Data Sheet

Cisco Unified Communications Manager Session Management Edition Version 8.0

Transform and consolidate your existing telephony and data infrastructure into a scalable, rich-media communications system for your business.

Product Overview

Cisco® Unified Communications Manager Session Management Edition (SME) is a key enterprise transformation solution that builds upon Cisco Unified Communications Manager's heritage of multi-protocol support (SIP, SCCP, H.323, Q.SIG, MGCP) and its native SIP back-to-back user agent (B2BUA) capabilities. It enables and coordinates progressive, rich-media communications for your users through the "point" introduction of applications and features.

For example, employees could start a chat session after noting one another's real-time presence on the Cisco Quad™ enterprise collaboration portal. They might then escalate this interaction to a phone call using Cisco Unified Communications Manager, and eventually engage other interested parties in an audio-visual conference in a Cisco Unified MeetingPlace®, Cisco WebEx®, or a Cisco TelePresence® session. This is the progression of a conversation through successively richer communication channels that today's enterprise users have come to expect.

You can centrally deploy any or all of the Cisco Quad™ enterprise collaboration platform, Cisco Unified MeetingPlace, and Cisco TelePresence applications with Session Management Edition. In turn, this edition uses its session-management and multi-protocol interworking capabilities to make each of these applications available to all your users, whether they are located at your primary site, a branch office, or are telecommuting from home or from on the road.

You can make these applications available to users of your legacy PBX infrastructure without the need for a "rip and replace" strategy. Upgrades to or replacements of existing equipment (for example, improved desk sets or new video endpoints) can be made on your schedule, as your budget and resources allow. In the interim, all phones and other endpoints - whether they be based on SIP, SCCP, H.323 or legacy TDM or analog standards - can participate in the use of these centrally deployed features according to the native capabilities of the device and their call processing agents. Other vendors' offerings simply do not have the flexibility or multi-protocol interoperability to offer this.

Session Management Edition is neither a mere SIP overlay nor is it just a SIP proxy network. It is a complete SIP call-control server capable of interworking multiple IP and legacy digital protocols. When you integrate Session Management Edition into your enterprise network, you will leverage your existing infrastructure to take advantage of the rapid roll out of new applications, you will save money by centrally deploying and managing these applications, and you will establish an architecture that will be compatible with future versions and help ensure your ability to adopt some of the new paradigms of SIP:

- Public-switched-telephone-network (PSTN) access cost savings from the replacement of over provisioned, fixed-capacity TDM trunks with fewer, more flexible, and lower priced SIP trunks to your service providers
- Operational cost savings through the consolidation of dial plans and routing control, which may be further optimized through the deployment of the Cisco Service Advertisement Framework (SAF)
- Enablement of inter-enterprise video, high-definition audio, and other applications with the tandem deployment of the Cisco Intercompany Media Engine (IME)
- Increased mobility for users, who can more easily take advantage of single number reach (SNR), Dial Via Office (DVO), and for users with Wi-Fi capable handsets - greater roaming capabilities
- Enablement of centralized policy control for applications such as least-cost routing (LCR); black list, white list, and priority-treatment call lists; call recording; and call chaperone
- Enhanced calling quality and user satisfaction through fewer call-blocking constraints, media shuffling, and improved codec selection and adaptation
- Enhanced reachability with SIP uniform-resource-identifier (URI) dialing

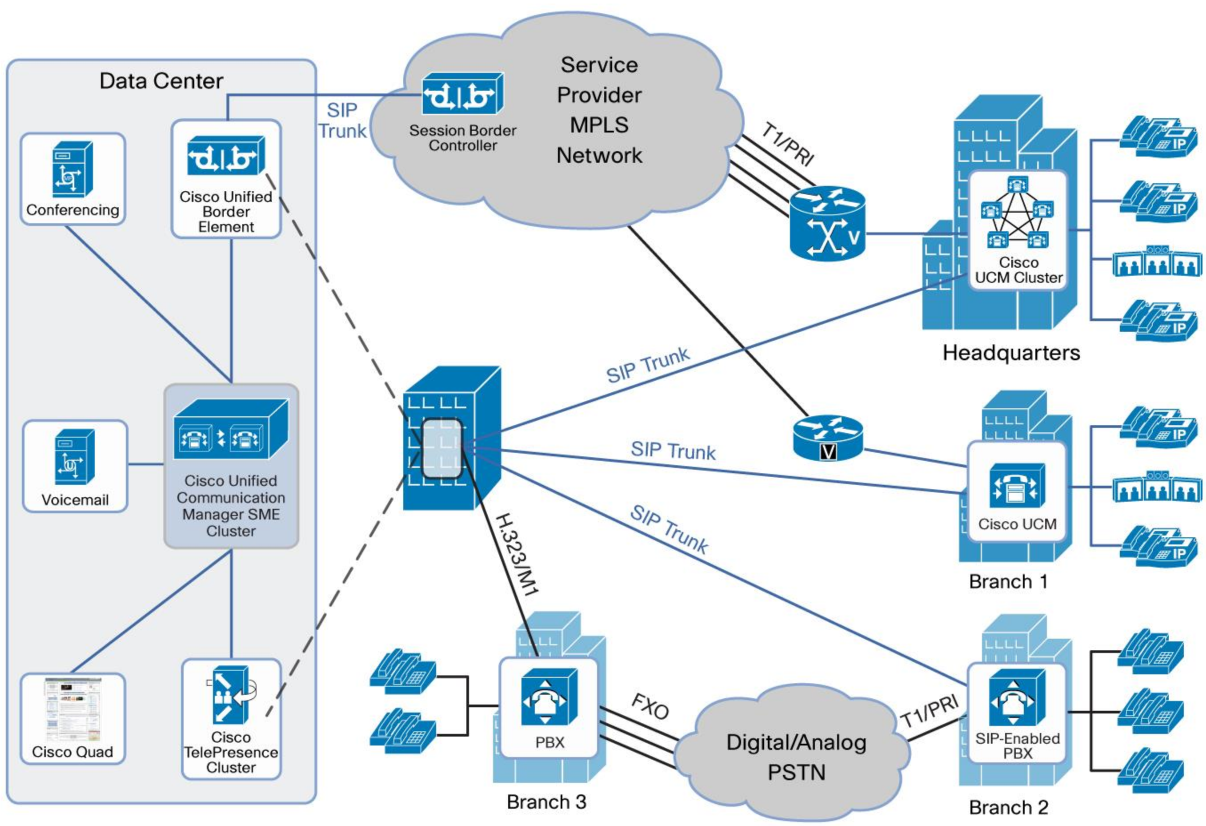
Solutions and Services

Session Management Edition routes among SIP-compliant elements and interworks with older components using legacy protocols (for example, H.323 and Q.SIG) to enable a broad range of unified communications services and to realize operational savings while providing an optimal user experience. One of the most frequently desired solutions is the centralized data center deployment of collaborative applications. As shown in Figure 1, you can use such a deployment to host and provide a diverse set of services:

- Video conferencing: A common Cisco TelePresence Manager can be shared among all Cisco TelePresence endpoints (from Cisco TelePresence System 500 Series through TelePresence 3200 Series) associated with communications managers connected within the enterprise. And with deployment of optional Cisco Intercompany Media Engine (IME) equipment, you can stream high-definition (HD) video between any of your organization's telepresence units and those of your suppliers, customers, partners, and advisors. For larger audiences or for those requiring more extensive multiparty sharing and editing of media, you can share web conferencing throughout your organization with the data center deployment of Cisco Unified MeetingPlace servers or even provisioned access to cloud conferencing solutions such as Cisco WebEx meetings through centralized SIP trunks. As described later in this document, you can also extend this audio and web conferencing to your users' mobile phones and smartphones through single number reach (SNR) and Cisco Unified Mobility Advantage.
- Voice messaging and visual voicemail: A common Cisco Unity® Connection cluster can provide mailboxes for up to 20,000 users, and the load-balancing and highly available configurations (active-active or active-standby) of the Cisco Unity Connection clusters help ensure your users will have access to their messages **whenever** they need them. And users can visually access their messages **wherever** they need them through Cisco Unified Mobility Client on their smartphones with deployment of a centralized Cisco Unified Mobility Advantage system. Message-waiting-indicator (MWI) lamps can be serviced on Cisco Unified Mobility Client-supported smartphones and on desk sets associated with Cisco Unified Communications Manager, Cisco Unified Communications Manager Express (CME), and other SIP and Q.SIG (H.323 Annex M1)-compliant PBXs, including current and older systems from Avaya, Nortel, Siemens, Alcatel, and many other vendors. For more information about third-party PBX compatibility, please refer to the Cisco Interoperability Portal.

- Enterprise social networking: In conjunction with Cisco Unified Communications Manager Session Management Edition and other applications mentioned previously, a common Cisco Quad enterprise collaboration platform can offer your users a rich and powerful enterprise social networking platform that integrates collaborative workspaces, contact information, and presence status with real-time posting, instant message, “click to call”, “click to video call”, and “click to conference” communication services.
- Mobility: Using the remote destination profiles integrated into Session Management Edition, you can enable single number reach (SNR) for salespeople, part-time telecommuters, and other mobile workers within your enterprise. This capability extends not only simple voice calling to your mobile users, but also audio conferencing, and for Cisco Unified Mobility Client-supported smartphones (Apple iPhone, Android, Blackberry and Nokia) visual voice mail, web conferencing (with WebEx client), and presence. You can also take advantage of the latest mobility features such as session handoff when you use a third-party PBX phone as your desktop endpoint.

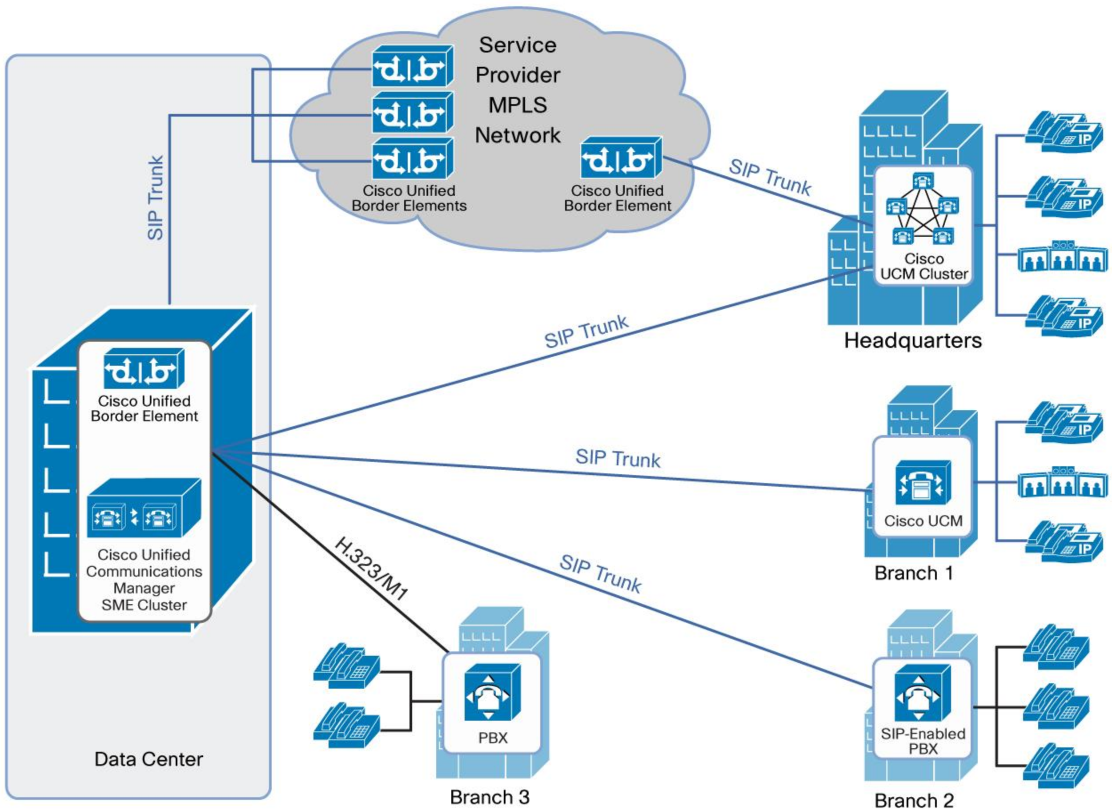
Figure 1. A fully configured Cisco Unified Communications Manager Session Management Edition deployment hosting diverse services



Another powerful capability of SIP is the consolidation and replacement of TDM trunks with the newer service provider SIP trunk offerings. These changes can produce significant return on investment (ROI) through savings on access charges and operating expenses, more efficient use of trunk resources (in terms of both provisioned bandwidth and least-cost routing (LCR)), and the flexibility to dynamically size trunks for periods of peak usage. Cisco Unified Communications Manager SME supports many type of deployments including centralized and distributed SIP trunking. Figure 2 shows but one example of a myriad of choices involving centralized or hybrid-centralized SIP trunking.

Other benefits of such deployment topologies could include the savings from managing a consolidated, global dial plan that integrates Cisco Unified Communications Manager as well as third-party PBXs (please refer to the Cisco Interoperability Portal for compatibility information), the utility of centralized billing and call detail reporting, and the consistency of call forking to centralized recording servers. Also, the simplified troubleshooting of SIP call problems using either SIP tracing on Session Management Edition or the various demarc debugging and quality-of-service (QoS) monitoring tools on the deployed Cisco Unified Border Elements could be other benefits.

Figure 2. A deployment example with centralized SIP Trunking



Feature Highlights and Benefits

New with Version 8.0

Built on Cisco Unified Communications Manager Session Management Edition 7.1(3), Session Management Edition Version 8.0 aims to lower the total cost of ownership (TCO) for organizations and improve the communications experience for end users as well as system administrators. Some of the important features of this release follow:

- Cisco Intercompany Media Engine (IME): Cisco IME allows intercompany, boundary-less communications among business partners and customers. It enables enterprise video telephony and high-fidelity wideband audio between companies. It is easy to use because it self-learns IP routes to business partners. When making SIP calls to business partners, you have the same experience as you do within your own company. It enables innovative collaboration capabilities and applications to be shared between partners.

- Cisco Service Advertisement Framework (SAF) Call Control Discovery: SAF is a network-based, scalable, bandwidth-efficient, real-time approach to service advertisement and discovery. It allows Session Management Edition and Cisco Unified Communications Manager to advertise directory-number ranges that they own, discover ranges owned by others, and dynamically create routes for calls between clusters, between PBXs, or to service provider SIP trunks.
- Resource Reservation Protocol (RSVP) SIP preconditions: Cisco Unified Communications Manager Session Management Edition 8.0 enhances RSVP to enable RSVP agents to provide end-to-end dynamic Call Admission Control (CAC) for calls between Session Management Edition and Cisco Unified Communications Manager clusters or Cisco Unified Border Elements (CUBE) based on the conditions of a dynamic network rather than those of a static, locations-based model.
- External call control with Cisco Unified Routing Rules Interface (CURRI): The Cisco Unified Routing Rules Interface enables web-service applications to remotely permit, deny, or redirect calls based on regulatory rules and business logic. When dialed digits match specified patterns, Session Management Edition issues a Web 2.0 Route Request over the Cisco Unified Routing Rules Interface. The web application evaluates the information and returns call-handling instructions as a decision (allow or deny) with an obligation (specific call-routing instructions and treatment).

Table 1 presents a complete list of new features included in Cisco Unified Communications Manager Session Management Edition Version 8.0.

Table 1. New Features in Session Management Edition 8.0

Feature	Benefits
Cisco Intercompany Media Engine	• Support for Cisco IME
Total Cost of Ownership and complexity-reduction features	• Cisco SAF Call Control Discovery • Support for Cisco Unified Analysis Manager: ◦ Product nodes inventory and grouping ◦ Trace setting ◦ Trace collection on demand ◦ Scheduled trace collection ◦ Templates for setting trace levels and trace file server for Cisco Unified Communications Manager ◦ Collect and view system-configuration information ◦ System call tracking ◦ Uploading of trace files to Cisco Technical Assistance Center (TAC) FTP server and preferences • Daylight savings time and time-zone structure
Serviceability enhancements	• Clarity and consistency of alarms and events • New performance-monitoring counters (external call control, Cisco SAF client)
Call Admission Control	• SIP preconditions: End-to-end RSVP • Application identification (ID)
Application-programming-interface (API) enhancements	• External Call Control (ECC) service and Cisco Unified RRI • Cisco Administrative XML[AXL) enhancements (changes to Cisco AXL versioning, partial response, list of API improvements, member associations, and schema improvements)
Cisco Unified Communications Manager infrastructure enhancements	• System-diagnostic-interface (SDI) trace reduction and improved overall Cisco Unified Communications Manager performance by reducing I/O wait associated with redundant trace

Product Capabilities and Requirements

Centralized Application Availability and Policy-Enabled Application Presentation Through SIP Connectivity

Cisco Unified Communications Manager Session Management Edition provides the session routing intelligence to enable the delivery of SIP-connected applications to Cisco Unified Communications Manager users anywhere within your organization, to the level of their connected endpoint capability (for example, not all phones support video calling), even if those endpoints are SCCP- or H.323. This intelligence includes an ability to deliver these applications to users of compatible smartphones from leading manufacturers and consortia - Apple, Blackberry, Android, and Nokia. And when combined with a Cisco Unified Routing Rules Interface-compliant server, this advanced routing intelligence can be further enhanced to include operational and business policy rules in the presentation of applications; for example, determining whether an outbound call should be blocked, connected, or routed through an alternate, lower-cost trunk. SIP-connected applications include:

- Cisco Unified Communications Manager and Cisco Unified Communications Manager Business Edition
- Cisco Unified Communications Manager Express
- Cisco Unity and Cisco Unity Connection messaging
- Cisco TelePresence meeting applications
- Cisco Intercompany Media Engine (HD video)
- Cisco Unified MeetingPlace conferencing applications and Cisco WebEx conferencing
- Cisco Quad (Enterprise Collaboration Portal)
- Cisco Unified Mobility Advantage (UMA)
- Cisco Unified SIP Proxy (USP)

Third-Party PBX Endpoint Support

For many SIP applications, the session routing and application presentation capabilities of Cisco Unified Communications Manager Session Management Edition can also be extended to other vendors' traditional digital and IP PBXs, to the level of their capabilities (for example, digital PBXs do not support video calling), when they are enabled to support SIP, H.323, Q.SIG over H.323 (Annex M1) or MGCP (via T1/PRI through IOS gateways) trunk connections. Voice messaging, conferencing, and mobility enablement are typical of applications that can be served by Session Management Edition and supported by PBXs from Avaya, Nortel, Siemens, Alcatel Lucent, and others. Cisco continuously tests interoperability with numerous vendors' products; for the latest information about compatibility with third-party systems, please refer to the Cisco Interoperability Portal.

Global Dial-Plan Consolidation and Topology-Aware Routing

Cisco Unified Communications Manager Session Management Edition can support a variety of different dial plans. At one end of the spectrum, it enables organizations to consolidate disparate, multiple-length dial plans into a single, consistent, enterprise-wide numbering plan that can be centrally administered, saving on operations, administration, maintenance, and provisioning (OAM&P) costs. At the other end, Cisco Unified Communications Manager Session Management Edition can interwork the existing dial plans where greater numbering flexibility and local number reuse are important.

Cisco Unified Communications Manager Session Management Edition also supports sophisticated enterprise network topologies and policies that enable access cost savings through least-cost and time-of-day routing, PSTN toll avoidance through tail-end hop off or IME, toll-fraud prevention and government inter-LATA restrictions through partitioned provisioning, and increased resiliency through automatic alternate routing around failures and bandwidth limitations.

Reliability and Security

You can cluster Cisco Unified Communications Manager Session Management Edition servers to provide session load balancing and trunk redundancy and resiliency with up to N+8 server support. The servers themselves support NIC teaming for added redundancy at the link layer (Layer 2), helping ensure five 9's reliability of the application. If a node or network connection fails, however, active calls are preserved and new session traffic is automatically directed to the redundant nodes. And although for most deployments of this application it is preferable to have all nodes of a cluster in a centralized data center, the extended latency tolerance of the system means that nodes could be deployed in redundant data centers thousands of miles apart.

Your enterprise data remains secure when transiting Cisco Unified Communications Manager Session Management Edition networks. The application supports encrypted transport layer security (TLS) for SIP connections, and if you choose to share applications and services with partner organizations, you may rest assured that this support (along with SRTP encrypted bearer channels) extends to IME.

Capacity and Network Scalability

You can cluster Cisco Unified Communications Manager Session Management Edition servers in groups of up to nine to support session routing with load balancing and redundancy (active-active) at high capacities:

- Up to 28 calls per second (cps) per server, and 224 cps per cluster
- Up to 3,840 concurrent calls per server, and 30,000 concurrent calls per cluster
- More than 2,000 trunks per cluster for interconnecting instances of Cisco Unified Communications Manager, third-party PBXs, and SIP (and H.323) applications within your enterprise network

At such high session-processing capacities, Cisco UCM SME could overwhelm bandwidth-constrained WAN links within your enterprise if excessive loads are offered to them. To prevent this possibility, the application supports a rich set of bandwidth-efficient codecs and both locations-based call admission control (LCAC) and end-to-end Resource Reservation Protocol (RSVP) for call admission. These methods help ensure voice QoS by automatically diverting calls across alternate available routes (including PSTN connections).

For greater flexibility, RSVP now supports Application ID, which associates RSVP reservations with specific applications and sub-applications. Initially, voice and video are supported. Supported voice and video codecs include:

- G.711 (mu-law and a-law)
- Internet Low Bitrate Codec (iLBC)
- G.729A/B, G.723.1, and G.728
- Global System for Mobile-Enhanced Full Rate (GSM-EFR)
- Global System for Mobile-Full Rate (GSM-FR)
- G.722 and G.722.1
- Wideband audio (proprietary 16-bit resolution; 16-kHz sampled audio)

- Advanced Audio Codec (AAC) for use with Cisco TelePresence devices
- H.261, H.263, H.264
- Cisco Wideband Video Codec (Cisco Unified Video Advantage)

Platforms

- Cisco MCS 7845 Media Convergence Servers
- Selected third-party servers that are Cisco MCS 7845-equivalent; for details, please visit: http://www.cisco.com/go/swonly

The appliance model provides a platform for call processing with the software preloaded on a Cisco MCS platform; the software is optionally available as a DVD kit for equivalent customer-provided servers. The appliance comes with a single firmware image that includes the underlying operating system as well as the Cisco Unified Communications Manager Session Management Edition application. The appliance is accessed through a GUI, and a command-line interface (CLI) has been added to facilitate diagnostics and basic system management, such as the starting or stopping of services and rebooting of the appliance. No access to the underlying operating system is necessary. All system management activities - for example, disk-space monitoring, system monitoring, and upgrades - are controlled through the GUI.

SIP Trunk and Endpoint Support

SIP trunk support provides interoperability and enables the development of innovative applications. Cisco Unified Communications Manager Session Management Edition includes the following major SIP functions:

- Native support of SIP devices
- Presence information for SIP and SCCP devices, including support for PUBLISH
- Fault, configuration, accounting, performance, and security (FCAPS) enhancements to support SIP
- SIP trunk enhancements for external applications, such as conferencing and presence
- Third-party SIP devices supporting RFC 3261
- SIP line-side RFCs: RFCs 3261, 3262, 3264, 3265, 3311, 3515, and 3842
- SIP trunk RFC support: RFCs 2833, 2976, 3261, 3262, 3264, 3265, 3311, 3323, 3325, 3515, 3842, 3856, and 3891

Licensing

The basic unit of licensing for Cisco Unified Communications Manager Session Management Edition is number of sessions. In Version 8.0 there is no enforced licensing. Future versions of the application will have licensing enforcement.

- A session is any SIP, H.323, or SCCP call that exits a trunk in the Cisco Unified Communications Manager Session Management Edition. Examples of sessions include:
  - Calls that are in tandem with the Cisco Unified Communications Manager Session Management Edition between two different PBXs or clusters
  - Calls that go from a PBX or cluster to a SIP trunk
  - Calls that originate from the Cisco Unified Communications Manager Session Management Edition and go to another PBX or trunk

- Although Cisco Unified Communications Manager Session Management Edition is optimized for PBX aggregation and SIP trunking, it can also register SIP and SCCP phones. Each device (Cisco Unified IP Phones, soft phones, third-party devices, and video devices) provisioned in the system corresponds to a User Connect License (Enhanced IP User, Basic IP User, Public Space, or Analog). Device licensing is enforced in Cisco Unified Communications Manager Session Management Edition.

Cisco Unified Workspace Licensing

Enterprises that order Cisco Unified Workspace Licensing (UWL) Pro are entitled to one session for every five Cisco UWL Pro users. This rule applies to Cisco UWL Pro users who are in Cisco Unified Communications Manager leaf clusters. Please visit http://www.cisco.com/go/workspace_licensing for more information and to determine whether Cisco Unified Workspace Licensing is appropriate for your customer.

Ordering Information

New Installations

Please refer to Chapter 8 of the Ordering Guide for information about ordering Cisco Unified Communications Manager Session Management Edition.

Software Upgrades

You can order Cisco Unified Communications Manager Session Management Edition 8.0 installation CDs and DVDs for existing systems.

Customers with a Cisco Unified Communications Software Subscription running Cisco Unified Communications Manager Session Management Edition 7.1(3) who want to upgrade to Cisco UCM Session Management Edition 8.0 can order upgrades using the Product Upgrade Tool located at: http://www.cisco.com/upgrade.

If you are planning an upgrade to Cisco Unified Communications Manager Version 8.0, please refer to the upgrade program for supported servers at: http://www.cisco.com/go/swonly.

Hard disk capacities of 72 GB or greater and 2 GB of RAM are required.

Cisco Unified Communications Services

Cisco Unified Communications Services allows you to accelerate cost savings and productivity gains associated with deploying a secure, resilient Cisco Unified Communications Solution. Delivered by Cisco and our certified partners, our portfolio of services is based on proven methodologies for unifying voice, video, data, and mobile applications on fixed and mobile networks. Our unique lifecycle approach to services can enhance your technology experience to accelerate true business advantage.

For More Information

For more information about Cisco Unified Communications Manager Session Management Edition, please visit http://www.cisco.com/go/unifiedcm or contact your local Cisco account representative.

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