



Chapter 1: Cisco Unity Express Deployment Models

Cisco Unity Express is an entry-level auto-attendant (AA) and voice-mail system that is integrated with a Cisco IOS router for small-to-medium business or enterprise branch offices requiring up to 100 mailboxes.

Cisco Unity Express provides a distributed AA and voice-mail application to an IP Telephony solution where the large offices may have a high-end voice mail solution (such as Cisco Unity) and one or more of the smaller remote offices may use a local or distributed Cisco Unity Express for voice mail. Cisco Unity Express is deployed with one of two call-processing models that are based on either Cisco CME or Cisco CallManager.

The call processing engine (Cisco CME or Cisco CallManager) manages the IP phones and features such as call-forward-busy (CFB) and call-forward-no-answer (CFNA) to the voice mail pilot number, while Cisco Unity Express provides the AA menus, scripts and voice mail telephony user interface (TUI) sessions for callers retrieving or leaving voice messages. Cisco Unity Express stores the AA scripts and prompts, voice mail subscriber spoken names, greetings and voice mail messages.

Cisco Unity Express interfaces with Cisco CME call control via a Session Initiation Protocol (SIP) interface and to Cisco CallManager via a Java Telephony Applications Programming Interface (JTAPI) interface.

Cisco Unity Express Overview

Cisco Unity Express is offered in two forms: a Network Module (NM) and an Advanced Integration Module (AIM) that is added to the router in the office. Cisco Unity Express hardware includes a CPU to offload AA and voice-mail processing from the router CPU such that Cisco Unity Express has minimal impact on the router CPU, as well as storage (hard disk on the NM and compact flash on the AIM) for the AA menus, prompts, voice-mail greetings, and messages.

The NM-CUE offers up to 100 hours of voice mail storage and is available as of Cisco Unity Express 1.0. The AIM-CUE offers up to 8 or 14 hours of storage (depending on which flash card is equipped) and is available as of Cisco Unity Express 1.1. Cisco Unity Express is available in four different product tiers (or licenses), including a 12-mailbox, 25-mailbox, 50-mailbox and 100-mailbox license.

The key benefits of Cisco Unity Express include the following:

- Provides a cost-effective, Cisco-integrated voice mail solution for IP phones in the branch or remote office as part of the full-service branch (FSB) evolution.
- Delivers a router-integrated AA and voice-mail application for Cisco CME.

- Enables a decentralized voice-mail solution in a centralized Cisco CallManager network. It keeps voice-mail traffic off the WAN (does not require the WAN to have the bandwidth and quality of service (QoS) to carry real-time traffic). As a result, voice-mail availability continues during WAN failures—via Cisco Survivable Remote Site Telephony (SRST).

Standalone Office

The deployment model shown in [Figure 1](#) is a single site representing a standalone (autonomous) small-medium business office, such as a medical or legal office. While it is possible to create this deployment with either Cisco CME or Cisco CallManager as the call controller, it is almost certain to be Cisco CME (shown in [Figure 1](#)) as Cisco CallManager is not cost-effective for a standalone environment where there are fewer than 250 people in the office.

This type of office has very limited or no IT expertise. A Cisco partner, reseller, or systems integrator (SI) is likely to install, configure, and maintain the system. An Internet Service Provider (ISP) or local exchange carrier (LEC) is likely to provide the Internet and PSTN service. It is possible that this organization may also sell, install, and maintain the system for the small-medium business. If the end customer performs configuration changes on the Cisco Unity Express system, end-customer network administration staff will access the system via the GUI (not the CLI). The SI or ISP may well access the system via the CLI.

Figure 1 Standalone Office

The office environment depicted in [Figure 1](#) includes the following characteristics:

- All offnet voice connectivity is via the PSTN to customers or clients of the business.
- Onnet voice calls are made between IP phones connected in the office.
- The data connectivity from the site is purely for Internet access and is connected to a local ISP service.

Multisite Networks

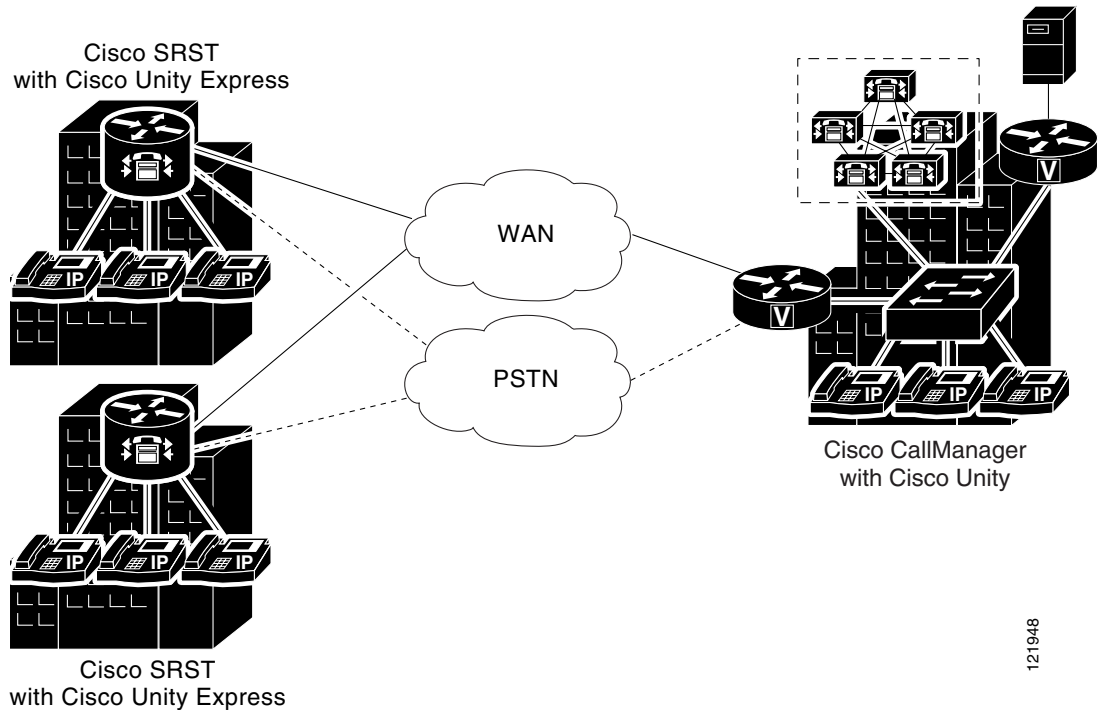
In this deployment model, there are multiple sites of the same business or enterprise networked together. The networking may be very loose (very little inter-site business is conducted, such as restaurants of the same chain) or fairly tight (a branch office of a larger enterprise where there is significant business traffic between sites).

The sites in the network may be of two forms:

- All Cisco CME sites (a distributed call-control model) with a Cisco Unity Express providing local AA and voice-mail service at each site (shown in [Figure 2](#)).
- The IP phones and PSTN calls at all sites in the network are controlled by a centralized Cisco CallManager while Cisco Unity Express in the router (with Cisco SRST configured) at each remote site provides distributed voice mail (shown in [Figure 3](#)).

Figure 2 *Multisite Network with Cisco CME*

Figure 3 Multisite Network with Cisco CallManager



The network may also be a hybrid of the two preceding deployment models and contain a mix of sites of any of the types summarized in Table 2.

Table 2 Site Configurations in a Multisite Network

	Call Control	
Voice Mail	Centralized	Distributed
Centralized	Cisco CallManager and Cisco Unity	Cisco CME and Cisco Unity
Distributed	Cisco CallManager and Cisco Unity Express	Cisco CME and Cisco Unity Express

The decision points in choosing between distributed and centralized call control are very often the same as those that govern choosing between distributed and centralized voice mail. These include the following:

- Management philosophy of your company
 - Who makes decisions on the technology used at which sites?
 - Is this a centralized IT decision or, are these more often left up to the sites themselves in a “remote ownership” or “franchise” model?
- Redundancy and availability considerations determine whether a distributed solution is the right fit when AA and voice-mail access must be available during WAN failures.
- WAN bandwidth availability and WAN QoS provisioning— A centralized voice mail model requires that all calls into voice mail traverse the WAN. This requirement might not be optimal for your application or your WAN architecture might not yet be ready for this traffic.

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Sites Networked with PSTN

The first type of multisite deployment is where the coupling between the sites is either loose (based on your business model), or the WAN is not QoS-enabled and therefore not capable of carrying VoIP traffic. It might not be in your best interest to upgrade the WAN at this time. For sites fitting this model, the WAN between the sites is used purely for low-bandwidth data access. Any voice calling between sites is via the PSTN.

This type of network has the following characteristics:

- All offnet voice connectivity is via the PSTN to customers or clients of the business.
- Onnet voice between sites are made between IP phones connected to the Cisco CME or Cisco CallManager system resident at the site and the PSTN. All other sites are reached via PSTN dialing.
- WAN connectivity between sites is used purely for data traffic.

Each site in this deployment model operates essentially as a standalone site. Small sites will tend to be Cisco CME and Cisco Unity Express (cost-effective at the low end) while large sites are likely to be Cisco CallManager and Cisco Unity (high-end solutions).

A variation of this deployment occurs when the WAN is QoS-enabled but has limited bandwidth for VoIP traffic. Perhaps inter-site onnet calls can be carried via VoIP, but AA and voice-mail calls cannot. This is another situation where a distributed AA and voice-mail solution is a good fit.

Sites Networked with H.323 VoIP

A second type of multisite deployment features a tighter coupling between sites and a VoIP-capable WAN between sites. For sites fitting this model, the WAN between the sites is used for both voice and data access, and the PSTN is used only for off-net calls.

This type of office has the following characteristics:

- All offnet voice connectivity is via the PSTN to customers or clients.
- Onnet voice between sites are made between IP phones connected to the Cisco CME or Cisco CallManager system resident at the site via VoIP to all other sites in the enterprise.
- The WAN connectivity between sites is used for voice and data traffic.

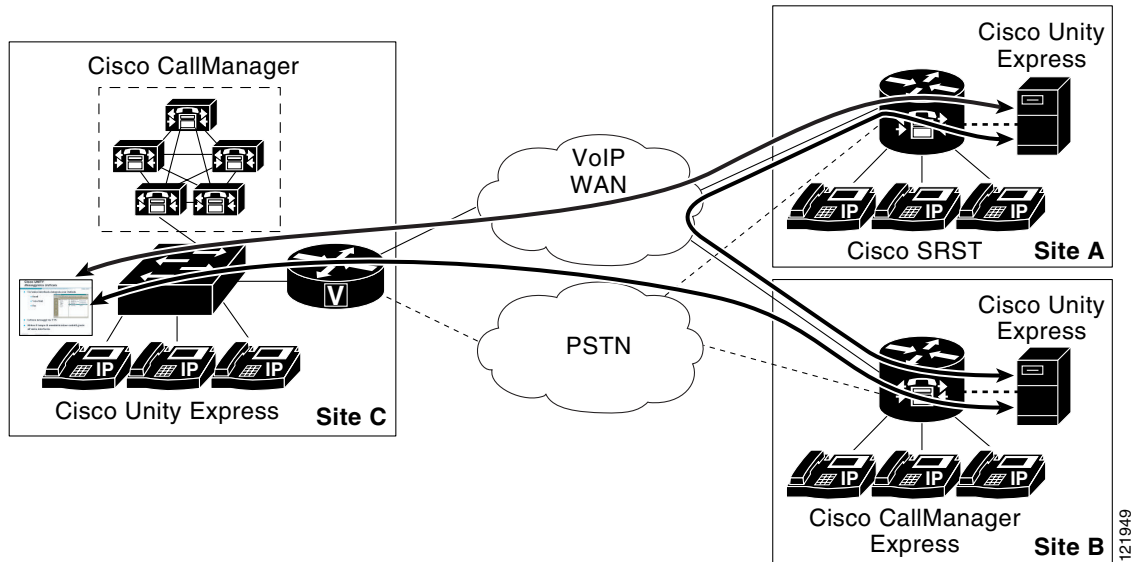
When networking together, mixed Cisco CME and Cisco CallManager sites with VoIP, deploy Cisco CME 3.1 (or later releases) and ensure the H.450 and other internetworking dependencies between these call controllers are addressed in your network.

Cisco Unity Express Voice Mail Networking

Cisco Unity Express 2.0 and later releases support voice-mail networking via Voice Profile for Internet Mail (VPIM) Version 2. While this protocol is widely supported by voice mail systems in general, Cisco Unity Express supports networking only with other sites running Cisco Unity Express and Cisco Unity Release 4.03 or later releases.

As in the previous sections, a network can be deployed with various configurations for each site, including Cisco CME with Cisco Unity Express, Cisco CallManager with Cisco Unity Express and Cisco CallManager with Cisco Unity. In such a network, all the Cisco Unity Express and Cisco Unity sites can be networked together via VPIM, as shown in [Figure 4](#).

Figure 4 Cisco Unity Express Voice Mail Networking



Unsupported Deployments

The following deployment scenarios are not supported (up to Cisco CME 3.2 and Cisco Unity Express 2.1):

- Sites networked with SIP as the VoIP protocol. While Cisco CME by itself supports SIP trunking, these deployments are not yet supported if Cisco Unity Express is used in conjunction with Cisco CME. Only H.323 VoIP trunking is supported.
- For implementations prior to Cisco CME 3.2 and Cisco Unity Express 2.0, these two components must be collocated in the same router chassis. As of Cisco CME 3.2 and Cisco Unity Express 2.0, this is no longer a requirement. While you should have LAN connectivity between Cisco CME and Cisco Unity Express (and not low-speed WAN links), these two components may now reside in physically different routers.
- Cisco SRST, Cisco Unity Express and the PSTN gateway at the site must be collocated in the same router chassis up to and including Cisco SRST Release 3.1. As of Cisco SRST Release 3.2, the PSTN gateway portion can be separated out onto a different router.
- Multiple Cisco Unity Express systems in the same router chassis.
- A “centralized” Cisco Unity Express model where a single Cisco Unity Express system provides voice mail to IP phones at more than one Cisco CME or Cisco SRST site.
- SIP phones as Cisco Unity Express voice mail subscribers—only SCCP IP phones are supported.
- Analog phones as Cisco Unity Express voice mail subscribers. Analog phones (FXS ports) can connect to Cisco Unity Express as “outside” callers into the AA and voice-mail systems, but cannot receive subscriber features such as call-forward busy (CFB), call-forward no answer (CFNA), or message waiting indicator (MWI.)

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