CHAPTER 4

Voice and IP Communications

Voice and IP Communications at a Glance

Product	Features	Page
Cisco CallManager 4.1	• The software-based call processing and call control component of Cisco's IP Telephony solution • Resides on Cisco Media Convergence Servers (MCS), or selected third-party servers (CallManager 4.1)	4-3
Cisco Unity—Unified Messaging and Voice Mail	Voicemail and unified messaging system delivers all messages into single inbox for access via phone, email or Internet	4-7
Cisco CallManager Express	• IOS ® -based call control - supported on standard Cisco multiservice routers (1751,1760,2600XM,3600,3700, and IAD 2400 series) • Configurable Key-system or Hybrid-PBX feature set for up to 100 users • Intuitive GUI for system administration and administration • Supports all Cisco standard IP phones	4-9
Cisco Unity Express	• Integrated voice mail and automated attendant services locally delivered for either Cisco CallManager or Cisco CallManager Express solution • Leverages the data infrastructure; supported on broad range of Cisco access routers and new Cisco Integrated Services Routers • Increases employee productivity and customer service for businesses of various sizes and operational models	4-10
Cisco Emergency Responder	• Works with Cisco CallManager to automatically provide E9-1-1 features in North America, and is compatible with any emergency number including 112 in Europe, 999 in UK, and 000 in Australia. • Dynamically tracks the location of IP phones, routes emergency calls to the appropriate E9-1-1 network, and provides the current location information to E9-1-1 call center dispatchers. • Provides real-time alert notifications to on-site or contracted security groups, to facilitate a timely response to emergency situations.	4-11
Cisco IP Contact Center (IPCC) Enterprise Edition	Delivers intelligent call routing, network-to-desktop CTI, and multi-channel contact management to contact center agents over an IP infrastructure. • Pre-routing and post-routing of contacts • Ability to handle voice, Web and e-mail contacts, as well as inbound and outbound campaigns • Computer telephony integration, universal queue, IVR integration and remote agent support • Complete supervisory, administration and reporting for the contact center	4-12
Cisco IPCC Hosted Edition	Offers full IP-based contact center services from its network. The only required customer premises equipment (CPE) are Cisco IP Phones, agent PCs, and a Cisco AVVID network enabled for quality of service (QoS).	4-13
Cisco IP Contact Center Express Edition	• A software-based ACD, IVR, and CTI application for departmental, enterprise branch, or small to mid-sized contact centers with Cisco IP Telephony networks based on Cisco AVVID. • Provided in Standard, Enhanced and Premium versions • Offers an open systems platform allowing ease of installation and configuration • Provides a graphically driven workflow editor providing a common interface for creating interactions, or call flows, and creates business logic between IVR and ACD functions	4-14
Cisco IP IVR	Provides a feature-rich foundation for the creation of an IP-based interactive voice response (IVR) system that is open and expandable. • Provides an IP-empowered application-generation environment for self service and call treatment solutions • Offers Web-based activation and administration • HTML, XML, ODBC, JDBC integrations supported • Offers optional Automatic Speech Recognition and Text to Speech • Extendable through custom Java steps	4-15
Cisco Customer Voice Portal	Offers a Web-based prompt, collect, queuing, and call control service, using standard Internet technologies • Offers a Web-based, self-service IVR option enabling customers to; pay a bill, order products and track delivery, locate a dealer, schedule a pickup, change name and address information, make travel arrangements, check payment status, receive notification of unusual activity, or request literature or product information • Supports automatic speech recognition (ASR) and text to speech (TTS) • Integrates fully with both traditional TDM and IP-based contact centers	4-16

Product	Features	Page
Cisco ICM Enterprise Edition	Enables interaction with a contact center via phone, Web, voice over IP (VoIP), text chat, or e-mail. • Provides centralized management control over customer contacts • Cisco Pre-Routing and Post-Routing; Customer Profile Routing • Computer telephony integration, Web collaboration, E-mail management, IVR integration • Inbound and outbound campaign management • Enterprise reporting • Distributed fault tolerance • Multi-carrier, multi-vendor capabilities	4-16
Cisco Personal Assistant	Software application allows users to browse voicemail, dial by name, and conference from any phone using voice commands instead of telephone keypad via speech recognition	4-18
CiscoWorks Manager IP Telephony Environment Monitor	A suite of management applications that helps ensure the readiness and manageability of converged networks that are supporting VoIP and IP telephony traffic and applications. The bundle includes: Voice Health Monitor; Default Fault Manager; CiscoView • CCO Downloadable Modules: IP Phone Information Utility; IP Phone Help Desk Utility; Fault History Manager	9-17
Cisco IP Communicator	• Softphone client that gives computers the capabilities and features of a Cisco IP phone • A supplemental telephone that provides access to phone extensions and services outside the office • Unified administration with Cisco IP phones and automatic software updates makes it easy to manage	4-17
Cisco VT Advantage	• Brings video telephony functionality to Cisco IP phones • Adds video to phone calls through the use of a personal computer without button-pushing or mouse-clicking • Video telephony is now just a phone call	4-18
Cisco Survivable Remote Site Telephony Software (SRST)	• IOS software that runs on local branch office router provides IP Telephony backup redundancy for IP phones in that office when IP phones detect that WAN is down or/and CallManager is unreachable	4-20
Cisco Conference Connection (CCC)	• Enables enterprises to bring geographically dispersed employees and customers together to facilitate meetings and collaboration. • Provides a cost-effective and time-efficient method of doing business without the hassle of travel.	4-21
Cisco IP Phones IP 7900 Series	An exciting, distinctively stylish, and pure Voice over IP phone portfolio to meet the wide range of business communication needs at affordable prices • Display-based technology provides ease-of-use • Integrated inline power and 2-port Ethernet Switch provides end-to-end infrastructure integration • Rich application environment enabled by open APIs based on XML	4-21
Cisco ATA Series of Analog Telephone Adaptors	Turns any analog telephone into an IP telephone. Each of the two voice ports supports independent telephone numbers, providing two separate lines. • Interoperable with multiple standards including H.323, SIP, MGCP and SCCP • Enables analog devices, such as phones and fax machines, to support Voice over IP services by converting the analog signal into an IP signal	4-23
Cisco MCS 7800 Series Media Convergence Servers	High availability server platform for Cisco IP telephony systems • Turnkey solution, includes CallManager or other software • For large- and medium-sized enterprise IP telephony deployments	4-24
Cisco Voice Gateways[1]	The Cisco VG248 dedicated voice gateway provides connectivity between IP networks and legacy telephony systems/PSTN • VG248 supports up to 48 Foreign eXchange Station (FXS) ports for connecting analog telephones. • Fully manageable by Cisco CallManager, a CLI interface via Telnet, or via SNMP	4-26
Cisco MeetingPlace	• Integrated voice, video and web conferencing • On-network conferencing solution that integrates directly into an organization's private voice and data networks • Cost-effective and secure with a superior user experience	4-27
Cisco IP/VC 3500 Series Videoconferencing Products	• Videoconferencing over IP solution • Cost-effective, easy-to-manage • Translates between H.323 and H.320 systems • Management and Quality of Service	4-28
Cisco IP/TV 3400 Series Video Servers	• High-quality video communications over enterprise networks • Support live and scheduled video broadcasts, recording of live broadcasts and video on demand • MPEG-1, MPEG-2, and ISMA MPEG-4 support • Enables e-training, corporate communications, business TV, and distance learning	4-29
Cisco Voice Bundles[2]	Cisco 1760, 2600, 2800, 3700, and 3800 Voice Bundles for handling toll bypass applications or PSTN/PBX to IP Connectivity for Cisco CallManager deployments.	4-30
Cisco CallManager Express Bundles [2]	Cisco 1760, 2600, 2800, 3700, and 3800 Cisco CallManager Express Bundles for customers wanting a complete all in one router IP Telephony Solution for the Small Office or Enterprise Branch location.	4-30

Product	Features	Page
Cisco Survivable Remote Site (SRST) Bundles[2]	Cisco 1760, 2600, 2800, 3700, and 3800 Voice Bundles for customers' branch locations where they need data connectivity plus local PSTN call handling with a centralized Cisco CallManager deployment.	4-31
Cisco V3PN Bundles[2]	Cisco 1760, 2800 and 3800 Voice Bundles for remote branch locations needing VPN and IP Telephony using Cisco CallManager Express or SRST with centralized Cisco CallManager Express deployment.	4-32

1. Cisco's full line of multiservice routers also provide analog and digital voice gateway functionality through use of network modules and voice interface cards. Please see the 1700, 2600XM, 3600, 7200, 5x00 series in Chapter 1—Routers, as well as Chapter 7—Access Products.
2. Bundles provide a discount compare to ordering the parts individually. With each bundle, the customer needs to choose the country explicit PSTN module required plus any additional features or modules for the applications.

Cisco CallManager 4.1

Cisco CallManager Version 4.1 provides a scalable, distributable, and highly available enterprise IP telephony call-processing solution. 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.

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.

Cisco CallManager 4.1 has 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.

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.

Key Features

- Supplementary and enhanced services such as hold, transfer, forward, conference, multiple line appearances, automatic route selection, speed dial, last-number redial, and other features are extended to IP phones and gateways
- Capabilities enhancements are achieved though software upgradeability, avoiding expensive hardware costs traditional to legacy PBX systems

- Cisco CallManager Attendant Console—This Web-enabled application supports the traditional role of a manual attendant console and allows the attendant to quickly accept and dispatch calls to enterprise users. An integrated directory service provides traditional busy lamp field (BLF) and direct station select (DSS) functions for any line in the system. It monitors the state of every line in the system without requiring hardware-based line monitoring devices, thereby saving costs
- Software-only applications such as the Cisco Interactive Voice Response system, Cisco IP Contact Center, Cisco Automated Attendant, and Cisco SoftPhone are applications that interact with the CallManager through telephony APIs
- Video telephony gives users the ability to place video calls using the same user model as a voice call; Traditional telephony features available for video calls such as hold, resume, park, transfer, conference; Provides a common dial plan and administration between voice and video
- Comprehensive and wide range of security features; Encryption and Authentication are available across the most widely used Cisco IP phones and gateways; Secure CallManager system and user administration
- New call coverage/call routing features allow calls to be routed based on time of day, source of call and destination of call to handle the complex routing requirements of many of our customers
- By supporting a wide set of Q.SIG features, installable along with a TDM based PBX to allow peaceful co-existance and migration to an all IP environment while delivering some of the key features needed between the TDM PBX and CallManager; Path replacement and call completion allow Cisco CallManager to integrate with other Q.SIG compatible systems; H.323 Annex M.1 support now gives users improved feature transparency between CallManager clusters
- Easily configure a Cisco Unity voice mail box while configuring an IP phone for that user using Cisco Unity User Integration

Specifications

Feature	Cisco CallManager 4.1[1]
Platforms	Media Convergence Server (MCS)
	Selected third-party servers
Pre-Installed Software	Auto attendant
	Bulk Administration Tool (BAT)
	CDR Analysis and Reporting (CAR) tool
	Cisco Attendant Console
	Cisco CallManager Administration 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 Conference Bridge
	Cisco Customer Directory Configuration Plugin
	Cisco Dialed Number Analyzer—Serviceability tool that analyzes the dialing plan for specific numbers.
	Cisco IP Manager Assistant (IPMA)
	Cisco IP Phone Address Book Synchronizer
	Cisco IP Telephony Locale Installer
	Cisco JTAPI
	Cisco TAPS
	Cisco Telephony Service Provider
	Real Time Monitoring Tool RTMT

Feature	Cisco CallManager 4.1[1]
Sample Subset of System Capabilities	Anti-Virus checker certification Cisco Intrusion Detection System (IDS) Host-Based Sensor certification Cisco Security Agent (CSA) bundled with CallManager. Data integrity with TLS cipher "NULL-SHA" Device Authentication with X. 509v3 certificates. Distributed Call Processing Encryption of signaling and media on many Cisco IP endpoints and gateways. Forced Authorization Codes/Client Matter Codes H.323 FastStart H.323 scalability improvements - 1,000 H.323 calls per CallManager server in cluster H.323 T.38 fax support Multi-level Precedence and Pre-emption (MLPP) Secure HTTP (HTTPS) support Session Initiation Protocol (SIP) trunk Survivable Remote Site Telephony (SRST)
Summary of Administrative Features	Application discovery and registration to SNMP manager Automated Alternate Routing Groups AXL SOAP API with performance and real-time informationBulk Administration Call Back 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 Dialed Number Identification Service (DNIS) Dialed number translation table (inbound/outbound translation) Distinctive ring per line Dynamic Host Configuration Protocol (DHCP) block IP assignment-phones and gateways Enhanced 911 service H.323-compliant interface to H.323 clients, gateways, and gatekeepers Individual line Call Waiting Alert Configuration Integration with Unity Voicemail JTAPI 1.2 computer telephony interface LDAP version 3 directory interface to selected vendor's LDAP directories • Active Directory • Netscape Directory Server Manager Assistant Mappable softkeys MGCP signaling and control to selected Cisco VoIP gateways Multilevel Administration Access (MLA) Native supplementary services support to Cisco H.323 gateways Network Specific Facilities Paperless phone DNIS-display driven button labels on phones Performance Monitor Performance monitoring SNMP statistics from applications to SNMP manager or to operating system Q.SIG Support 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 Toll Fraud preventation Video telephony XML API into IP phones (794X/6X) Zero cost automated phone moves; Zero cost phone adds

Feature	Cisco CallManager 4.1[1]
Summary of User Features	Abbreviated Dialing or speed dials
	Answer/answer release
	Auto-answer/intercom
	Barge
	Call back
	Call connection
	Call coverage
	Call forward-all (off-net/on-net); Call forward-busy; Call forward-no answer
	Call hold/retrieve
	Call Join
	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)
	Calling Line Identification Restriction (CLIR)
	Direct inward dial (DID; Direct outward dial (DOD)
	Direct Transfer
	Directories-missed, placed, received calls list stored on selected IP phones
	Directory dial from phone-corporate, 2personal
	Distinctive ring (on-net vs. off-net); Distinctive ring per phone
	Drop last conference party (ad-hoc conferences)
	Extension Mobility
	Extension mobility support
	H.264 video Codec support
	Hands-free, full-duplex speakerphone
	HTML help access from phone
	Immediate Divert to voicemail
	Last number redial (off-net/on-net)
	List and drop conference participate
	Manager-assistant service (IPMA application)
	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
	Service URL—Single button access to IP phone service
	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
	Video support for continuous presence
	Video support of H.323 endpoints, gateways, and MCUs
	Video Telephony
	VT Advantage with mid call invocation.
	Web services access from phone
	Wideband audio codec support—proprietary 16-bit resolution, 16 kHz sampling rate codec

1. Additional RAM may be required in Media Convergence Servers to support existing and enhanced services in Cisco CallManager 4.1

For More Information

See the Cisco CallManager Web sites: **http://www.cisco.com/go/callmgr**

Cisco Unity—Unified Messaging and Voice Mail

Cisco Unity is a powerful Unified Communications server that provides advanced, convergence-based communication services and integrates them with the desktop applications business professionals use everyday, improving customer service and productivity. Designed for enterprise-scale organizations, Cisco Unity delivers unified messaging that gives subscribers the ability to access and manage messages and calls from anywhere, at any time, regardless of device or media type. Subscribers can listen to e-mail over the telephone, check voice messages from the Internet, and if a fax server is present, forward faxes to any local fax machine. Cisco Unity voice messaging features robust automated attendant functionality that includes intelligent routing, and easily customizable call screening and message notification options. Cisco Unity supports localized versions in multiple languages and supports multiple languages on a single system.

Cisco Unity's optional digital networking module enables connectivity to other Cisco Unity servers at the same site via the LAN or remotely via WAN. Digital networking gives users the ability to send subscriber-to-subscriber messages anywhere in the world.

Cisco Unity supports both Cisco CallManager and leading legacy telephone systems—even simultaneously—to help smooth the transition to IP telephony and protect existing infrastructure investments. Built on a scalable platform, it uses streaming media and an intuitive HTML browser-style system administration interface. Costs are minimized when Cisco Unity's server architecture is truly unified with an organization's data network.

Key Features

- Architecture allows IT staff to set one back-up procedure, one message storage policy, and one security policy
- Enhanced scalability allows up to 72 ports per server; up to 7,500 subscribers per server; or a total of 250,000 users in an Exchange environment or 100,000 users in a Domino environment up
- Support for Exchange 2000/Active Directory as the single message store and directory; AMIS-A and VPIM interoperability for Exchange systems.
- Enhanced networking for large deployments
- Support for multiple CCM clusters; ability to light Message Waiting Indicators
- With Exchange/Domino off-line, utilizes pre-MTA queue to take messages and give basic message access; Support for Lotus Domino as the single message store
- Fault-tolerant system tools—robust security, file replication, and event logging
- Scalable, high-availability hardware platforms—Server options with multiple processors, redundant fans, redundant power supplies and RAID hard disk drives
- Support for Windows 2000 in a mixed/native mode
- Unity Inbox/VMI (Visual Messaging Interface) is an Internet Explorer-based voice mail inbox providing unified messaging
- Unity Bridge provides advanced message interchange functionality with legacy Avaya/Octel voice mail systems—unlocking proprietary networking to deliver open standards-based IP migration

Specifications

Feature	Cisco Unity 4.0
Unity Voice Mail (VM) and Unified Messaging (UM) Possible Configurations	16, 32 and Max sessions Configured for CallManager or configured for legacy PBX/dual integration[1]
Options	Voice Mail; Voice Mail with Multi-lingual option; Unified Messaging with Text-to-Speech (TTS) option Unified Messaging with Multi-lingual option; Exchange or Domino; AMIS for Exchange; VPIM for Exchange; Unity Inbox/Visual Messaging Interface; Failover for Exchange; Unity Bridge for Exchange

1. Contact your Cisco Software Sales Representative for integration information.

Selected Part Numbers and Ordering Information[1]

Cisco Unity Servers	
MCS-7815-I1-ECS1	MCS 7815 tower; Unity and Unity Bridge; 1GB; Win2K
MCS-7825-I1-ECS1	MCS 7825-IBM rack; 1GB RAM; sATA RAID; Win2K
MCS-7825-H1-ECS1	MCS 7825-HP rack; 1GB RAM; sATA RAID; Win2K
MCS-7835-I1-ECS1	MCS 7835-IBM; rack; 2GB; RAID 1; Win2K
MCS-7835-H1-ECS1	MCS 7835-HP; rack; 2GB; RAID 1; Win2K
MCS-7845-I1-ECS1	MCS 7845-IBM; VM-6HDD; rack; 4GB; RAID 1(x3) DUAL CPU; Win2K
MCS-7845-I1-ECS2	MCS 7845-IBM; UM-4HDD; rack; 4GB; RAID 1(x2) DUAL CPU; Win2K
MCS-7845-H1-ECS1	MCS 7845-HP; VM-6HDD; rack; 4GB; RAID 1(x3) DUAL CPU; Win2K
MCS-7845-H1-ECS2	MCS 7845-HP; UM-4HDD; rack; 4GB; RAID 1(x2) DUAL CPU; Win2K
UNITY-MCS-ML-DAT=	Hot-Swap tape drive for MCS-7835H and MCS-7845H
DAT-7835-I1	Hot swap IBM DAT tape drive for MCS-7835-I1
DAT-7845-I1	Hot swap IBM DAT tape drive for MCS-7845-I1
MCS-EXT-DAT	External DAT tape drive for MCS servers
MCS-EXT-SDLT	External SDLT drive for MCS servers
MCS-EXT-SCSI	SCSI card option for external DAT drive
UNITY-EXP-CHAS=	Expansion chassis
Cisco Unity 4.0 Unified Messaging and Voicemail Software	
UNITYU4-50USR-E=	Unity UM Exchange, 50 users, 16 session, 2 TTS
UNITYU4-100USR-E=	Unity UM Exchange, 100 users, 16 session, 2 TTS
UNITYU4-200USR-E=	Unity UM Exchange, 200 users, 16 session, 2 TTS
UNITYU4-300USR-E=	Unity UM Exchange, 300 users, 16 session, 2 TTS
UNITYV4-50USR=	Unity VM, 50 users, 16 sessions
UNITYV4-100USR=	Unity VM, 100 users, 16 sessions
UNITYV4-200USR=	Unity VM, 200 users, 16 sessions
UNITYV4-300USR=	Unity VM, 300 users, 16 sessions
UNITY-DVD-4.0=	UNITY 4.0 DVD set
UNITY-CD-4.0=	UNITY 4.0 CD set
Cisco Unity Integrations	
UNITY-D/41U-LS	D/41 Universal PCI - US/CANADA
UNITY-D/41U-EU	D/41 Universal PCI - Europe, Aus, NZ
UNITY-D/120U-LS	D/120JCT Universal PCI rev 2 - US/CANADA
UNITY-D/120U-EU	D/120JCT Universal PCI Euro - Europe, Aus, NZ
UNITY-PBXLINK-1	PBXLink, 4-24 ports for Lucent; 4-20 ports for Nortel
UNITY-PBXLINK-2	PBXLink, 4-40 ports for Lucent; 4-48 ports for Nortel
UNITY-PIMG-DIG	PBX IP Media gateway for digital integrations
UNITY-RLM-GW-2.4	Rolm GW for Unity Integration with Rolm PBX
Cisco Unity Options	
UNITY-AMIS	Unity, AMIS-A networking
UNITY-ADDL-LANG	Support for an additional language. May order up to 17.
UNITY-RS-ML	Unity, one session Real Speak TTS
UNITY-VPIM	Unity, VPIM - Per Server
Cisco Unity Bridge	
UNITY-BRIDGE-3.0	Unity Bridge 3.0, requires Unity 4.0.3 or later
UNITY-BRIDGE-2.1	Unity Bridge 2.1, requires Unity 4.0.2 or earlier

1. This is only a small subset of all parts available via URL listed under “For More Information.” Some parts have restricted access or are not available through distribution channels.

For More Information

See the Cisco Unity site: **http://www.cisco.com/go/unity**

Cisco CallManager Express

Cisco CallManager Express is a solution embedded in Cisco IOS Software that provides call processing for Cisco IP phones. This solution enables the large portfolio of Cisco multiservice access routers to deliver features similar to those of a key system or low-end private branch exchange (PBX), thereby enabling deployment of a cost-effective, highly reliable, IP Communications solution for the small office. Customers can now scale IP telephony to a small site or branch office with a solution that is very simple to deploy, administer, and maintain. The Cisco CallManager Express solution provides customers with a low-cost, reliable, feature-rich solution for a deployment of up to 100 users.

Key Features

- Cost-effective operations through a single, integrated voice-and-data platform for all branch office needs
- Provides integrated IP telephony services for multiservice access routers that allows customers to deploy one device in their office to address all their business needs, simplifying management, maintenance, and operations, and delivering a lower total cost of ownership (TCO)
- Robust set of key system and low-end PBX capabilities—Delivers a robust set of telephony features for the small office, and delivers unique value-added capabilities through Extensible Markup Language (XML) that enhance the productivity of the end user and of the business, and that cannot be delivered by traditional solutions
- Investment protection and ease of upgrade to centralized call-processing solutions—Simple software or firmware upgrade capabilities
- Remote maintenance and troubleshooting using Cisco IOS Software command-line interface (CLI) or Web-based GUI—Acts as a standalone call-processing engine for IP phones located in the branch office

Specifications

Feature	Cisco IP Manager Assistant
Platform	Cisco IAD2400, 1751, 1760, 2600XM, 2691, 3640, 3640A, 3660, and 3700 Series Multi-Service Routers.
Phones supported	Cisco 7960 7940, 7935, 7912, 7905, 7902, 7910, ATA
IOS Images	Cisco Call Manager Express 3.1 supported in 12.2(15)ZJ3

For More Information

See the Cisco CallManager Express Web site:
http://www.cisco.com/en/US/products/sw/voicesw/ps4625/index.html

Cisco Unity Express

Cisco ® Unity Express provides voice mail and automated attendant services specifically for the small and medium office or branch. It's application is delivered on either a network module or advanced integration module which are supported on a variety of Cisco access router platforms and the Cisco Integrated Services Router family.

In a Cisco CallManager environment, Cisco Unity Express provides local storage and processing of voice mail and automated attendant services for the branch office, thereby alleviating WAN bandwidth and Quality of Service concerns. Cisco CallManager Express and Unity Express combine to create the Cisco IP Communications Express solution which provides voice, data, and telephony management services integrated on a single, router-based platform. Both solutions offers a core set of phone, voice mail and automated attendant features to meet everyday business needs, while providing the rich telephony feature sets that customers have grown to expect.

Key Features

- Cost-effective operations through a single, integrated voice-and-data platform for all small and medium office or branch needs
- Boosts the professionalism, productivity and customer service available to the small and medium office through built-in automated attendant and voice mail capabilities; Customizable automated attendant to support business hours, day-of-week treatment and multi-level, nested automated attendant menus with up to 5 separate automated attendants per system
- Both standard and advanced voice mail features such as remote message access, save and delete, forward, pause, fast forward or rewind messages, alternate greetings, zero-out to designated alternate phone number or extension, per user storage allocation, plus much, much more
- Support for VPIMv2 for voice mail message exchange with other Cisco Unity or Unity Express systems.
- Remote maintenance and troubleshooting using Cisco command-line interface (CLI) or Web-based GUI

Specifications

Feature	Cisco CallManager Express
Hardware	NM-CUE or AIM-CUE, fully self-contained modules with onboard storage, memory and processing supporting a variety of mailbox densities, storage capacities and concurrent sessions to meet the needs of every small and medium office or branch
Software	Cisco Unity Express release 1.1.2and Cisco IOS 12.3.4T for NM-CUE, Cisco IOS 12.3.8T for the AIM-CUE
Platform Supported	Cisco 2600XM, 2691,3700 Series Multi-Service Routers and Cisco 2800, 3800 Integrated Services Routers

For More Information

See the Cisco Unity Express Web site: **http://www.cisco.com/go/ccmecue**

Cisco Emergency Responder

Cisco Emergency Responder revolutionizes enterprise telephony support for E9-1-1 in North America, E1-1-2 in Europe, and other emergency telephone services across the globe. Traditional PBX E9-1-1 implementations in North America support "automatic location identification" of emergency callers through daily manual database update processes, which limit the frequency of location updates and increase the likelihood of update errors. The Cisco Emergency Responder software application works with Cisco CallManager to automatically track the location of Cisco IP phones in enterprise campuses, route emergency calls to an appropriate public safety answering point (PSAP), and provide the location of the caller to the Public Safety Answering Point (PSAP).

Cisco ER performs these functions without requiring tedious manual database updates after phone moves/adds/changes, which significantly reduces the time, headcount, and costs associated with traditional PBX E9-1-1 maintenance. While some vendors may automate location updates, they still require manual PBX configuration changes to trigger the updates. The Cisco ER solution, when coupled with the automated phone moves/adds/changes features in Cisco CallManager, is the first in the industry to completely automates the phone move process while maintaining E9-1-1 and location data integrity.

In addition, Cisco ER can use email/pager messaging, telephone calls, and auto-refreshing webpage updates to notify on-site security operations personnel and third-party agencies of emergency calls in progress.

Key Features

- Meets and exceeds traditional E9-1-1 requirements
- Automates all user and phone moves, adds, and changes; Enables users and phones to move an unlimited number of times per day
- Avoids the expense and burden of daily PS-ALI record uploads
- Avoids daily error-prone documentation and database updates
- Enables quicker and more effective emergency response from onsite personnel and public agencies
- Provides configuration auditing to facilitate responsible change management and investigative or legal processes
- Provides call history logs for capacity planning, management of emergency call abuse, and incident documentation
- Compatible with any emergency number

Specifications

Feature	Cisco Emergency Responder
Supported Platform	Cisco Media Convergence Server MCS-7815I-3.0-IPC1, MCS-7825H-3.0-IPC1, MCS-7825I-3.0-IPC1, MCS-7835H-3.0-IPC1, MCS-7835I-3.0-IPC1, MCS-7845H-3.0-IPC1
System Capacity	A single Cisco Emergency Responder server can (depending on server platform) support 30,000 phones, 120,000 Ethernet switch ports and 10,000 manually entered endpoints such as analog or proprietary phones or H.323 clients. Cisco recommends a second Emergency Responder server to form a fully redundant Cisco Emergency Responder Group with the same capacity and increased availability compared with a single Cisco Emergency Responder server. Larger campuses and distributed systems are supported via a network of Cisco Emergency Responder groups called a Cisco Emergency Responder Cluster.
Configurable Elements	
Cisco CallManager	Call routing and digit manipulation to forward user-initiated emergency calls and PSAP return calls to and from Cisco Emergency Responder as appropriate
Cisco Emergency Responder	System administration interface-for access to all configuration components or oversight of outsourced vendors LAN administration interface-for IT LAN group or an outsourced vendor Emergency Response Location (ERL) administration interface-for IT telecom group or an outsourced vendor

Feature	Cisco Emergency Responder
Other Components	Configure e-mail account on a Simple Mail Transfer Protocol (SMTP) Internet mail server for use by Cisco Emergency Responder Configure an email-to-pager gateway, or use an email paging service Configure a PS-ALI transfer application provided by the PS-ALI database service provider (often requires a dialup modem connection) Provision an E9-1-1 capable voice trunk (Centralized Automated Message Accounting [CAMA] or Primary Rate Interface [PRI]) through a local exchange carrier
Supported Switches[1]	Cisco Catalyst 2900 XL, 2950, 3500 XL, 3550, 3560, 3750, 4000, 4500, 5000, 5500, 6000, and 6500 Series switches. Cisco 3700 and 3800 series routers. (See the latest Release Notes for specific devices supported within each series.)

1. Check for updates on CCO, and following is list of tested switch platforms at time of printing

Selected Part Numbers and Ordering Information[1]

Cisco Emergency Responder

SW-CER1.2-SVR=	Cisco Emergency Responder software (MCS platforms) on CD, including 100 user licenses
KEY-CER1.2-100=	CER 1.2 user license for 100 phones
KEY-CER1.2-500=	CER 1.2 user license for 500 phones
KEY-CER1.2-1K=	CER 1.2 user license for 1,000 phones
KEY-CER1.2-5K=	CER 1.2 user license for 5,000 phones

1. Redundant user licenses are not required when ordering redundant CER servers for a single CER group.

For More Information

See the Cisco Emergency Responder Web site: **http://www.cisco.com/go/cer**

Cisco IP Contact Center (IPCC) Enterprise Edition

Cisco IP Contact Center (IPCC) Enterprise Edition delivers intelligent contact routing, call treatment, network-to-desktop computer telephony integration (CTI), and multi-channel contact management over an IP infrastructure. By combining multi-channel automatic call distributor (ACD) functionality with IP telephony in a unified solution, it enables companies to rapidly deploy a distributed contact center infrastructure.

Cisco IPCC Enterprise Edition segments customers, monitors resource availability, and delivers each contact to the most appropriate resource anywhere in the enterprise. The software profiles each customer using contact-related data such as dialed number and calling line ID, caller-entered digits, data submitted on a Web form, and information obtained from a customer profile database lookup. At the same time, the system knows which resources are available to meet the customer's needs based on real-time conditions (agent skills and availability, interactive voice response (IVR) status, queue lengths, and so on) continuously gathered from various contact center components.

It provides a state of the art VoIP contact center solution that allows seamless integration of inbound and outbound voice applications with Internet applications including real-time chat, Web collaboration and e-mail. This integration allows for unified capabilities, enabling a single agent to support multiple interactions simultaneously regardless of the communications channel the customer has chosen. Since each interaction is unique and may require individualized service, Cisco provides contact center solutions to manage each interaction based on virtually any contact attribute. Furthermore, Cisco can bridge the gap between TDM and IP infrastructures, providing a seamless integration of voice, chat, e-mail, and Web collaboration applications over both of these technology platforms.

Key Features

- Full Scalability from less than a hundred to thousands of seats
- Multi-channel interaction—Web collaboration with chat and callback, email, voice mail and fax routing
- Multi-site Contact Centers support; CRM Integration
- Cradle-to-grave contact call detail records
- Common agent and supervisor desktops across all Cisco customer interaction management products
- Pre-defined and custom historical reports; Real-time reports integrated in the agent and supervisor desktops
- Support for custom call treatment for calls in queue includes support for music in queue and custom messaging; Standard screen pop allows any caller-entered information to be popped to the agent
- Support for agent/supervisor interaction via chat; pre-define agent-supervisor messages

Selected Part Numbers and Ordering Information[1]

Cisco IP Contact Center	
IPCE-Bundle	Requires the purchase of one IPCC Server license. This license entitles the user to deploy the following necessary components: (redundant) Router; (redundant) Logger; (redundant) CallManager Peripheral Gateway(s); (redundant) IVR Peripheral Gateway(s) for connection to Cisco IVR systems ISN and IP-IVR (Non-Cisco IVRs require Third part IVR port licenses); Administrative Workstation(s); Historical Database server(s); WebView server and WebView user connections; Internet Script Editor (ISE) server and ISE user connections; Application Gateways; (redundant) CTI Server for 3rd party CTI connections (only for non-agent desktop application)
IPCE-SVR	IPCC ENTERPRISE SERVER LICENSE

1. This is only a small subset of all parts available via URL listed under “For More Information.” Some parts have restricted access or are not available through distribution channels.

For More Information

See the Cisco IPCC Enterprise Edition Web site: **http://www.cisco.com/go/ipcc**

Cisco IPCC Hosted Edition

The Cisco IPCC Hosted Edition is suitable for both enterprise customers and service providers. For service providers, the IPCC Hosted Edition creates a new, high-margin service revenue stream for incumbent providers as well as new service carriers. The service provider hosts the contact center infrastructure software-which is shared by multiple business customers-in its central office or data center. Subscribing business customers can have IP infrastructures, TDM infrastructures, or a combination of the two. For the enterprise customer with multiple branch offices or divisions the value is a centralized contact center infrastructure with the ability to offer services to its various divisions or satellite offices. The solution includes a suite of integrated services that can be introduced all at once or incrementally.

Key Features

- Virtual call center—Calls are routed to contact center agents independent of their location, a service especially appealing to businesses with branch offices or home agents
- Network routing with computer telephony integration-Network-based ACD is combined with computer telephony integration (CTI) services
- Network IVR—IVR functionality is located in the network to provide information to callers or to collect information from callers before they speak with a live agent
- Intelligent call routing—Calls are routed between contact centers based on call context information (dialed number and caller ID), agent availability, and customer information from databases

Selected Part Numbers and Ordering Information[1]

Cisco IPCC Hosted Edition

IPCH-BUNDLE	The IPCC Hosted Edition bundle consists of two separately licensed sets of components: The Base licenses for the shared infrastructure, such as the NAM, the network interfaces, shared network IVRs, CICMs and Hosted IPCC Agent licenses. "(redundant) Router "(redundant) Logger "(redundant) CallManager Peripheral Gateway(s) "(redundant) IVR Peripheral Gateway(s) for connection to Cisco IVR systems ISN and IP-IVR (Non-Cisco IVRs require Third part IVR port licenses) "Administrative Workstation(s) "Historical Database server(s) "WebView server and WebView user connections "Internet Script Editor (ISE) server and ISE user connections "Application Gateways "(redundant) CTI Server for 3rd party CTI connections (only for non-agent desktop application)

1. This is only a small subset of all parts available via URL listed under “For More Information.” Some parts have restricted access or are not available through distribution channels.

For More Information

See the Cisco IPCC Enterprise Edition Web site: **http://www.cisco.com/go/ipcchosted**

Cisco IPCC Express Edition

Cisco IPCC Express Edition meets the needs of departmental, enterprise branch, or small to medium-size companies planning to deploy an entry-level or mid-market contact center solution. Designed for formal and informal contact centers, IPCC Express delivers sophisticated call routing, contact management, and administration features with the ease of installation, configuration, and application hosting.

Cisco IPCC Express Edition simplifies business application integration, eases agent administration, increases agent flexibility, and provides efficiency gains in network hosting. Provides independence in agent location, improves agent scalability, and provides powerful automatic call distributor (ACD) features, such as conditional routing, call-in-queue and expected-wait-time messages, enterprise data displays, real-time data, and historical reporting together with integrated Interactive Voice Response (IVR) services.

Cisco IPCC Express Edition is provided in three versions, Standard, Enhanced and Premium. Upgrading from standard to enhanced or premium versions provides additional features such as skills-based routing and priority queuing, computer telephony integration (CTI) screen pop and supervisory features such as silent monitor, coaching, barge in, and intercept and full self-service application support. All Cisco IPCC Express Edition solutions are tightly integrated with Cisco AVVID and Cisco CallManager.

Key Features[1]

- Browser-based Cisco IPCC Express administration is fully integrated with Cisco CallManager browser-based administration
- Cradle-to-grave contact call detail records
- Standard screen allows any caller-entered information to be popped to the agent
- Pre-defined or custom historical reports
- Real-time reports within the agent and supervisor desktops
- Full support for agent/supervisor interaction via chat; Ability to pre-define agent-supervisor messages
- Full IP call queue points and prompt; Collect voice interaction capabilities
- Optional Automatic Speech Recognition (ASR) and Text to Speech (TTS) capabilities
- Support for custom call treatment such as music for calls in queue

Selected Part Numbers and Ordering Information[1]

Cisco IPCC Express Edition	
IPCX-3.Y-STANDARD	PCX 3.5 or later 3.Y Standard All-in-One (Servers & SW)
IPCX-3.Y-ENHANCED	PCX 3.5 or later 3.Y Enhanced All-in-One (Servers & SW)
IPCX-3.Y-PREMIUM	PCX 3.5 or later 3.Y Premium All-in-One (Servers & SW)

1. This is only a small subset of all parts available via URL listed under “For More Information.” Some parts have restricted access or are not available through distribution channels.

For More Information

See the Cisco IPCC Express Edition Web site: **http://www.cisco.com/go/icd**

Cisco IP IVR

Cisco IP IVR, an interactive voice response (IVR) solution, provides a feature rich foundation for the creation of an IP-based Cisco IP IVR that is open and expandable.

Key Features

- Written in Java to provide customer flexibility
- Multimedia (voice/data/Web) IP-empowered application-generation environment
- Support for optional Automated Speech Recognition (ASR) and Text-to-Speech (TTS)
- Support for VoiceXML
- Multiple Language support
- Cisco IP IVR can be located in anywhere the IP network
- Offers web-based activation and administration
- Flows (the IP IVR applications) are stored in an industry standard LDAP directory
- Cisco IP IVR is sold with Cisco CallManager and can be co-resident on the same server as CallManager or can function on a separate, dedicated media convergence servers (MCSs) or Cisco-approved customer provided server
- Packages available to scale up to 300 ports

1. Abbreviated list of IPCC Features

Cisco Customer Voice Portal

The Cisco Customer Voice Portal redefines and sets the standard for extending automated self-service beyond the limits of traditional interactive voice response (IVR) systems and voice portal platforms by providing a return on investment (ROI) to lower operational costs in the contact center and help protect technology investments from obsolescence to provide a smooth and consistent multi-channel customer experience. It integrates with both traditional time-division multiplexing (TDM) and IP-based contact centers to provide an unparalleled call-management and call-treatment solution with a self-service IVR option that can use information available to customers on the corporate Web server. With support for automated speech recognition (ASR) and text-to-speech (TTS) capabilities, callers can obtain personalized answers to increasingly complex questions and can conduct business in new and innovative ways-all without the costs of interacting with a live agent.

The Customer Interaction Network is a suite of innovative multi-channel IP communications applications for the contact center that is accessible from any point on the network and integrated with core business processes.

Selected Part Numbers and Ordering Information[1]

Cisco Customer Voice Portal	
ISN-APSSVRLIC	ISN APPLICATION SERVER LICENSE
ISN-APSSES-T1-L	ISN APPLICATION SERVER SESSION - TIER 1
ISN-APSSES -T2-L	ISN APPLICATION SERVER SESSION - TIER 2
ISN-APSSES -T3-L	ISN APPLICATION SERVER SESSION - TIER 3
ISN-APSSES -T4-L	ISN APPLICATION SERVER SESSION - TIER 4
ISN-APSSES-LC=	Part number to order additional Application Server session licenses without ordering a server license

1. This is only a small subset of all parts available via URL listed under “For More Information.” Some parts have restricted access or are not available through distribution channels.

For More Information

See the Cisco Internet Service Node Web site: **http://www.cisco.com/go/cvp**

Cisco ICM Enterprise Edition

Through a combination of multi-channel contact management, intelligent routing, and network-to-desktop computer telephony integration (CTI), the Cisco ICM Enterprise Edition segments customers, monitors resource availability, and delivers each contact to the most appropriate resource anywhere in the enterprise. To complete this transaction, the software profiles each customer using contact-related data such as dialed number and calling line ID, caller-entered digits, data submitted on a Web form, or information obtained from a customer profile database lookup. Simultaneously, the system monitors the resources available in the contact center to meet customer needs, including agent skills and availability, interactive-voice-response (IVR) status, and queue lengths.

The Cisco ICM Enterprise Edition gives your customers the choice to interact with your contact center via phone, Web, voice over IP (VoIP), text chat, or e-mail. It provides centralized management control over customer contacts, allowing users to implement a single set of business rules that uniformly address customer needs independent of contact channel or resource location.

This combination of customer and contact center data is processed through user-defined routing scripts that graphically reflect business rules-enabling routing each contact to the optimum resource anywhere in the enterprise. Wherever an agent is based, the system delivers a unique and rich set of call event and customer profile data to the targeted desktop as a contact arrives, personalizing service and maximizing efficiency. Throughout the process, carrier-class, distributed fault tolerance from the network to the desktop ensures uninterrupted operation in the contact center.

Selected Part Numbers and Ordering Information[1]

Cisco ICM Enterprise Edition

ICME-BUNDLE	Requires the purchase of at least one ACD PG and one ICM Agent license. Functionality included: Monitor 3rd party ACD (call and agent) activity; Call routing, including pre-routing from a carrier network and post-routing from an ACD; Historical Database server(s); WebView server and WebView user connections; Internet Script Editor (ISE) server and ISE user connections; Application, SQL and ICM-to-ICM Gateways; Redundant deployment of all licensed components; Integrations through the ICM's CTI interfaces with 3rd party devices that are not intended for agent desktop usage; Connections to Cisco IVR systems (IP-IVR and ISN) and all related functions an ICM can perform, such as monitoring, reporting and routing.

1. This is only a small subset of all parts available via URL listed under “For More Information.” Some parts have restricted access or are not available through distribution channels.

For More Information

See the Cisco ICM Enterprise Edition Web site: **http://www.cisco.com/go/icmee**

Cisco IP Communicator

Cisco IP Communicator is a software-based application that gives computers the capabilities and features of Cisco IP phones, providing high-quality voice calls and connection to company services while on the road, in the office, or from wherever users may have access to the corporate network. When used remotely, users have access to the same familiar phone services they have in the office. While registered to the Cisco CallManager call processing system, Cisco IP Communicator has the features and functionality of a full-featured Cisco IP Phone including the ability to transfer calls, forward calls, and conference additional participants to an existing call.

Key Features

- Easy to use—Feature parity with Cisco’s advanced IP phones, IP phone services support (XML) to deliver company specific applications to the desktop, and auto-detection of Cisco VPN client
- Premium voice quality—Audio Tuning Wizard for setting audio levels properly, advanced jitter buffer and packet loss concealment algorithms, and echo suppression and noise cancellation
- Easy to manage—Unified administration with Cisco IP phones and automatic software upgrades; system administrators can provision as they would any other Cisco IP Phone

Selected Part Numbers and Ordering Information

Cisco IP Communicator

SW-IPCOMM-E1	Cisco IP Communicator Software
SW-CCM-UL-IPCOMM-E	Station User License for Cisco CallManager

For More Information

See the Cisco IP Communicator Web site:
http://www.cisco.com/en/US/products/sw/voicesw/ps5475/index.html

Cisco VT Advantage

Cisco VT Advantage brings video telephony functionality to Cisco IP Phones, providing users with the ability to easily add video to their communications experience. It's software application and Cisco VT Camera, an included USB camera, allows a personal computer co-located with a Cisco IP Phone to add video to phone calls without requiring any extra button-pushing or mouse-clicking. When registered to Cisco CallManager, the Cisco VT Advantage-enabled IP Phone has the features and functionality of a full-featured IP video phone.

Key Features

- Consistent user experience for voice and video—users to talk as they normally would on the phone with call features such as call forward, transfer, conference, hold, and mute are initiated through the phone while available video is viewed through the PC
- Provides ease of use, high-quality video, and more value at a lower cost
- Easy to manage—Unified administration with Cisco IP phones and automatic software upgrades

Selected Part Numbers and Ordering Information

Cisco VT Advantage	
CVT-ADV-E1=	Cisco VT Advantage (software and camera included)

For More Information

See the Cisco VT Advantage Web site:
http://www.cisco.com/en/US/products/sw/voicesw/ps5662/index.html

Cisco Personal Assistant

Cisco Personal Assistant streamlines communications with personal call rules, speech recognition, an optional enhanced text-to-speech engine and productivity services for IP phones. As an integral part of Cisco AVVID (Architecture for Voice, Video and Integrated Data), it interoperates with Cisco CallManager and scales to meet the present and future needs of your employees. Users can access voice mail, dial by name, and conference from any telephone using speech recognition instead of the telephone keypad. The Web-based and telephone user administration interfaces allow users to forward and screen calls in advance or in real time. The phone services enable users to check e-mail, voice mail, calendar, and personal contact information using the large, pixel-based LCD and interactive soft keys on the Cisco IP Phone 7940 or 7960.

Key Features

- Ubiquitous Access: Cisco Personal Assistant with Speech Recognition and IP Phone Productivity Services integrate with Cisco CallManager, Cisco Unity, and Microsoft Exchange within Cisco AVVID to streamline communications
- Automatic Speech Recognition (ASR): Speech recognition interface allows users to utilize simple voice commands to perform tasks such as retrieval, replying, recording, and deletion of voice messages; Entries can be dialed from personal address books or the corporate enterprise Lightweight Directory Access Protocol (LDAP) directory; Users can synchronize their Microsoft Exchange contact lists with their personal address books for quick name-dialing and ad-hoc group conferencing; Access to sensitive features such as voice mail is controlled by user authentication
- Manage Inbound and Outbound Calls (Rules-Based Routing): Using a Web interface to create rules, users can forward and screen calls based on caller identification, time of day, and meeting schedules; With "follow me," a special rule that uses speech recognition, users can forward all calls to a phone number immediately; Users can activate sets of pre-created rules from any telephone
- CalendarView: Users can keep track of appointments right on the IP phone, directly from the Microsoft Exchange server with no synchronization necessary. In addition, users can choose to be notified of an upcoming event on the phone display or by pager
- MailView: Cisco Personal Assistant presents users with access to e-mail and Cisco Unity voice-mail messages in the inboxes on the corporate messaging server. Users can access messages from a conference room, lobby phone, or colleague's phone, as well as their own. Any operation performed on the messages using MailView is automatically reflected in Microsoft Exchange and Cisco Unity; Cisco Personal Assistant interfaces with Microsoft Exchange and IMAP 4 message stores for MailView features.

Specifications

Feature	Cisco Personal Assistant
Platform	Cisco Media Convergence Server MCS-7825H-3.0, MCS-7835H-3.0, MCS-7825I-3.0, and MCS-7835I-3.0
Web Server Requirements for IP Phone Productivity Services Platform	Basic Web Server: Microsoft IIS 4.0 or later Separate server for Cisco Personal Assistant Server and Speech Recognition Server
Software Compatibility	Cisco CallManager 3.3+ and 4.1 Cisco Unity 2.46+,3.0+, and 4.0+ for voice-mail features Microsoft Exchange 5.5, Exchange 2000, and Exchange 2003 for calendar, e-mail, contact synchronization features

Selected Part Numbers and Ordering Information[1]

Cisco Personal Assistant	
PASR-BUNDLE	Cisco Personal Assistant Bundle
SW_PASR1.4-SVR2S=	Cisco Personal Assistant 1.4 Server Software with Speech Recognition[2]
SW-PASR1-USR=	PA User License
SW-PASR1-USR10=	PA User License
SW-PASR1-KX=	Cisco Personal Assistant 1.4, Expansion Speech Recognition Session[3]
SW-PASR1-TX=	Cisco Personal Assistant 1.4 Enhanced Text-to-Speech Session

1. This is only a small subset of all parts available via URL listed under "For More Information." Some parts have restricted access or are not available through distribution channels. Resellers: For latest part number and pricing info, see the *Distribution Product Reference Guide* at: http://www.cisco.com/dprg (limited country availability).
2. Cisco Personal Assistant can be purchased with an MCS-7825H-2.2-EVV1 or an MCS-7835H-2.4-EVV1 media convergence server.
3. Various session combinations available.

For More Information

See the Cisco Personal Assistant Web site: **http://www.cisco.com/go/personalassist**

Cisco Survivable Remote Site Telephony (SRST)

Cisco SRST provides key backup telephony functions at remote branch office routers if connectivity to the centrally-located Cisco CallManager fails (i.e. WAN link is interrupted). In this case the Cisco SRST-enabled router will take over and provide robust telephony service (including off-net calls to 911). It is ideal for enterprise organizations looking to cost-effectively deploy IP telephony in their branch office location. SRST v3.2 with 12.3(11)T is available on Cisco 1751V, 1760V, 2600-XM, 2691, 2800 , 3725/3745, and 3800 series of routers.The Cisco Communications Media Module and the Cisco 7200 also support previous versions of SRST.

Key Features

- SRST 3.2 features: Increased Phone support from 480 upto 720 phones; RFC2833 support for SCCP-SIP (vmail) DTMF conversion during SRST; External MOH from live feed; SRST alias command increase from 10 to 50 (Set non registered DN's to call forward on busy/NA); Call Pickup Ringing Extension; COR list increase from 10 to 20; Option to disable H225 TCP keep alive timer to maintain calls during WAN failure (new with 12.3(7)T)
- SRST 2.1 features: International language support; Call forward no-answer/busy to Unity server with Personal Greeting; Cisco 7914 and 7935 support; VG248 support
- SRST 2.0 features: Huntstop support; Music/Tone on Hold; Class of Restriction; Distinctive Ringing; Global forwarding to voicemail across PSTN during Cisco CallManager fallback; TCL based simple AA and IVR on local gateway (SRS Telephony router); Transfer across H323 network of Cisco endpoints; Alias lists for single number to be designated for unregistered phones.

Selected Part Numbers and Ordering Information

SRST 3.2 Platform Density and Feature License Part Numbers

Platform	Number of Phones Supported	Part Number	Part Number (Spare)
Cisco 1751V, 1760V, 2801	Up to 24 phones	FL-SRST-SMALL	FL-SRST-SMALL=
Cisco 261x-XM, 262x-XM, 2811	Up to 36 phones	FL-SRST-36	FL-SRST-36=
Cisco 265xXM, 2821	Up to 48 phones	FL-SRST-MEDIUM	FL-SRST-MEDIUM=
Cisco 2691	Up to 72 phones	FL-SRST-SMALL and FL-SRST-MEDIUM	FL-SRST-SMALL= and FL-SRST-MEDIUM=
Cisco 2851	Up to 96 phones	FL-SRST-96	FL-SRST-96=
Cisco 3725 router	Up to 144 phones	FL-SRST-144	FL-SRST-144=
Cisco, 3745, 7200 NPE300/400	Up to 240 phones	FL-SRST-240	FL-SRST-240=
Cisco 3825	Up to 336 phones	FL-SRST-336	FL-SRST-336 =
Cisco 7200NPE400/G1	Up to 480 phones	FL-SRST-480	FL-SRST-480=
Cisco 3845	Up to 720 phones	FL-SRST-720	FL-SRST-720=

SRST bundles

CISCO2801-SRST/K9	Feature License FL-SRST-SMALL: 8 G.711 channels of DSPs included
CISCO2811-SRST/K9	Feature License FL-SRST-36: 16 G.711 channels of DSPs included
CISCO2821-SRST/K9	Feature License FL-SRST-MEDIUM : 32 G.711 channels of DSPs included
CISCO2851-SRST/K9	Feature License FL-SRST-96: 48 G.711 channels of DSPs included
CISCO3825-SRST/K9	Feature License FL-SRST-168 : 64 G.711 channels of DSPs included
CISCO3845-SRST/K9	Feature License FL-SRST-240: 64 G.711 channels of DSPs included
CISCO1760-V-SRST	Feature License FL-SRST-SMALL: 4 G.711 channels of DSPs included
CISCO2651XM-V-SRST	Feature License FL-SRST-MEDIUM : 30 G.711 channels of DSPs included

For More Information

See the Cisco SRS Telephony Web site: **http://www.cisco.com/go/srs**

Cisco IP Manager Assistant

Cisco IP Manager Assistant (IPMA) provides all the call-routing and display capabilities required in a business environment while giving users a choice of phone devices that better fit their needs. By combining a PC-based console application and various soft keys and display panes on Cisco IP phones, Cisco IPMA can present users job-specific tools to more efficiently manage calls in this important environment.

Cisco IPMA can be configured in either proxy-line mode or shared-line mode. In proxy-line mode, managers and assistants have different directory numbers or lines on their phones, but calls to managers are usually diverted to the assistant's line. In shared-line mode, both managers and assistants share the same directory number, but assistants can handle calls without disturbing managers.

For More Information

See the Cisco Media Convergence Server Web site: **http://www.cisco.com/go/ipma**

Cisco Conference Connection

Cisco Conference Connection is an IP audio conferencing solution that is designed for small and medium businesses. Cisco Conference Connection integrates with Cisco CallManager and can scale from 20 ports to 180 ports in a single system. This application is ideal for enterprises trying to increase productivity while reducing expenses. A simple web-based interface enables employees to manage their conference schedules.

For More Information

See the Cisco Conference Connection Web site:
http://www.cisco.com/en/US/products/sw/voicesw/ps752/index.html

Cisco 7900 Series IP Phones

Cisco IP Phones provide unmatched levels of integrated business functionality and converged communications beyond today's conventional voice systems. The Cisco IP Phone 7970G includes a color, high-resolution backlit touch-screen display promoting easy access to feature-rich phone functionality and time-saving applications. The Cisco IP Phone 7960G "manager set" addresses the communication needs of the professional, with a high or busy amount of phone traffic. The Cisco IP Phone 7940G "business set" addresses the communication needs of a transaction type worker, in a office cubicle environment, who conducts medium to high telephone traffic. The Cisco IP Phone 7912G, 7910G+SW and 7905G "basic sets" address the communication needs of a cubicle worker who conducts low to medium telephone traffic. The Cisco IP Phone 7902G "entry set" addresses voice communication needs of a lobby, lab, manufacturing floor, and other areas where only a minimal amount of features are required. Cisco IP Phone Expansion Module 7914 extends the Cisco IP Phone 7960G with additional buttons and LCD, increasing the total number of buttons to 20 with one module, or 34 with two modules. Cisco IP Conference Station 7936, a high-quality hands-free conference station, is designed for use on desktops and offices and in small to medium-sized conference rooms.

Key Features

- Dynamic soft keys present calling options based on context
- Open APIs using XML to deliver applications to the display
- Automatic phone discovery, VLAN configuration, and registration
- QoS is provided via support of 802.1pq, in addition to configurable DIFFSERV and TOS
- Voice-activity detection, silence suppression, comfort-noise generation, and error concealment
- G.711a, G.711u, G.729ab audio-compression coder-decoders (codecs)
- Software upgrade support via Trivial File Transfer Protocol (TFTP) server
- Microsoft NetMeeting enabled—features such as application sharing and video conferencing
- Integrated Ethernet Switch supporting Ethernet connectivity for a downstream PC
- Integrated Inline power support allows the phone to receive power over the LAN
- A hearing-aid-compatible handset

Specifications

Feature	7960G	7970G	7914 Expansion Module	IP Conference Station 7936
Display	Pixel-Based	Pixel-based	Pixel-based	Pixel-Based
Dynamic Soft Keys	4	8	N/A	3
Inline Power	Yes	Yes	Yes	No
10/100Base-T Ethernet Switch	Yes	yes	N/A	No
Lines	6	8	14	1
Speaker Phone	Yes	Yes	N/A	Yes
Headset Jack	Yes	Yes	N/A	No
3rd Party XML Applications	Yes	Yes	N/A	No

Feature	7902G	7905G	7910G+SW	7912G	7940G
Display	None	Pixel-Based	Character-Based	Pixel-based	Pixel-Based
Dynamic Soft Keys	0	4	0	4	4
Inline Power	Yes	Yes	Yes	Yes	Yes
10/100Base-T Ethernet Switch	No	No	Yes	Yes	Yes
Lines	1	1	1	1	2
Speaker Phone	No	Monitor Only	Monitor Only	Monitor Only	Yes
Headset Jack	No	No	No	No	Yes
3rd Party XML Applications	No	Yes	No	Yes	Yes

Selected Part Numbers and Ordering Information[1]

Cisco 7900 Series IP Power and Phones

CP-7960G	Cisco IP Phone 7960G, Manager Set
CP-7940G	Cisco IP Phone 7940G, Business Set
CP-7912G	Cisco IP Phone 7912G, Basic Set w/ Switch
CP-7910G+SW	Cisco IP Phone 7910G+SW, Basic Set w/ Switch
CP-7905G	Cisco IP Phone 7905G, Basic Set
CP-7902G	Cisco IP Phone 7902G, Entry Set
CP-7936	Cisco IP Conference Station
CP-7914=	Cisco 7914 IP Phone Expansion Module for the 7960 IP Phone
CP-SINGLFOOTSTAND=	Single module footstand
CP-DOUBLFOOTSTAND=	Double module footstand
CP-WALLMOUNTKIT=	Non-Locking Wall Mount Kit for 7910/40/60G series IP phones
CP-LCKNGWALLMOUNT=	Locking Wallmount Kit for the 7910/40/60G series IP phones
CP-LCKNGWALLMNT2=	Locking Wallmount Kit for all 7900 series IP phones
CP-PWR-CUBE=	IP Phone power transformer for 7900 series IP phones
WS-PWR-PANEL	Catalyst 48 port Inline Power Patch Panel

1. This is only a small subset of all parts. Some parts have restricted access or are not available through distribution channels.

For More Information

See the Cisco IP Telephones Web site: **http://www.cisco.com/go/iptel**

Cisco ATA Series of Analog Telephone Adaptors

The Cisco ATA 186 and 188 Analog Telephone Adaptors bring analog telephones into the networked world. The Cisco ATA series of products address the low-end product portfolio need by targeting the enterprise, business local services, small-office environment and the emerging managed voice services market. These cost effective handset-to-Ethernet adaptors enable analog devices, such as phones and fax machines, to support voice-over-IP (VoIP) services. The Cisco ATA 186 is equipped with, and a single RJ-45 Ethernet port. The Cisco ATA 188 has two RJ-11 voice ports and two RJ-45 ports. The internal Ethernet switch allows for a direct connection to a 10/100BASE-T Ethernet network and connectivity to a co-located PC or other Ethernet-based device via the RJ-45 ports.

Both models ship with a bootload image and must be upgraded to a signaling firmware image available on Cisco.com before deployment. Cisco ATAs can be configured[1] to use the standards-based Voice over IP (VoIP) protocols H.323, Session Initiation Protocol (SIP), Media Gateway Control Protocol (MGCP) and Skinny Client Control Protocol (SCCP).

When to Sell

Sell This Product	When a Customer Needs These Features
Cisco ATA Series of Analog Telephone Adaptors	• Enable analog devices, such as phones and fax machines, to support Voice over IP services by converting the analog signal into an IP signal • Continue use of existing analog phones with IP network

Key Features

- Auto-provisioning with Trivial File Transfer Protocol (TFTP) provisioning servers
- Automatic assignment of IP address, network route IP, and subnet mask via Dynamic Host Configuration Protocol (DHCP)
- Optional web configuration through built-in Web server
- Optional touch-tone telephone keypad configuration with voice prompt
- Administration password to protect configuration and access
- Advanced pre-processing to optimize full-duplex voice compression
- High performance line-echo cancellation eliminates noise and echo
- Voice activity detection (VAD) and comfort noise generation (CNG) save bandwidth by delivering voice, not silence
- Dynamic network monitoring to reduce jitter artifacts such a packet loss

1. Two software image combinations are available on Cisco.com: H.323/SIP and MGCP/SCCP. MGCP/SCCP image includes the letters, ms, in its name.

Specifications

Feature	Cisco ATA 186	Cisco ATA 188
Telephone and network interfaces	2 RJ-11 FXS ports 1 RJ-45 interface for network connection	2 RJ-11 FXS ports 1 RJ-45 interface for network connection 1 RJ-45 “switch port” for connection to PC or another downstream Ethernet device Note: "Daisychaining" multiple ATAs together through the second RJ45 port is not supported
Dimensions (H x W x D)	1.5 x 6.5 x 5.75 in. (3.8 x 16.5 x14.6 cm)	1.5 x 6.5 x 5.75 in. (3.8 x 16.5 x14.6 cm)
Weights	15 oz (425 gm)	15 oz (425 gm)
Voice-over-IP (VoIP) protocols	H.323 v2; H.323 v4; SIP (RFC 2543); MGCP 1.0 (RFC 2705); MGCP 1.0/network-based call signaling (NCS) 1.0 Profile; MGCP 0.1; SCCP	H.323 v2; H.323 v4; SIP (RFC 2543); MGCP 1.0 (RFC 2705); MGCP 1.0/network-based call signaling (NCS) 1.0 Profile; MGCP 0.1; SCCP

Selected Part Numbers and Ordering Information[1]

Cisco ATA Series of Analog Telephone Adaptors	
ATA186-I1-A	Cisco ATA 186 2-port adaptor, 600 ohm impedance
ATA186-I2-A	Cisco ATA 186 2-port adaptor, complex impedance (270 ohm in series with 750 ohm and 150 nF in parallel)
ATA188-I1-A	Cisco ATA 188 2-port adaptor with switch, 600 ohm impedance
ATA188-I2-A	Cisco ATA 188 2-port adaptor with switch, complex impedance (270 ohm in series with 750 ohm and 150 nF in parallel)
Cisco ATA Series of Analog Telephone Adaptors Power Supply Cables	
ATACAB-NA	ATA power supply cable for North American-style power systems
ATACAB-EU	ATA power supply cable for Continental European-style power systems
ATACAB-UK	ATA power supply cable for United Kingdom
ATACAB-AR	ATA power supply cable for Argentina
ATACAB-JP	ATA power supply cable for Japan
ATACAB-AU	ATA power supply cable for Australia

1. Some countries have telephone networks that list multiple impedance requirements. It is important to closely approximate the impedance of the typical handsets used in the region when selecting the proper configuration. The incorrect choice may lead to poor echo cancellation performance.

For More Information

See the Cisco ATA Series Web site: **http://www.cisco.com/go/ata186**

Cisco MCS 7800 Series Media Convergence Servers

Cisco MCS 7815-I1

- Provides an entry level tower server equipped with an Intel Pentium™ 4 3060MHz processor, 80GB Serial Attached ATA (SATA) hard drive and single non-hot-swap power supply

Cisco MCS 7825-H1 and Cisco MCS 7825-I1

- Each provides an entry level rack mount server that occupies only one rack mounting space
- Equipped with an Intel Pentium™ 4 3.4GHz processor, two 80GB Serial Attached ATA (SATA) hard drives configured as RAID1 and a single non-hot-swap power supply
- Limited to Unity applications only

Cisco MCS 7825H-3000

- Provides an entry level rack mount server that occupies only one rack mounting space
- Equipped with an Intel Pentium™ 4 3060MHz processor, 40GB ATA hard drive and a single non-hot-swap power supply

Cisco MCS 7825I-3000

- Provides an entry level rack mount server that occupies only one rack mounting space.
- Equipped with an Intel Pentium™ 4 3060MHz processor, 80GB SATA hard drive and a single non-hot-swap power supply
- Not supported for Unity operation

Cisco MCS 7835-H1 and Cisco MCS 7835-I1

- Each provides a highly available mid-level rack mounted server solution that is equipped with an Intel Nocona Xeon™ 3400MHz processor, up to six hot-swap Small Computer Systems Interface (SCSI) hard disks, a Redundant Array of Independent Disks (RAID) 1/0 Controller, hot-swap redundant fans and redundant hot-swap power supplies.
- An optional tape backup is available

Cisco MCS 7845-H1 and Cisco MCS 7845-I1

- Each provides a powerful and highly reliable high level rack mounted server solutions
- Equipped with two Intel Nocona Xeon™ 3400MHz processors, up to six hot-swap SCSI hard disks, RAID 1/0 controller, redundant hot-swap fans and redundant hot-swap power supplies
- An optional tape backup is available for both

Specifications

MCS-7815-I1	MCS-7825-H1	MCS-7825-I1	MCS-7825H-3000	MCS 7825I-3000
Intel PentiumÆ 4 3060-MHz processor 512KB L2 Cache	Intel PentiumÆ 3400-MHz processor 1024KB L2 Cache	Intel PentiumÆ 3400-MHz processor 1024KB L2 Cache	Intel PentiumÆ 3060-MHz processor 512KB Cache	Intel PentiumÆ 3060-MHz processor 1024KB L2 Cache
1024MB SDRAM	1024MB SDRAM	1024MB SDRAM	1024MB SDRAM	1024MB SDRAM
80GB SATA Hard Disk	Dual 80GB SATA Hard Disk	Dual 80GB SATA Hard Disk	40GB ATA Hard Disk.	80GB SATA Hard Disk
1.44MB Floppy Disk	1.44MB Floppy Disk	1.44MB Floppy Disk	1.44MB Floppy Disk	1.44MB Floppy Disk
DVD Drive	DVD Drive	DVD Drive	DVD Drive	DVD Drive
Integrated SATA Controller	Integrated SATA Controller-RAID1	Integrated SATA Controller-RAID1	ATA/100 Controller	Integrated SATA Controller
Single 10/100/1000 Ethernet NIC	Dual 10/100/1000 Ethernet NIC1U Rack Mount System	Dual 10/100/1000 Ethernet NIC1U Rack Mount System	Dual 10/100/1000 Ethernet NIC2U Rack Mount System	Dual 10/100/1000 Ethernet NIC1U Rack Mount
Tower System with optional rack mount kit.				

Specifications (Con.)

MCS 7835-H1	MCS-7835-I1	MCS-7845-H1	MCS-7845-I1
Intel XeonÆ 3400-MHz processor 1024KB Cache	Intel XeonÆ 3400-MHz processor 1024KB Cache	Dual Intel XeonÆ 3400-MHz processor 1024KB Cache	Dual Intel XeonÆ 3400-MHz processor 1024KB Cache
2048MB SDRAM	2048MB SDRAM	SDRAM is configuration dependant	SDRAM is configuration dependant
Two-72GB SCSI Hard Disk	Two-72GB SCSI Hard Disk	Hard Disk is configuration dependant	Hard Disk is configuration dependant
1.44MB Floppy Disk	1.44MB Floppy Disk	1.44MB Floppy Disk	1.44MB Floppy Disk
DVD Drive	DVD Drive	DVD Drive	DVD Drive
SCSI Controller.	SCSI Controller.	SCSI Controller.	SCSI Controller.
Dual 10/100/1000 Ethernet NIC2U Rack Mount System	Dual 10/100/1000 Ethernet NIC2U Rack Mount System	Dual 10/100/1000 Ethernet NIC2U Rack Mount System	Dual 10/100/1000 Ethernet NIC2U Rack Mount System

Selected Part Numbers and Ordering Information[1]

Cisco Media Convergence Server 7800	
MCS-7815-I1-IPC1, MCS-7815-I1-ECS1, MCS-7815-I1-CC1	Cisco Media Convergence Server 7815-I1
MCS-7825-H1-ECS1	Cisco Media Convergence Server 7825-H1
MCS-7825-I1-ECS1	Cisco Media Convergence Server 7825-I1
MCS-7825H-3.0-IPC1, MCS-7825H-3.0-CC1	Cisco Media Convergence Server 7825H-3000
MCS-7825I-3.0-IPC1, MCS-7825I-3.0-CC1	Cisco Media Convergence Server 7825I-3000
MCS-7835-H1-IPC1, MCS-7835-H1-ECS1, MCS-7835-H1-CC1	Cisco Media Convergence Server 7835-H1
MCS-7835-I1-IPC1, MCS-7835-I1-ECS1, MCS-7835-I1-CC1	Cisco Media Convergence Server 7835-I1
MCS-7845-H1-IPC1, MCS-7845-H1-ECS1, MCS-7845-H1-ECS2, MCS-7845-H1-CC1	Cisco Media Convergence Server 7845-H1
MCS-7845-I1-ECS1, MCS-7845-I1-ECS2	Cisco Media Convergence Server 7845-I1

1. This is only a small subset of all parts available via URL listed under “For More Information.” Some parts have restricted access or are not available through distribution channels.

For More Information

See the Cisco Media Convergence Server Web site: **http://www.cisco.com/go/mcs**

Cisco Voice Gateways

Voice Gateways interface directly to PBXs or public telephone networks to carry voice traffic across IP networks—by converting IP calls to standard telephony calls and vice versa. They provide connectivity between packet telephony and legacy telephony such as PSTN, PBX, fax machines, and other devices. Cisco offers the Cisco VG 248 and VG 224 dedicated analog voice gateways.

Cisco’s full line of multiservice routers can also add analog and digital voice gateway functionality through the use of network modules and voice interface cards[1].

Cisco VG 224 Voice Gateway

The Cisco VG 224 is a Cisco IOS high-density 24-port gateway for analog phones, fax machines, modems, and speakerphones within an enterprise voice system based on Cisco CallManager. Having these devices tightly integrated with the IP-based phone system is advantageous for increased manageability, scalability, and cost-effectiveness. Commercial businesses can also use the Cisco VG 224 in conjunction with Cisco CallManager Express to effectively augment a Full Service Branch router environment. This topology environment supports a businesses needs of high concentration of analog voice ports for basic calls, with some supplementary services.

Cisco VG248 Voice Gateway

The Cisco VG248 Voice Gateway is a 1 unit high rack mountable device allowing 48 analog devices (phones, fax machines & modems) to be used with Cisco Call Manager. It enables organizations with large numbers of analog phones (hotels, universities, hospitals, etc.) to deploy IP Telephony while maintaining the investment in legacy handsets. The analog lines are full featured (caller id, message waiting lights, feature codes) and the price per port is competitive with a legacy PBX.

The VG 248 supports the legacy voicemail Simple Message Desk Interface (SMDI) mail interface that allows the connection of a Cisco Call Manager network to a legacy voicemail system. It also will allow the sharing of existing SMDI based voicemail systems between the Cisco Call Manager and the legacy PBX.

Selected Part Numbers and Ordering Information

Cisco VG 248 Voice Gateway	
VG 224	Cisco VG 224 Analog Phone Gateway
VG 248	48 Port Voice over IP analog phone gateway

For More Information

1. Please see the 1700, 2600, 3600, 7200, 5x00 series in Chapter 1—Routers, Chapter 7—Access Products.

See the Cisco Voice Gateways Web site: **http://www.cisco.com/go/voicegate**

Cisco MeetingPlace

Cisco MeetingPlace—part of the Cisco IP Communications system—is a complete rich-media conferencing solution that seamlessly integrates voice, video, and Web conferencing capabilities to make remote meetings as natural and effective as face-to-face meetings for unmatched productivity gains. It is deployed "on network" behind the firewall and integrated directly into an organization's private voice and data networks and collaborative applications to provide significant cost savings, the utmost in security, and a superior user experience. Intuitive interfaces make setting up, attending, and managing meetings easy.

Key Features

- Immediate or future voice, video, and Web conferences can be set up and attended in a single step—from Cisco IP phones, instant messaging clients, Web browsers, and Microsoft Outlook and Lotus Notes calendars; Meeting participants have total control over their voice, video, and Web conference from a single browser interface
- Allows enterprises to isolate their confidential meetings and content behind the firewall for secure data network transport while providing the flexibility to meet with external parties
- Deployed in a variety of ways—on premises or hosted, customer managed or outsourced; Cisco Managed Solutions and managed services partners can provide system administration and end-user help desk functions for on-premises or hosted systems

Selected Part Numbers and Ordering Information[1]

Cisco MeetingPlace 8100 Solution Bundles	
MP-8106-30IP	Cisco MeetingPlace 8106 30UL IP Solution Bundle
MP-8106-48	Cisco MeetingPlace 8106 48UL T1-CAS Solution Bundle
MP-8106-120IP	Cisco MeetingPlace 8106 120UL IP Solution Bundle
MP-8112-180IP	Cisco MeetingPlace 8112 180UL IP Solution Bundle
MP-8112-180PRI	Cisco MeetingPlace 8112 180UL PRI Solution Bundle
MP-8112-192	Cisco MeetingPlace 8112 192UL T1-CAS Solution Bundle
MP-8112-510IP	Cisco MeetingPlace 8112 510UL IP Solution Bundle
MP-8112-510PRI	Cisco MeetingPlace 8112 510UL PRI Solution Bundle
MP-8112-600	Cisco MeetingPlace 8112 600UL T1-CAS Solution Bundle
Cisco MeetingPlace Hardware and Software options	
MP-SMARTBLADE	Cisco MeetingPlace Audio Conference Module, 96 ports with 4xT1-CAS
MP-MA-4	Cisco MeetingPlace Gateway Module, 120 IP ports
MP-MA-4-PRI	Cisco MeetingPlace Gateway Mod., 4xPRI (T1/E1)
MP-MA-16	Cisco MeetingPlace Gateway Mod., 480 IP ports
MP-MA-16-PRI	Cisco MeetingPlace Gateway Module 16xPRI (T1/E1)
MP-AUDIO-UL-24	Cisco MeetingPlace Audio Conference user license bundle, 24 user licenses
MP-AUDIO-UL-30	Cisco MeetingPlace Audio Conference. user license bundle, 30 user licenses
MP-MTGTM-SITE-5.3	Cisco MeetingPlace MeetingTime Site License
MP-FLEX-MENU	Cisco MeetingPlace Flex Menu Option
MP-LANG-5.3	Cisco MeetingPlace Multilingual System License
Web Conferencing Options	
MP-WEBCONFSW-5.3	Cisco MeetingPlace Web Conferencing Server Application (includes 20 concurrent Web conferencing user licenses)
MP-RM-RECORDING	Cisco MeetingPlace Web Voice Recording System License
MP-CONF-UI-5.3	Cisco MeetingPlace Conferencing User Interface (included in Cisco MeetingPlace Solution Bundles)
MP-WEB-UL-20	Cisco MeetingPlace Web Conferencing 20 user licenses bundle

Integration Applications	
MP-VIDEO	Cisco MeetingPlace Video Integration
MP-IP-GW	Cisco MeetingPlace H.323/SIP Gateway Application (included in Cisco MeetingPlace Solution Bundles)
MP-EMAIL-5.3	Cisco MeetingPlace SMTP E-mail Gateway Application (included in Cisco MeetingPlace Solution Bundles)
MP-OUTLOOK-5.3	Cisco MeetingPlace for Outlook Integration Application
MP-NOTES-5.3	Cisco MeetingPlace for Lotus Notes Integration Application
MP-DIRSVCS-5.3	Cisco MeetingPlace Directory Services 5.3
MP-IM	Cisco MeetingPlace Instant Messaging Integration Application

1. Some countries have telephone networks that list multiple impedance requirements. It is important to closely approximate the impedance of the typical handsets used in the region when selecting the proper configuration. The incorrect choice may lead to poor echo cancellation performance.

For More Information

See the Cisco MeetingPlace Web site: **http://www.cisco.com/go/meetingplace**

Cisco IP/VC 3500 Series Videoconferencing Products

The Cisco IP videoconferencing (IP/VC) solution—a component of the Cisco rich-media communications solution—enables face-to-face discussions among conferencing participants. The Cisco IP/VC is for enterprises and service providers who want a reliable, easy-to-manage, cost-effective network infrastructure for rich-media conferencing, video telephony and videoconferencing.

Cisco IP/VC 3500 MCU series connects three or more videoconference endpoints (H.323, SCCP, H.320, SIP) in a single, multi-participant meeting. The Cisco IP/VC product family consists of the IP/VC 3511 Multipoint Control Unit (MCU, also known as a “video bridge”), the IP/VC 3521 and 3526 H.320 to H.323 Gateways and the IP/VC 3540—Series Videoconferencing System. Cisco IP/VC works with H.323-standards-based videoconference client devices from a variety of vendors and integrates with legacy H.320 networks. When used in conjunction with the Enhanced Media Processor (EMP), users experience best-in-class voice and video solutions. If needed, users with access to H.320 endpoints can also participate in a videoconference through the Cisco IP/VC videoconferencing gateway.

Key Features

- Enhanced video experience for users in their voice communications solutions as part of the latest releases of multipoint control unit (MCU) software; No longer restricted to traditional videoconferencing, users can extend the value of video and the IP/VC products by integrating with Cisco CallManager and Cisco MeetingPlace products to enable Video Telephony and Rich-media Conferencing solutions
- The Cisco IP/VC 3511 Multipoint Control Unit (MCU)—1RU stack/rack-mount system enabling ad-hoc videoconferences between three or more endpoints. It is suitable for small to medium enterprises and remote branch offices in larger enterprises
- The IP/VC 3521 and the IP/VC 3526 Videoconferencing Gateways—1RU stack/rack-mount systems that translate between H.320 and H.323 protocols; The IP/VC 3521 provides up to four BRI interfaces and the IP/VC 3526 provides one ISDN T1/E1 PRI interface
- The IP/VC 3540 Videoconferencing System integrates multipoint control units, and gateways in a single platform for cost-effective deployment of IP-centric videoconferencing networks

- The Multimedia Conference Manager (MCM) software—part of Cisco IOS Software and available across a wide range of Cisco router platforms, including the Cisco 2600/2600XM, 3600, 3700, and 7200 series; As a gatekeeper/proxy, it enables network managers to control and secure bandwidth and priority settings for H.323 videoconferencing services

Selected Part Numbers and Ordering Information[1]

Cisco IP/VC 3500 Series Videoconferencing Products

IPVC-3511-MCU	IP/VC 3511 Multipoint Control Unit
IPVC-3511-MCU-E	IP/VC 3511 Multipoint Control Unit and Enhanced Media Processor
IPVC-3544-CHAS	IPVC 3544 Four Slot Chassis
IPVC-3540-MC03A	IPVC 3540 Series MCU Modules - 30 Port Session
IPVC-3540-MC06A	IPVC 3540 Series MCU Modules - 60 Port Session
IPVC-3540-MC10A	IPVC 3540 Series MCU Modules - 100 Port Session
IPVC-3540-EMP	IP/VC 3540 Enhanced Media Processor
IPVC-3540-EMP3	IP/VC 3540 Enhanced Media Processor 3
IPVC-3540-XAM03	IP/VC 3540 MCU Audio Transcoder Cards - 30 Port MCU Daughter card
IPVC-3540-XAM06	IP/VC 3540 MCU Audio Transcoder Cards - 60 Port MCU Daughter card

1. This is only a small subset of all parts available. Some parts have restricted access or are not available through distribution channels.

For More Information

See the Cisco IP/VC 3500 series Web site: **http://www.cisco.com/go/ipvc**

Cisco IP/TV 5.2

Cisco IP/TV 5.2 delivers a complete, highly scalable, bandwidth-efficient solution for high-quality video communications over enterprise networks. Cisco IP/TV supports live video, scheduled video, video on demand (VOD), synchronized presentations and screen captures, and a wide range of video management functions. The solution enables a broad spectrum of applications for enterprise communications including training, corporate communications, business TV, and distance learning.

The Cisco IP/TV 5.2 solution is purchased as Cisco IP/TV 3400 Series Server appliances which will include the IP/TV client viewer software, SlideCast, Stream Watch and other applications to enable a turn-key enterprise IP broadcasting solution. The Cisco IP/TV 3400 Series servers contain pre-configured software, preinstalled capture cards, network interface cards, and device drivers. The Cisco IP/TV 3400 Series includes the IP/TV 3441 Broadcast Server, the IP/TV 3442 Broadcast Server, the IP/TV 3427 Broadcast Servers, and the IP/TV Program Manager device mode of the ACNS 566 Content Engine. This product family offers a range of choices to best suit large-scale enterprise applications, performance requirements, and bandwidth availability.

Selected Part Numbers and Ordering Information[1]

Cisco IP/TV 3400 Series Video Servers

IPTV-3441-BCAST	Cisco IP/TV 3441 Broadcast Server, (1) MPEG-1/2
IPTV-3442-BCAST	Cisco IP/TV 3442 Broadcast Server, (4) MPEG-4
IPTV-3427-BCAST-C1	Cisco IP/TV 3427 Broadcast Server, (6) MPEG-4
IPTV-3427-BCAST-C2	Cisco IP/TV 3427 Broadcast Server, (6) MPEG-4, (2) MPEG-1/2
IPTV-3427-BCAST-C3	Cisco IP/TV 3427 Broadcast Server, (2) MPEG-4, (3) MPEG-1/2
CE-566-K9	IP/TV Program Manager device mode of the Cisco ACNS 566 Content Engine (standard configuration, no extra licenses needed)

1. This is only a small subset of all parts available via URL listed under “For More Information.” Some parts have restricted access or are not available through distribution channels.

For More Information

See the IP/TV 3400 series Web site: **http://www.cisco.com/go/iptv**

Cisco Voice Router Bundles

Cisco Voice router bundles based on the Cisco 1760, 2600XM, 2691, and 3800 access router platforms provide customers with a single part number for ease of ordering. These voice bundles are especially designed for the application of toll bypass, PSTN or PBX to IP connectivity with Cisco CallManager, or other voice applications where connectivity from IP to PSTN is required.

Each Voice bundle provides DSP resources to support PSTN connectivity to IP, necessary memory, and recommended Cisco IOS feature set..

When to Sell

Sell This Product	When a Customer Needs These Features
Cisco Voice Router Bundles	• Deploying access router where toll bypass application is required • Deploying Cisco CallManager and PSTN or PBX gateway is required • Customer may need voice application in the future and wants investment protection on initial purchase

Selected Part Numbers and Ordering Information[1]

Cisco Voice Router Bundles	
CISCO1760-V	1760 Router, 32MB Flash, 96MB DRAM, PVDM-256K-4, IOS IP Voice Feature Set
CISCO2651XM-V	2651 Router, 32MB Flash, 128MB DRAM, AIM-VOICE-30 (DSP for 30 calls), IOS IP Voice.
CISCO2801-V/K9	2801 Router, 64MB Flash, 256MB DRAM, PVDM2-8 (DSP for 8 calls), IOS SP Services
CISCO2811-V/K9	2811 Router, 64MB Flash, 256MB DRAM, PVDM2-16 (DSP for 16 calls), IOS SP Services
CISCO2821-V/K9	2821 Router, 64MB Flash, 256MB DRAM, PVDM2-32 (DSP for 32 calls), IOS SP Services
CISCO2851-V/K9	2851 Router, 64MB Flash, 256MB DRAM, PVDM2-48 (DSP for 48 calls), IOS SP Services
CISCO3825-V/K9	3825 Router, 64MB Flash, 256MB DRAM, PVDM2-64 (DSP for 64 calls), IOS SP Services
CISCO3845-V/K9	3845 Router, 64MB Flash, 256MB DRAM, PVDM2-64 (DSP for 64 calls), IOS SP Services

1. This is only a small subset of all parts available via URL listed under “For More Information.” Some parts have restricted access or are not available through distribution channels.

For More Information

See Routing, page 1-1 for product details.

Cisco CallManager Express Bundles

Designed for enterprise branch offices, small and midsize businesses, and managed network service providers who seek consolidated, easy to order CallManager Express enabled routers. Each of these bundles provide a versatile, cost-effective solution that offer integrated IP Telephony and data services via a single router platform and include Cisco CallManager Express feature license, DSP resources to support PSTN connectivity to IP, necessary memory and recommended Cisco IOS feature set.

When to Sell

Sell This Product	When a Customer Needs These Features
Cisco CCME Router Bundles	• SMB or Enterprise branch locations looking for data and distributed call processing using Cisco CallManager Express in a all-in-one solution based on Cisco access router • Support for 24 to 240 Phones per location • Customers who may need IP Telephony application in the future and wants investment protection on initial purchase

Selected Part Numbers and Ordering Information[1]

Cisco CCME Router Bundles

CISCO1760-V-CCME	1760 Router, 32MB Flash, 96MB DRAM, PVDM-256K-4, IOS IP Voice Feature Set
CISCO-2611XM-V-CCME	2611XM router, 32 MB Flash, 128 MB DRAM, NM-HD-2V (for 8 calls) Cisco IOS IP Voice
CISCO2651XM-V-CCME	2651 Router, 32MB Flash, 128MB DRAM, AIM-VOICE-30 (DSP for 30 calls), IOS IP Voice.
CISCO2801-CCME/K9	2801 Router, 64MB Flash, 256MB DRAM, PVDM2-8 (DSP for 8 calls), IOS SP Services
CISCO2811-CCME/K9	2811 Router, 64MB Flash, 256MB DRAM, PVDM2-16 (DSP for 16 calls), IOS SP Services
CISCO2821-CCME/K9	2821 Router, 64MB Flash, 256MB DRAM, PVDM2-32 (DSP for 32 calls), IOS SP Services
CISCO2851-CCME/K9	2851 Router, 64MB Flash, 256MB DRAM, PVDM2-48 (DSP for 48 calls), IOS SP Services
CISCO3825-CCME/K9	3825 Router, 64MB Flash, 256MB DRAM, PVDM2-64 (DSP for 64 calls), IOS SP Services
CISCO3845-CCME/K9	3845 Router, 64MB Flash, 256MB DRAM, PVDM2-64 (DSP for 64 calls), IOS SP Services

1. This is only a small subset of all parts available via URL listed under "For More Information." Some parts have restricted access or are not available through distribution channels.

For More Information

See Routing, Page 1-1 for detailed product information.

See Cisco CallManager Express, page 4-9 for product details.

Cisco Survivable Remote Site (SRST) Bundles

Designed for enterprise branch offices, small and midsize businesses, and managed network service providers who seek consolidated, easy to order SRST enabled routers. Each of these bundles provide a versatile, cost-effective solution that offer voice connectivity, call processing redundancy and data services via a single Enterprise Router platform and include Cisco Survivable Remote Site Telephony (SRST) feature license, DSP resources to support PSTN connectivity to IP, necessary memory and recommended Cisco IOS feature set.

When to Sell

Sell This Product	When a Customer Needs These Features
Cisco Survivable Remote Site (SRST) Bundles	• Survivable Remote Site Telephony (SRST) for remote branch locations with Centralized Cisco CallManager deployments • Support for 24 to 720 phones in SRST mode • Customers who may deploy Cisco CallManager in the future and wants investment protection on initial purchase

Selected Part Numbers and Ordering Information[1]

Cisco Survivable Remote Site (SRST) Bundless

CISCO1760-V-SRST	1760 Router, 32MB Flash, 96MB DRAM, PVDM-256K-4, 24 Phone SRST FL, IOS IP Voice
CISCO2651XM-V-SRST	2651 Router, 32MB Flash, 128MB DRAM, AIM-VOICE-30 (DSP for 30 calls), 48 Phone FL, IOS IP Voice.
CISCO2801-SRST/K9	2801 Router, 64MB Flash, 256MB DRAM, PVDM2-8 (DSP for 8 calls), IOS SP Services
CISCO2811-SRST/K9	2811 Router, 64MB Flash, 256MB DRAM, PVDM2-16 (DSP for 16 calls), IOS SP Services
CISCO2821-SRST/K9	2821 Router, 64MB Flash, 256MB DRAM, PVDM2-32 (DSP for 32 calls), IOS SP Services
CISCO2851-SRST/K9	2851 Router, 64MB Flash, 256MB DRAM, PVDM2-48 (DSP for 48 calls), IOS SP Services
CISCO3825-SRST/K9	3825 Router, 64MB Flash, 256MB DRAM, PVDM2-64 (DSP for 64 calls), IOS SP Services
CISCO3845-SRST/K9	3845 Router, 64MB Flash, 256MB DRAM, PVDM2-64 (DSP for 64 calls), IOS SP Services

1. This is only a small subset of all parts available via URL listed under "For More Information." Some parts have restricted access or are not available through distribution channels.

For More Information

See Cisco Survivable Remote Site Telephony (SRST), page 4-20 for product details.

Cisco V3PN Bundles

Designed for enterprise branch offices, small and midsize businesses, and managed network service providers who seek security and IPCommunications capabilities in the same platform. V3PN bundle includes CME Features license, DSP resources for PSTN to IP calls, Advanced IP Services and AIM-VPN Accelerator card for maximum performance of security features including VPN tunneling

When to Sell

Sell This Product	When a Customer Needs These Features
Cisco V3PN Bundles	• SMB or Enterprise branch locations looking for data, IP Telephony and VPN in a all in one solution based on Cisco Access router • Support for 24 to 240 Phones per location • Customers who may need IP Telephony application in the future and wants investment protection on initial purchase

Selected Part Numbers and Ordering Information[1]

Cisco V3PN Bundles	
CISCO2801-V3PN/K9	2801 Router, 64MB Flash, 256MB DRAM, PVDM2-8 (DSP for 8 calls), IOS SP Services
CISCO2811-V3PN/K9	2811 Router, 64MB Flash, 256MB DRAM, PVDM2-16 (DSP for 16 calls), IOS SP Services
CISCO2821-V3PN/K9	2821 Router, 64MB Flash, 256MB DRAM, PVDM2-32 (DSP for 32 calls), IOS SP Services
CISCO2851-V3PN/K9	2851 Router, 64MB Flash, 256MB DRAM, PVDM2-48 (DSP for 48 calls), IOS SP Services
CISCO3825-V3PN/K9	3825 Router, 64MB Flash, 256MB DRAM, PVDM2-64 (DSP for 64 calls), IOS SP Services
CISCO3845-V3PN/K9	3845 Router, 64MB Flash, 256MB DRAM, PVDM2-64 (DSP for 64 calls), IOS SP Services

1. This is only a small subset of all parts available via URL listed under “For More Information.” Some parts have restricted access or are not available through distribution channels.

For More Information

See individual product pages for more detail