Unified Communications Endpoints

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A variety of endpoints can be used in a Cisco Unified Communications deployment. These endpoints range from gateways that support ordinary analog phones in an IP environment to an extensive set of native IP phones offering a range of capabilities.

When deploying endpoints, you need to consider several factors, including authentication, upgrades, signaling protocol, Quality of Service (QoS), and so forth. The Unified Communications system must be designed appropriately to accommodate these factors.

This chapter summarizes various types of Unified Communications endpoints and covers design and deployment considerations including high availability and capacity planning. The Unified Communications endpoints covered in this chapter can be categorized into the following major types:

The sections listed above provide information about each endpoint type, including deployment considerations. That information is followed by a discussion related to high availability, capacity planning, and design considerations for effectively deploying endpoints.

Use this chapter to understand the range of available endpoint types and the high-level design considerations that go along with their deployment.

What's New in This Chapter

Table 18-1 lists the topics that are new in this chapter or that have changed significantly from previous releases of this document.

Table 18-1   New or Changed Information Since the Previous Release of This Document

New or Revised Topic	Described in:	Revision Date
A few minor updates	Various sections	April 30, 2013
Minor updates related to QoS, WLAN video call capacity, and lack of Cisco EX Series endpoint registration redundancy	Various sections throughout this chapter	October 31, 2012
Removal of product-specific content such as product support lists, QoS example configurations, and endpoint feature summary tables	Various sections throughout this chapter	June 28, 2012

Unified Communications Endpoints Architecture

Call signaling in Cisco Unified Communications Manager (Unified CM), Cisco Business Edition, and Cisco Unified Communications Manager Express (Unified CME) distinguishes between line-side signaling and trunk-side signaling. Whereas trunk-side signaling is used for connecting the entire call processing cluster or router to other servers and gateways, the line side is used for connecting end-user devices to the call processing platform. The two interfaces are distinct in the services they offer, with the line side offering a rich set of user-oriented features.

Session Initiation Protocol (SIP) and Skinny Client Control Protocol (SCCP) are the two main line-side signaling protocols supported by Cisco call processing platforms. All Cisco endpoints support either or both of these protocols. The set of features supported in both protocols is roughly equivalent, and the choice of which protocol to use is essentially a personal preference in a deployment. Cisco endpoints must be configured with several operating parameters before they can be used to make or receive calls or to run applications. This configuration must be performed in advance on the call processing server or router. Once configured, the call processing platform generates a configuration file for the endpoint to use, and it stores that file on a Trivial File Transfer Protocol (TFTP) server. The endpoints themselves go through a boot-up sequence when powered on. They retrieve this configuration file before they register with the appropriate server, and then they are ready to be used. The endpoints execute the following steps as part of the boot-up sequence:

1. When connected to the access switch, if the endpoint is not plugged in to a power source, it attempts to obtain power from the switch (Power over Ethernet).

2. Once power is obtained, if device security is enabled, the endpoint presents its credentials to the security server.

3. If it is allowed to use the network, the endpoint obtains its network parameters such as IP address, Domain Name Service (DNS) servers, gateway address, and so forth, either through static provisioning in the endpoint or through Dynamic Host Control Protocol (DHCP).

4. The endpoint also obtains a TFTP server address either through static provisioning or through DHCP options.

5. The endpoint then uses the TFTP server address to obtain its configuration files that, among other parameters, details the call processing server(s) or router(s) that the endpoint may associate with, the directory numbers that the endpoint must support, and so forth.

6. The endpoint registers with the call processing platform and is available for use.

Analog Gateways

An analog gateway typically is used to connect analog devices such as fax machines, modems, telecommunications device for the deaf (TDD)/teletypewriter (TTY), and analog phones, to the VoIP network so that the analog signal can be packetized and transmitted over the IP network. Analog gateways also provide physical connectivity to the PSTN and other traditional telephony equipment such as PBXs and key systems. Analog gateways include Cisco IOS router-based analog interface or service modules as well as fixed-port standalone gateways Generally analog gateways rely on Cisco Unified CM, Cisco Business Edition, Unified CM Express, and even Survivable Remote Site Telephony (SRST) for call control, supplementary services, and in some cases interface registration and configuration. Call control protocol support across Cisco analog gateways includes SIP, H.323, SCCP, and Media Gateway Control Protocol (MGCP).

Standalone Analog Gateways

Cisco standalone analog gateways, including the Cisco Analog Telephony Adapter (ATA) and Cisco VG200 Series Gateway, provide connectivity for analog devices such as fax machines, modems, TDD/TTY and analog phones, as well as one or more Ethernet ports for connecting to the IP network. Cisco standalone analog gateways support the FXS analog telephony interface port type only.

For more information on Cisco ATAs, refer to the data sheets and documentation at

http://www.cisco.com/en/US/products/hw/gatecont/ps514/index.html

For more information on Cisco VG200 Series Gateways, refer to the data sheets and documentation at

http://www.cisco.com/en/US/products/hw/gatecont/ps2250/prod_literature.html

Analog Interface Module

Cisco IOS router-based analog interface modules, including network modules (NMs) and voice interface cards (VICs), connect the PSTN and other legacy telephony equipment, including PBXs, analog telephones, fax machines, and key systems, to Cisco multiservice access routers such as the Cisco Integrated Services Router (ISR). Cisco IOS analog interface modules support a wide range of analog telephony interface port types, including FXS, FXO, T1/E1, E&M, and BRI.

Cisco IOS version support is critical for successful deployment of analog interface modules. For more information on Cisco IOS-based analog interface modules, including interface port type and Cisco IOS version support, refer to the data sheets and documentation listed at

http://www.cisco.com/en/US/products/ps10537/products_relevant_interfaces_and_modules.html#analogdigital

Analog Gateway Quality of Service Considerations

When configuring network-level quality of service (QoS), Cisco analog gateways such as the standalone Cisco VG200 Series and the Cisco IOS-based analog interface modules can be trusted and their packet markings honored. By default they mark their voice media and signaling packets with appropriate Layer 3 values (voice media as DSCP 46 or PHB EF; call signaling as DSCP 24 or PHB CS3), which match Cisco QoS recommendations for appropriate voice media and signaling marking, so as to ensure end-to-end voice quality on a converged network.

Desk Phones

The Cisco Unified IP Phone portfolio includes the following family of desk phones:

Cisco Unified IP Phone 7900 Series

The Cisco Unified IP Phone 7900 Series of endpoints consists of a wide range of models and feature sets. Models range from small single-line phones such as the Cisco Unified IP Phone 7911G to large eight-line phones such as the Cisco Unified IP Phone 7975G. In addition to the obvious and expected size differences between the various phone models, they vary in many other ways including: whether they have an LCD display and, if so, what size; whether they have a built-in speakerphone; what speed the network port(s) supports and whether there is a port for PC network attachment; how many phone lines they support; how many fixed feature keys they have and whether they are programmable; and so forth. In general, all phones in the Unified IP Phone 7900 Series provide the same basic set of enterprise IP telephony features such as call hold, call transfer, call forwarding, and so forth. However, some phone models provide features and functions well beyond the traditional enterprise IP telephony feature set, including support for IP-based phone services to enable presence, messaging, mobility, security, and other network-based applications and services.   The Cisco Unified IP 7900 Series supports both SCCP and SIP signaling protocols for registering and communicating with the Cisco call processing platforms.

In some cases additional line keys can be added to Unified IP Phone 7900 Series devices by physically attaching a key expansion module such the Cisco Unified IP Phone Expansion Module 7916. This gives administrative assistants and other users the ability to answer and/or determine the status of a number of lines beyond the current line capability of their desk phone. Some Unified IP Phone 7900 Series models are capable of supporting up to two Cisco Unified IP Phone Expansion Modules, but the use of an external power adaptor may be required.

Note   When two Expansion Modules are used with a single phone, the second module must be the same model as the first one.

The Cisco Unified IP Conference Station 7937G, with its 360-degree room coverage, provides conference room speaker-phone technology for use in conferencing environments. The Unified IP Conference Station provides an external speaker and built-in microphones. Optional extension microphones can be added to extend microphone coverage in larger rooms. These devices support SCCP signaling protocols for registering and communicating with the Cisco call processing platforms.

For more information about the Cisco Unified IP Phone 7900 Series, refer to the data sheets and documentation at

http://www.cisco.com/en/US/products/hw/phones/ps379/index.html

Cisco Unified IP Phone 6900 Series

The Cisco Unified IP Phone 6900 Series of endpoints includes a number of models. Models range from small, basic single-line phones such as the Cisco Unified IP Phone 6901 to larger, more advanced 12-line phones such as the Cisco Unified IP Phone 6961. In addition to the obvious and expected size differences between these various phone models, they vary in many other ways, including whether they have LCD displays, built-in speakerphone, PC port, and so forth. In general, all of the phones in the Unified IP Phone 6900 Series provide the same basic set of enterprise IP telephony features like such as hold, call transfer, call forwarding, and so forth. The Cisco Unified IP 6900 Series supports both SCCP and SIP signaling protocols for registering and communicating with the Cisco call processing platforms.

For more information about the Cisco Unified IP Phone 6900 Series, refer to the data sheets and product documentation at

> http://www.cisco.com/en/US/products/ps10326/index.html

Deployment Considerations for the Cisco Unified IP Phone 6900 Series

The Cisco Unified IP Phone 6900 Series provides call features such as Direct Transfer and Direct Transfer Across Lines as well as the Join and Join Across Lines features. These features can operate over calls spanning multiple lines, and their operation can be opaque to CTI applications that monitor only the primary line on the phone. Therefore, in order for these applications to work properly and maintain control over the phone functions, it might be necessary to disable the call features. These features may be disabled, in decreasing priority order, in either the specific phone configuration, the Common Device Profile configuration applicable to a group of phones that share the profile, or the enterprise-wide phone configuration.

Cisco Unified IP Phone 8900 and 9900 Series

The Cisco Unified IP Phone 8900 and 9900 Series of endpoints provide a wide range of form-factors and physical characteristics, including models with and without LCD displays and speaker phones and with varying numbers of line keys. Likewise, some models in this series provide support for Bluetooth and/or 802.11, such as the Cisco Unified IP Phone 9971, while others do not. In general, all of the phones in the Unified IP Phone 8900 and 9900 Series provide the same set of enterprise IP telephony features such as call hold, call transfer, call forwarding, and so forth. The Cisco Unified IP Phone 8900 Series supports either SIP only or both SCCP and SIP signaling protocols (dependent on phone model) for registering and communicating with Cisco call processing platforms. The Cisco Unified IP Phone 9900 Series models support only SIP signaling when registering and communicating with call control.

For more information about the Cisco Unified IP Phone 8900 Series, refer to the data sheets and product documentation at

> http://www.cisco.com/en/US/products/ps10451/index.html

For more information about the Cisco Unified IP Phone 9900 Series, refer to the data sheets and product documentation at

> http://www.cisco.com/en/US/products/ps10453/index.html

The Cisco Unified IP 8900 and 9900 Series devices may also be equipped with up to three (dependent on phone model) Cisco Unified IP Color Key Expansion Modules for administrative assistants and others user who need to answer and/or determine the status of a number of lines beyond the current line capability of their phone. These modules extend the capability of the Cisco Unified IP Phone 8900 and 9900 Series desk phones by adding an additional LCDs and buttons.

Some Cisco Unified IP Phone 8900 and 9900 Series models provide video capabilities either through a built-in camera for the Cisco Unified IP Phone 8900 Series or through the Cisco Unified Video Camera add-on accessory for the Cisco Unified IP Phone 9900 Series.

For more information about the Cisco Unified IP Phone 9900 and 8900 Series accessories, refer to the data sheets and product documentation at

http://www.cisco.com/en/US/products/ps10655/index.html

Deployment Considerations for the Cisco Unified IP Phone 8900 and 9900 Series

The Cisco Unified IP Phone 8900 and 9900 Series provide call capabilities that generate JTAPI events that must be handled by applications that monitor the phone through CTI. These call features allow the user to cancel an in-progress transfer or conference, or to perform a join or direct transfer of calls across the same or different lines. If the monitoring applications have not been upgraded to versions that properly handle these events, unexpected application behavior could result, including applications that no longer have their view of the phone or call state in synchronization with the phone itself. Therefore, by default, all applications are restricted from monitoring or controlling these phones.

For applications that have been upgraded to properly handle these new events, or for applications that have verified that they are not impacted by these events, the administrator may enable the role of **Standard CTI Allow Control of Phones supporting Connected Xfer and conf** in the application or end-user configuration associated with the application. Only after this role has been enabled can the application monitor or control these phones.

Cisco Unified SIP Phone 3900 Series

The Cisco Unified SIP Phone 3900 Series provides cost-effective, entry-level endpoints that support a single line and provide a basic set of enterprise IP telephony capabilities and basic supplementary features such as mute, call hold, and call transfer. The Cisco Unified SIP Phone 3900 Series has a two-line liquid crystal display (LCD) screen and a half-duplex or full-duplex speakerphone (depending on the model). The Cisco Unified SIP Phone 3900 Series supports the SIP signaling protocols for registering and communicating with Cisco call processing platforms. For more information about the Cisco Unified SIP Phone 3900 Series, refer to the data sheets and documentation at

http://www.cisco.com/en/US/products/ps7193/index.html

Deployment Considerations for Cisco Desk Phones

The following sections list important design considerations when deploying Cisco desk phones.

For the latest information on features and functions supported for Cisco desk phone models, refer to the product data sheets and documentation available at

http://www.cisco.com/go/ipphones

Firmware Upgrades

Most commonly, and by default, IP phones upgrade their images using TFTP, which is a UDP-based protocol, from TFTP servers integrated into one or more of the call processing platforms. With this arrangement, all the phones obtain their images directly from these TFTP servers. This method works well for a relatively small number of phones or if all of the phones are located in a single campus region that has a LAN environment with essentially unlimited bandwidth.

For larger deployments that use centralized call processing, upgrading phones in branch offices that are connected to the central data center by low-speed WAN links, can require a large amount of data traffic over the WAN. The same set of files will have to traverse the WAN multiple times, once for each phone. Transferring this amount of data is not only wasteful of the WAN bandwidth but can also take a long time as each data transfer competes with the others for bandwidth. Moreover, due to the nature of TFTP protocol, some phones might be forced to abort their upgrades and fall back to the existing version of the code.

Note   During the upgrade, the Cisco Unified IP Phones 9900 and 8900 Series stay in service, unlike the 7900 Series phones. The 9900 and 8900 Series phones download and store the new firmware in their memory while still maintaining their active status, and they reboot with the new firmware only after a successful download.

Two methods are available to alleviate problems created by the need to upgrade phones over the WAN. One method is to use a local TFTP server just for the upgrades. The administrator can place a TFTP server in branch offices (particularly in branches that have a larger number of phones, or whose WAN link is not speedy or robust), and can configure the phones in those offices to use that particular TFTP server just for new firmware. With this change, phones will retrieve new firmware locally. This upgrade method would require the administrator to pre-load the phone firmware on the TFTP server in the branch and manually configure the TFTP server address in the **load server** parameter in the affected phone configurations. Note that the branch router may be used as a TFTP server.

The second method to upgrade phones without using the WAN resources excessively is to use the Peer File Sharing (PFS) feature. In this feature, typically only one phone of each model in the branch downloads each new firmware file from the central TFTP server. Once the phone downloads the firmware file, it distributes that file to other phones in the branch. This method avoids the manual loading and configuration required for the load server method.

The PFS feature works when the same phone models in the same branch subnet arrange themselves in a hierarchy (chain) when asked to upgrade. They do this by exchanging messages between themselves and selecting the "root" phone that will actually perform the download. The root phone sends the firmware file to the second phone in the chain using a TCP connection; the second phone sends the firmware file to the third phone in the chain, and so on until all of the phones in the chain are upgraded. Note that the root phone may be different for different files that make up the complete phone firmware.

Power Over Ethernet

Deploying desk phones with inline power-capable switches enables these endpoints to derive power over the Ethernet network connection, thus eliminating the need for an external power supply as well as a wall power outlet. Inline power-capable switches with uninterruptible power supplies (UPS) ensures that power over Ethernet (PoE) capable IP desk phones continue to receive power during power failure situations. Provided the rest of the telephony network is available during these periods of power failure, then IP phones should be able to continue making and receiving calls.

Depending on the type of desk phone and the PoE standard supported by both the desk phone and the inline power-capable switch, in some cases the power budget of the inline powered switch port may be exceeded. This typically occurs when attaching key extension modules or other power consuming attachments such as USB cameras. In these situations, the phone may need to be powered using a wall outlet and external power supply or else the switch providing the power may need to be upgraded.

> **Note**   In addition to using the inline power from the access switch or local wall power, a Cisco Unified IP Phone can also be supplied power by a Cisco Unified IP Phone power injector. The Cisco Unified IP Phone power injector connects Cisco Unified IP Phones to Cisco switches that do not support inline power or to non-Cisco switches. The Cisco Unified IP Phone power injector is compatible with most Cisco Unified IP Phones. It has two 10/100/1000 Base-T Ethernet ports. One Ethernet port connects to the switch access port and the other connects to the Cisco Unified IP Phone.

Quality of Service

When configuring network-level quality of service (QoS), Cisco desk phones such as the Cisco Unified IP Phone 7900, 8900, and 9900 Series can be trusted and their packet markings honored. By default these endpoints mark their voice media and signaling packets with appropriate Layer 3 values (voice media as DSCP 46 or PHB EF; call signaling as DSCP 24 or PHB CS3), which match Cisco QoS recommendations for appropriate voice media and signaling marking, to ensure end-to-end voice quality on a converged network.   While many Cisco desk phones support the attachment of a desktop computer, Cisco desk phones are capable of separating the voice and data traffic, placing voice traffic onto the voice VLAN and data traffic from the desktop onto the data VLAN. This enables the network to extend trust to the phone but not to the PC port of the phone, as per Cisco recommendations.

> **Note**   While many Cisco desk phones support Link Layer Discovery Protocol for Media Endpoint Devices (LLDP-MED), they do so only for VLAN and Power over Ethernet negotiation. Cisco Unified IP Phones do not honor DSCP and CoS markings provided by LLDP-MED.

SRST and Unified CME as SRST

When deploying Cisco desk phones in branch locations separated from a centralized call processing platform by a low-speed or unreliable WAN link, it is important to consider local call processing redundancy. By leveraging Survivable Remote Site Telephony (SRST) or Cisco Unified Communications Manager Express (Unified CME) as SRST on a Cisco IOS router in each branch location, basic IP telephony services can be maintained for the desk phones when connectivity to the centralized call processing platform is lost. However, the set of available user-facing features is much smaller when a device is registered to SRST than when the phone is registered to Unified CM.

Software-Based Endpoints

A software-based endpoint is an application installed on a client desktop computer that registers and communicates with Cisco call processing platforms for voice and video services. In addition, these endpoint software client applications may provide collaboration features and services such as messaging, presence, directory access, and conferencing. Software-based endpoint desktop client applications include Cisco IP Communicator as well as Cisco WebEx Connect and Cisco Jabber, all of which use the Cisco Unified Client Services Framework (CSF).

Cisco IP Communicator

Cisco IP Communicator is a Microsoft Windows-based application that provides enterprise IP phone functionality to desktop computers. This application provides enterprise-class IP voice calling for remote users, telecommuters, and other mobile users. Cisco IP Communicator supports both SCCP and SIP signaling protocols for registering and communicating with Cisco call processing platforms. For more information about Cisco IP Communicator, refer to the data sheets and product documentation at

http://www.cisco.com/en/US/products/sw/voicesw/ps5475/index.html

Cisco Unified Client Services Framework

Cisco Unified Client Services Framework (CSF) is a software application that provides an underlying framework for integration of Unified Communications services, including audio, video, web collaboration, visual voicemail, and so forth, into a software-based desktop application. The Cisco Unified Client Services Framework allows desktop application users to access a variety of communication and collaboration services as provided by back-end collaboration application servers such as Cisco Unified Communications Manager (Unified CM), Cisco Unity Connection, Cisco WebEx, and Lightweight Directory Access Protocol (LDAP)-compliant directories. The Cisco Unified Client Services Framework is a device type in Cisco Unified CM that enables phone registration and communication for Cisco Unified Communications Integration for Cisco WebEx Connect and Cisco Jabber desktop applications, and it operates in either softphone mode or deskphone mode to control a Cisco Unified IP Phone.

Softphone Mode of Operation

For the Cisco WebEx Connect and Cisco Jabber desktop applications to operate in softphone mode, a Cisco Unified Client Services Framework device must be configured in Cisco Unified CM. The Cisco Unified Client Services Framework will then enable the Cisco Jabber and Cisco Unified Communications Integration for Cisco WebEx Connect applications to operate as a SIP-based single-line Cisco Unified IP Phone and will support the full registration and redundancy mechanisms of a Cisco Unified IP Phone.

Deskphone Control Mode of Operation

When the Cisco Jabber or Cisco WebEx Connect desktop application operates in deskphone control mode, the application uses CTI/JTAPI to control an associated Cisco Unified IP Phone. The Unified Client Services Framework uses the Cisco CallManager Cisco IP Phone Services (CCMCIP) service from Unified CM to provide a listing of valid Cisco Unified IP Phones to control.

The following design considerations should be taken into account when deploying Cisco Jabber and other desktop applications that use the Cisco Unified Client Services Framework:

- The administrator must determine how to install, deploy, and configure the Unified Client Services Framework desktop applications in their organization. Cisco recommends using a well-known installation package such as Altris to install the desktop application, and use Group Policies to configure the user registry settings for the required components such as TFTP, CTI Manager, CCMCIP, and LDAP server IP addresses and other pertinent information.

- The user ID and password configuration of the Cisco Unified Client Services Framework desktop application user must match the user ID and password of the user stored in the LDAP server to allow for seamless integration of the Unified Communications and back-end directory components.

- The directory number configuration on Cisco Unified CM and the telephoneNumber attribute in LDAP should be configured with a full E.164 number. A private enterprise dial plan can be used, but it might involve the need to use application dial rules and directory lookup rules.

- The deskphone mode for control of a Cisco Unified IP Phone uses CTI; therefore, when sizing a Unified CM deployment, you must also account for other applications that require CTI usage. For more information on CTI system sizing, refer to the section on .

For additional information about the Cisco WebEx Messenger service (formerly Cisco WebEx Connect service), Cisco WebEx Connect, Cisco Jabber, and Cisco Unified Client Services Framework, see the chapter on .

For more information about Cisco Jabber for Windows, refer to the data sheets and product documentation at

http://www.cisco.com/en/US/products/ps12511/index.html

For more information about the Cisco Jabber for Mac, refer to the data sheets and product documentation at

http://www.cisco.com/en/US/products/ps11764/index.html

For more information about the Cisco WebEx Messenger service, Cisco WebEx Connect, and Cisco Unified Communications Integration, refer to the product information at

http://www.cisco.com/en/US/products/ps10528/index.html

General Deployment Considerations for Software-Based Endpoints

The following sections list important design considerations for deploying software-based endpoints.

For the latest information on the features and functions supported for software-based endpoints, refer to the product data sheets and documentation available at

http://www.cisco.com/en/US/products/ps6789/Products_Sub_Category_Home.html

Quality of Service

While some software-based client applications do mark their traffic in accordance with QoS marking best practices, many applications do not. Further, even when the application does properly mark traffic, the underlying operating system or hardware may not honor the markings. Given the general unpredictability and unreliability of traffic marking coming from desktop computers, as a general rule these traffic markings should not be trusted. This means that all traffic flows must be re-marked by the network based on protocol and/or port numbers, with real-time traffic flows being marked based on best practices. This includes re-marking of voice-only call media with DSC 46 or PHB EF, video call media (including voice) with DSCP 34 or PHB AF41, and call signaling with DSCP 24 or PHB CS3. These markings along with a properly configured network infrastructure ensure priority treatment for voice-only call media and dedicated bandwidth for video call media and call signaling. In addition to re-marking of software-based endpoint traffic, Cisco recommends using network-based policing and rate limiting to ensure that the software-based endpoint does not consume too much network bandwidth. This can occur when the desktop computer generates too much data traffic or when the endpoint application misbehaves and generates more voice and/or video media and signaling traffic than would be expected for a typical call. In cases where third-party software is used to fully control desktop computer network traffic marking, administrators may decide to trust desktop computer marking, in which case re-marking of packets would not be required. Network-based policing and rate limiting is still recommended to protect the overall network in case of a misbehaving endpoint.

Inter-VLAN Routing

Because software-based endpoints run on a desktop computer usually deployed on a data VLAN, when software-based endpoints are deployed on networks with voice and data VLAN separation, inter-VLAN routing should be configured and allowed so that voice traffic from these endpoints on the data VLAN can reach endpoints on the voice VLAN.

SRST and Unified CME as SRST

When deploying Cisco software-based endpoint desktop applications in branch locations separated from a centralized call processing platform by a low-speed or unreliable WAN link, it is important to consider local call processing redundancy. By using SRST or Unified CME as SRST on a Cisco IOS router in each branch location, basic IP telephony services can be maintained for software-based endpoints when

connectivity to the centralized call processing platform is lost. However, the set of available user-facing features is much smaller when a desktop software-based endpoint is registered to SRST than when the application is registered to Unified CM.

Wireless Endpoints

Cisco wireless endpoints rely on an 802.11 wireless LAN (WLAN) infrastructure for network connectivity and to provide IP telephony functionality and features. This type of endpoint is ideal for mobile users that move around within a single enterprise location or between enterprise locations or environments where traditional wired phones are undesirable or problematic. Cisco offers the following voice and video over WLAN (VVoWLAN) IP phones:

- Cisco Unified Wireless IP Phones, including the Cisco Unified Wireless IP Phone 7925G and 7926G
- Cisco Unified IP Phone 9971

All are hardware-based phones with built-in radio antenna. The Cisco Unified Wireless IP Phones as well as the wirelessly attached Cisco Unified IP Phone 9971 enable 802.11b, 802.11g, or 802.11a connectivity to the network. The Cisco Unified Wireless IP Phones register and communicate with Cisco call processing platforms using SCCP signaling protocol, while the Cisco Unified IP Phone 9971 uses the SIP signaling protocol to register and communicate with Cisco call processing platforms.

For more information about the Cisco Unified Wireless IP Phones, refer to the data sheets and product documentation available at

http://www.cisco.com/en/US/products/hw/phones/ps379/index.html

For more information about the Cisco Unified IP Phone 9971, refer to the data sheets and product documentation available at

http://www.cisco.com/en/US/products/ps10453/index.html

General Deployment Considerations for Wireless Endpoints

The following sections list important design considerations for deploying wireless endpoints.

For the latest information on features and functions of wireless IP endpoints such as the Cisco Unified Wireless IP Phone 7925G, and for more information about deploying wireless IP endpoints, refer to the deployment guides at

http://www.cisco.com/en/US/products/hw/phones/ps379/products_implementation_design_guides _list.html

For the latest information on features and functions of the Cisco Unified IP Phone 9971 and for more information about deploying it wirelessly, refer to the deployment guide at

http://www.cisco.com/en/US/products/ps10453/products_implementation_design_guides_list.html

Network Radio Frequency Design and Site Survey

Before deploying wireless endpoints, you must ensure your WLAN radio frequency (RF) design minimizes same-channel interference while also providing sufficient radio signal levels and non-adjacent channel overlap so that acceptable voice and video quality can be maintained as the device moves from one location to another. In addition, you must perform a complete WLAN site survey to verify network RF design and to ensure that appropriate data rates and security mechanisms are in place. Your site survey should take into consideration which types of antennas will provide the best coverage,

as well as where sources of RF interference might exist. Even when using third-party site survey tools, Cisco highly recommends that you verify the site survey using the wireless endpoint device itself because each endpoint or client radio can behave differently depending on antenna sensitivity and survey application limitations. Cisco Unified Wireless IP Phones and the Cisco Unified IP Phone 9971 provide a built-in site survey tool that enables easy verification of the surrounding WLAN network channels and signal strength. Cisco recommends relying on the 5 GHz WLAN band (802.11a/n) whenever possible for connecting wireless endpoints capable of generating voice and video traffic. 5 GHz WLANs provide better throughput and less interference for voice and video calls. Refer to the section on Wireless LAN Infrastructure, page 3-54, for more information about wireless network design.

Security: Authentication and Encryption

When deploying wireless endpoints, it is important to consider the security mechanisms used to control access to the network and to protect the network traffic. Cisco wireless endpoints support a wide range of authentication and encryption protocols including WPA, WPA2, EAP-FAST, PEAP, and so forth. Choose an authentication and encryption method that is supported by the WLAN infrastructure and the endpoint devices you deploy, and one that aligns with IT security policies. In addition, ensure that the authentication and encryption method chosen supports a fast rekeying method such as Cisco Centralized Key Management (CCKM) so that active voice and video calls can be maintained when the device is roaming from one location in the network to another.

Note In dual-band WLANs (those with both 2.4 GHz and 5 GHz bands), it is possible to roam between 802.11b/g and 802.11a with the same SSID, provided the client is capable of supporting both bands. However, with some devices this can cause gaps in the voice or video path. In order to avoid these gaps, use only one band for voice and video communications.

Wireless Call Capacity

When deploying wireless devices and enabling wireless device roaming within the enterprise WLAN, it is also important to consider the device connectivity and call capacity of the WLAN infrastructure. Oversubscription of the WLAN infrastructure in terms of number of devices or number of active calls will result in dropped wireless connections, poor voice and video quality, and delayed or failed call setup. The chances of oversubscribing a deployment of voice and video over WLAN are greatly minimized by deploying sufficient numbers of WLAN access points (APs) to handle required call capacities. AP call capacities are based on the number of simultaneous bidirectional streams that can be supported in a single channel cell area. The general rule for VVoWLAN call capacities is as follows:

- Maximum of 27 simultaneous VoWLAN bidirectional streams per 802.11g/n (2.4 GHz) channel cell with Bluetooth disabled or per 802.11a/n (5 GHz) channel and 24 Mbps or higher data rates enabled.

- Maximum of 8 simultaneous VVoWLAN bidirectional streams per 802.11 g/n (2.4 GHz) channel cell with Bluetooth disabled or per 802.11 a/n (5 GHz) channel cell assuming a video resolution of 720p (high-definition) and video bit rate of up to 1 Mbps.

These call capacity values are highly dependent upon the RF environment, the wireless handset features, and underlying WLAN system features. Actual capacities for a particular deployment could be less.

Note A single call between two wireless endpoints associated to the same AP is considered to be two simultaneous bidirectional streams.

The above capacities are based on voice activity detection (VAD) being disabled and a packetization sample size of 20 milliseconds (ms). VAD is a mechanism for conserving bandwidth by not sending RTP packets while no speech is occurring during the call. However, enabling or disabling VAD, also referred to as Silence Suppression, is sometimes a global configuration depending on the Cisco call control platforms. Thus, if VAD is enabled for wirelessly attached Cisco Unified IP Phones, then it may be enabled for all devices in the deployment. Cisco recommends leaving VAD (Silence Suppression) disabled to provide better overall voice quality.

At a sampling rate of 20 ms, a voice call will generate 50 packets per second (pps) in either direction. Cisco recommends setting the sample rate to 20 ms for almost all cases. By using a larger sample size (for example, 30 or 40 ms), you can increase the number of simultaneous calls per AP, but a larger end-to-end delay will result. In addition, the percentage of acceptable voice packet loss within a wireless environment decreases dramatically with a larger sample size because more of the conversation is missing when a packet is lost. For more information about voice sampling size, see the section on Bandwidth Provisioning, page 3-45.

Bluetooth Support

The Cisco Unified Wireless IP Phones 7925G, 7925G-EX, and 7926G, and the Cisco Unified IP Phone 9971 are Bluetooth-enabled devices. The Bluetooth radio or module within these wireless Cisco Unified IP Phones provides the ability to support Bluetooth headsets with the phones. Because Bluetooth devices use the same 2.4 GHz radio band as 802.11b/g devices, it is possible that Bluetooth and 802.11b/g devices can interfere with each other, thus resulting in connectivity issues.

While the Bluetooth and 802.11 WLAN radios co-exist in the Cisco Unified Wireless IP Phones and Cisco Unified IP Phone 9971, greatly reducing and avoiding radio interference between the Bluetooth and 802.11b/g radio, the Bluetooth radio in these wirelessly attached phones can cause interference for other 802.11b/g devices deployed in close proximity. Due to the potential for interference and disruption of 802.11b/g WLAN voice and video devices (which can result in poor voice and video quality, deregistration, and/or call setup delays), Cisco recommends deploying all WLAN voice and video devices on 802.11a, which uses the 5 GHz radio band. By deploying wireless phones on the 802.11a radio band, you can avoid interference caused by Bluetooth devices.

Note   Using Bluetooth wireless headsets with the battery-powered Cisco Unified Wireless IP Phones will increase battery power consumption on your phone and will result in reduced battery life.

Quality of Service

When configuring network-level quality of service (QoS), Cisco wireless endpoints (including Cisco Unified Wireless IP Phones and the Cisco Unified IP Phone 9971) can be trusted and their packet markings honored. By default these endpoints mark the recommended and appropriate Layer 3 values for voice media and signaling (voice media as DSCP 46 or PHB EF; voice signaling as DSCP 24 or PHB CS3). Likewise, these devices mark appropriately at Layer 2 (voice media WMM User Priority (UP) of 6; call signaling WMM UP of 4). With these packet markings, end-to-end voice quality on the converged network will be acceptable.

SRST and Unified CME as SRST

When deploying wireless endpoints in branch locations separated from a centralized call processing platform by a low-speed or unreliable WAN link, it is important to consider local call processing redundancy. By deploying SRST or Unified CME as SRST on a Cisco IOS router in each branch location, basic IP telephony services can be maintained for wireless endpoints when connectivity to the centralized call processing platform is lost. However, the set of available user-facing features is much smaller when a wireless endpoint is registered to SRST than when it is registered to Unified CM.

Device Mobility

When wireless endpoints move between locations in a multi-site centralized call processing deployment, the Cisco Unified CM Device Mobility feature may be used to dynamically update the location of the device based on the IP address the device uses to register to Unified CM. This prevents issues with call routing, PSTN egress, and codec and media resource selection typically encountered when devices move between locations. For more information on Device Mobility, see the section on Device Mobility, page 25-14.

Mobile Endpoints

Cisco mobile endpoint devices and mobile endpoint client applications register and communicate with Unified CM for voice and video calling services. These devices and clients also enable additional features and services such as enterprise messaging, presence, and corporate directory integration by communicating with other back-end systems such as Cisco Unity Connection, Cisco IM and Presence, and LDAP directories. Cisco offers the following mobile endpoint devices and clients:

- Cisco Cius, page 18-15
- Cisco Jabber for Android and Apple iOS, page 18-16
- Cisco Jabber IM, page 18-16, for Android, BlackBerry and Apple iOS devices

Cisco Cius

Cisco Cius is an Android-based enterprise tablet that provides native voice and video calling over a WLAN or mobile data network when registered to Unified CM as a SIP device. In addition to enabling enterprise voice and video calling, Cius has native applications for XMPP-based enterprise instant messaging (IM) and presence, corporate directory access, and visual voicemail. For more information about Cisco Cius, refer to the data sheets and product documentation available at

http://www.cisco.com/en/US/products/ps11156/index.html

For more information about deploying Cisco Cius wirelessly, refer to the *Cisco Cius Deployment Guide* available at

http://www.cisco.com/en/US/products/ps11156/products_implementation_design_guides_list.html

Cisco Jabber for Android and Apple iOS

The Cisco Jabber mobile clients for Android and Apple iOS devices including the iPhone and iPad enable smartphones and tablets to make and receive enterprise calls using voice or voice and video over IP. The Cisco Jabber mobile client application running on the Android or Apple iOS device registers and communicates with Unified CM using the SIP signaling protocol. In some cases the device may register and communicate with the Cisco Video Communications System (VCS) instead. The Cisco Jabber mobile client also enables additional features such as corporate directory access, enterprise visual voicemail, and in some cases enterprise instant messaging and presence.

For more information about Cisco Jabber for Android, refer to the data sheet and product documentation at

http://www.cisco.com/en/US/products/ps11678/index.html

For more information about Cisco Jabber for iPhone, refer to the data sheet and product documentation at

http://www.cisco.com/en/US/products/ps11596/index.html

For more information about Cisco Jabber for iPad, refer to the data sheet and product documentation at

http://www.cisco.com/en/US/products/ps12430/index.html

Cisco Jabber IM

The Cisco Jabber IM client runs on specific BlackBerry and Android smartphones and on various Apple iOS devices including the iPhone, and it communicates via XMPP with on-premises Cisco IM and Presence services or off-premises cloud-based Cisco WebEx Messenger service.

Note   Cisco Jabber for iPad provides native XMPP-based IM and presence capabilities.

For more information about Cisco Jabber IM for Android, refer to the data sheet and product documentation at

http://www.cisco.com/en/US/products/ps11678/index.html

For more information about Cisco Jabber IM for BlackBerry, refer to the data sheet and product documentation at

http://www.cisco.com/en/US/products/ps11763/index.html

For more information about Cisco Jabber IM for iPhone, refer to the data sheet and product documentation at

http://www.cisco.com/en/US/products/ps11596/index.html

Deployment Considerations for Mobile Endpoints and Clients

The following sections list important design considerations for deploying mobile endpoints and clients.

For additional design and deployment information about Cisco Cius and Cisco Jabber mobile clients, refer to the section on Cisco Mobile Clients and Devices, page 25-60.

Cisco Jabber Client Application Interaction

Although Cisco Jabber for Android and iPhone smartphones require separate applications for enterprise voice and enterprise XMPP-based IM and presence services, both mobile client applications can be installed on the same device (co-resident). While they are separate client applications, they are aware of each other and will cross-launch each other as required. For example, an IM conversation on the Cisco Jabber IM application can be escalated to a voice call, causing the Cisco Jabber client application to become active in order to handle the call.

WLAN Design

Because Cisco Jabber mobile clients and Cisco Cius are often attached to a WLAN, all of the previously mentioned WLAN deployment considerations apply to mobile clients and devices, including WLAN RF design and verification by site survey. In particular, Cisco recommends relying on the 5 GHz WLAN band (802.11a/n) whenever possible for connecting wireless endpoints capable of generating voice and video traffic. 5 GHz WLANs provide better throughput and less interference for voice and video calls. If the 2.4 GHz band is used for mobile clients and devices, Bluetooth should be avoided.   Likewise, the WLAN channel cell voice-only and video call capacity numbers covered in the section on Wireless Call Capacity, page 18-13, should be considered when deploying these clients and devices.

Secure Remote Enterprise Attachment

If appropriately deployed, Cisco mobile endpoints and clients can also connect to the enterprise from remote locations by using public or private 802.11 Wi-Fi hot spots or over the mobile data network. In these scenarios the Cisco AnyConnect mobile VPN client can be used to connect the device or client to the enterprise with a secure SSL tunnel.

Quality of Service

Cisco mobile client applications and devices generally mark Layer 3 QoS packet values in accordance with Cisco collaboration QoS marking recommendations. This includes marking voice-only call media traffic with DSCP 46 or PHB EF, video call media (including voice) traffic with DSCP 34 or PHB AF41, and call signaling traffic with DSCP 24 or PHB CS3. Despite appropriate mobile client and device application Layer 3 packet marking, Layer 2 802.11 WLAN packet marking (User Priority, or UP) presents further challenges. Some devices such as Cisco Cius appropriately mark wireless Layer 2 802.11 User Priority (UP) values (voice-only call media UP 6, video call media UP 5, and call signaling UP 3). However, because Cisco mobile clients run on a variety of mobile devices, Layer 2 wireless QoS marking is inconsistent and therefore cannot be relied upon to provide appropriate treatment to traffic on the WLAN. In deployments with Cisco Unified Wireless LAN Controllers, enabling wireless SIP call admission control (CAC) might provide some relief for incorrect or nonexistent Layer 2 WLAN marking. SIP CAC utilizes media session snooping and ensures that downstream voice and video frames are prioritized and/or treated correctly. Even assuming appropriate mobile client application Layer 3 or even Layer 2 packet marking, mobile devices present many of the same challenges as desktop computers in terms of generating many different types of traffic, including both data and real-time traffic. Given

this, mobile devices generally fall into the untrusted category of collaboration endpoints. For deployments where mobile client devices are not considered trusted endpoints, packet re-marking based on traffic type and port numbers is required to ensure that network priority queuing and dedicated bandwidth are applied to appropriate traffic. In addition to re-marking the mobile device traffic, Cisco recommends using network-based policing and rate limiting to ensure that the mobile client devices do not consume too much network bandwidth.

Note   Mobile clients and devices may attach remotely to the enterprise using Cisco AnyConnect client over the mobile data network or public or private Wi-Fi hot spots. Because these connections traverse the Internet, there is no end-to-end QoS on the IP path and therefore all traffic is treated as best-effort. Voice and video quality cannot be guaranteed over these types of connections.

SRST and Unified CME as SRST

When deploying mobile endpoints and clients such as Cisco Cius or Cisco Jabber for iPhone in branch locations separated from a centralized call processing platform by a low-speed or unreliable WAN link, it is important to consider local call processing redundancy. By deploying SRST or Unified CME as SRST on a Cisco IOS router in each branch location, basic IP telephony services can be maintained for mobile endpoints when connectivity to the centralized call processing platform is lost. However, the set of available user-facing features is much smaller when a mobile device is registered to SRST than when the device is registered to Unified CM. Not all Cisco Jabber mobile clients support SRST; however, because most Cisco Jabber mobile clients run on smartphones with cellular voice radios, users may still be able to make call using the mobile provider network.

Video Endpoints

Cisco video endpoints provide IP video telephony features and functions similar to IP voice telephony, enabling users to make point to point and point to multi-point video calls. Cisco offers the following desktop video-capable endpoints:

- Cisco Unified IP Phone 9900 Series with the optional USB camera attachment.
- Cisco Unified IP Phones 8941 and 8945 with built-in camera
- Cisco TelePresence System EX Series
- Cisco Unified Client Services Framework (CSF) software-based desktop clients such as Cisco Jabber for Windows

For additional information on video telephony, see the chapter on .

Cisco Unified IP Phone 8900 and 9900 Series

The Cisco Unified IP Phones 8900 and 9900 Series are capable of transmitting video and receiving and displaying video natively on their screens. With the built-in camera (8941 and 8945) or optional, specially designed USB camera attachment (9900 Series), they can also transmit video. The screens on these phones can display a variety of video resolutions and frame rates. The video capabilities of these phones can be enabled and disabled or tuned as desired from the Cisco call control platform configuration pages. These devices register and communicate with Unified CM using either SCCP or

SIP signaling protocols in the case of the 8900 Series video-capable phones or via SIP only in the case of the 9900 Series phones. For more information about the Cisco Unified IP Phone 8900 and 9900 Series video capabilities, refer to the data sheets and product documentation available at

http://www.cisco.com/en/US/products/ps10453/index.html

Cisco TelePresence System EX Series

The Cisco TelePresence System EX Series video endpoints provide personal telepresence or video calling for the desktop and include the Cisco TelePresence System EX60 and EX90. The EX Series video endpoint models vary in screen size as well as viewing angle and video resolution, but both deliver near equivalent telephony features. The Cisco TelePresence System EX Series video endpoints register and communicate with Unified CM by means of the SIP signaling protocol.

Note   The EX Series video endpoints do not support registration redundancy with Unified CM. If the primary Unified CM node to which the EX Series endpoint is registered becomes unreachable, the endpoint will not fail-over its registration to a secondary Unified CM node.

For more information about the Cisco TelePresence System EX Series video endpoints, refer to the data sheets and product documentation available at

http://www.cisco.com/en/US/products/ps11327/index.html

Cisco Unified Client Services Framework (CSF) Video

Some Cisco Unified CSF software-based desktop clients such as Cisco Jabber for Windows are able to send and receive video when running on a desktop computer with an integrated or USB-attached camera. These video-capable software-based endpoints register and communicate with Unified CM call control and operate as a SIP single-line voice and video enabled phone. These endpoints support the primary and backup registration redundancy mechanisms as provided by Unified CM. The Cisco Unified CSF software-based endpoint processes video on the computer where it is installed. The quality of the decoding and encoding depends on the availability of CPU and memory resources on that computer.

For more information about the video capabilities of Cisco Jabber for Windows, refer to the data sheet and product documentation available at

http://www.cisco.com/en/US/products/ps12511/index.html

General Deployment Considerations for Video Endpoints

The following sections list important design considerations for deploying video endpoints.

Quality of Service

When configuring network-level quality of service (QoS), Cisco video endpoints (including Cisco Unified IP Phone 8900 and 9900 Series and Cisco TelePresence System EX Series devices) generally mark traffic at Layer 3 according to Cisco general QoS guidelines related to voice and video packet marking (video media as DSCP 34 or PHB AF41; call signaling as DSCP 24 or PHB CS3) and therefore these devices can be trusted. Even when trusting the endpoint marking, Cisco recommends using network-based policing and rate limiting to ensure that the video endpoint does not consume too much

network bandwidth. Software-based video-capable endpoints do present challenges when they do not or cannot mark traffic appropriately. In these situations, typical guidance is to re-mark media and signaling traffic within the network from best-effort to appropriate and recommended values (voice media as DSCP 46 or PHB EF; video media as DSCP 34 or PHB AF41; call signaling as DSCP 24 or PHB CS3) based on protocols and/or port numbers. However, in some cases software-based applications may send voice and video media on the same ports. This means that it would not be possible to provide differentiated packet marking for the voice stream of an audio-only call from the voice stream of a video call.

Note   While some Cisco video-capable endpoints support Link Layer Discovery Protocol for Media Endpoint Devices (LLDP-MED), they do so only for VLAN and Power over Ethernet negotiation. Cisco video endpoints do not honor DSCP and CoS markings provided by LLDP-MED.

Inter-VLAN Routing

When deploying video endpoints on networks with voice and data VLAN separation, it is important to consider software-based video-capable endpoints as well as hardware-based video endpoints that need to access resources. Because software-based endpoints running on a desktop computer are primarily attached to the data VLAN, inter-VLAN routing should be configured and allowed so that voice traffic from these endpoints on the data VLAN can reach endpoints on the voice VLAN. Likewise, if hardware-based video endpoints such as the Cisco TelePresence System EX60 need access to network resources such as directory or management services deployed on the data VLAN, inter-VLAN routing must be allowed.

SRST and Unified CME as SRST

When deploying video endpoints in branch locations separated from a centralized call processing platform by a low-speed or unreliable WAN link, it is important to consider local call processing redundancy. By deploying SRST or Unified CME as SRST on a Cisco IOS router in each branch location, basic IP telephony services can be maintained for most video endpoints when connectivity to the centralized call processing platform is lost. However, the set of available user-facing features is much smaller when a video endpoint is registered to SRST than when the application is registered to Unified CM. Specifically, video endpoint devices registered to SRST will be capable of making and receiving only voice calls (audio-only). SRST is not supported with the Cisco TelePresence System EX Series video endpoints.

Cisco Virtualization Experience Clients

The Cisco Virtualization Experience Clients (VXC) are the integral collaboration components of the Cisco Virtualization Experience Infrastructure (VXI). The VXCs provide user access to data applications and services across various network environments, as well as user preferences and device form factors for a fully integrated voice, video, and virtual desktop environment.

Cisco offers the following VXC endpoints:

- Cisco Virtualization Experience Client 2000 Series
- Cisco Virtualization Experience Client 4000 Series
- Cisco Virtualization Experience Client 6000 Series

For additional information about Cisco Virtualization Experience Clients, see the section on Cisco Virtualization Experience Client Architecture, page 24-29.

Cisco Virtualization Experience Client 2000 Series

The Cisco Virtualization Experience Client 2000 Series provides simple devices with a small firmware footprint (also known as a "zero client") for virtual desktop access to a Citrix or VMware environment. The VXC 2112 and 2212 are specifically designed to work in a Citrix environment, while the VXC 2111 and 2211 work in a VMware environment. The VXC 2111 and 2112 integrated form factor devices are designed to replace the footstand on the Cisco Unified IP Phones 8961, 9951, or 9971 to provide a fully integrated voice, video, and virtual desktop environment. The VXC 2211 and 2212 standalone form factor devices are designed to operate as a simple virtual desktop environment, or they can be paired with any third-generation Cisco Unified IP Phone to provide a full voice, video, and virtual desktop environment.

For more information about the Cisco Virtualization Experience Client 2000 Series, refer to the data sheets and product documentation available at

http://www.cisco.com/en/US/products/ps11499/index.html

Cisco Virtualization Experience Client 4000 Series

The Cisco Virtualization Experience Client (VXC) 4000 Series software appliance, when used in conjunction with a repurposed PC, allows for secure access to a remote hosted virtual desktop while supporting rich media locally. Windows 7 and Windows XP are the only operating systems supported for the repurposed PC. The hosted virtual desktop is supported, using Citrix XenDesktop or VMware View, through a locally installed thick-client Citrix Receiver and VMware View Client, respectively.

For more information about the Cisco Virtualization Experience Client 4000 Series, refer to the data sheets and product documentation available at

http://www.cisco.com/en/US/products/ps11498/index.html

Cisco Virtualization Experience Client 6000 Series

The Cisco Virtualization Experience Client (VXC) 6000 Series thin client provides a fully integrated voice, video, and virtual desktop solution in a single device. The VXC 6215 is a Linux platform that can be used in Virtual Desktop Infrastructure (VDI) mode to support Citrix XenDesktop or VMware View, or it can be enabled for Unified Communications with a software appliance add-on that allows for full voice, video, and virtual desktop support of Citrix XenDesktop.

For more information about the Cisco Virtualization Experience Client 6000 Series, refer to the data sheets and product documentation available at

http://www.cisco.com/en/US/products/ps11976/index.html

Third-Party IP Phones

Some third-party IP phones and devices may be integrated with Cisco call control to provide basic IP telephony functionality, as described in this section.

Third-Party SIP IP Phones

Third-party phones have specific local features that are independent of the call control signaling protocol, such as features access buttons (fixed or variable). Basic SIP RFC support allows for certain desktop features to be the same as on Cisco Unified IP Phones and also allows for interoperability of certain features. However, these third-party SIP phones do not provide the full feature functionality of Cisco Unified IP Phones.

Cisco works with key third-party vendors who are part of the Cisco Developer Network and who are developing solutions that leverage Cisco Unified CM and Unified CME SIP capabilities. For example, Cisco worked with Research In Motion to integrate their BlackBerry Mobile Voice System (MVS) solution with Cisco call control platforms in order to enable Cisco Unified Communications and enterprise calling natively on BlackBerry smartphones. Another third-party vendor is Tenacity Operating, which provides a software-based endpoint called accessaphone ipTTY that enables terminal teletype (TTY) or text-based communications for IP telephony. This software-based endpoint can register and communicate with Cisco Unified CM as a third-party SIP phone.

For more information on Cisco's line-side SIP interoperability, refer to the Cisco Unified Communications Manager programming guides at

http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_programming_reference_guides_list.html

For more information on the Cisco Developer Network and third-party development partners, refer to the information available on the Cisco Developer Community at

http://developer.cisco.com

High Availability for Unified Communications Endpoints

To stay in service even during failure of the Unified CM subscriber or other servers, Cisco Unified Communications endpoints are capable of being configured with multiple servers. For example, either through direct configuration or through DHCP during the boot-up phase, the endpoints can accept and process more than one TFTP server address. In case the primary TFTP server is down when the endpoint boots up, the endpoint can get its configuration files from the secondary TFTP server.

Each of the endpoints is also associated with a device pool. The device pool contains a Unified CM Group that has one or more Unified CM subscribers. A list of these subscribers is sent to the endpoints in their configuration files. The endpoints attempt to register with the first (the primary) subscriber in the list. If that Unified CM subscriber is unavailable, the endpoint attempts to register with the second subscriber in the list (the secondary), and so on. Once registered to a subscriber, an endpoint can fail-over to another subscriber in the priority list in the Unified CM Group if the current subscriber fails. When a higher-priority subscriber comes back up, the endpoint will re-register to it.

Note The EX Series video endpoints do not support registration redundancy as described above. If the primary Unified CM node is unreachable, the endpoint will not fail-over its registration to a secondary Unified CM node.

To protect against network failure for endpoints located across a WAN from the Unified CM cluster, a locally available Cisco Integrated Services Router (ISR) equipped with SRST or Unified CME acting as SRST may also be configured in the list of servers with which the endpoint may register. In case of a WAN failure, the endpoints register to the SRST router and provide uninterrupted telephony services (although the set of features they support in SRST mode might be smaller). Note that some endpoints might not support SRST.

Endpoints should be distributed uniformly across servers in the cluster to avoid overloading of any single server. For more information on redundancy methods between cluster subscribers, see the chapter on Call Processing, page 8-1.

Capacity Planning for Unified Communications Endpoints

Cisco call control platforms support the following high-level endpoint capacities:

- A Cisco Unified CM cluster supports a maximum of 40,000 SCCP or SIP endpoints.
- Cisco Business Edition supports a maximum of between 400 and 1,200 SCCP or SIP endpoints, depending on the version.
- Cisco Unified CM Express supports a maximum of 450 SCCP or SIP endpoints.

The above numbers are nominal maximum capacities. The maximum number of endpoints that the call control platform will actually support depends on all of the other functions that the platform is performing, the busy hour call attempts (BHCA) of the users, and so forth, and the actual capacity could be less than the nominal maximum capacity.

In addition to call control platform capacity, network capacity must be considered when it comes to 802.11 wireless attached devices such as the Cisco Unified Wireless IP Phone 7925G or an Android smartphone running Cisco Jabber for Android. See Wireless Call Capacity, page 18-13, for voice and video call capacities per 802.11 channel cell.

For more information on endpoint capacity with Cisco call control, including platform-specific endpoint capacities per node, see the chapter on Unified Communications Design and Deployment Sizing Considerations, page 29-1.

Design Considerations for Unified Communications Endpoints

The following list summarizes high-level design recommendations for deploying Cisco Unified Communications endpoints:

- Analog gateways are available both as standalone devices and as integrated interface modules on Cisco IOS multiservice routers, and both types can be used within the same deployment. Select the analog gateway or gateways that meet analog port density requirements across company locations. Ensure that appropriate port capacity is provided for all locations in order to accommodate the required analog devices.
- Enable the role of St**andard CTI Allow Control of Phones supporting Connected Xfer and conf** for the end-user configuration associated with the device in order to enable CTI monitoring and control of Cisco Unified IP Phone 8900 and 9900 Series endpoints. Only after this role has been enabled can CTI applications monitor or control these phones.

- To minimize endpoint firmware upgrade times over the WAN to remote branches, consider deploying a local TFTP server at the remote location and point endpoints located in that branch to this local TFTP server using the **load server** parameter. Alternatively, consider the use of the Peer File Sharing (PFS) feature when all or most of the devices at a particular remote location are the same phone model.

- Cisco Unified IP desk phones can be powered by power over Ethernet (PoE) when plugged into inline power-capable switches or when deployed with an inline power injector. Consider the use of inline power to reduce downtime and eliminate the need for an external power supply and wall power outlet.

- When deploying Cisco endpoints in branch locations separated from a centralized call processing platform by a low-speed or unreliable WAN link, it is important to consider local call processing redundancy. By using SRST or Unified CME as SRST on a Cisco IOS router in each branch location, basic IP telephony services can be maintained for the desk phones when connectivity to the centralized call processing platform is lost. However, the set of available user-facing features is much smaller when a device is registered to SRST than when the phone is registered to Unified CM.

- For deployments with network voice and data VLAN separation, ensure that inter-VLAN routing has been configured and allowed so that Cisco software-based endpoints that run on desktop computers usually connected to data VLANs can communicate with endpoints on the voice VLAN. This is also important for endpoints on the voice VLAN that may be dependent on data VLAN-based resources that provide services such as directory and management.

- A WLAN site survey must be conducted to ensure appropriate RF design and to identify and eliminate sources of interference prior to deploying wireless and mobile endpoints capable of generating real-time traffic on the wireless network. This is necessary to ensure acceptable voice and video quality for calls traversing the WLAN.

- Select a WLAN authentication and encryption method that not only adheres to company security policies but also enables fast rekeying or authentication so that audio and video calls are not interrupted when wireless endpoints move from one location to another.

- Cisco recommends relying on the 5 GHz WLAN band (802.11a/n) whenever possible for connecting wireless endpoints and mobile client devices capable of generating voice and/or video traffic. 5 GHz WLANs provide better throughput and less interference for voice and video calls. If the 2.4 GHz band is used for connecting wireless client devices and endpoints, Bluetooth should be avoided.

- Provide appropriate network and call control capacity to support the number of endpoints deployed. First, consider the endpoint registration and configuration capacities per call control platform (maximums between 40,000 endpoints per Unified CM cluster and 400 endpoints per Cisco Business Edition). Next, consider call capacities per wireless channel cell for wireless attached endpoints, and the maximum of 27 bidirectional voice-only streams or maximum of 8 simultaneous voice and video streams or calls per WLAN channel cell.

- Cisco TelePresence System EX Series video endpoints do not currently support registration redundancy with Unified CM. If the primary Unified CM node to which the EX series endpoint is registered becomes unreachable, the endpoint will not fail-over its registration to a secondary Unified CM node.