CHAPTER 14

Cisco Unity Express

Last revised on: January 29, 2008

Cisco Unity Express provides fully integrated voicemail and automated attendant (AA) services on a Cisco IOS router for small and medium-sized businesses (SMBs) or enterprise branch offices of up to 250 users. Cisco Unity Express can be integrated with Cisco Unified Communications Manager Express (Unified CME) or Cisco Communications Manager (Unified CM) for distributed or centralized call control environments.

Overview of Cisco Unity Express

Cisco Unity Express is delivered on a network module (Cisco Unity Express NM-CUE, NM-CUE-EC, or NME-CUE) or advanced integration module (Cisco Unity Express AIM-CUE), which directly integrates into the Cisco 2800 and 3800 Series Integrated Services Routers (ISRs) and Cisco Unified Communications 500 (UC 500) platforms to support from 12 to 250 users with up to 24 ports of concurrent voicemail and integrated messaging, automated attendant sessions, or optional IVR services. Cisco Unity Express 3.0 supports NME-CUE and 24 ports.

While Cisco Unity Express uses SIP to communicate with Cisco Unified Communications Manager Express (Unified CME), Cisco Unity Express uses JTAPI to connect to Cisco Unified Communications Manager (Unified CM). When connected with Unified CM, Cisco Unity Express auto-detects the version of Unified CM that is running and compares that with its own JTAPI library version. If Cisco Unity Express detects that the Unified CM release has changed, Cisco Unity Express automatically reboots and reconfigures itself with the correct JTAPI library for the version of Unified CM to which it is connected.

This chapter discusses Cisco Unity Express as a distributed voicemail and automated attendant (AA) solution for Cisco Unified Communications Manager (Unified CM). Use Cisco Unity Express as a distributed voicemail solution if any of the following conditions apply to your Unified CM network deployment:

- Survivability of voicemail and AA access must be ensured regardless of WAN availability.
- Available WAN bandwidth is insufficient to support voicemail calls traversing the WAN to a central voicemail server.
- There is limited geographic coverage of the AA or branch site PSTN phone numbers published to the local community, and these numbers cannot be dialed to reach a central AA server without incurring toll charges.

- The likelihood is high that a PSTN call into a branch office will be transferred from the branch AA to a local extension in the same office.
- Management philosophy allows remote locations to select their own voicemail and AA technology.

Note Interoperability with Cisco Unified CM 6.*x* requires a minimum of Cisco Unity Express Release 3.0. Cisco Unity Express 2.0 provides interoperability with Cisco Unified CM 4.0, Cisco Unity Express 2.1 provides interoperability with Cisco Unified CM 4.1, and Cisco Unity Express 2.3 provides interoperability with Cisco Unified CM 4.2.

For more information about Cisco Unity Express, refer to the product documentation available at

http://www.cisco.com/en/US/products/sw/voicesw/ps5520/tsd_products_support_series_home.html

Deployment Models for Cisco Unity Express

Cisco Unity Express is supported with all of the Cisco Unified CM deployment models, including single-site deployments, multisite WAN deployments with centralized call processing, and multisite WAN deployments with distributed call processing. Figure 14-1 shows a centralized call processing deployment incorporating Cisco Unity Express, and Figure 14-2 shows a distributed call processing deployment.

Cisco Unity Express can also be deployed as a voicemail solution with Cisco Unified Communications Manager Express (Unified CME). Cisco Unity Express sites controlled by Unified CME, as well as other sites controlled by Unified CM, can be interconnected in the same network using H.323 or SIP trunking protocol. Although Cisco Unity Express can integrate with either Unified CM or Unified CME, it cannot integrate with both simultaneously.

In prior releases, Cisco Unity Express was limited to a co-resident deployment with Unified CME or a Survivable Remote Site Telephony (SRST) router. However, Cisco Unity Express and SRST or Unified CME can reside on two separate routers when deployed with Unified CM or Unified CME, respectively, with the H.323-to-SIP call routing capability introduced in Cisco IOS Release 12.3(11)T.

For details on interoperability of Unified CM and Unified CME, see Interoperability of Unified CM and Unified CM Express, page 8-27.

For additional information on supported deployment models with Unified CME, refer to the appropriate Cisco Unified Communications Manager Express design documentation available at

http://www.cisco.com

Figure 14-1 Cisco Unity Express in a Centralized Call Processing Deployment

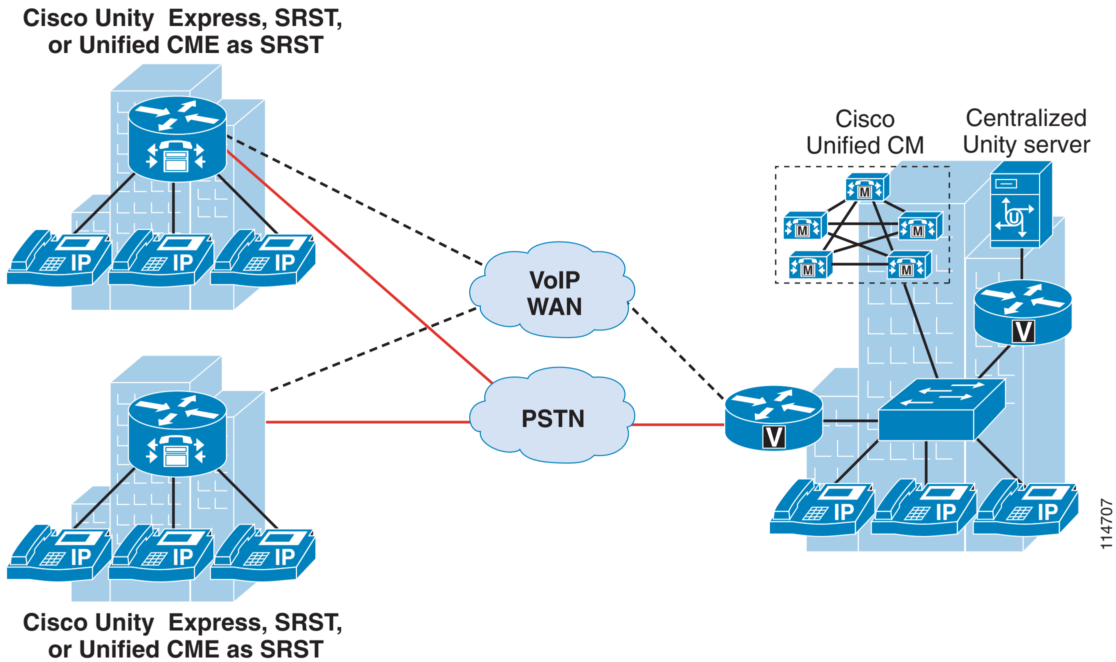
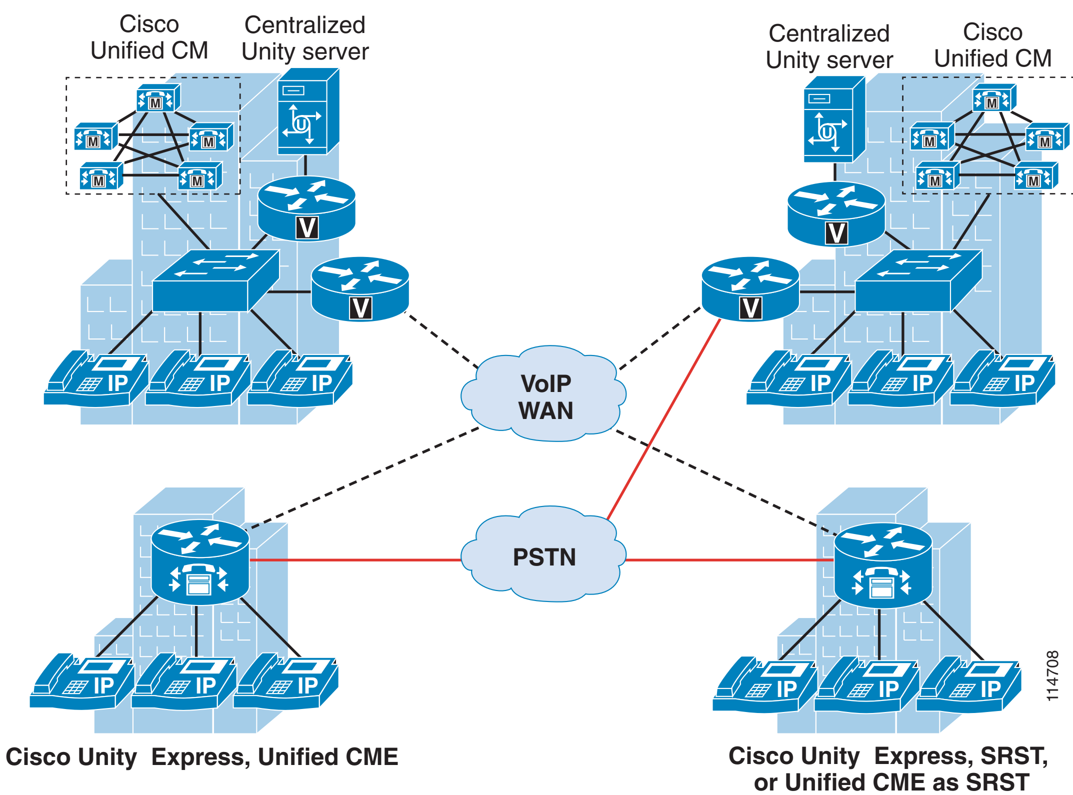


Figure 14-2 Cisco Unity Express in a Distributed Call Processing Deployment



The most likely deployment model to use Cisco Unity Express is the multisite WAN model with centralized call processing, where Cisco Unity Express provides distributed voicemail at the smaller remote offices and a central Cisco Unity system provides voicemail to the main campus and larger remote sites.

The following characteristics and guidelines apply to Cisco Unity Express in either a centralized or distributed Cisco Unified Communications Manager deployment:

- A single Cisco Unity Express can be integrated with a single Unified CM cluster.
- Cisco Unity Express integrates with Unified CM using a JTAPI application and Computer Telephony Integration (CTI) Quick Buffer Encoding (QBE) protocol. CTI ports and CTI route points control the Cisco Unity Express voicemail and automated attendant applications.
- Cisco Unity Express provides voicemail functionality to Cisco Unified IP Phones running Skinny Client Control Protocol (SCCP). Cisco Unity Express 2.3 and later releases also provide support for Session Initiation Protocol (SIP) IP phones with Unified CM 5.*x* and later.
- The following CTI route points are defined on Unified CM for Cisco Unity Express:
  - Automated attendant entry point (Cisco Unity Express can contain up to five distinct AAs and may therefore require up to five different route points.)
  - Voicemail pilot number
  - Greeting management system (GMS) pilot number (Optional; if the GMS is not used, then this route point need not be defined.)
- The following CTI ports are defined on Unified CM for Cisco Unity Express:
  - A 12 or 25-mailbox system (4 ports)
  - A 50-mailbox AIM-CUE system (6 ports)
  - A 100-mailbox NM-CUE system (8 ports)
  - A 250-mailbox NM-CUE-EC system (16 ports)
  - A 250-mailbox NME-CUE system (24 ports)
- Each Cisco Unity Express site has 250 mailboxes or less. For deployments that require more than 250 mailboxes, consider using Cisco Unity or other voicemail solutions.
- Each Cisco Unity Express mailbox can be associated with a maximum of two different extensions, if needed.
- The automated attendant function for any office deployed with Cisco Unity Express can be local to the office (using the AA application in Cisco Unity Express) or centralized (using Cisco Unity Express for voicemail only).
- Cisco Unity Express can be networked with other Cisco Unity Expresses or with Cisco Unity via Voice Profile for Internet Mail (VPIM) version 2. Thus, a Cisco Unity Express subscriber can send, receive, or forward messages to or from another remote Cisco Unity Express or Cisco Unity subscriber.
- Cisco Unity Express allows you to specify up to three Unified CMs for failover. If IP connectivity to all three Unified CMs is lost, Cisco Unity Express switches to Survivable Remote Site Telephony (SRST) call control, thus providing AA call answering service as well as mailbox access to IP phones and PSTN calls coming into the branch office.
- Cisco Unity Express automated attendant supports dial-by-extension and dial-by-name functions. The dial-by-extension operation enables a caller to transfer a call to any user endpoint in the network. The dial-by-name operation uses the directory database internal to Cisco Unity Express and does not interact with external LDAP or Active Directory databases.

- Centralized Cisco Unity Express is not supported.
- Cisco Unity Express is not supported in pure SIP networks that do not have either Cisco Unified CME or Cisco Unified CM controlling the SIP phones.

Figure 14-3 shows the protocols involved in the call flow between Unified CM and Cisco Unity Express.

Figure 14-3 Protocols Used Between Cisco Unity Express and Unified CM

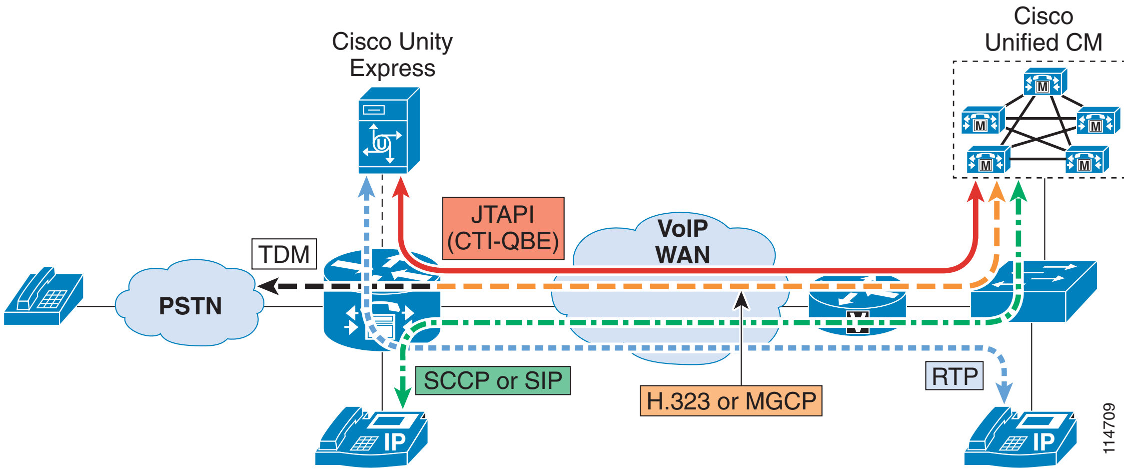


Figure 14-3 illustrates the following signaling and media flows:

- Phones are controlled via SCCP from Unified CM.
- Cisco Unity Express is controlled via JTAPI (CTI-QBE) from Unified CM.
- The Message Waiting Indicator (MWI) on the phone is affected by Cisco Unity Express communicating a change of mailbox content to Unified CM via CTI-QBE, and by Unified CM in turn sending a MWI message to the phone to change the state of the lamp.
- The voice gateway communicates via either H.323, SIP, or MGCP to Unified CM.
- Real-Time Transport Protocol (RTP) stream flows carry the voice traffic between endpoints.

Figure 14-4 shows the protocols involved in the call flow between the SRST router and Cisco Unity Express when the WAN link is down.

Figure 14-4 Protocols Used Between Cisco Unity Express and the SRST Router

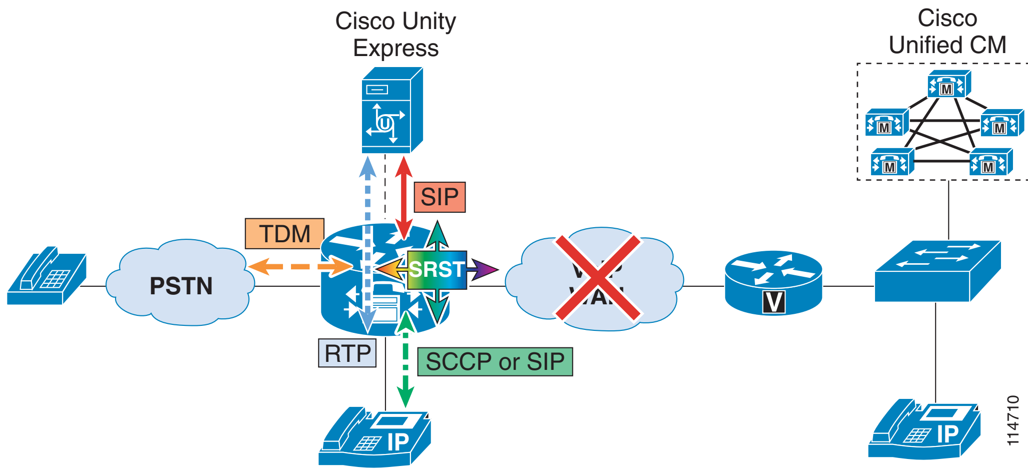


Figure 14-4 illustrates the following signaling and media flows:

- Phones are controlled via SCCP from the SRST router.
- Cisco Unity Express communicates with the SRST router via an internal SIP interface.
- Although MWI changes are not supported in SRST mode with previous releases of Cisco Unity Express, voice messages can be sent and retrieved as during normal operation, but the MWI lamp state on the phone remains unchanged until the phone registers again with Unified CM. At that time, all MWI lamp states are automatically resynchronized with the current state of the users' Cisco Unity Express voicemail boxes. Cisco Unity Express 3.0 and later releases support MWI for SRST mode
- Cisco Unity Express 2.3 and later releases now support SIP Subscriber/Notify and Unsolicited Notify to generate MWI notifications, in both Unified CME and SRST modes.
- When not located on the same router, the voice gateway communicates via H.323 to the SRST router. Configure H.323-to-SIP conversion with the command **allow-connection h323 to sip** if Cisco Unity Express is located on a separate SRST router.
- RTP stream flows carry the voice traffic between endpoints.
- SRST subscribes to Cisco Unity Express for MWI for each of the ephone-dns registered to receive MWI notifications.

Note Unified CM MWI (JTAPI) is independent of the SIP MWI methods.

Cisco Unity Express Voicemail Networking

Cisco Unity Express voicemail networking is the ability to send voicemail messages between Cisco Unity Express systems and between Cisco Unity Express and Cisco Unity systems, using an embedded Simple Mail Transfer Protocol (SMTP) server and a subset of the Voice Profile for Internet Mail (VPIM) version 2 protocol.

Cisco Unity Express communicates with Cisco Unity 4.0.3 and later. Cisco Unity Express uses SMTP (RFC 2821) for message delivery between remote sites.

Cisco Unity Express voicemail networking provides the following capabilities:

- Subscribers can receive, send, and forward messages to or from another remote Cisco Unity Express or Cisco Unity for locations configured on the originating system.
- Subscribers can also reply to a remote message received from a remote system.
- Subscribers can be recipients of a Unity-originated distribution list or individual message.

The following characteristics and design considerations apply to Cisco Unity Express voicemail networking:

- Cisco Unity Express voicemail networking uses blind addressing with known location ID and user extension.
- Cisco Unity Express voicemail networking is able to define remote locations by means of a five-letter location abbreviation (used for confirmation of addressing only).
- Message encoding can be dynamic, and SMTP session negotiation determines either .wav or G.726 encoding format.
- Cisco Unity Express voicemail networking is able to exchange G.711 .wav file and G.726 Message Encoding.
- Use G.726 for sites with low bandwidth. G.726 forces 32 kbps Adaptive Differential Pulse Code Modulation (ADPCM) format regardless of SMTP negotiation.
- The sender's spoken name may or may not be contained in the VPIM message. If the spoken name is not included in a message, then when a VPIM message is played, the name confirmation will state only the extension and location of the sender. Cisco Unity Express includes the sender's spoken name in the VPIM message if **send spoken name** is configured for a particular remote location.
- Cisco Unity Express voicemail networking notifies the sender if the message has not been sent for an hour. It also sends a Non-Delivery Receipt (NDR) if it fails to deliver for 6 hours or if the recipient's mailbox is full or non-existent.
- Cisco Unity Express voicemail networking uses an SMTP server, which accepts only incoming sessions from locations defined in the remote location tables.
- Each Cisco Unity Express must have its own email domain or sub-domain name, so that messages can eventually be routed to it.
- For interworking with Cisco Unity, configure Cisco Unity Express locations with a domain or sub-domain name.
- Cisco Unity Express tries every 15 minutes for 6 hours to deliver a message to a recipient. (Neither value is configurable.)
- Each Cisco Unity Express allows only 2 simultaneous SMTP send and receive sessions.
- Cisco Unity Express supports a total of 500 remote sites.

Voicemail Networking with a Messaging Gateway

Smaller voicemail 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.

A messaging gateway can provide the following benefits:

- It enables interworking for multiple autonomous voicemail networks with VPIM.
- It provides scalable voicemail networks for VPIM.
- It simplifies voicemail VPIM network additions and expansion.

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 networked fully without requiring the overhead of a full-mesh 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 product documentation available on http://www.cisco.com.

Fax Deployment with Cisco Unity Express

Cisco Unity Express 3.0 and later releases support both inbound and outbound faxing. Outbound faxing enables faxes to be printed to the fax machine. Cisco Unity Express supports both a single DID number and separate DID numbers for voice and fax calls. The single DID approach allows each user to have one number for both voicemail and fax, while separate DID numbers provide each user with two numbers, one for fax and one for their voice extension. In both cases, the same mailbox is used for storing both voice and fax messages.

The fax support relies on T.37 fax support from the Cisco IOS Gateway. The T.37 store-and-forward Cisco IOS fax gateway uses the T.37 store-and-forward fax application, which consists of on-ramp and off-ramp processes. Cisco Unity Express relies on the fax detection application to support single DID functionality but requires that the on-ramp application be configured on the fax gateway when separate DIDs for fax and voice calls are used. The Cisco fax gateway, acting as the on-ramp gateway, receives faxes from end users, converts them into TIFF files, creates standard Multipurpose Internet Mail Extension (MIME) email messages with attached TIFF files, and forwards the fax mail messages to the designated SMTP server (which is Cisco Unity Express) for storage. The off-ramp gateway handles calls going out from the network to a fax machine or the PSTN by dialing the POTS to communicate with a remote fax machine (Group 3 fax device).

Cisco IOS fax gateways support the following fax capabilities:

- Fax pass-through and fax pass-through with upspeed
- Cisco Fax Relay
- T.38 Fax Relay
- T.37 Store-and-Forward Fax
- IVR applications for fax

On-ramp and off-ramp fax processes can reside on the same gateway or separate gateways.

Note Cisco Unity Express supports only Cisco IOS fax gateways. Third-party fax gateways are not supported.

For more details on Cisco IOS fax gateway capabilities, refer to the related product documentation available at http://www.cisco.com.

Cisco Unity Express supports the following fax deployment scenarios:

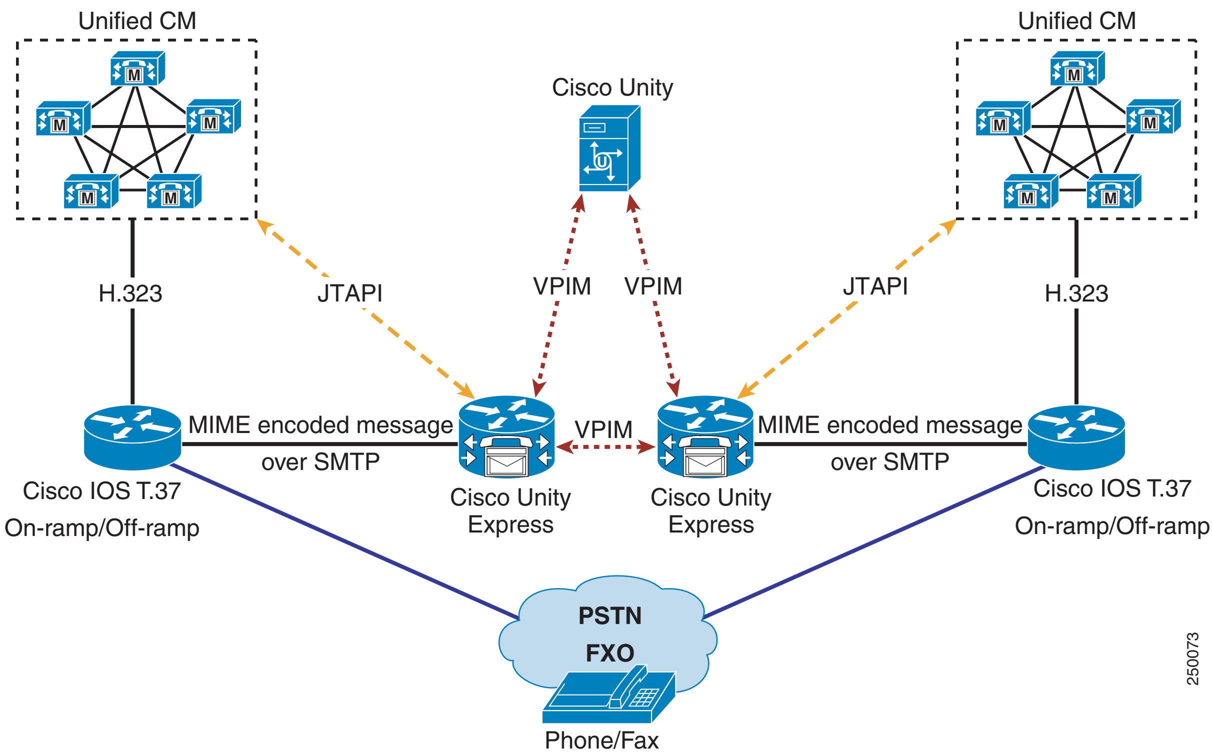
- Cisco Unity Express, on-ramp, and off-ramp applications all reside on the same voice gateway.
- Cisco Unity Express and the on-ramp application reside on the same voice gateway, but the off-ramp application runs on a separate voice gateway.
- Cisco Unity Express and the off-ramp application reside on the same voice gateway, but the on-ramp application runs on a separate voice gateway.

- Cisco Unity Express, the on-ramp application, and the off-ramp application each runs on a separate voice gateway.

Note Two or more instances of Cisco Unity Express cannot integrate with the same gateway for inbound fax calls.

Figure 14-5 shows a sample fax deployment scenario for Cisco Unity Express with Unified CM.

Figure 14-5 Fax Deployment Scenario for Cisco Unity Express with Unified CM



If the call is a fax call, whether Cisco Unity Express is integrated with Unified CME or Unified CM, the fax application configured on the outbound Multimedia Mail over IP (MMoIP) dial peer converts the fax into an email message with TIFF attachment(s) and sends the message over SMTP to Cisco Unity Express.

Cisco Unity Express uses the fax detection application to support single DID functionality. The fax detection application has a limitation that causes the fax call to get disconnected and requires the fax to be resent if either of the following cases occurs:

- The fax call is picked up and disconnected before the application detects that it is the fax call.
- After picking up the call and hearing CNG (fax) tones, the user tries to transfer the call to the fax (MMoIP) dial peer.

Cisco recommends that you configure the on-ramp application on the fax gateway when you use separate DIDs for fax and voice calls.

The following characteristics and guidelines apply when deploying fax capability with Cisco Unity Express and Unified CM:

- Cisco Unity Express supports only analog fax machines to fax services.
- The administrator should enable fax on the mailboxes requiring the fax support.
- The fax detection application required to support single DID usage for fax and voicemail must run on the originating gateway
- Cisco Unity Express sends VPIM-encoded voice messages or faxes over SMTP to another Cisco Unity Express or Cisco Unity server in the network.
- Regardless of whether Cisco Unity Express is integrated with Unified CME or Unified CM, Cisco Unity Express can interoperate with Cisco Unity to receive, send, and forward fax messages as well as voice messages from Unity, or vice versa, via VPIM.
- A fax message can be printed using the fax number configured at the system level. This number is displayed when the user is trying to print the fax using the Telephony User Interface (TUI) or Voice View Express (VVE), and it is changeable to allow users to print the fax messages at an alternative fax machine.
- Cisco Unity Express does not support broadcast fax messages.
- Cisco Unity Express send a Delayed Delivery Record (DDR) or Non-Delivery Receipt (NDR) when fax message delivery is delayed or failed, respectively.
- Cisco Unity Express supports the same number of the total TUI sessions as the total number of fax sessions.
- Cisco Unity Express integrates with one Cisco fax gateway for outbound fax and one for inbound fax calls. Inbound and outbound calls can use the same gateway or different gateways. Two or more instances of Cisco Unity Express cannot integrate with the same fax gateway for inbound fax calls
- Assign a unique fax DID number to a user when separate DIDs are used for fax and voice.
- Cisco Unity Express supports mailbox login using the voice number but not the fax DID number.
- Cisco Unity Express supports the fax feature when Cisco Unity Express is in SRST mode. When in SRST mode, Cisco Unity Express fax functionality works between the users within the same SRST site without needing to go across WAN.

Best Practices for Deploying Cisco Unity Express

Use the following guidelines and best practices when deploying the Cisco Unity Express.

- Ensure that the IP phones having Cisco Unity Express as their voicemail destination are located on the same LAN segment as the router hosting Cisco Unity Express.
- When Cisco Unity Express is deployed on a router separate from Unified CME or SRST, configure the command **allow-connections h323 to sip** for H.323-to-SIP routing.
- If uninterrupted AA and voicemail access is required for a site deployed with Cisco Unity Express, ensure that Cisco Unity Express, SRST, and the PSTN voice gateway are all located at the same physical site. Hot Standby Router Protocol (HSRP) or other redundant router configurations are not currently supported with Cisco Unity Express.

- Each mailbox can be associated with a primary extension number and a primary E.164 number. Typically, this number is the direct-inward-dial (DID) number that PSTN callers use. If the primary E.164 number is configured to any other number, use Cisco IOS translation patterns to match either the primary extension number or primary E.164 number so that the correct mailbox can be reached during SRST mode.
- Each Cisco Unity Express site must be associated with a CTI route point for voicemail and one for AA (if licensed and purchased), and you must configure the same number of CTI route points as Cisco Unity Express ports licensed. Ensure that the number of sites with Cisco Unity Express does not exceed the CTI scalability guidelines presented in the chapter on Call Processing, page 8-1.
- Cisco Unity Express is associated with a JTAPI user on Unified CM. Although a single JTAPI user can be associated with multiple instances of Cisco Unity Express in a system, Cisco recommends associating each dedicated JTAPI user in Unified CM with a single Cisco Unity Express.
- Cisco Unity Express can be deployed on a separate SRST router or a separate PSTN gateway.
- If Unified CM is upgraded from a previous version, the password of the JTAPI user automatically gets reset on the Unified CM. Therefore, after the upgrade, the administrator must make sure that the JTAPI password is synchronized between Cisco Unity Express and Unified CM so that Cisco Unity Express can register with Unified CM.
- Calls into Cisco Unity Express use G.711 only. Cisco recommends using a local transcoder to convert the G.729 calls traversing the WAN into G.711 calls. You can configure Unified CM regions with the G.711 voice codec for intra-region calls and the G.729 voice codec for inter-region calls.
- If transcoding facilities are not available at the Cisco Unity Express site, provision enough bandwidth for the required number of G.711 voicemail calls over the WAN. Configure the Unified CM regions with the G.711 voice codec for calls between the IP phones and Cisco Unity Express devices (CTI ports and CTI route points).
- Cisco Unified CM 5.1 and later releases support SIP trunking protocol; however, Cisco Unity Express uses JTAPI to communicate with Unified CM. Cisco Unity Express supports both SCCP and SIP phones.
- Configure a SIP trunk for SRST and Unified CM for support of SIP phones (through JTAPI).
- Cisco Unity Express does not support in-band DTMF tones; it supports only DTMF relay.
- With Cisco Unity Express, DTMF is carried out-of-band via either the SIP or JTAPI call control channels.
- Cisco Unity Express 2.3 supports G.711 SIP calls with RFC 2833 into Cisco Unity Express.
- Cisco Unity Express 3.0 supports G.729 SIP calls via a transcoder, with the ability added in Cisco IOS Release 12.3(11) XW for RFC 2833 to pass through a transcoder.
- Cisco Unity Express supports delayed media (no SDP in the INVITE message) for call setup in case of a slow-start call from Unified CM.
- Cisco Unity Express supports both blind and consultative transfer, but the default transfer mode is consultative transfer (semi-attended) using REFER in SIP calls. Use the Cisco Unity Express command line interface to explicitly change the transfer mode to consultative transfer using REFER or blind transfer using BYE/ALSO. If REFER is not supported by the remote end, BYE/ALSO will be used.
- Cisco Unity Express supports outcall for voice message notifications. It also supports consultative transfers. During both of these call setups, Cisco Unity Express can receive 3*xx* responses to the INVITE. Cisco Unity Express processes only 301 (Moved Permanently) and 302 (Moved Temporarily) responses to the INVITE. This requires that the URL from the Contact header from the 3*xx* response be used to send a new INVITE. 305 (Use Proxy) responses are not supported.

- The CTI ports and CTI route points can be defined in specific locations. Cisco recommends using locations-based call admission control between Unified CM and Cisco Unity Express. RSVP may also be used.
- Ensure proper Quality of Service (QoS) and bandwidth for signaling traffic that traverses the WAN between Cisco Unity Express and Unified CM. Provision 20 kbps of bandwidth for CTI-QBE signaling for each Cisco Unity Express site. See the chapter on Network Infrastructure, page 3-1, for more details.
- The CTI-QBE signaling packets from Unified CM to Cisco Unity Express are marked with a DSCP value of AF31 (0x68). Unified CM uses TCP port 2748 for CTI-QBE signaling.
- The Unified CM JTAPI library sets the proper IP Precedence bits in all outgoing QBE signaling packets. As a result, all signaling between Cisco Unity Express and Unified CM will have the proper QoS bits set.