Data Sheet

Cisco CallManager Version 4.0

Cisco IP Communications—a comprehensive system of powerful, enterprise-class solutions including IP telephony, unified communications, IP video and audio conferencing, and customer contact—helps organizations realize business gains by improving operational efficiencies, increasing organizational productivity, and enhancing customer satisfaction. Cisco CallManager—an integral component of the Cisco IP Communications system—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 the Cisco 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 Phones Support (TAPS); and the IP Manager Assistant (IPMA) application.

Key Features and Benefits

Cisco CallManager Version 4.0 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 1 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) ensures 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-browsable 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 Version 4.0 offer improved security, interoperability, functionality, supportability, and productivity as well as the new Video Telephony function. CallManager 4.0 has many security features that give CallManager users the ability to verify identity of the devices or servers that they communicate, ensure the integrity of data it is receiving, and provide privacy of communications via encryption. 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. 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. CallManager 4.0 introduces Video Telephony that includes support for SCCP and H.323 video and gives the same administration and user experience for voice and video. Common system administration and call behavior with existing audio phone calls help truly merge voice and video. New CallManager 4.0 features like Multiple calls per lines, call join, direct transfer, immediate divert, and ad-hoc conference list and drop any member improve the usability of the phones.

Specifications

Platforms

- Cisco Media Convergence Server (MCS)
- Cisco Integrated Communication System (ICS) 7750
- Selected third-party servers

Bundled Software

- Cisco CallManager Version 4.0 (call-processing and call-control application)
- Cisco CallManager Version 4.0 configuration database (contains system and device configuration information, including dial plan)
- Cisco CallManager administration software
- Auto Attendant—bundled with CallManager via the Extended Services CD.
- Cisco CDR Analysis and Reporting (CAR)—provides reports for calls based on CDR records. Reports that are provided include Calls on a user basis, Calls through gateways, Simplified Call Quality, and CDR search mechanism. In addition, CAR provides limited database administration; for example, deleting records based on DB size.
- Cisco 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, a client-server application, on a PC that runs Windows 98, Windows ME, Windows NT 4.0 (Service Pack 4 or greater), Windows 2000, or Windows 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.
- 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 CTI applications. It also 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, Security), Dr.Watson log, Cisco Update, Prog Logs, RIS DC Logs, SQL and IIS Logs.
- Cisco Conference Bridge—provides software conference bridge resources that can be used by CallManager.
- Cisco CTL Client—retrieves the CTL file from the Cisco TFTP server. It digitally signs the CTL file by using a security token and then updates the file on the Cisco TFTP server.
- 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 access this feature from the Cisco IP Phone Configuration web page.
- Cisco IP Telephony Locale Installer—provides user and network locales for Cisco CallManager through the Cisco IP Telephony Locale Installer, which adds 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 plugin 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 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, a standard programming interface for telephony applications, 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 Tool for Auto-Registered Phone Support (TAPS) loads a preconfigured phone setting on a phone.
- 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)
- AVVID XML Layer (AXL) Simple Object Access Protocol (SOAP) Application Programming Interface with performance and real-time information.
- CAC—intercluster and intracluster
- 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 CallManagers for scalability, redundancy, and load balancing
  - Maximum of 7,500 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 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
- H.323 interface to selected devices
- Hotline and private line automated ringdown (PLAR)
- Hunt groups—*broadcast, *circular, longest idle, linear
- Interface to H.323 gatekeeper for scalability, CAC, and redundancy
- Language support for client user interfaces (languages specified separately)
- *Multi-Level Precedence and Preemption (MLPP)
- 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 Real Time Monitor Tool (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
  - Cluster wide 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])
  - Basic call
  - ID services
  - General functional procedures
  - Call Diversion (SS-CFB (Busy), SS-CFNR (No Answer), SS-CFU (Unconditional))
  - Call Transfer by Join
  - Identification Restriction (CNIR (Calling Name Identification Restriction), COLR (Connected Line Identification Restriction), CONR (Connected Name Identification Restriction))
  - Loop prevention, Diversion Counter & Reason, Loop Detection, Diverted to Number, Diverting Number, Original Called Name& Number, Original Diversion Reason, Redirecting Name.
  - MWI—Message Waiting Indication
- 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)
  - Java Telephony API (JTAPI) and Telephony API (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

- Security
  - Configurable operation modes: non-secure or secure
  - Device authentication: Embedded X.509v3 certificate in new model phones. CAPF used to install locally significant certificate in phones.
  - Data Integrity: TLS cipher "NULL-SHA" supported. Messages appended with SHA1 hash of the message to ensure that the message is not altered on the wire and can be trusted.
  - Privacy: CallManager supports encryption of signaling and media.
  - 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: TFTP files (configuration and firmware loads) are signed with the self-signed certificate of the TFTP server. The CallManager system admin will be able to disable http and telnet on the IP phones
- Session Initiation Protocol (SIP) Trunk
- Survivable Remote Site Telephony (SRST)
- Third-party applications support
  - Broadcast paging—through foreign exchange station (FXS)
  - Simple Messaging Desktop Interface (SMDI) for message waiting indication
  - Hook-flash feature support on selected FXS gateways
  - TAPI 2.1 service provider (TSP) interface
  - JTAPI 2.0 service provider interface
  - Billing and call statistics
  - Configuration database API (Cisco AVVID XML Layer)
- 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 RFC2833
  - Annunciator
- Silence suppression, voice activity detection
- Simplified North American Numbering Plan (NANP) and non-NANP support
- Toll restriction—dial-plan partition
- Unified device and system configuration
- Unified dial plan
- Video (SCCP and H.323)

**Indicates new feature or service for Cisco CallManager Version 4.0*

Summary of User Features

- Abbreviated Dial
- Answer and answer release
- Autoanswer and intercom
- *Barge
- Callback 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 call by call (CLIR)
- 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
- Hypertext Markup Language (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 their managers; View manager status and calls; Create speed dials for frequently used numbers; Search for people in the Corporate/Call Manager 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 managers phone.
  - System capabilities: Multiple managers per assistant (up to 33 lines), redundant service
- Message waiting indication
- Multiparty conference—Ad hoc with add-on, meet-me features
- Multiple calls per line appearance
- Multiple line appearances per phone
- Music-on-hold
- 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-button data collaboration on softphone—chat, whiteboard, and application sharing
- 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
- WebDialer—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 4.0*

Summary of Administrative Features

- Application discovery and registration to SNMP manager
- AVVID XML Layer (AXL) Simple Object Access Protocol (SOAP) Application Programming Interface with performance and real-time information.
- Bulk Admin Tool (BAT)
- Call detail records (CDRs)
- CDR Analysis and Reporting Tool (CAR)
- 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
- Multilevel Administration Access (MLA)
- 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 computer telephony interface
- Time-zone configurable per phone
- Tool for Auto-Registered Phones Support (TAPS)
- Extended 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 4.0*

Cisco CallManager Version 4.0 Enhancements

User Feature Enhancements

- Abbreviated Dialing is a new softkey in CallManager 4.0 that allows expanded speed dial functionality. With 3 button dialing a user can have speed dials for up to 99 entries. A phone user can dial 2 digits and hit the "AbbrDial" to dial the configured speed dial. Configuration of speed dials is accomplished through the CallManager user speed dial page http://<ipaddress>/CCMUser/speeddial.asp.
- Attendant Console 1.3(1) which will be released with CallManager 4.0 will support the following new features: configuration tool, basic call queuing, broadcast hunting, support for shared lines, multiple calls on line, direct transfer and join, and the reimplementation of transfer to voicemail feature.
- Barge—allows a user on a shared line to barge into an existing call on the shared line. Barge existed in previous release but is enhanced in 4.0 so that if the person who had their call barged into hangs up, but original caller and the barger are left in a point to point call. The concept of conference barge or "cBarge" is introduced. cBarge creates an adhoc conference bridge that can be joined by multiple users that share the same line. A new service parameter "Party Entrance Tone" enables or disables tone for Barge and cBarge.
- Call Join allows a phone user to join multiple callers on a line into an ad-hoc conference. The user can select the callers and easily add them to an ad-hoc conference by hitting the Join Softkey.
- Conference List and drop any party: Participants in an ad-hoc conference can list the participants of the conference call. The conference originator will be able to select and drop any participant in the conference.

- Direct Transfer allows a user to directly transfer two parties on a line to each other. The user that is performing the direct transfer is not included in the new call. No conference bridge resources are used.
- Immediate Divert: A new “iDivert” softkey allows a user to transfer an incoming call to their voicemail. Also while on a call or on hold, a user can hit the iDivert softkey to transfer the call to voicemail. A system administrator can assign the iDivert softkey in the following states: Alerting, Call Active, On Hold.
- IP Manager Assistant service (IPMA application) now supports shared line support in addition to existing proxy line support.
  - Shared Line support
    - Manager features: Immediate divert or transfer, do not disturb, intercom, speed dials, barge, direct transfer, join.
    - Assistant features: Handle calls for their managers; View manager status and calls; Create speed dials for frequently used numbers; Search for people in the Corporate/Call Manager 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 managers phone.
  - System capabilities: Multiple managers per assistant (up to 33 lines), redundant service
- Malicious Call ID: A new MCID softkey is available for a system administrator to add to phones. The softkey can be added to phones in the cluster or phones in high risk areas. When the MCID button is hit, the CDR is marked as malicious, an alert can be generated via RTMT or SNMP trap can be generated. The connected network is notified through facility message with MCID invocation encoded in facility Information Element via Q.932, Q.951.7, and EN 300 130-1.
- Multiple calls per line: Multiple calls can exist on the same line. This eliminates the need to create multiple instances of the same directory in different partitions to allow users to share a line and still be able to receive and place multiple calls out of the same line. CallManager will now support up to 200 active calls on a single line. A new user interaction model has been introduced to allow a user to easily manage more than one call on the line and view calling name and number of the calls on the line. The system administrator will be able to on a line by line basis be able to do the following:
  - Provision a call forward no answer timer
  - Provision the maximum number of calls that will be allowed on the line
  - Provision the maximum number of incoming calls allowed on the line.
- Privacy: A system administrator can assign a Privacy button to a line button. The privacy button will toggle on and off when the user presses the button. A user can enable privacy so that when they pick up a new call, privacy is automatically invoked. They can also invoke privacy after a call has been established. When the privacy button is disabled (default), other users that share the line will see calling name and number info on the shared line. They will be able to Barge into the call as well. When privacy is invoked, the call instance does not appear on other instances of the shared line. A user can not barge into the call.
- Service URL: Any line or speed dial button can be used as a short cut to a selected XML services such as MyFastDials, Weathers, Stock Quote, etc. The system administrator can create a phone button template that assigns a button as a service URL button. The CallManager user or system administrator can assign a URL to the button and have service activation with one button push.

- WebDialer is an add-on application that enables click-to-dial functionality from web and desktop based applications. It is supported in CallManager 3.3(3), 4.0 and later. Supported platforms are on Windows platform with IE5.5 and later, Netscape 4.7x and later, Opera 7.0 and later, Mozilla1.3 and later. Two interfaces are supported HTTP/HTML and SOAP. HTTP/HTML is used by web based applications to create click to dial functionality. This gives the system administrator the ability to add click-to-dial to existing corporate directory HTTP/HTML pages. SOAP can be used by desktop applications to create add-ins to use create click to dial functionality. The CCMUser web page also includes a directory page that can be easily implemented. The system administrator can create dial rules for digit translation of numbers placed by WebDialer users.

System Capabilities Enhancements

- The AXL SOAP API now includes access to serviceability information. The information available includes CallManager perfmon counters, real-time information device status, and CTI information. Throttling of messages is included to protect core CallManager services running on the cluster.
- CiscoTSP (Telephony Service Provider) will support TAPI version 2.1 and has the following new features:
  - Dynamic Port Registration
  - Media Termination at RP
  - Transfer to VM
  - Barge/Privacy Release
  - Auto-install
  - Multiple Calls per Line Appearance
  - Shared Line Appearance
  - Direct Transfer
  - Call Join
  - Blind Transfer Enhancements
  - QoS Enhancements
  - Presentation Indication Flag.
- Configurable Call Forward Display: The system administrator can configure the information that is displayed when a phone is forwarded. The following information is available for display: Original Dialed Number (ODN) (Enable); Redirected Dialed Number (RDN) (Disable); Calling Line ID (CLID) (Enable); Calling Name ID (CNID) (Disable). Each of the listed parameters is configurable on a line appearance basis.
- H.323 enhancements for wholesale voice applications
  - SendTCS handling has been changed in CallManager 4.0 for H323.
  - Active Caps is supported for H.323. This allows improved H.323 performance. Active Caps is configurable and can be used per H323 GW\Trunk\Client. All Inter Cluster Calls are now active caps calls.
- Hunt Groups: Native hunt group capability in CallManager. Hunt groups support broadcast (ring all members), top down, circular, and longest idle hunting. A ring no answer timer can be applied to determine the time to wait before proceeding to the next point.

- JTAPI 2.0 will be released with CallManager 4.0 to provide support for many of the new Parche features. This support includes the following:
  - Multiple Calls per Line Appearance
  - Shared Line Appearance
  - Direct Transfer
  - Call Join
  - Barge/Privacy Release.
  - The following new JTAPI features are also in JTAPI 2.0: Changes to Blind Transfer
    - Transfer to VoiceMail
    - CTIPort Dynamic Registration
    - CTIRoutePoint with Media Termination
    - Terminal Event Filtering
    - Auto-update API
    - Auto accept for CTIPorts and Route Points
    - SelectRoute enhancements
    - Enhanced scalability—30,000 IP phones per cluster (configuration dependent)
- Multi-Level Precedence and Preemption (MLPP) allows the CallManager system administrator to assign a maximum precedence level to users. Users with higher precedence level authorization can originate precedence calls that can preempt calls of users with lower precedence. The MLPP functionality will play a tone to indicate to users on an IP to IP phone call to tell them that their call is being preempted. A user with higher precedence can also preempt a call with lower precedence that is being made out a gateway. The functionality also provides an Alternate Party Diversion functionality which will allow the CallManager administrator to allow the call to be routed to a different phone incase preemption is not possible or the preempted party does not answer the preempted call.
- Q.SIG Enhancements: CallManager supports the ISO variant of Q.SIG and supports the following. This added Q.SIG capability will allow CallManager to connect with other Q.SIG PBX systems and provide some feature transparency between the two systems.
  - Name Restriction (CNIR, CONR) configurable by gateway, route pattern or translation pattern.
  - Q.SIG MWI on/off application processor data units (APDU) to be passed from a Q.SIG capable CallManager cluster and another Q.SIG PBX. CallManager will both send and receive the MWI messages.
  - Q.SIG Call Diversion (Forward) by forward switch supported. CallManager supports SS-CFU (Call Forward Unconditional/ Call Forward All); SS-CFB (Call Forward Busy); and SS-CFNR (Call Forward No Reply/Call Forward No Answer). CallManager will pass forwarding information that can be presented to the users.
  - Diversion counters are used to protect against forwarding loops.
  - Transfer by join allowing connecting two legs of a call together.
- Security enhancements were made in the following areas:
  - CallManager configurable operation: CallManager can operate in two modes
    - Non-Secure Mode: default mode after installation.

    - Secure Mode: security features are enabled. This mode is turned on by using a USB eToken and a new Certificate Trust List (CTL) Client utility. A USB eToken can be purchased separately that will contain a Cisco rooted X.509v3 Certificate. This token is used to generate the CTL file for the phones as well as configuring the security mode of the cluster. Customers will be required to have two or more eTokens for redundancy.
- Device configurable operation: Devices that support security can be enabled or disabled on a device by device basis.
- Authentication: CallManager and IP phones will authenticate with each other to verify the identity of each component in the system.
    - Each CallManager server will have a self signed digital certificate.
    - New models of Cisco IP phones will have an embedded X.509v3 digital certificate that will be used for authentication.
    - When CallManager 4.0 releases, the 7970 phone will be the only phone that will contain a certificate installed during manufacturing.
    - A new security service called The Certificate Authority Proxy Function (CAPF) will be provided. Using CAPF, locally significant certificates can be installed on the 7940 and 7960. CAPF can generate its own certificates or work with other corporate or third-party Certificate Authorities.
    - All other phone models will not support these security features.
    - Phones containing a digital certificate will use the TLS (Transport Layer Security) Protocol for authentication and/or encryption of the SCCP signaling protocol and the SRTP (Secure RTP) protocol for authentication and encryption of the voice stream.
    - IP phones can be configured as non-secure or authenticated or encrypted.
- Modes of operation:
    - Authenticated Mode provides Data Integrity.
    - During registration, RSA signatures are used in TLS for mutual authentication between phones and CallManager.
    - All TLS packets are appended with an HMAC-SHA1 hash of the message to ensure that the message is not altered on the wire and can be trusted.
    - In Authenticated Mode, no encryption is used.
    - Phones will receive a Certificate Trust List (CTL) that will tell the phone what servers it can trust. This file is signed by a USB eToken and the CTL Client. The phones download this file using authenticated TFTP. In CallManager 4.0, the 7940, 7960 and 7970 can operate in Authenticated mode.
    - Encrypted Mode provides Data Confidentiality/Privacy and Integrity. Same functionality as authenticated mode plus encryption.
    - Signaling encryption will be implemented to ensure that information between CallManager and the phones is private.
    - IP phones that are configured with security mode "encrypted" will connect to CallManager using TLS cipher "AES128-SHA".

    - All signaling messages will be encrypted.
    - Enabling signaling encryption will cause a 15-20% performance impact to the CallManager.
    - Media between IP phones will be encrypted using SRTP.
    - Phones that are capable of doing media encryption will indicate its capability when the phone registers with the Call Manager.
    - The phones will use AES-128 Counter Mode to encrypt the RTP payload and authenticate the packet.
    - At the release of CallManager 4.0, only the 7970 will support media encryption. Other phone models that support encryption will be released at a later time.
  - Several phone security features will be implemented.
    - TFTP files (configuration and firmware loads) are signed.
    - The configuration is signed with the self-signed certificate of the TFTP server.
    - Firmware is signed using a Cisco-rooted certificate in manufacturing.
    - The phones will verify the validity of the file before accepting it.
    - The CallManager system admin will be able to disable http on the IP phones.
- Session Initiation Protocol (SIP) Trunk: The SIP Trunk is designed to connect CallManager to distributed SIP networks.
  - SIP standards RFC 2543 bis4 is supported; RFC 3261is partially supported. The SIP trunk uses RFC 2833 DTMF. The Cisco MTP is required for calls through the SIP trunk and provides 2833 DTMF relay functionality so that in band 2833 DTMF can interoperate with the SCCP, MGCP, and H.323 out of band DTMF that CallManager uses. The following devices are supported for SIP trunk interoperability with CallManager: Cisco SIP Proxy Server (CSPS), Cisco BTS 10200, Cisco PGW 2200, 26xx, 36xx, 37xx, 53xx, 54xx, 58xx, Microsoft Messenger, inter-cluster communications to another CallManager cluster.
  - Basic calls from SCCP phones can be made to SIP phones connected through a proxy server or vice versa. Once these calls are established the SCCP phones registered to CallManager can initiate the following supplementary services: Hold, Transfer, Ad Hoc Conference, Call Forward, Call Pickup, Call Park, Join, etc. The SIP endpoint can initiate Hold or Call Forward.
  - The SIP trunk provided identification services: RDNIS for voice mail support; Support for bi-directional passage of ID services on initial call (CLID/CLIR, CNID/CNIR); and Connected party information (COLD/COLR, COND/CONR). The restriction can be configured on the SIP trunk level, or call-by-call basis. The trunk level restriction will take precedence.
- Video enhancements have been made to H.323 and SCCP signaling and call processing within CallManager to allow video calls to be placed with the same user model as audio calls.
  - CallManager supports new video end points that will be released over the next few months.
  - SCCP Video endpoints inherit the functionality of audio calls which gives the user the same call model for both video and audio calls.
  - CallManager has the infrastructure to handle codec and video capabilities of the endpoints, bandwidth negotiation to determine if video/audio call can take place, single point of administration, management of media devices such as gateways and MCUs.

- CallManager provides a common control agent for signaling, configuration, and serviceability for voice or video end points.
- Planned endpoints include
  - SCCP-controlled voice-video IP phones
  - Expanding existing IP phone capability to include video
  - 3rd party SCCP Room and Executive Systems.
- Using CallManager 4.0, the system administrator has a single unified dial plan. CDR records now include video information. Call Admission control now supports video and will allow the administrator to configure WAN bandwidth for audio and video.
- Supplementary services such as Park, Hold, Resume, Transfer, Forward, Conference, Far End Camera Control, Music on Hold, Join, Direct Transfer, etc. are supported using video.
- CallManager 4.0 allows preservation of existing video infrastructure with support for H.323 video interoperability through H.323 logical trunks to IP/VC H.323 MCU and MCM H.323 gatekeeper. Interoperability with the IP/VC MCU Plus 3.1 will be available when it is released. For H.323 endpoints, common dialing, call forward, shared lines, and hunt groups will be supported. Full capabilities of H.323 end points are dependant on their support of Empty Capability Set (ECS).
- A video call will consist of audio (supports current codecs plus G.722 & G.728), video (H.261, H.263), and far end camera control. Common H.261 and H.263 parameters and typical values are bitrate (64k, 320k—can be any multiple of 100 bps), resolution (QCIF, CIF, Custom Picture Format), and frame rate (15 fps, 30 fps).
- CallManager 4.0 will interoperate with Cisco's current video portfolio:
  - Cisco IP/VC 3511 (Video Bridge)
  - Cisco IP/VC 3520 (V.35/BRI H.323/H.320 GW)
  - Cisco IP/VC 3525 (PRI H.323/H.320 GW)
  - Cisco IP/VC 3540 (Chassis based Bridge/GW unit)
  - IOS H.323 Gatekeeper.
- A complete set of serviceability capabilities will exist including perfmon counters for video, and Call Detail Record information.

- New Cisco IP Phone (Cisco IP Phone 7905, 7912, 7920, 7970) support

Administrative Enhancements

- Bulk Administration Tool (BAT)—BAT-TAPS5.0(1) will be released with CallManager 4.0. This version of BAT will have the following improvements: installation improvements, a wizard for user friendly bulk configuration, use of flexible CSV (data input file) format, a reporting utility, support for master phone templates, device summary report, validate feature that checks for dependencies on imported information, and custom file support.
- CDR Analysis and Reporting Tool (CAR) will have support for new hunting features, MLPP, Malicious Call, and new conference reports. CDR management has been enhanced to export CDR records for date range and allows date specific purging.

- Dialed Number Analyzer is a tool that will be released in the same time frame as CallManager 4.0. DNA can be used to troubleshoot unexpected call routing or to test new CallManager route plans. DNA will exercise configured CallManager dial plans and provide valuable information in troubleshooting problems. DNA will not be installable via a CallManager plugin in 4.0, but will be available in a separate download from CCO.
- Extension Mobility architecture has changed to use Tomcat. New service URLs have been defined. The service has been renamed to ‘Cisco Extension Mobility’. The CallManager service will now be controlled from the serviceability pages. Core extension mobility functionality has not changed.
- MLA has now been integrated with the CallManager installation. There is no longer a need to separately install MLA. The login page and logout menu have been removed. MLA gives the system administrator the ability to assign privileges to users to allow them full access, read only access, or no access to all of the CallManager configuration pages.
- Performance monitoring support has added several new counters to support the new features in CallManager. Perfmon has added counters for the application-controlled bridge device, annunciator, security, video, hunt lists, SIP, transcoders, MTP, software conference bridge, and hardware conference bridge. Some system performance counters have been renamed.
- Real Time Monitoring Tool (RTMT) has been enhanced to provide client-server functionality that did not exist in previous versions of CallManager. This delivers improved RTMT performance and usability. With the 4.0 version of RTMT, typical prepackaged alerts, monitors and reports are delivered. The prepackaged monitoring allows the system administrator to have a comprehensive view of the entire CallManager cluster with little RTMT configuration. The prepackaged alerts allow the system administrator to be alerted via email or email pager on typical CallManager, server, or service problems. The prepackaged reporting allows for some minor trending as well as troubleshooting problems after they occur.
- Shared resource and application management and configuration (Media Resources). CallManager Media Resources were enhanced to support Annunciator and RFC2833 MTP. The Annunciator is media device that is used to play voice prompts for MLPP and cases such as dialing a number that is not in the database. RFC 2833 MTP provides DTMF relay between out of band SCCP DTMF and In band SIP RFC2833 DTMF. Required when using the SIP Trunk.
- SNMP has added new MIB objects: ccmSystemVersion and ccmInstallationId. A new table to provide SIP Trunk information is added. The ability for dynamic population of new product types was added for quick implementation of new product types. Two new traps were added to alert the system administrator of a Malicious Call or Quality Report. SIP and H.323 information was added. Support for 7970 was added. Additional media device types have been added to keep pace with the media devices that are supported by CallManager.
- Tomcat 4.1.12 is installed and used as the web server for Cisco IPMA, Cisco WebDialer, Cisco Extension Mobility, Cisco CDR Analysis and Reporting etc. Tomcat was deployed to improve performance and give the system administrator additional control over web services through the Tomcat Web Application Manager Application. The Tomcat manager supports deploying a new web application that has the ability to do the following:
  – List the currently deployed web applications
  – Force an existing application to be reloaded
  – Stop and start existing applications.

- Trace Collection tool runs on a separate Windows 2000, 98 or XP PC and collects traces for CallManager services, application and system traces from the cluster. After collecting the traces, it zips the traces into a file that can be delivered to TAC for further troubleshooting.
- Trace setting tool allows the system administrator to set/reset the troubleshooting trace for selected services in the CallManager cluster from a single page.

Ordering Information

Description

- Base Cisco MCS installation—CD-ROMs, documentation shipped with ordered Cisco MCS servers
- Base Cisco ICS 7750 installation—Operating system, database, and documentation preinstalled to ordered Cisco ICS 7750 platform
- Upgrade CD-ROM package—Upgrade from Cisco CallManager 3.2(X) and 3.3(X) to Cisco CallManager 4.0

Part Numbers

- Base Cisco MCS installation—Ordered as software option to Cisco MCS servers; see Cisco MCS data sheets for detail. http://www.cisco.com/en/US/products/hw/voiceapp/ps378/index.html
- Base Cisco ICS 7750 installation—Ordered as component software to Cisco ICS 7750 platform; see Cisco ICS 7750 data sheet for detail
- Cisco CallManager 4.0 upgrade—CD-ROM package, including supporting software (operating system upgrade and database server upgrade) and documentation (part number provided separately). List Pricing by Platform:
  - CM4.X-U-K9-7815SE=
  - CM4.X-U-K9-7815=
  - CM4.X-U-K9-7825SE= (used on MMIPC bundles only)
  - CM4.X-U-K9-7825=
  - CM4.X-U-K9-7835=
  - CM4.X-U-K9-7845=
  - CM4.X-U-K9-DL320=
  - CM4.X-U-K9-DL380=
  - CM4.X-U-K9-DL380D=
  - CM4.X-U-K9-X345=
- Base installation to selected third-party servers CD-ROM package. See http://www.cisco.com/go/swonly for details.

Cisco IP Communications Services and Support

Cisco IP Communications services and support reduce the cost, time, and complexity associated with implementing a converged network. Cisco and its partners have designed and deployed some of today's largest and most complex IP communications networks—meaning that they understand how to integrate an IP communications solution into your network.

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