DATA SHEET

CISCO CALLMANAGER VERSION 4.1

Cisco® IP Communications is a comprehensive system of powerful, enterprise-class solutions, including IP telephony, unified communications, IP video and audio conferencing, and customer contact. It helps organizations realize business gains by improving operational efficiencies, increasing organizational productivity, and enhancing customer satisfaction. Cisco CallManager is the software-based call-processing component of the Cisco enterprise IP telephony solution; it is enabled by Cisco AVVID (Architecture for Voice, Video and Integrated Data).

Cisco CallManager software extends enterprise telephony features and capabilities to packet telephony network devices such as IP phones, media processing devices, voice over IP (VoIP) gateways, and multimedia applications. Additional data, voice, and video services such as unified messaging, multimedia conferencing, collaborative contact centers, and interactive multimedia response systems interact with the IP telephony solution through Cisco CallManager open telephony application programming interfaces (APIs). Cisco CallManager is installed on Cisco 7800 Series media convergence servers (MCSs) and selected third-party servers. Cisco CallManager software is shipped with a suite of integrated voice applications and utilities, including the Cisco CallManager Attendant Console—a software-only manual attendant console; a software-only ad-hoc conferencing application; the Bulk Administration Tool (BAT); the CDR Analysis and Reporting (CAR) tool; the Real-Time Monitoring Tool (RTMT); a simple, low-density Cisco CallManager Auto Attendant (CM-AA); the Tool for Auto-Registered Phone Support (TAPS); and the IP Manager Assistant (IPMA) application.

FEATURES AND BENEFITS

Cisco CallManager Version 4.1 provides a scalable, distributable, and highly available enterprise IP telephony call-processing solution. Multiple Cisco CallManager servers are clustered and managed as a single entity. Clustering multiple call-processing servers on an IP network is a unique capability in the industry, and highlights the leading architecture provided by Cisco AVVID. Cisco CallManager clustering yields scalability of from 1 to 30,000 IP phones per cluster, load balancing, and call-processing service redundancy. By interlinking multiple clusters, system capacity can be increased up to one million users in a 100+ site system. Clustering aggregates the power of multiple, distributed Cisco CallManagers, enhancing the scalability and accessibility of the servers to phones, gateways, and applications. Triple call-processing server redundancy improves overall system availability.

The benefit of this distributed architecture is improved system availability, load balancing, and scalability. Call Admission Control (CAC) helps ensure that voice quality of service (QoS) is maintained across constricted WAN links, and automatically diverts calls to alternate public switched telephone network (PSTN) routes when WAN bandwidth is not available. A Web interface to the configuration database enables remote device and system configuration. HTML-based online help is available for users and administrators.

The enhancements provided by Cisco CallManager Version 4.1 offer improved security, interoperability, capability, supportability, and productivity as well enhancements to video telephony introduced in Cisco CallManager 4.0.

Cisco CallManager 4.1 has many security features that give Cisco CallManager servers and IP phones the ability to verify identity of the devices or servers that they communicate with, ensure the integrity of data they are receiving, and provide privacy of communications via encryption. The devices that can participate in secure communications now includes the Cisco IP Phone 7940G, IP Phone 7960G, IP Phone 7970G, and Media Gateway Control Protocol (MGCP) gateways. Secure administration and troubleshooting is now capable with CallManager 4.1 using HTTPS.

Improvements in the CallManager Q.SIG signaling interface expands the range of functions with which Cisco CallManager can connect to other Q.SIG compatible systems. Features like path replacement and call completion allow Cisco CallManager to integrate with other Q.SIG compatible systems closer than ever before. H.323 Annex M.1 support now gives users improved feature transparency between CallManager clusters.

Enhancements to the CallManager APIs (AXL, JTAPI, TSP) provide customers and 3rd party vendors increased ability to develop improved applications that can be integrated with CallManager and IP Phones.

Other key features provided by Cisco CallManager 4.1 include call coverage, time-of-day routing and restrictions, forced authorization codes (FAC) and client matter codes (CMC) and enhancements for Video Telephony that was provided in Cisco CallManager 4.0.

New administration features such as Cisco Unity User Integration allow a CallManager system administrator to easily configure a Cisco Unity voice mail box while configuring an IP phone for that user.

SPECIFICATIONS

Platforms

- Cisco 7815, 7825, 7835, and 7845 MCSs
- Selected third-party servers; for details, visit: http://www.cisco.com/go/swonly

Bundled Software

- Cisco CallManager Version 4.1—Call-processing and call-control application.
- Cisco CallManager Version 4.1 configuration database—Contains system and device configuration information, including dial plan.
- Cisco CallManager administration software.
- Auto attendant—Bundled with Cisco CallManager via the Extended Services CD.
- Cisco CallManager CDR Analysis and Reporting Tool (CAR)—Provides reports for calls based on call detail records (CDRs). Reports that are provided include calls on a user basis, calls through gateways, simplified call quality, and a CDR search mechanism. In addition, CAR provides limited database administration; for example, deleting records based on database size.
- Cisco CallManager Bulk Administration Tool (BAT)—Allows the administrator to perform bulk add, delete, and update operations for devices and users.
- Cisco CallManager Attendant Console—Allows a receptionist to answer and transfer/dispatch calls within an organization. The attendant can install the attendant console, which is a client-server application, on a PC that runs Windows 98, ME, NT 4.0 (Service Pack 4 or greater), 2000, or XP. The attendant console connects to the Cisco Telephony Call Dispatcher (TCD) server for login services, line state, and directory services. Multiple attendant consoles can connect to a single Cisco TCD server. In Cisco CallManager Version 4.1, Attendant Console now supports accessibility enhancements for sight-impaired individuals.
- Cisco CallManager Real-Time Monitoring Tool (RTMT)—A client tool that monitors real-time behavior of the components in a Cisco CallManager cluster. RTMT uses HTTP and TCP to monitor device status, system performance, device discovery, and computer telephony integration (CTI) applications. It connects directly to devices by using HTTP for troubleshooting system problems.
- Cisco CallManager Trace Collection Tool—Collects traces for a Cisco CallManager cluster into a single zip file. The collection includes all traces for Cisco CallManager and logs such as Event-Viewer (application, system, and security), Dr. Watson log, Cisco Update, Prog logs, RIS DC logs, and SQL and IIS logs.
- Cisco Conference Bridge—Provides software conference bridge resources that can be used by Cisco CallManager.
- Cisco Customer Directory Configuration Plugin—Guides the system administrator through the configuration process for integrating Cisco CallManager with Microsoft Active Directory and Netscape Directory Server.
- Cisco IP Phone Address Book Synchronizer—Allows users to synchronize Microsoft Outlook or Outlook Express address books with Cisco Personal Address Book. The synchronizer provides two-way synchronization between the Microsoft and Cisco products. After the user installs and configures Cisco Personal Address Book, users can access this feature from the Cisco IP Phone Configuration Website.

- Cisco IP Telephony Locale Installer—Provides user and network locales for Cisco CallManager, adding support for languages other than English. Locales allow users to view translated text, receive country-specific phone tones, and receive TAPS prompts in a chosen language when working with supported interfaces. Install the Cisco IP Telephony Locale Installer on every server in the cluster. Click the icon to download one or more locale installers from the Web (you must have an Internet connection and a Cisco.com user account and password to download the executable).
- Cisco JTAPI—This plug-in is installed on all computers that host applications that interact with Cisco CallManager via JTAPI. JTAPI provides the standard programming interface for telephony applications written in the Java programming language. JTAPI reference documentation and sample code are included. Cisco CallManager Version 4.1 adds support for new features as well as the ability to disable device validation, which would allow applications to monitor or control a large amount of devices without requiring the devices to be specified in those applications' controlled device lists. JTAPI Device State Server is new in Cisco CallManager Version 4.1, as is notification of status (busy, idle, etc.) of a CTI device without having to monitor individual lines.
- Cisco Telephony Service Provider—Contains the Cisco TAPI service provider (TSP) and the Cisco Wave Drivers. Install the application on the Cisco CallManager server or on any other computer that is running a Microsoft Windows operating system that interacts with the Cisco CallManager server via TCP/IP (TAPI runs on the Microsoft Windows operating system). The Cisco TAPI Developer's Guide describes the TAPI interfaces that are currently supported. Install the Cisco TSP and the Cisco Wave Drivers to allow TAPI applications to make and receive calls on the Cisco IP Telephony Solution.
- Cisco TAPS—Loads a preconfigured phone setting on a phone.
- Cisco Dialed Number Analyzer—Serviceability tool that analyzes the dialing plan for specific numbers.
- Cisco IP Manager Assistant (IPMA)—Provides "boss"/administration features along with administration Web pages for improved call handling.

SYSTEM CAPABILITIES SUMMARY

- Alternate automatic routing (AAR)
- Attenuation and gain adjustment per device (phone and gateway)
- Automated bandwidth selection
- Auto route selection (ARS)
- AXL Simple Object Access Protocol (SOAP) API with performance and real-time information
- Basic Rate Interface (BRI) endpoint support; registers BRI endpoints as SCCP devices*
- CAC—intercluster and intracluster
- Call coverage*
  - Forwarding based on internal/external calls*
  - Forwarding out of a coverage path*
  - Timer for maximum time in coverage path*
  - Time of day*
- Call display restrictions*
- Coder-decoder (codec) support for automated bandwidth selection
  - G.711 mu-law, a-law
  - G.723.1
  - G.729A/B
  - GSM-EFR, FR
  - Wideband audio—Proprietary 16-bit resolution, 16-kHz sampled audio

- Digit analysis and call treatment (digit string insertion, deletion, stripping, dial access codes, digit string translation)
- Distributed call processing
  - Deployment of devices and applications across an IP network
  - "Clusters" of Cisco CallManager servers for scalability, redundancy, and load balancing
  - Maximum of 7500 IP phones per Cisco CallManager server (configuration-dependent)
  - Maximum of 100,000 busy-hour call completions (BHCCs) per Cisco CallManager server (configuration-dependent)
  - Eight Cisco CallManager servers per cluster
  - Maximum of 250,000 BHCCs per Cisco CallManager cluster (configuration-dependent)
  - Maximum of 30,000 IP phones per cluster (configuration-dependent)
  - Intercluster scalability to more than 100 sites or clusters through H.323 gatekeeper
  - Intracluster feature transparency
  - Intracluster management transparency
- Fax over IP—G.711 pass-through and Cisco Fax Relay
- Forced authorization codes/client matter codes (account codes) *
- H.323 interface to selected devices
- H.323 FastStart (inbound and *outbound)
- Hotline and private line automated ringdown (PLAR)
- Hunt groups—broadcast, circular, longest idle, and linear
- Interface to H.323 gatekeeper for scalability, CAC, and redundancy
- Language support for client user interfaces (languages specified separately)
- Multilevel precedence and pre-emption (MLPP)—*Enhancements made in Cisco CallManager Version 4.1
- Multilocation—Dial-plan partition
- Multiple ISDN protocol support
- Multiple remote Cisco CallManager platform administration and debug utilities
  - Prepackaged alerts, monitor views, and historical reports with RTMT
  - Real-time and historical application performance monitoring through operating system tools and Simple Network Management Protocol (SNMP)
  - Monitored data collection service
  - Remote terminal service for off-net system monitoring and alerting
  - Real-time event monitoring and presentation to common syslog
  - Trace setting and collection utility
  - Browse to onboard device statistics
  - Clusterwide trace setting tool
  - Trace collection tool

- Multisite (cross-WAN) capability with intersite CAC
- Dial-plan partitioning
- Off-premises extension (OPX)
- Outbound call blocking
- Out-of-band dual tone multifrequency (DTMF) signaling over IP
- PSTN failover on route nonavailability—AAR
- Q.SIG (International Organization for Standardization [ISO])
  - Alerting name specified in ISO 13868 as part of the SS-CONP feature. *
  - Basic call
  - ID services
  - General functional procedures
  - Call back—ISO/IEC 13870: 2nd Ed, 2001-07 (CCBS, CCNR) *
  - Call diversion (SS-CFB [busy], SS-CFNR [no answer], SS-CFU [unconditional]); service ISO/IEC 13872 and ISO/IEC 13873, first edition 1995
    - Call diversion by forward switching
    - Call diversion by reroute*
  - Call transfer by join
  - H.323 Annex M.1 (Q.SIG over H.323) —ITU recommendation for Annex M.1*
  - Identification restriction (Calling Name Identification Restriction [CNIR], Connected Line Identification Restriction [COLR]), Connected Name Identification Restriction [CONR])
  - Loop prevention, diversion counter and reason, loop detection, diverted to number, diverting number, original called name and number, original diversion reason, redirecting name
  - Message waiting indicator (MWI)
  - Path replacement ISO/IEC 13863: 2nd Ed. 1998 and ISO/IEC 13974: 2nd Ed. 1999. *
- Redundancy and automated failover on call-processing failure
  - Call preservation on call-processing failure
- Station to station
- Station through trunk (Media Gateway Control Protocol [MGCP] gateways)
  - JTAPI and TAPI applications enabled with automated failover and automatic update
  - Triple Cisco CallManager redundancy per device (phones, gateway, applications) with automated failover and recovery
  - Trunk groups
  - MGCP BRI support (ETSI BRI basic-net3 user-side only) *
- Security
  - Configurable operation modes—Nonsecure or secure

- Device authentication—Embedded X.509v3 certificate in new model phones; certificate authority proxy function (CAPF) used to install locally significant certificate in phones
- Data integrity—TLS cipher "NULL-SHA" supported. Messages are appended with SHA1 hash of the message to ensure that the message is not altered on the wire and can be trusted.
- Secure HTTP (HTTPS) support for the following applications: Cisco CallManager Admin, Cisco CallManager Serviceability, Cisco CallManager User, RTMT, Cisco CallManager TraceAnalysis, Cisco CallManager Service, Trace Collection Tool, and CAR. *
- Privacy—Cisco CallManager supports encryption of signaling and media. Phone types include Cisco IP Phone 7940, 7960, and 7970; Survivable Remote Site Telephony (SRST), and MGCP gateways*
- Secure Sockets Layer (SSL) for directory—Supported applications include BAT, CAR, Cisco CallManager Admin User Pages, Cisco CallManager Admin IPMA Pages, Cisco CallManager User Pages / IP Phone Options Pages, Cisco Conference Connection, CTI Manager, Extension Mobility, IP Manager Assistant, and Multilevel Administration (MLA). *
- USB eToken containing a Cisco rooted X.509v3 certificate is used to generate a Certificate Trust List (CTL) file for the phones as well as configuring the security mode of the cluster.
- Phone security—Trivial File Transfer Protocol (TFTP) files (configuration and firmware loads) are signed with the self-signed certificate of the TFTP server. The Cisco CallManager system administrator will be able to disable HTTP and Telnet on the IP phones.

- Session Initiation Protocol (SIP) trunk
- SRST
- Shared resource and application management and configuration
  - Transcoder resource
  - Conference bridge resource
  - Topological association of shared resource devices (conference bridge, music on hold [MoH] sources, transcoders)
  - Media termination point (MTP)—Support for SIP trunk and RFC 2833
  - Annunciator
- Silence suppression, voice activity detection
- Simplified North American Numbering Plan (NANP) and non-NANP support
- T.38 fax support (H.323 only) *
- Third-party applications support
  - Broadcast paging—through foreign exchange station (FXS)
  - SMDI for MWI
  - Hook-flash feature support on selected FXS gateways
  - TSP 2.1 interface
  - JTAPI 2.0 service provider interface
  - Billing and call statistics
  - Configuration database API (Cisco AXL)

- Time of day, day of week, day of year routing/restrictions*
- Toll restriction—Dial-plan partition
- Toll fraud prevention
  - Prevent trunk to trunk transfer*
  - Drop conference call when originator hangs up*
  - Forced authorization codes*
- Unified device and system configuration
- Unified dial plan
- Video (SCCP and H.323)

*Indicates new feature or service for Cisco CallManager Version 4.1

Summary of User Features

- Abbreviated dial
- Answer and answer release
- Autoanswer and intercom
- Barge
- Call-back busy, no reply to station
- Call connection
- Call coverage*
- Call forward—All (off-net and on-net)
- Call forward—Busy
- Call forward—No answer
- Call hold and retrieve
- Call join
- Call park and pickup
- Call pickup group—Universal
- Call status per line (state, duration, number)
- Call waiting and retrieve (with configurable audible alerting)
- Calling line identification (CLID)
- Calling line identification restriction (CLIR) call by call
- Calling party name identification (CNID)
- Conference barge
- Conference list and drop any party (ad-hoc conference)
- Direct inward dial (DID)
- Direct outward dial (DOD)
- Directory dial from phone—Corporate, personal
- Directories—Missed, placed, received calls list stored on selected IP phones
- Distinctive ring (on-net vs. off-net)
- Distinctive ring per line appearance
- Distinctive ring per phone
- Drop last conference party (ad-hoc conferences)

- Extension mobility support
- Hands-free, full-duplex speakerphone
- HTML help access from phone
- Immediate divert to voicemail
- Last number redial (off-net and on-net)
- Malicious call ID and trace
- Manager-assistant service (IPMA application)
  - Proxy line support

    Manager features—Immediate divert or transfer, do not disturb, divert all calls, call intercept, call filtering on CLID, intercom, speed dials.

    Assistant features—Intercom, immediate divert or transfer, divert all calls, manager call handling through assistant console application.
  - Shared line support

    Manager features—Immediate divert or transfer, do not disturb, intercom, speed dials, barge, direct transfer, join.

    Assistant features—Handle calls for managers; view manager status and calls; create speed dials for frequently used numbers; search for people in the corporate/Cisco CallManager directory; handle calls on their own lines; immediate divert or transfer, intercom, barge, privacy, multiple calls per line, direct transfer, join; send DTMF digits from console, MWI status of manager phone.
  - System capabilities—Multiple managers per assistant (up to 33 lines), redundant service.
- MWI
- Multiparty conference—Ad-hoc with add-on, meet-me features
- Multiple calls per line appearance
- Multiple line appearances per phone
- MoH
- Mute capability from speakerphone and handset
- On-hook dialing
- Operator attendant—Cisco Attendant Console
  - Call queuing
  - Broadcast hunting
  - Shared line support
- Privacy
- Real-time QoS statistics through HTTP browser to phone
- Recent dial list—Calls to phone, calls from phone, autodial, and edit dial
- Service URL—Single button access to IP phone service
- Single directory number, multiple phones—Bridged line appearances
- Speed dial—Multiple speed dials per phone
- Station volume controls (audio, ringer)
- Transfer
  - Blind
  - Consultative
  - Direct transfer of two parties on a line.

- User-configured speed dial and call forward through Web access
- Video (SCCP and H.323)
- Web services access from phone
- Web dialer—Click to dial
- Wideband audio codec support—Proprietary 16-bit resolution, 16-kHz sampling rate codec

*Indicates new feature or service for Cisco CallManager Version 4.1

Summary of Administrative Features

- Application discovery and registration to SNMP manager
- AXL SOAP API with performance and real-time information
- BAT
- CDRs
- CAR tool
- Call forward reason code delivery
- Centralized, replicated configuration database; distributed Web-based management viewers
- Configurable and default ringer WAV files per phone
- Configurable call forward display
- Database automated change notification
- Date and time display format configurable per phone
- Debug information to common syslog file
- Device addition through wizards
- Device-downloadable feature upgrades—Phones, hardware transcoder resource, hardware conference bridge resource, VoIP gateway resource
- Device groups and pools for large system management
- Device mapping tool—IP address to Media Access Control (MAC) address
- Dynamic Host Configuration Protocol (DHCP) block IP assignment—Phones and gateways
- Dialed Number Analyzer (DNA)
- Dialed number translation table (inbound and outbound translation)
- Dialed number identification service (DNIS)
- Enhanced 911 service
- H.323-compliant interface to H.323 clients, gateways, and gatekeepers
- JTAPI 2.0 computer telephony interface
- Lightweight Directory Access Protocol (LDAP) Version 3 directory interface to selected vendor's LDAP directories
    - Active Directory
    - Netscape Directory Server
- MLA access
- MGCP signaling and control to selected Cisco VoIP gateways
- Native supplementary services support to Cisco H.323 gateways
- Paperless phone DNIS—Display-driven button labels on phones
- Performance-monitoring SNMP statistics from applications to SNMP manager or to operating system performance monitor
- QoS statistics recorded per call
- Redirected DNIS (RDNIS), inbound, outbound (to H.323 devices)

- Select specified line appearance to ring
- Select specified phone to ring
- Single CDR per cluster
- Single point system and device configuration
- Sortable component inventory list by device, user, or line
- System event reporting—To common syslog or operating system event viewer
- TAPI 2.1 CTI
- Time-zone configurable per phone
- Cisco Unity™ software user integration*
- TAPS
- Extensible Markup Language (XML) API into IP phones (Cisco IP Phone 794x/796x)
- Zero-cost automated phone moves
- Zero-cost phone adds

*Indicates new feature or service for Cisco CallManager Version 4.1

CISCO CALLMANAGER VERSION 4.1 ENHANCEMENTS

User Feature Enhancements

- Attendant Console has been "accessibility enabled" to simplify use for visually handicapped attendants.
  - Works in conjunction with JAWS screen reader software.
  - Shortcut keys provided for easy navigability; mouseless operation of Attendant Console is possible.
  - Audible alerts provided to alert the user when certain events occur.
  - Attempt to transfer, consult transfer, or conference a call results in the call being put on hold while the dial pad is displayed.
- Video has added the following new capabilities:
  - Support for the SCCP H.264 video codec. The H.264 video codec delivers significantly higher quality at a given bandwidth than H.263. H.264 will be supported for intracluster SCCP calls only. Devices that will support SCCP H.264 include:
    - Tandberg SCCP phones (550 and T1000)
    - IPVC 3.6plus (3511 and 3540)
  - Midcall video for Cisco VT Advantage—If both parties are SCCP video endpoints, the call will immediately become a video call. If one party is an H.323 endpoint, the call will become a video call. But if the H.323 endpoint rejects the incoming channel or does not open a channel, the video call will be either one-way video or no video.
  - Video display mode for IPVC 3.6plus. IPVC 3.6plus will include a SCCP version of the message control unit (MCU) that supports both voice-activated and continuous presence videoconferencing modes. Video display mode is a softkey (called VidMode) that allows users on a SCCP videophone to toggle incoming videos between voice-activated and continuous presence mode.
  - Participant information for IPVC 3.6plus. Cisco CallManager will provide participant information to the SCCP version of the IPVC MCU, including the user name and number. The IPVC MCU will display this information in the participant list on its Web management interface. The IPVC MCU can overlay this information in the video if it has an Enhanced Media Processor (EMP) module.

- Dynamic H.323 addressing (E.164 addressing)—H.323 clients can be configured in Cisco CallManager via an E.164 address. This simplifies H.323 client administration for deployments where H.323 clients are configured for DHCP. E.164 addressing can be paired with an Cisco IOS Software Release 12.3(8)T gatekeeper to simplify H.323 client dialing.

System Capability Enhancements

- BRI support—Support for secure communications using legacy BRI and analog secure endpoints (STE/STU) and support for IP-STE.
  - V.150.1 Modem-Relay-over-IP support—Cisco CallManager will respond to the Session Description Protocol (SDP) sent by the gateway with default parameters. V.150.1 is required by the secure mode of IP-STEs and BRI-STEs.
  - BRI station pre-emption.
- Call coverage—Cisco CallManager Version 4.1 provides the ability to set up coverage paths to route calls to individuals or groups, helping to ensure that calls are answered. Call coverage features include:
  - Forwarding out of a coverage path.
  - Ability to set up different coverage paths based on time of day, day of week, or day of year.
  - Ability to provide separate forwarding treatment for internal versus external Call Forward No-Answer (CFNA) calls.
  - Ability to provide separate forwarding treatment for internal versus external Call Forward Busy (CFB) calls.
  - Support of a maximum timer for hunt lists.
  - Ability to allow a line to appear in multiple line groups, which was a limitation in previous versions of Cisco CallManager.
  - Ability to allow a gateway to appear in multiple route groups, which was a limitation in previous versions of Cisco CallManager.
  - Ability to divert to a final forwarding location when a hunt list terminates, either through exhaustion or expiration of its maximum hunt timer. This location may be a dialed number (voicemail pilot, another hunt pilot, a route pilot, or any allowed dialed number) or a checkbox to select personal treatment based on settings for the original called party’s line.
  - Splitting the existing route-list/hunt-list GUI into two separate forms—one for hunt list and one for route list.
- Call display restrictions—Ability for the system administrator to block calling/called/connected name/number between certain phones. This is frequently used in areas where, for security reasons, this calling information cannot be displayed on phones. A hotel is an example where calls from room to room would not display calling information.
- Forced authorization codes/client matter codes
  - Forced authorization codes—Allows a system administrator to require that an authorization code be entered prior to extending a call to a specific route pattern. This is often used as a mechanism to prevent fraudulent toll calls by individuals that might have access to the phones. The system administrator can assign authorization levels to allow some codes to have full calling capability and others to not be able to call certain numbers.
  - Client matter codes or account codes—Ability for a system administrator to require that a client matter code be entered prior to extending a call to a specific route pattern. This code is often used by companies to track calls made to specific accounts and use this data for billing purposes.
  - Client matter code and forced authorization code (CMC/FAC) information is recorded in the Cisco CallManager CDR database.

- H.323 FastStart—Support for inbound and outbound H.323 FastStart. This feature reduces the number of signals exchanged before voice is extended on a call. By using H.323 FastStart voice will be extended or cut-through using 10 less message exchanges. This can eliminate voice clipping connections that are separated by large distances with more than 50ms WAN delay. MTPs are required for outbound H.323 FastStart.
- MGCP BRI ETSI BRI basic-net3 (user-side only)—Allows a smaller, more cost-effective ISDN connection between Cisco CallManager and the PSTN for small offices where the cost of a Primary Rate Interface (PRI) is prohibitive.
- MLPP enhancements
  - BRI station pre-emption.
  - Support for MLPP-enabled, user-to-user information element (UUIE)-based PRI-4ESS interface.
  - Executive override precedence level; gives an additional precedence level that was previously unsupported.
  - Location-based MLPP through locations over intracluster and intercluster limited bandwidth WAN links.
  - Support for intercluster MLPP.
  - The signal information element, as described in 4.5.24 of ANSI T1.607.
- Q.SIG enhancements
  - Alerting name—Support of alerting name presentation and restriction for Q.SIG facilities. "alerting name" is the capability to send and receive a CalledName application protocol data unit (APDU) encapsulated in a facility information element within the ISDN Q.931 message. This is an optional capability specified in ISO 13868 as part of the SS-CONP feature; it provides "alerting on ring" only. "Alerting on busy" (an optional service that provides the name of the called user who cannot be reached) is not supported by this feature.
  - Call back/call completion—Support for ISO/IEC 13870: 2nd Ed, 2001-07 for Call Completion Supplementary Service:
    - Call Completion to Busy Subscriber (CCBS)
    - Call Completion on No Reply (CCNR)
  - Call diversion by reroute in addition to call diversion by forward switching—Call Diversion Supplementary Service ISO/IEC 13872 and ISO/IEC 13873, first edition 1995.
  - H.323 Annex M.1 (Q.SIG over H.323)—Delivers the Q.SIG feature set across intercluster trunks by the tunneling of Q.SIG messages by the Cisco CallManager over H.323, based on recommendations in the ITU recommendation for Annex M.1: "Tunneling of QSIG in H.323 07/2003". This development is limited to Q.SIG tunneling over Cisco CallManager H.323 intercluster trunks (both gatekeeper- and nongatekeeper-controlled). Interoperability between Cisco CallManager servers is the focus of this first release of Annex M.1.
  - Path replacement—Replace the existing time-division multiplexing (TDM) circuit(s) in use between two parties on an active call with new ones, to use TDM resources more efficiently. The Q.SIG path replacement feature will be implemented based on ISO/IEC 13863 – 2nd Ed. 1998 and ISO/IEC 13974 – 2nd Ed. 1999.
- Security features
  - Encryption to additional Cisco IP Phone 7940 and IP Phone 7960 devices, in addition to the already supported Cisco IP Phone 7970.
  - Encryption for MGCP gateways.
  - Encryption for SRST.
  - HTTPS for secure administration of Cisco CallManager. Supported by the following applications:
    - Cisco CallManager Admin

  - Cisco CallManager Service
  - Cisco CallManager User
  - RTMT
  - Cisco CallManager Trace Analysis
  - Cisco CallManager Service Trace Collection Tool
  - CDR Analysis and Reporting Tool
- Locally significant certificates on Cisco IP Phone 7970G systems.
- SSL for secure transport of user information between Cisco CallManager applications and directories. Supported by the following applications:
  - BAT
  - CAR
  - Cisco CallManager Admin User Pages
  - Cisco CallManager Admin IPMA Pages
  - Cisco CallManager User Pages and IP Phone Options Pages
  - Cisco Conference Connection
  - CTI Manager
  - Extension Mobility
  - IPMA
  - MLA

- T.38 fax support (H.323 only)—Support for T.38 fax when using H.323 gateways. When a fax call is placed, the call is initially established as a voice call. The gateways advertise capabilities during connection establishment. If both gateways support T.38, they will attempt to switch to T.38 upon fax tone detection by either gateway.
- Time of day, day of week, day of year routing/restrictions
  - Ability to assign time schedules to partitions to determine when a phone, gateway, translation pattern, or route pattern can be reached. The time schedule can be based on time of day, day of week, or day of year. Using partitions, this feature can be used to assign time schedules for outbound calls (TOD restrictions) or inbound calls (TOD routing).
- Toll fraud improvements
  - Ability to drop an ad-hoc conference when the conference originator hangs up
  - Ability to drop an ad-hoc conference when all internal callers hang up
  - Ability to block transfers from external trunks or gateways to external trunks or gateways
- Video enhancements
  - SCCP support for H.264 video

- Midcall video for Cisco VT Advantage
- Video display mode for IPVC 3.6plus
- Participant information for IPVC 3.6plus
- Dynamic H.323 addressing (E.164 addressing)

Administrative Enhancements

- BAT has been enhanced to provide support for the following:
  - FAC/CMC
  - CAPF configuration
  - Option for removing duplicate IP services
  - Option for deleting unassigned dialed number
  - Call coverage
  - Video
  - Call display restrictions
  - MLPP DOD enhancements
  - Security
  - Q.SIG alerting name
  - Trunk-to-trunk transfer and drop conference feature
  - CTI super provider
- Serviceability enhancements
  - HTTPS support for secure troubleshooting.
  - New services added to the service activation and control center pages
    - Cisco dial number analyzer is not shown
    - Cisco CAPF has been added.
  - Dialed number analyzer enhancements
- Security
  - CAPF improvements
    - Runs as a Windows NT service
    - Can be managed from Cisco CallManager Administration interface
    - CAPF device database integrated in to Cisco CallManager database
    - Support for automatic certificate install/upgrade

– Support for certificates as phone credentials for CAPF operation (MIC/LSC) [EXPAND ACRONYMS HERE ON FIRST INSTANCE]

– Support for external certificate authority; includes KEON CA, Microsoft CA

– BAT support for CAPF

– CTL client direct support for CAPF

– The phone find list has new search options.

- Toll fraud improvements—Ability to mark gateways and trunks as internal or external.
- Cisco Unity user integration—Allows easy integration between Cisco CallManager directory number or user admin pages and Cisco Unity voice mailbox configuration. This helps shorten the time it takes for a system administrator to complete the task of adding a phone and voice mailbox for a user.

ORDERING INFORMATION

Software Upgrades

A downloadable upgrade package is available for Cisco CallManager clusters that are already running Cisco CallManager Version 4.0 at:

http://www.cisco.com/cgi-bin/tablebuild.pl/callmgr-41

For all other upgrades or new Cisco CallManager 4.1 installations, Cisco CallManager CDs can be ordered.

Customers with a Cisco Software Application Support plus Upgrades (SASU) contract that is running Cisco CallManager versions 3.2, 3.3, or 4.0 who want to upgrade to Cisco CallManager Version 4.1 can order free upgrades using the Product Upgrade Tool (PUT) located at:

http://www.cisco.com/upgrade

For customers with no upgrade maintenance contract or upgrades from a previous version of Cisco CallManager one of the part numbers in Table 1 can be ordered.

Table 1. Cisco CallManager Part Numbers

SKU	Description
CM4.0-4.1-K9-UPG=	Cisco CallManager 4.0 to 4.1 upgrade
CM4.1-U-K9-7815SE=	Cisco CallManager 3.3 to 4.1 upgrade, MCS-7815s, 100-server user license
CM4.1-U-K9-7815=	Cisco CallManager 3.3 to 4.1 upgrade, MCS-7815s, 300-server user license
CM4.1-U-K9-7825SE	Cisco CallManager 3.3 to 4.1 upgrade, MCS-7825s, 100-server user license. Please note that Cisco CallManager 3.3 for a MCS-7825 with 100-server user license only shipped with MMIPC bundles MID-MKT-IPC-B and MID-MKT-IPC-C.
CM4.1-U-K9-7825=	Cisco CallManager 3.3 to 4.1 upgrade, MCS-7825s, 1000-server user license
CM4.1-U-K9-7835=	Cisco CallManager 3.3 to 4.1 upgrade, MCS-7835s, 2500-server user license
CM4.1-U-K9-7845=	Cisco CallManager 3.3 to 4.1 upgrade, MCS-7845s, 5000-server user license
CM4.1-U-K9-DL320=	Cisco CallManager 3.3 to 4.1 upgrade, HP DL320s, 1000-server user license
CM4.1-U-K9-DL380=	Cisco CallManager 3.3 - 4.1 upgrade, HP DL380s/1-CPU, 2500-server user license
CM4.1-U-K9-DL380D=	Cisco CallManager 3.3 - 4.1 upgrade, HP DL380s/2-CPU, 5000-server user license
CM4.1-U-K9-X306=	Cisco CallManager 3.3 to 4.1 upgrade, IBM xSeries 306, 1000-server user license

SKU	Description
CM4.1-U-K9-X345=	Cisco CallManager 3.3 to 4.1 upgrade, IBM xSeries 345/1-CPU, 2500-server user license
CM4.1-U-K9-X345D=	Cisco CallManager 3.3 to 4.1 upgrade, IBM xSeries 345/2-CPU, 5000-server user license

New Installations

For new Cisco CallManager installations, Cisco CallManager software and server hardware must be ordered. Table 2 lists these part numbers.

Table 2. New Cisco CallManager Order Numbers

Server Model	SKU	Number of Phones
HP DL320-G2	CM4.1-K9-DL320=	1000
HP DL380-G3 with a single CPU	CM4.1-K9-DL380=	2500
HP DL380-G3 with dual CPUs	CM4.1-K9-DL380D=	5000
IBM x306	CM4.1-K9-X306=	1000
IBM x345 with a single CPU	CM4.1-K9-X345=	2500
IBM x345 with dual CPUs	CM4.1-K9-X345D=	5000
Cisco MCS 7825H-3000 or Cisco MCS 7825I-3000	CM4.1-K9-7825=	1000
Cisco MCS 7835H-3000 or Cisco MCS 7835I-3000	CM4.1-K9-7835=	2500
Cisco MCS 7845H-3000	CM4.1-K9-7845=	5000
Cisco MCS 7845H-3000	LIC-CCM4.X-2500=	2500 additional; 7500 total

The following servers will support Cisco CallManager Version 4.1:

- MCS-7815-1000
- MCS-7815I-2.0-EVV1
- MCS-7815I-3.0-IPC1
- MCS-7825-1133
- MCS-7825-800
- MCS-7825H-2.2-EVV1
- MCS-7825H-3.0-IPC1
- MCS-7825I-3.0-IPC1
- MCS-7835
- MCS-7835-1000
- MCS-7835-1266
- MCS-7835H-2.4-EVV1
- MCS-7835H-3.0-IPC1
- MCS-7835I-2.4-EVV1
- MCS-7845-1400
- MCS-7845H-2.4-EVV1
- MCS-7845H-3.0-IPC1
- HP DL320*
- HP DL380/1CPU*
- HP DL380/2CPU*

- IBM x306*
- IBM x330 1.2GHz only*
- IBM x342*
- IBM x345/1CPU*
- IBM x345/2CPU*

*See http://www.cisco.com/go/swonly for details.

If you don't have one of supported servers, but wish to upgrade to Cisco CallManager Version 4.1, please refer to the server upgrade program can be found at:

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

Non-MCS servers that are supported with Cisco CallManager Version 4.1 can be found at:

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

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