Cisco CallManager Version 3.1

Cisco CallManager is the software-based call-processing component of the Cisco enterprise IP telephony solution and a product 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's open telephony application programming interfaces (API). Cisco CallManager is installed on the Cisco Media Convergence Server (MCS). Cisco CallManager software is shipped with a suite of integrated voice applications and utilities, including the Cisco WebAttendant, which is a software-only manual attendant console; a software-only conferencing application; the Bulk Administration Tool (BAT); and the Administrative Reporting Tool (ART).

Key Features and Benefits

Cisco CallManager version 3.1 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 up to 10,000 users per cluster. 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 and scalability. Call admission control 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.

Enhancements in version 3.1 include:

- Music-on-hold service
- JTAPI/TAPI call processing redundancy support
- Survivable Remote Site Telephony support
- Simple Network Management Protocol (SNMP) and serviceability
- Wideband audio coder/decoder (codec) support
- Mobile NETwork (MNET) Global System for Mobile Communications (GSM) pico-cell support
- Database API
- Overlap sending and T1-CAS support in a variety of voice-over-IP (VoIP) gateways

A more complete summary of enhancements is included below.

Additional RAM may be required in Media Convergence Servers to support existing and enhanced services in CiscoCallManager 3.1.

Specifications

Platforms

- Media Convergence Server (MCS)
- Integrated Communications Server (ICS-7750)

Bundled Software

- Cisco CallManager version 3.1 (call processing and call-control application)
- Cisco CallManager version 3.1 configuration database (contains system and device configuration information, includingdial plan)
- Cisco CallManager Administration software
- Cisco Conference Bridge
- Cisco WebAttendant
- Bulk Administration Tool (BAT)
- Administrative Reporting Tool (ART)

System Capabilities Summary

- Alternate Automatic Routing (AAR)
- Attenuation/gain adjustment per device (phone and gateway)
- Automated bandwidth selection
- Automated Route Selection (ARS)
- Call Admission Control (CAC)—intercluster and intracluster
- Comfort Noise Generation (Cisco IP Phones 7900 Series, Catalyst® 6XXX gateways, Cisco IOS® gateways)
- 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 2500 devices per Cisco CallManager server
  - Maximum BHCC 50,000 per Cisco CallManager server
  - Eight Cisco CallManager servers per cluster
  - Maximum BHCC 125,000 per Cisco CallManager cluster
  - Maximum 10,000 devices per cluster
  - Intercluster scalability to 100 sites or clusters through H.323 gatekeeper
  - Intracluster feature transparency
  - Intracluster management transparency
- FAX over IP—G.711 pass-through

- H.323 interface to selected devices
- Hot line and Private Line Automated Ringdown (PLAR)
- Interface to H.323 gatekeeper for scalability and call admission control
- Multilocation—dial plan partition
- Multiple ISDN protocol support
- Multiple remote CallManager platform administration and debug utilities
  - Real-time and historical application performance monitoring through operating system tools and SNMP
  - Monitored data collection service
  - Remote terminal service for off-net system monitoring and alerting
  - Telnet relay application
  - Platform and database debugging tools—Supports **show** command using command-line interface
  - Real-time event monitoring and presentation to common syslog
  - Call trace utility
  - Browse to onboard device statistics
- Multisite (cross-WAN) capability with intersite call admission control
- Dial plan partitioning
- Off-premise station (OPX)
- Outbound call blocking system
- Out-of-band DTMF signaling over IP
- PSTN failover—AAR
- Redundancy
  - Triple Cisco CallManager redundancy per device (phones, gateway, *applications) with automated failover and recovery
  - Trunk groups
- Survivable Remote Site Telephony (SRST)
- Third-party applications support
  - Broadcast paging—through FXS
  - SMDI for message waiting indication
  - Hook-flash support on selected FXS gateways
  - TAPI 2.1 service provider (TSP) interface
  - JTAPI 1.3 service provider interface
  - Billing and call statistics
- Shared resource/application management and configuration
  - Transcoder resource
  - Conference bridge resource
  - Topological association of shared resource devices (conference bridge, MoH sources, transcoders)
- Silence suppression, voice activity detection
- Simplified North American Numbering Plan (NANP) and Non-NANP support
- SMDI interface for message waiting indication
- Toll restriction—dial plan partition
- Unified device and system configuration
- Unified dial plan

*Indicates new feature or service for Cisco CallManager version 3.1

Summary of User Features

- Answer/answer release

- Auto-answer
- Call connection
- Call coverage
- Call forward—all (off-net/on-net)
- Call forward—busy
- Call forward—no answer
- Call hold/retrieve
- Call park/pickup
- Call pickup group-universal
- Call status per line (state, duration, number)
- Call waiting/retrieve
- Calling Line Identification (CLID)
- Calling party name identification (CNID)
- 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 phone
- *Extension mobility support
- Hands-free, full-duplex speakerphone
- HTML help access from phone
- Last number redial (off-net/on-net)
- Message waiting indication
- Multiparty conference—Ad-hoc with add-on, Meet-me
- Multiple line appearances per phone
- *Music-on-hold
- Mute capability from speakerphone and handset
- *On-hook dialing
- Operator attendant—Web-browser interface, loop key notification, logon/logoff, busy/available, left/right hand access, headphone access, busy lamp field, direct station select, drag and drop transfer, call status (state, duration, and number)
- Privacy
- Real-time QoS statistics through http browse to phone
- Recent dial list—calls to phone, calls from phone, auto-dial, and edit dial
- Single button data collaboration on SoftPhone—chat, whiteboard, and app sharing
- Single directory number, multiple phones—bridged line appearances
- Speed dial—multiple speed dials per phone
- Station volume controls (audio, ringer)
- Transfer—with consultation hold
- User-configured speed dial and call forward through Web access
- Web services access from phone
- *Wideband audio codec support

*Indicates new feature or service for Cisco CallManager version 3.1

Summary of Administrative Features

- Application discovery and registration to SNMP manager
- Call Detail Records (CDR)
- Call forward reason code delivery
- Centralized, replicated configuration database, distributed Web-based management viewers
- Configurable and default ringer WAV files per phone
- *Configuration database API
- Database automated change notification
- Date/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 MAC address
- Dynamic Host Configuration Protocol (DHCP) block IP assignment—phones and gateways
- Dialed number translation table (inbound/outbound translation)
- Dialed Number Identification Service (DNIS)
- Enhanced 911 service
- H.323-compliant interface to H.323 clients, gateways, and gatekeepers
- JTAPI 1.3 computer telephony interface
- LDAP version 3 directory interface to selected vendor's LDAP directories
  - Active Directory
  - Netscape Directory Server
- *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 DNISSNMP 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/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
- Zero cost automated phone moves
- Zero cost phone adds

*Indicates new feature or service for Cisco CallManager version 3.1

Cisco CallManager Version 3.1 Enhancements

User Feature Enhancements

• *Wideband audio codec support in Cisco IP Phone 7910, 7940, and 7960 handsets (16 kHz sampling, 16-bit sampleresolution)

• *Extension mobility support—user login, then profile and class of service restriction delivery to Cisco 7940 and7960IPphones

• *Intelligent dialing support RDNIS missed, received, placed calls from Cisco 7940 and 7960 IP Phones

• *On-hook dialing

• *Support for personal directory service at Cisco 7940 and 7960 IP Phones

System Capabilities Enhancements

• *Shared resource management enhancements

– *Conference bridge, transcoder, and music-on-hold topological association

– *Single shared resource per cluster

– *More efficient use of transcoder devices for IP WAN calls

• *Phone XML tag enhancements

– *Idle URL

– *Audio streaming

• *Support for low-cost hardware conference bridge

• *Music-on-hold multicast and unicast streaming service

– *Music streaming service for “user” hold and “network” hold

– *51 sources per MCS server

Fifty continuously looping .WAV file sources

One real-time streaming source

Each source configurable as either unicast or multi-cast stream

– *Support for audio streaming to selected devices

Gateways (unicast and multicast)—VG200 (MGCP), Access Gateway Module (Catalyst 4000 card), Catalyst 4224, Catalyst 4248, WS-6608-T1/E-1 (Catalyst 6000 8-port PRI card), DE-30+, and DT-24+

Gateways (unicast only)—AT-2/-4/-8, AS-2/-4/-8, and all other Cisco IOS VoIP gateways including 1750, 2600/3600, 5300, 58XX, and 72XX

IP phones (unicast, multicast)—Cisco 7910, 7940, and 7960

IP phones (unicast only)—7935, 12 SP+, and 30 VIP phones

Cisco SoftPhone (unicast only)

– *Maximum 250 simultaneous on-hold streaming sessions per server

– *Multiple server instances for application scalability

– *Multiple server instances for server load balancing and redundancy

– *G.711, G.729A, and wide-band audio codec support

– *Off-line audio translation utility

• *T1-CAS support in selected VoIP gateways

– *VG200, Access Gateway Module (Catalyst 4000 card), DT-24+, WS-6608-T1 (Catalyst 6000 8-port card), and Catalyst4224

• *T1/E1 Primary Rate Interface (PRI) support for selected VoIP gateways

– *VG200

• *Extend MGCP protocol support to selected VoIP gateways

– *WS-6608-T1/E1 (Catalyst 6000 8-port card), DE-30+, DT-24+, Access Gateway Module (Catalyst 4000 card), Catalyst4224, and Catalyst 4248

- *Configuration database API
- *Support for centralized voice messaging application with multiple Cisco CallManager clusters
  - *RDNIS—outbound through H.323, MGCP gateways, H.323 interface to uOne version 4.2.2S
  - *Call forward number expansion
  - *Call forward reason code delivery
- *Serviceability enhancements
  - *Monitoring and alarm collection utility (RIS) with SNMP, http external interfaces
  - *Simple Administrative Serviceability Tool (AST) for device monitoring
  - *Broadened SNMP MIB support for devices, applications
  - *Real-time diagnostics for devices (gateways and IP phones) through html server on devices
- *Survivable Remote Site Telephony support
- *Telephony Call Distributor (TCD) hunt group enhancement—least-used hunt group logic
- *WebAttendant enhancement—pop-to-top on new call
- *Gateway survivability/availability enhancement through MGCP signaling/control protocol
- *Cisco CallManager redundancy for TAPI/JTAPI applications
- *Overlap sending support through PRI ISDN
  - *non Cisco IOS gateways—DT-24+, DE-30+, and WS-6608-T1/E1
  - *Cisco IOS gateways (MGCP only)—VG200, AGM, and Catalyst 4224
- *Cisco MNET-JetCell integration

Administrative Enhancements

- *Enable/disable speaker/microphone mute per IP phone
- *ART enhancements—new gateway, route list/group inventories
- *BAT enhancements—support for new CallManager 3.1 configuration database scheme
- *Device, user, and line list enhancements

Ordering Information

Description

Base MCS installation—CD-ROMs, documentation shipped with ordered MCS servers

Base ICS-7750 installation—Operating system, database, and documentation preinstalled to ordered ICS-7750 platform

Upgrade CD-ROM package—Upgrade from Cisco CallManager 3.0(X) to CallManager 3.1

Part Numbers

Base MCS installation—ordered as software option to MCS servers. See MCS data sheets for detail.

Base ICS-7750 installation—ordered as component software to ICS-7750 platform. See ICS-7750 data sheet for detail.

Cisco CallManager 3.1 upgrade—CD-ROM package includes supporting software (operating system upgrade and database server upgrade) and documentation (Part Number provided separately)

Cisco Service and Support Solutions

The Cisco AVVID (Architecture for Voice, Video and Integrated Data) IP Telephony Service and Support Solutions are designed for one purpose—to ensure customer success by delivering a suite of proactive services. Rapid deployment, core, and advanced service and support covering the entire network life cycle can be delivered directly by Cisco Systems, or through its ecosystem of best-in-class partners.

The award-winning Cisco Service and Support offerings provide presales network audit planning, design consulting, network implementation, operational support, and network optimization. By including service and support when purchasing Cisco AVVID IP Telephony Solutions, customers can confidently deploy a converged network architecture using Cisco expertise, experience, and resources.

Corporate Headquarters
Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
www.cisco.com
Tel: 408 526-4000
800 553-NETS (6387)
Fax: 408 526-4100

European Headquarters
Cisco Systems Europe
11, Rue Camille Desmoulins
92782 Issy-les- Moulineaux Cedex 9
France
www-europe.cisco.com
Tel: 33 1 58 04 60 00
Fax: 33 1 58 04 61 00

Americas Headquarters
Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
www.cisco.com
Tel: 408 526-7660
Fax: 408 527-0883

Asia Pacific Headquarters
Cisco Systems Australia, Pty., Ltd
Level 17, 99 Walker Street
North Sydney
NSW 2059 Australia
www.cisco.com
Tel: +61 2 8448 7100
Fax: +61 2 9957 4350

Cisco Systems has more than 200 offices in the following countries and regions. Addresses, phone numbers, and fax numbers are listed on the Cisco Web site at www.cisco.com/go/offices

Argentina • Australia • Austria • Belgium • Brazil • Bulgaria • Canada • Chile • China PRC • Colombia • Costa Rica • Croatia • Czech Republic • Denmark • Dubai, UAE Finland • France • Germany • Greece • Hong Kong SAR • Hungary • India • Indonesia • Ireland • Israel • Italy • Japan • Korea • Luxembourg • Malaysia • Mexico The Netherlands • New Zealand • Norway • Peru • Philippines • Poland • Portugal • Puerto Rico • Romania • Russia • Saudi Arabia • Scotland • Singapore • Slovakia Slovenia • South Africa • Spain • Sweden • Switzerland • Taiwan • Thailand • Turkey • Ukraine • United Kingdom • United States • Venezuela • Vietnam • Zimbabwe

4/01LW2243