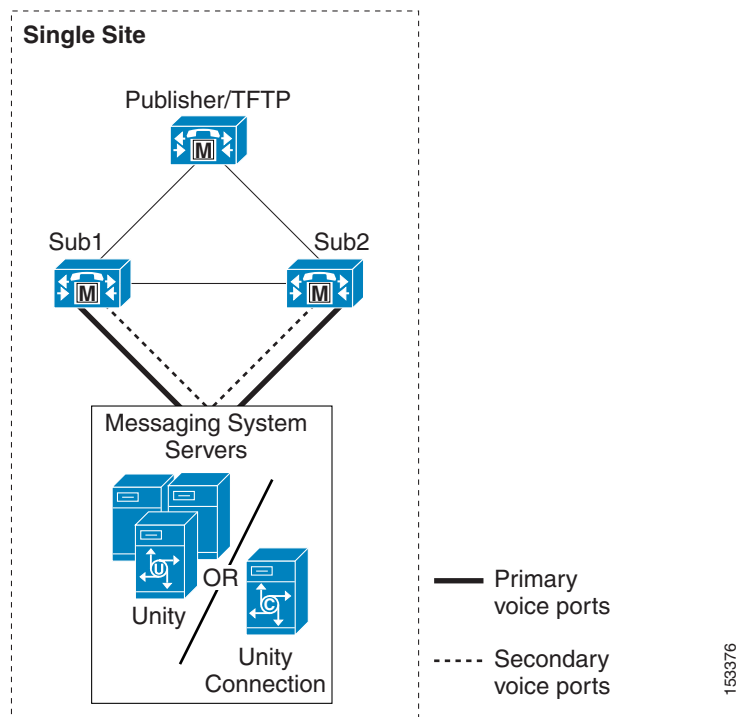
Step 5 Create a Voice Mail Pilot number that is the same number as the route pattern configured in [Step 4](#).

Step 6 Create a Voice Mail Profile using the appropriate Voice Mail Pilot number.

Voice Port Integration with a Unified CM Cluster

When deploying Cisco Unity in a single-site messaging environment, integration with the Unified CM cluster occurs through the SCCP voice ports or SIP ports. Design considerations must include proper deployment of the voice ports among the Unified CM subscribers so that, in the event of a subscriber failure (Unified CM failover), users and outside calls can continue to access voice messaging. (See [Figure 13-4](#).)

Figure 13-4 Cisco Unity Server(s) Integrated with a Unified CM Cluster (No Dedicated Backup Servers)



The Unified CM cluster in [Figure 13-4](#) employs 1:1 server redundancy and 50/50 load balancing. During normal operations, each subscriber server is active and handles up to 50% of the total server call processing load. In the event of a subscriber server failure, the remaining subscriber server takes up the load of the failed server.

This configuration uses two groups of voicemail ports, with each group containing one-half of the total number of licensed voice ports. One group is configured so that its primary server is Sub1 and its secondary (backup) server is Sub2. The second group is configured so that Sub2 is the primary server and Sub1 is the backup.

Make sure that MWI-only ports or any other special ports are equally distributed between the two groups. During the configuration of the voice ports, pay special attention to the naming convention. When configuring the two groups of ports in the Cisco Unity Telephony Integration Manager (UTIM) utility, make sure that the device name prefix is unique for each group and that you use the same device name when configuring the voicemail ports in Unified CM Administration. The device name prefix is unique for each group of ports in this example, with group Sub1 using CiscoUM1 as the device name prefix and Sub2 using CiscoUM2 in this example.

For additional design information on the ratio of inbound to outbound voicemail ports (for MWI, message notification, and TRaP), refer to the *Cisco Unity System Administration Guide* available at <http://www.cisco.com>.

**Note**

The device name prefix is unique for each group of ports and must match the same naming convention for the voicemail ports configured in Unified CM Administration.

In Unified CM Administration, half of the ports in this example are configured to register using the unique device name prefix of CiscoUM1, and the other half are configured to register using the unique device prefix. (See [Table 13-1](#).) When the ports register with Unified CM, half will be registered with subscriber server Sub1, and the other half will be registered with Sub2, as shown in [Table 13-1](#).

Table 13-1 Voicemail Port Configuration in Unified CM Administration

Device Name	Description	Device Pool	SCCP Security Profile	Status	IP Address
CiscoUM1-VI1	Unity1	Default	Standard Profile	Registered with sub1	1.1.2.9
CiscoUM1-VI2	Unity1	Default	Standard Profile	Registered with sub1	1.1.2.9
CiscoUM1-VI3	Unity1	Default	Standard Profile	Registered with sub1	1.1.2.9
CiscoUM1-VI4	Unity1	Default	Standard Profile	Registered with sub1	1.1.2.9
CiscoUM2-VI1	Unity1	Default	Standard Profile	Registered with sub2	1.1.2.9
CiscoUM2-VI2	Unity1	Default	Standard Profile	Registered with sub2	1.1.2.9
CiscoUM2-VI3	Unity1	Default	Standard Profile	Registered with sub2	1.1.2.9
CiscoUM2-VI4	Unity1	Default	Standard Profile	Registered with sub2	1.1.2.9

**Note**

The naming convention used for the voicemail ports in Unified CM Administration must match the device name prefix used in Cisco UTIM, otherwise the ports will fail to register.

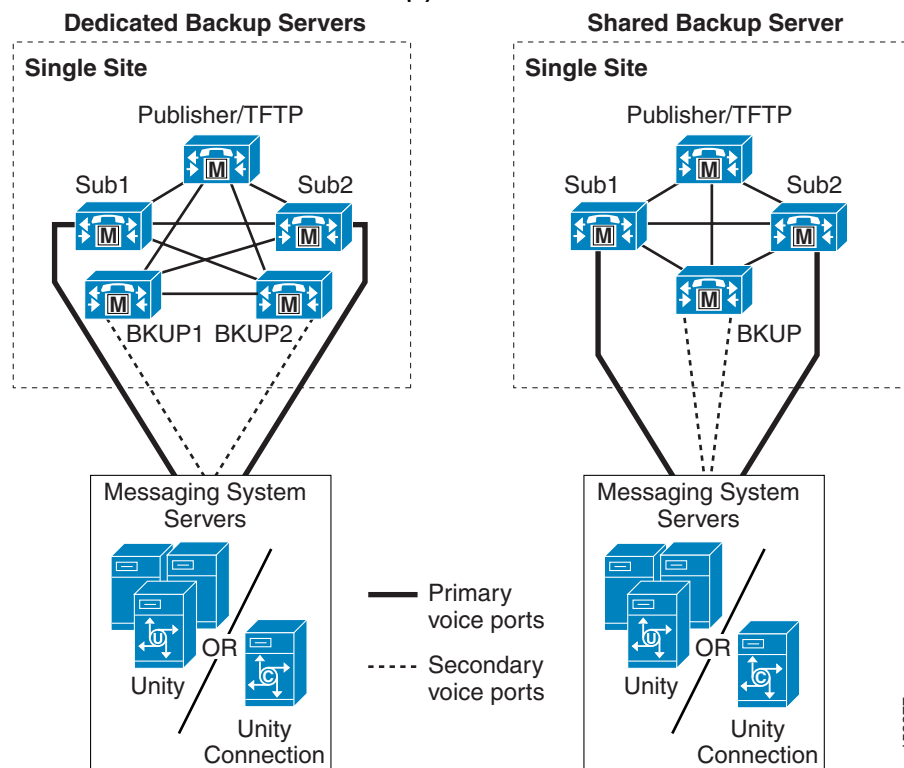
Cisco Unified CM 4.0 introduced numerous changes to the configuration of SCCP voicemail ports with regard to how they hunt and forward calls. For information on how these changes impact the voicemail port operation and configuration, refer to the Cisco Unified CM 4.0 documentation available at

<http://www.cisco.com>

Voice Port Integration with Dedicated Unified CM Backup Servers

This Unified CM cluster configuration allows each subscriber server to operate at a call processing load higher than 50%. Each primary subscriber server has either a dedicated or shared backup server. (See Figure 13-5.) During normal operation the backup server processes no calls, in the event of failure or maintenance of a Subscriber server, the backup server will then take the full load of that server.

Figure 13-5 Cisco Unity Server(s) Integrated with a Single Unified CM Cluster with Backup Subscriber Server(s)



Configuration of the voicemail ports in this case is similar to the 50/50 load-balanced cluster. However instead of configuring the voice ports to use the opposite subscriber server as the secondary server, the individual shared or dedicated backup server is used. In the Unified CM clustered with a shared backup server, both of the secondary ports for the subscribers servers are configured to use the single backup server.

The voice port names (device name prefix) must be unique for each Cisco UTIM group and must be the same as the device names used on the Unified CM server.

To configure the voicemail ports on Cisco Unity, use the UTIM tool. For Cisco Unity Connection, use the Telephony Integration section of the Unity Connection Administration console. For details, refer to the Cisco Unity or Cisco Unity Connection administration guides available at <http://www.cisco.com>.

Messaging and Unified CM Deployment Model Combinations

This section discusses the design considerations for integrating the various messaging deployment models with the Unified CM call processing deployment models. [Table 13-2](#) lists the various combinations of messaging and call processing deployment models supported by Cisco Unity, Unity Connection, and Unified CM.

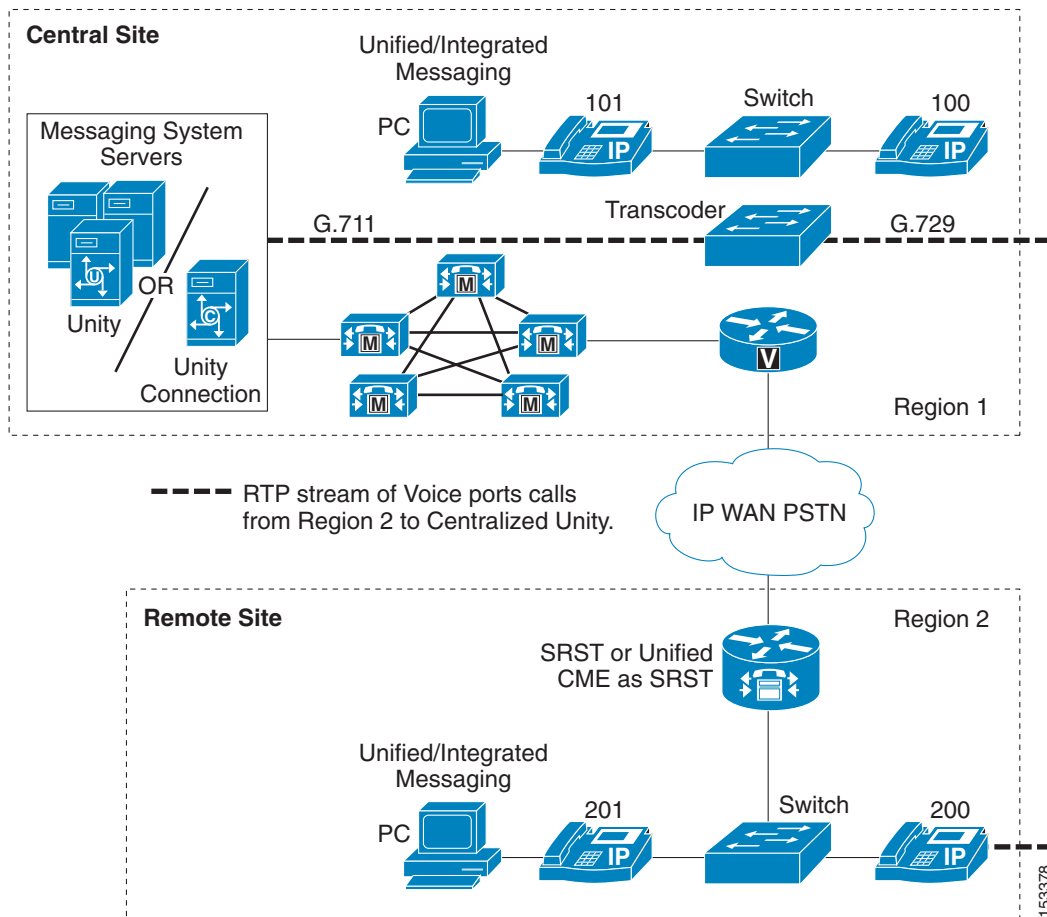
Table 13-2 *Supported Combinations of Messaging and Call Processing Deployment Models*

Model Type	Cisco Unity	Cisco Unity Connection
Single-site messaging and single-site call processing	Yes	Yes
Centralized messaging and centralized call processing	Yes	Yes
Distributed messaging and centralized call processing	Yes	No
Centralized messaging and distributed call processing	Yes	Yes
Distributed messaging and distributed call processing	Yes	No

Refer to the Cisco Unity and Cisco Unity Connection design guides available at <http://www.cisco.com> for further details on site classification and a detailed analysis of supported combinations of messaging and call processing deployment models.

Centralized Messaging and Centralized Call Processing

In centralized messaging, the Cisco Unity server is co-located with the Unified CM cluster. With centralized call processing, subscribers may be located either remotely and/or locally to the cluster and messaging server(s). (See [Figure 13-6](#).) When remote users access resources at the central site (such as voice ports, IP phones, or PSTN gateways, as in Tail-End Hop-Off (TEHO)), these calls are transparent to gatekeeper call admission control. Therefore, regions and locations must be configured in Unified CM for call admission control. (See [Managing Bandwidth, page 13-5](#).) When making inter-region calls to IP phones or MGCP gateways, IP phones automatically select the inter-region codec that has been configured. Native transcoding should be disabled so that the Cisco Unity voice ports use Unified CM transcoding resources for calls that transverse the WAN (inter-region).

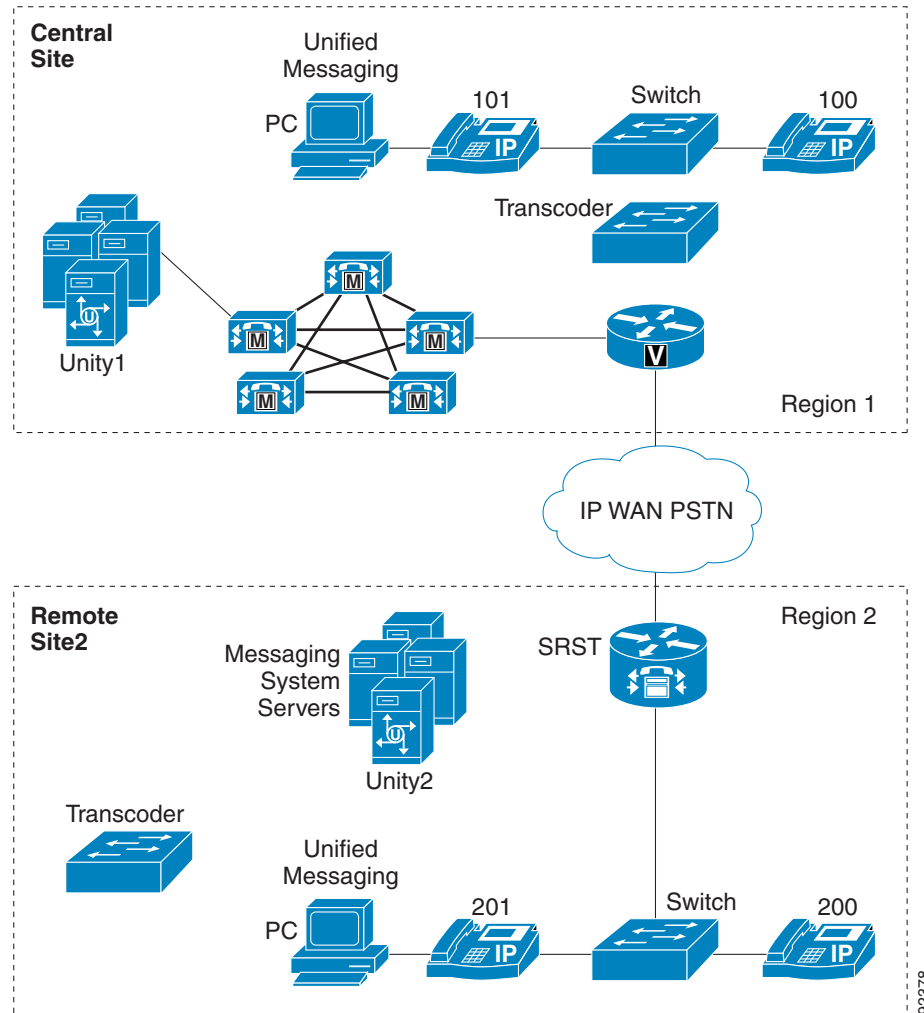
Figure 13-6 Centralized Messaging with Centralized Call Processing

In Figure 13-6, regions 1 and 2 are configured to use G.711 for intra-region calls and G.729 for inter-region calls. Native transcoding has been disabled on the Cisco Unity server.

As Figure 13-6 shows, when a call is made from extension 200 to the voicemail ports in Region 1, the inter-region G.729 codec is used at the endpoint but the RTP stream is transcoded to use G.711 on the voice ports. Native transcoding on the Cisco Unity server has been disabled in this example. Unified CM transcoding resources must be located at the same site as the voicemail system.

Distributed Messaging with Centralized Call Processing

Distributed messaging means that there are multiple messaging systems distributed within the telephony environment, and each messaging system services only local messaging clients. This model differs from centralized messaging, where clients are both local and remote from the messaging system. Figure 13-7 illustrates the distributed messaging model with centralized call processing. As with other multisite call processing models, the use of regions and locations is required to manage WAN bandwidth. For this model, you must also disable Cisco Unity native transcoding.

Figure 13-7 Distributed Messaging with Centralized Call Processing

For the configuration in Figure 13-7, transcoder resources must be local to each Cisco Unity message system site. Regions 1 and 2 are configured to use G.711 for intra-region calls and G.729 for inter-region calls. Native transcoding has been disabled on the Cisco Unity servers.

Voice messaging ports for both Cisco Unity servers must be assigned the appropriate region and location by means of calling search spaces and device pools configured on the Unified CM server. In addition, to associate telephony users with a specific group of voicemail ports, you must configure Unified CM voicemail profiles. For details on configuring calling search spaces, device pools, and voicemail profiles, refer to the applicable version of the *Cisco Unified Communications Manager Administration Guide*, available at

<http://www.cisco.com>

The messaging systems are "networked" together to present a single messaging system to both inside and outside users. For information about Cisco Unity Networking for the distributed Unity servers, Refer to the *Networking in Cisco Unity Guide*, available at

http://www.cisco.com/en/US/products/sw/voicesw/ps2237/products_feature_guides_list.html

In addition, Cisco Unified Survivable Remote Site Telephony (SRST) can be deployed at a remote site (for example, Region 2 in [Figure 13-7](#)) to allow IP phones and local Unity voicemail ports at the remote site to fall back to the Cisco IOS router with SRST enabled. Thus, SRST provides backup call processing in the event that the phones lose connectivity with Unified CM, such as during a WAN failure. This configuration provides users at the remote site with access to the local Unity server as well as MWI support during WAN failure. Likewise, Cisco Unified Communications Manager Express (Unified CME) has SRST fallback support for both IP phones and voicemail ports, and it can be used in lieu of SRST to provide backup call processing at remote sites. For more details on Unified CME and SRST, refer to the product documentation on <http://www.cisco.com>.

Combined Messaging Deployment Models

It is possible to combine messaging models in the same deployment, provided that the deployment adheres to all the guidelines listed in the preceding sections. [Figure 13-8](#) shows a user environment in which both centralized and distributed messaging are employed simultaneously.

Figure 13-8 Combined Deployment Models

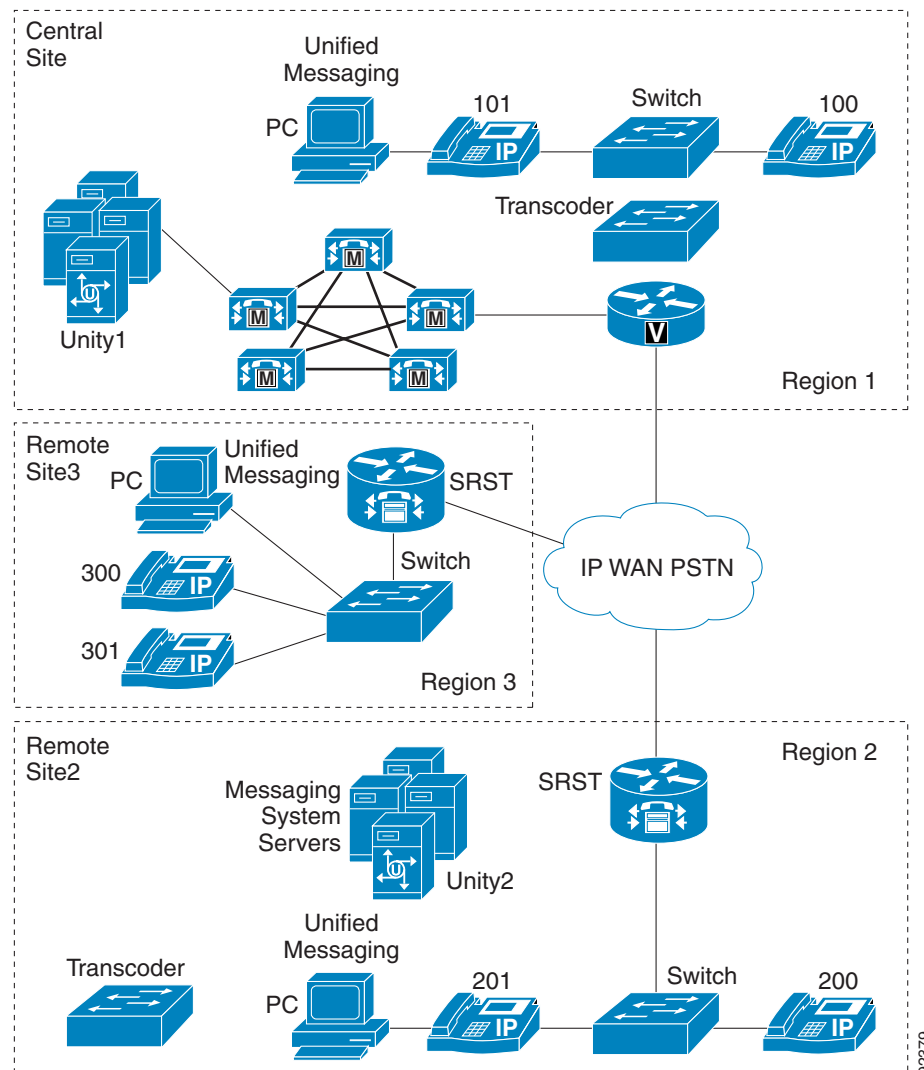


Figure 13-8 shows the combination of two messaging models. Regions 1 and 3 use centralized messaging with centralized call processing, while Region 2 uses distributed messaging with centralized call processing. All regions are configured to use G.711 for intra-regions calls and G.729 for inter-region calls.

In Figure 13-8, centralized messaging and centralized call control are used between the Central Site and Site3. The messaging system at the Central Site provides messaging services for clients at both the Central Site and Site3. Site2 uses the distributed messaging model with centralized call control. The messaging system (Unity2) located at Site2 provides messaging services for only those users located within Site2. In this deployment, both models adhere to their respective design guidelines as presented in this chapter. Transcoding resources are located locally to each messaging system site, and they support clients who access messaging services from a remote site (relative to the messaging system), as in the case of a Site2 user leaving a message for a Central Site user.

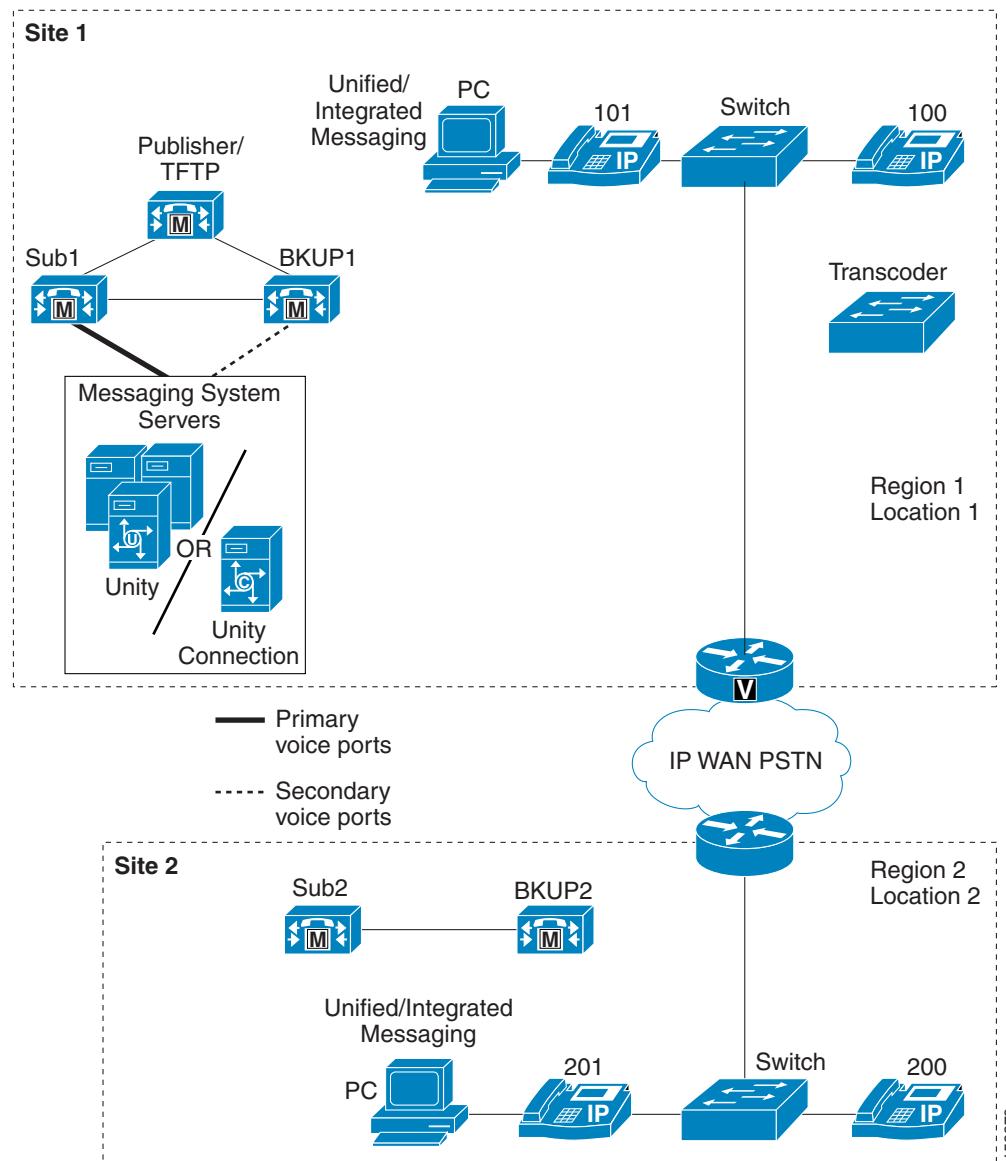
In addition, Cisco Unified Survivable Remote Site Telephony (SRST) can be deployed at a remote site (for example, Region 2 in Figure 13-8) to allow IP phones and local Unity voicemail ports at the remote site to fall back to the Cisco IOS router with SRST enabled. Thus, SRST provides backup call processing in the event that the phones lose connectivity with Unified CM, such as during a WAN failure. This configuration provides users at the remote site with access to the local Unity server as well as MWI support during WAN failure. Likewise, Cisco Unified Communications Manager Express (Unified CME) has SRST fallback support for both IP phones and voicemail ports, and it can be used in lieu of SRST to provide backup call processing at remote sites. For more details on Unified CME and SRST, refer to the product documentation on <http://www.cisco.com>.

Centralized Messaging with Clustering Over the WAN

This section addresses Cisco Unity design issues for deploying centralized messaging with Unified CM clustering over the WAN with local failover. In the case of a WAN failure with this model, all remote messaging sites will lose voicemail capability until the WAN is restored. (See Figure 13-9.)

Clustering over the WAN supports local failover. With local failover, each site has a backup subscriber server physically located at the site. This section focuses on deploying Cisco Unity centralized messaging with local failover for clustering over the WAN.

For additional information, refer to the section on [Clustering Over the IP WAN](#), page 2-19.

Figure 13-9 Cisco Unity Centralized Messaging and Clustering Over the WAN with Local Failover

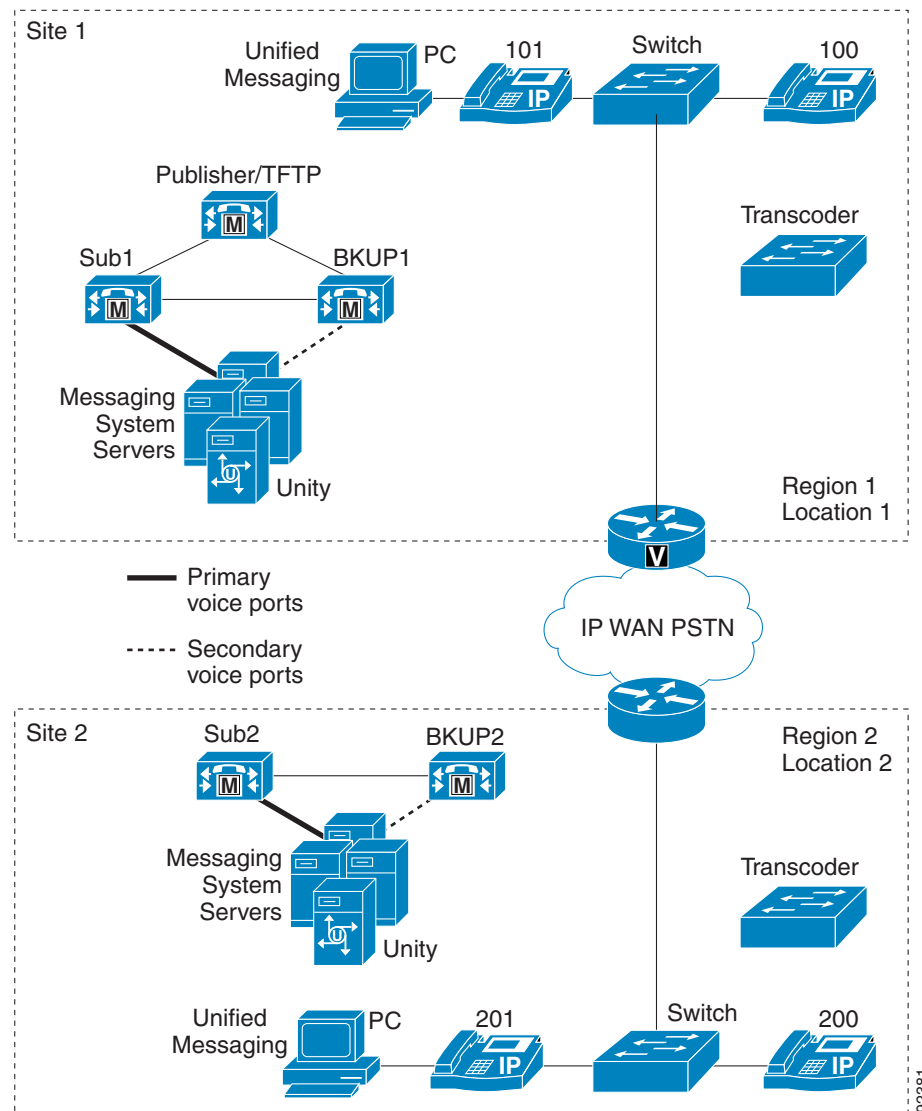
The minimum bandwidth required between cluster sites is a T1 line (1,544 MHz). This amount of bandwidth can support signaling and database traffic for up to 10,000 busy hour call attempts (BHCA), but it does not include the required media bandwidth.

Clustering over the WAN with Cisco Unified CM Release 3.3(3) and earlier supports up to four sites per cluster, but Cisco Unified CM Release 4.1 and higher supports up to eight sites. Cisco Unity supports up to the maximum number of sites in either case. The voicemail ports are configured only at the site where the Cisco Unity messaging system is located (see Figure 13-9). Voicemail ports do not register over the WAN to the remote site(s). Messaging clients at the other site(s) access all voicemail resources from the primary site. There is no benefit to configuring voice ports over the WAN to any of the remote sites because, in the event of a WAN failure, remote sites would lose access to the centralized messaging system. Because of bandwidth consideration, the voicemail ports should have TRaP disabled and all messaging clients should download voicemail messages to their local PCs (unified messaging only).

Distributed Messaging with Clustering Over the WAN

Local failover sites that also have Cisco Unity messaging server(s) deployed would have voice ports registered to the local Unified CM subscriber server(s), similar to the centralized messaging model. For information about configuring the voice ports, see [Voice Port Integration with a Unified CM Cluster](#), page 13-10, and [Voice Port Integration with Dedicated Unified CM Backup Servers](#), page 13-12.

Figure 13-10 Cisco Unity Distributed Messaging and Clustering over the WAN with Local Failover



In a purely distributed messaging implementation with clustering over the WAN, each site in the cluster would have its own Cisco Unity messaging server with messaging infrastructure components. If not all of the sites have local Cisco Unity messaging systems but some sites have local messaging clients using a remote messaging server(s), this deployment would be a combination model with both distributed messaging and centralized messaging. (See [Combined Messaging Deployment Models](#), page 13-16.) In the event of a WAN failure in this model, all remote sites that use centralized messaging will lose voicemail capability until the WAN is restored.

Each site that does not have a local messaging server must use a single messaging server for all of its messaging clients, but all such sites do not have to use the same messaging server. For example, suppose Site1 and Site2 each have a local messaging server. Site3 can then have all of its clients use (register with) the messaging server at Site2, while Site4 can have all of its clients use the messaging server at Site1. Transcoder resources are required at sites that have local Cisco Unity messaging server(s).

As with other distributed call processing deployments, calls going between these sites are transparent to gatekeeper call admission control, therefore you must configure regions and locations in Unified CM to provide call admission control. (See [Managing Bandwidth](#), page 13-5.)

The distributed Cisco Unity servers may also be networked digitally. For more information on this topic, refer to the *Cisco Unity Networking Guide*, available at <http://www.cisco.com>. The networking guides are specific to the particular messaging store deployed.

Cisco Unity Messaging Failover

Cisco Unity failover provides voicemail survivability in the case of a Cisco Unity server failure. (See [Figure 13-1](#).) With Cisco Unity local failover, both the primary and secondary Unity messaging servers reside at the same physical location, and the messaging infrastructure components are co-located with the primary server. Optionally, messaging infrastructure components (such as messaging store servers, domain controller and global catalog (DC/GC) servers, and DNS servers) can also have redundant components that may be physically co-located with the Cisco Unity secondary server.

Cisco Unity failover is supported by all of the messaging deployment models. Cisco Unity Connection currently does not support failover.

For information on configuring Cisco Unity failover, refer to the *Cisco Unity Failover Configuration and Administration Guide*, available at

<http://www.cisco.com>

Cisco Unity's SIP trunk implementation requires *call forking* for Unity failover functionality. Currently, Unified CM does not support call forking on SIP trunks, therefore Unity failover is not available when SIP trunks are used with Unified CM.

The Unified CM SIP trunk implementation also requires a media termination point (MTP) to handle DTMF Relay between the SIP trunk and SCCP endpoints. When Unity is integrated with SCCP, it can receive digits from both SCCP and SIP endpoints without an MTP.

Cisco Unity Failover and Clustering Over the WAN

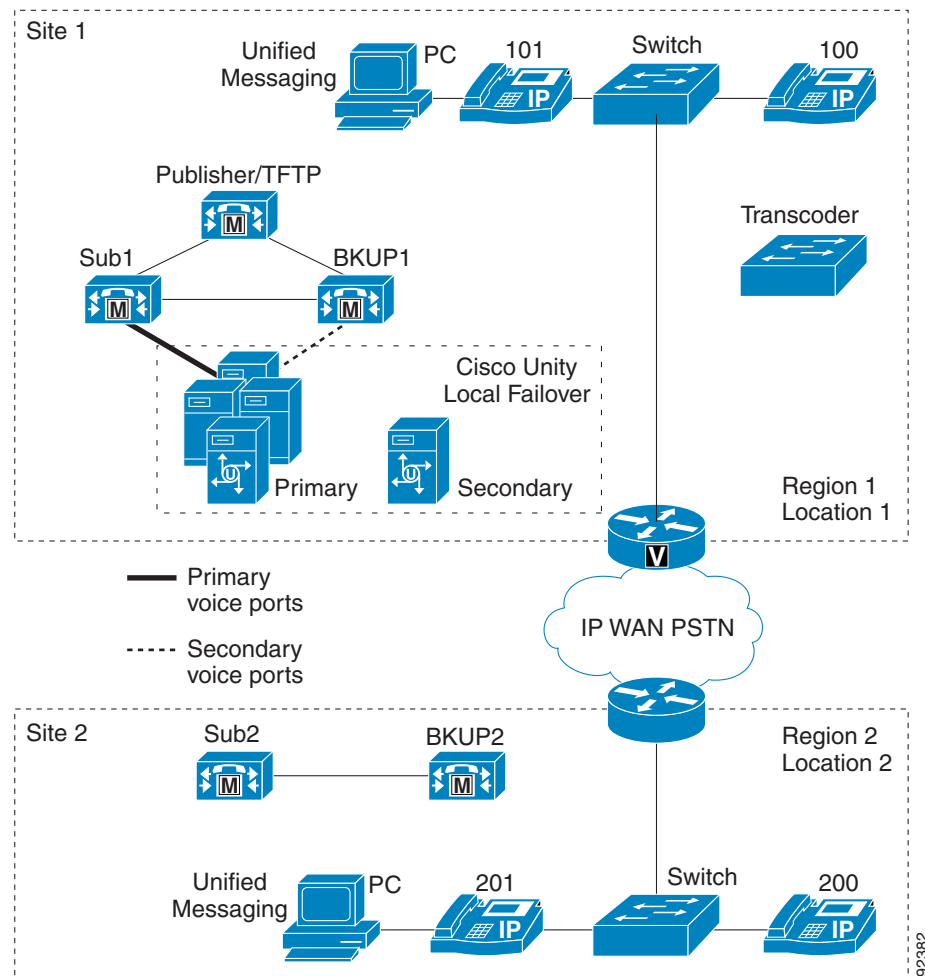
When deploying Cisco Unity local failover with clustering over the WAN, apply the same design practices described in [Centralized Messaging with Clustering Over the WAN](#), page 13-17, and [Distributed Messaging with Clustering Over the WAN](#), page 13-19. The voice ports from the primary Cisco Unity server should not cross the WAN during normal operation.

[Figure 13-11](#) depicts Cisco Unity local failover. Note that the primary and secondary Cisco Unity servers are both physically located at the same site. Cisco Unity failover supports up to the maximum number of remote sites available with clustering over the WAN for Unified CM.



Note

Unity Connection does not currently support failover.

Figure 13-11 Cisco Unity Local Failover and Clustering Over the WAN

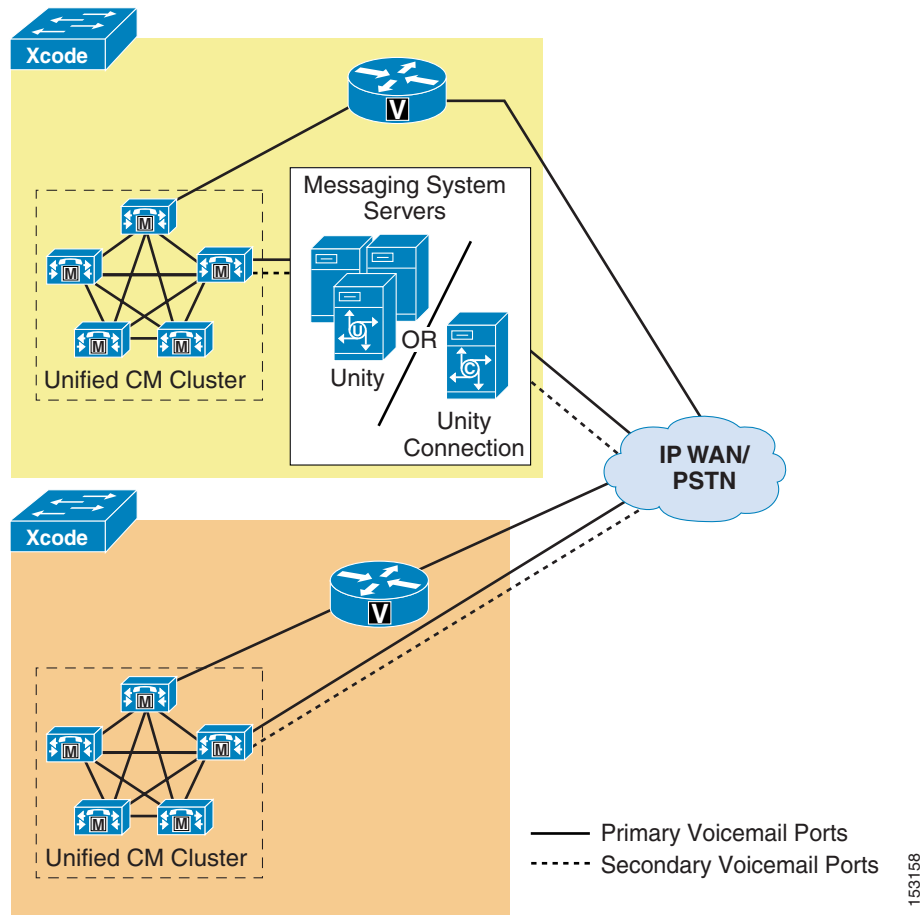
For information on configuring Cisco Unity failover, refer to the *Cisco Unity Failover Configuration and Administration Guide*, available at

<http://www.cisco.com>

Centralized Messaging with Multiple Unified CM Servers

Cisco Unity can also be deployed in a centralized messaging configuration with multiple Unified CM clusters (see [Figure 13-12](#)). See the section on [Telephony Integrations, page 13-8](#), for details on multiple integrations and MWI considerations with multiple Unified CM clusters.

Figure 13-12 Integrating Cisco Unity or Unity Connection with Multiple Unified CM Clusters



For the configuration in [Figure 13-12](#), messaging clients at both Cluster 1 and Cluster 2 sites use the Cisco Unity or Unity Connection messaging infrastructure physically located at Cluster 1.