Data Sheet

Cisco 1861 and Cisco 2800, 3800, 2900, 3900, and 3900E Series Integrated Services Router Interoperability with Cisco Unified Communications Manager

Cisco® 1861 and Cisco 2800, 3800, 2900, 3900, and 3900E Series Integrated Services Routers can be deployed as unified communications routers as part of the Cisco Unified Communications and Collaboration Solution. New and existing deployments can benefit by using any of these routers as unified communications gateways with Cisco Unified Communications Manager.

Cisco 1861 and Cisco 2800, 3800, 2900, 3900, and 3900E Series unified communications routers communicate directly with Cisco Unified Communications Manager, allowing for the deployment of unified communications solutions that are ideal for small and medium-sized businesses, large enterprises, and service providers that offer managed network services. The Cisco 1861 and Cisco 2800, 3800, 2900, 3900, and 3900E Series Routers provide a highly flexible and scalable solution for small and medium-sized branch and regional offices.

The Cisco 1861 and Cisco 2800, 3800, 2900, 3900, and 3900E Series unified communications routers support a wide range of packet telephony-based voice interfaces and signaling protocols within the industry, providing connectivity support for more than 90 percent of the world's private branch exchanges (PBXs) and public-switched-telephone-network (PSTN) connection points. Signaling support includes T1/E1 Primary Rate Interface (PRI), T1 channel associated signaling (CAS), E1-R2, T1/E1 QSIG protocol, T1 Feature Group D (FGD), Basic Rate Interface (BRI), foreign exchange office (FXO), ear and mouth (E&M), and foreign exchange station (FXS). You can configure these unified communications routers to support from 2 to 720 voice channels. Additionally, you can use these unified communications routers to terminate Session Initiation Protocol (SIP) trunking into the enterprise or branch office by enabling the Cisco Unified Border Element features. Additional details are available in the Cisco Unified Border Element data sheet.

As your enterprise seeks to deploy an expanding list of unified communications applications and services, Cisco 1861 and Cisco 2800, 3800, 2900, 3900, and 3900E Series unified communications routers - interoperating with Cisco Unified Communications Manager - can provide a solution that will grow with your changing needs.

Interoperability Using SIP, H.323, or MGCP

The unified communications routers can communicate with the Cisco Unified Communications Manager using Session Initiation Protocol (SIP), H.323, or Media Gateway Control Protocol (MGCP):

- In SIP and H.323 mode, the unified communications routers communicate with Cisco Unified Communications Manager as intelligent gateway devices.
- In MGCP mode, these routers operate as stateless clients, giving Cisco Unified Communications Manager full control. Dial plans are configured centrally in Cisco Unified Communications Manager. Then you can automatically configure voice-gateway routers by downloading XML files.

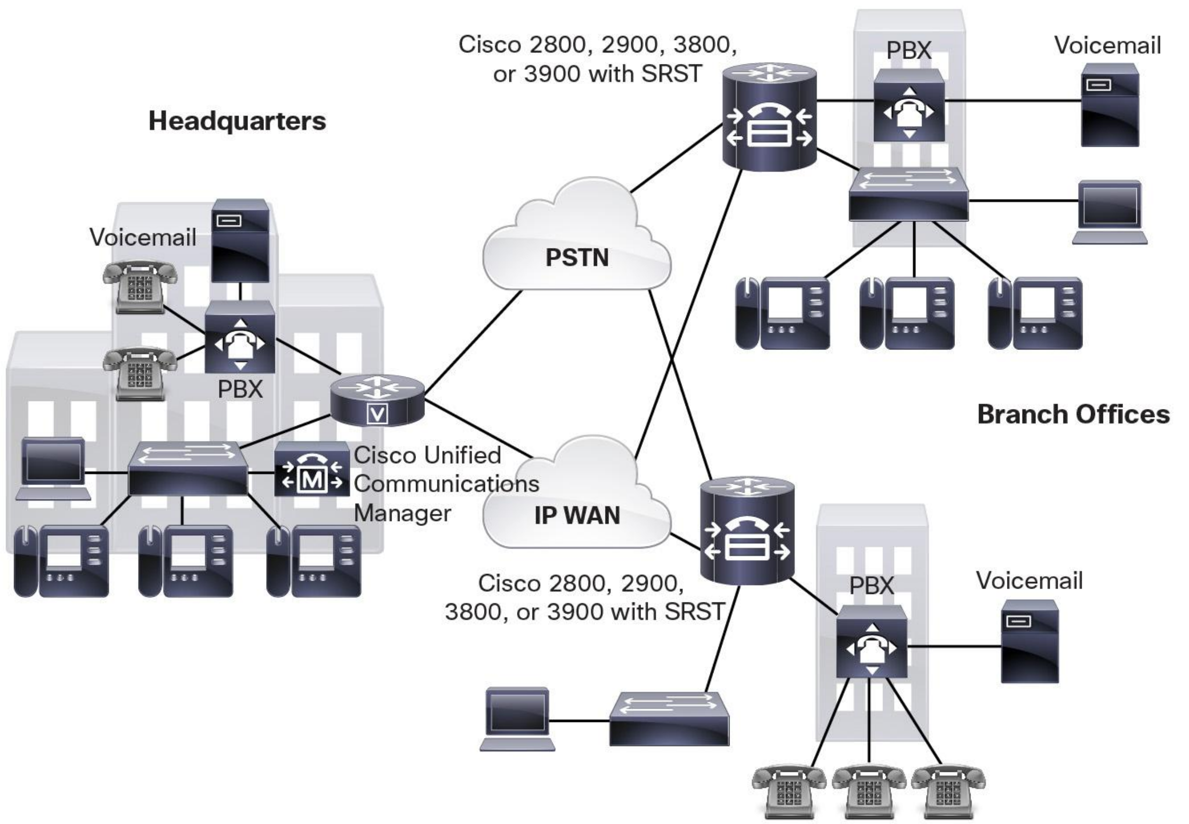
IP Telephony Phased Migration

The Cisco 1861 and Cisco 2800, 3800, 2900, 3900, and 3900E Series unified communications routers can help you immediately deploy an end-to-end unified communications network architecture or gradually shift voice traffic from traditional circuit-switched networks to a single infrastructure carrying data, voice, and video over packet networks.

Initially, you can use these unified communications routers to interconnect older PBXs over the packet infrastructure and still maintain PSTN (off-net) connectivity through your circuit-switched PBXs. Later, you can migrate PSTN (off-net) connectivity to the unified communications routers and start to incorporate IP phones at larger sites (Figure 1). After all sites are running IP telephony, you can begin deploying IP-based applications such as IP unified messaging, personal assistants, and extension mobility.

The unified communications routers are an ideal solution for circuit-switched PBX and PSTN access within a Cisco Unified Communications Manager-based IP telephony architecture.

Figure 1. IP Telephony Phased Migration: Migrate Circuit-Switched PSTN and PBX Connectivity to Unified Communications



As companies seek to deploy unified communications solutions across the entire enterprise - converging voice, video, and data across potentially thousands of sites - they require a solution that offers simple administration, virtually unlimited scalability, and high availability. The unified communications routers work in concert with the Cisco Unified Communications Manager, deployed in either a distributed or centralized call-processing model, to provide the unified communications solutions that enterprises require.

Centralized Call Processing

Demand for technology to help increase employee productivity and reduce costs is at an all-time high. At the same time, many organizations are struggling to deploy new applications and services because of unavailable capital budgets. The centralized call-processing model can provide technology to users who require it, while simultaneously providing ease of centralized management and maintenance of applications to network administrators.

Instead of deploying and managing key systems or PBXs in small offices, applications are centrally located at a corporate headquarters or data center, and accessed through the IP LAN and WAN. This deployment model allows branch-office users to access the full enterprise suite of communications and productivity applications for the first time, while lowering total cost of ownership (TCO). There is no need to "touch" each branch office each time a software upgrade or new application is deployed, accelerating the speed in which organizations can adopt and deploy new technology solutions.

The ability to quickly roll out new applications to remote users can provide a sustainable competitive advantage versus having to visit each of many branch-office sites to take advantage of new applications. An architecture in which a Cisco Unified Communications Manager and other Cisco IP Communications applications are located at the central site offers the following benefits:

- Centralized configuration and management
- Access at every site to all Cisco Unified Communications Manager features, next-generation contact centers, unified messaging services, personal productivity tools, mobility solutions, and software-based phones all the time
- IT staff not required at each remote site
- Ability to rapidly deploy applications to remote users
- Easy upgrades and maintenance
- Lower TCO

Survivable Remote Site Telephony

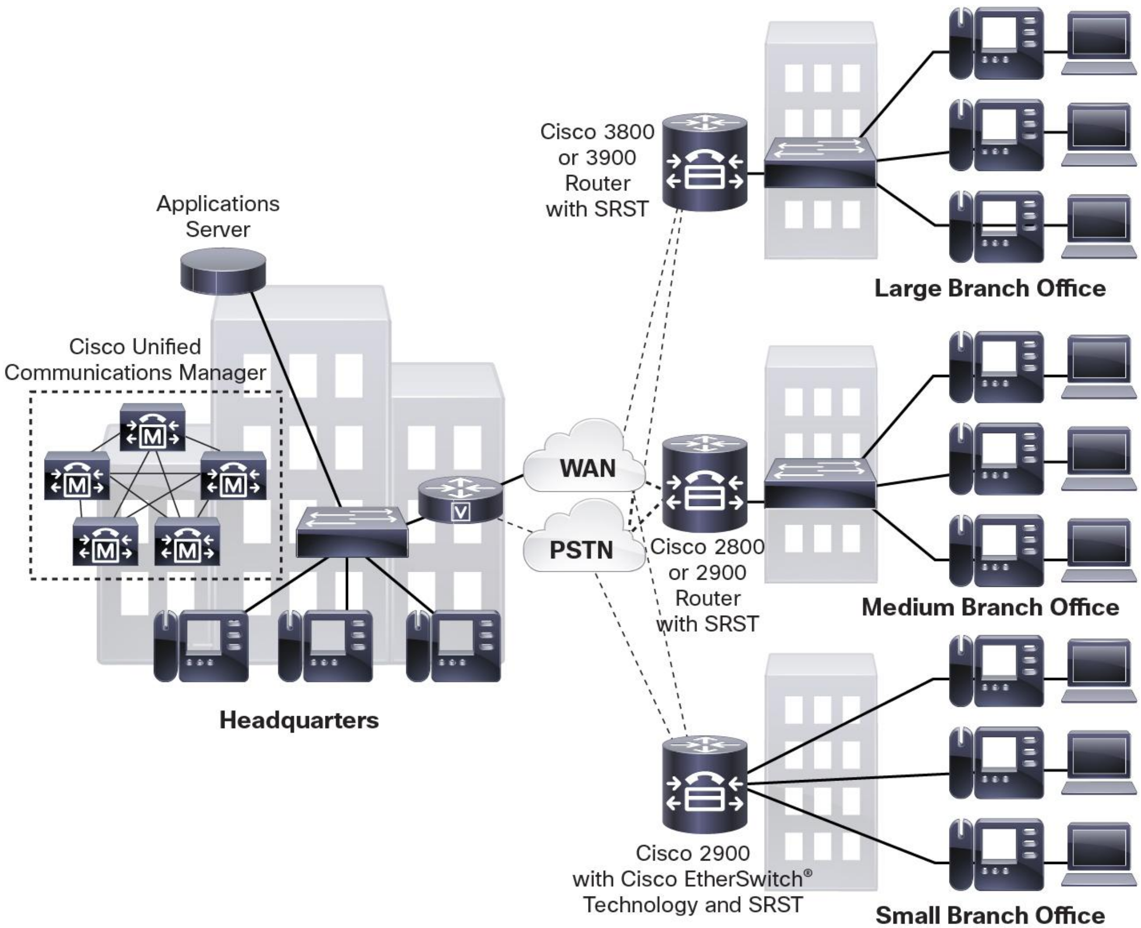
As enterprises extend their IP telephony deployments from central sites to remote offices, an important consideration is the ability to cost-effectively provide failover capability at remote branch offices. However, the size and number of these small-office sites preclude most enterprises from deploying dedicated call-processing servers, unified messaging servers, or multiple WAN links to each site to achieve the required high availability.

Cisco Unified Communications Manager with Survivable Remote Site Telephony (SRST) allows companies to extend high-availability IP telephony to their remote branch offices with a cost-effective solution that is easy to deploy, administer, and maintain. The SRST capability is embedded in the Cisco IOS® Software that runs on the Cisco 1861 and Cisco 2800, 3800, 2900, 3900, and 3900E Series unified communications routers.

SRST software automatically detects a connectivity failure between Cisco Unified Communications Manager and IP phones at the branch office. Using the Cisco Simple Network Automated Provisioning capability, SRST initiates a process to automatically configure the unified communications routers to provide call-processing backup redundancy for the IP phones and PSTN access in the affected office. The router provides essential call-processing services for the duration of the failure, helping ensure that critical phone capabilities are operational.

Upon restoration of the connectivity to the Cisco Unified Communications Manager, the system automatically shifts call-processing functions back to the primary Cisco Unified Communications Manager cluster. Configuration for this capability is performed only once in the Cisco Unified Communications Manager at the central site (Figure 2).

Figure 2. Centralized Cisco Unified Communications Manager Deployment with SRST



Cisco Unified Communications Router Features and Benefits

Simple Administration

- Provides centralized administration and management
- Enables administration of large dial plans
- Provides a single point of configuration for a Cisco IP Telephony network

Availability

- Provides for Cisco Unified Communications Manager redundancy; if a primary host Cisco Unified Communications Manager fails, call control fails over to the next available Cisco Unified Communications Manager server
- Offers branch-office survivability using SRST when connection to the Cisco Unified Communications Manager cluster is lost

Scalability

- Meets enterprise office requirements of small offices to large corporations
- Scales up to 30,000 users per cluster with Cisco Unified Communications Manager clustering

Investment Protection

- Provides a modular platform design with a growing list of more than 90 interface combinations
- Allows you to increase voice capacity while taking advantage of your existing investments in Cisco Unified Communications routers

Unified Communications Router with Cisco Unified Communications Manager Feature Summary

Table 1 summarizes the features of the unified communications routers with Cisco Unified Communications Manager.

Table 1. Cisco Unified Communications Routers with Cisco Unified Communications Manager Feature Summary

SIP	MGCP	H.323	Feature	Benefits
Y	Y[1]	Y	Analog FXS interfaces loop-start and ground-start signaling	This signaling facilitates direct connection to phones, fax machines, and key systems.
Y	N	Y	Analog E&M (wink, immediate, and delay) interfaces	These interfaces make direct connection to a PBX possible.
Y	Y	Y	Analog FXO interfaces loop-start and ground-start signaling	This feature facilitates connection to a PBX or key system and provides off-premises connections to or from the PSTN. Calling line ID (CLID) is available in MGCP mode.[2]
Y	N	Y	Analog direct inward dialing (DID)	Analog DID enables connection to the PSTN with DID operation.
Y	N	Y	Analog Centralized Automated Message Accounting (CAMA)	Analog CAMA facilitates analog PSTN connection for E-911 support.
Y	Y	Y	BRI Q.931 user side (NET3)	This feature enables connection to PSTN.
Y	N	Y	BRI Q.931 network side (NET3)	This feature enables connection to a PBX.
Y	Y	Y	BRI Q.SIG-basic call (including calling number)	This feature facilitates connection to a PBX or key system.
Y	N	N[3]	BRI Q.SIG forward, transfer, and conference	These services enable connection to a PBX or key system.
N	Y[4]	N	T1 E&M hookflash	This feature is used to transfer a call from time-division multiplexing (TDM) interactive voice response (IVR) to a PSTN or IP phone destination.
Y	Y	Y	T1-CAS E&M (wink-start and immediate-start) interfaces	These interfaces facilitate connection to a PBX, key system, or PSTN.
Y	N	Y	T1-CAS E&M (delay dial) interfaces	These interfaces facilitate connection to a PBX, key system, or PSTN.
Y	N	Y	T1-CAS feature group D[5]	This feature is used to connect to a PBX or PSTN.
Y	N	Y	T1-CAS FXO (ground-start and loop-start) interfaces	These interfaces are used to connect to a PBX or key system and to provide off-premises connections.
Y	N	Y	T1-CAS FXS (ground-start and loop-start) interfaces	These interfaces are used to connect to a PBX or key system.
Y	N	Y	E1 CAS	E1 CAS enables connection to a PBX or PSTN.

[1] Supports loop-start signaling only
[2] Requires Cisco IOS Software Release 12.4(20)T or later and Cisco Unified Communications Manager 8.0 or later
[3] Supported between gateways in the absence of Cisco Unified Communications Manager
[4] Requires Cisco IOS Software Release 12.4(4)T or later and Cisco Unified Communications Manager 4.2 or later
[5] Not supported on the Cisco 1700 Series unified communications routers

SIP	MGCP	H.323	Feature	Benefits
Y	N	Y	E1 MelCAS	E1 MelCAS facilitates connection to a PBX or PSTN.
Y	N	Y	E1 R2 (more than 30 country variants)	E1 R2 enables connection to a PBX or PSTN.
Y	Y	Y	T1/E1 ISDN PRI Q.931 interfaces	These interfaces are used to connect to a PBX or key system and to provide off-premises connections to or from the PSTN or post, telephone, and telegraph (PTT).
Y	Y	Y	T1/E1 Q.SIG basic call (including calling number)	This feature is used to connect to a PBX.
Y[6]	Y	N[3]	T1/E1 Q.SIG, including call diversion and forward, transfer, calling and connected ID services, and message-waiting indicator	This feature is used to connect to a PBX.
Y	Y	Y	Out-of-band dual-tone multifrequency (DTMF)	This feature carries DTMF tones and information out of band for clearer transmission and detection.
N	Y	N	Single point of gateway configuration for a Cisco IP Telephony network	This feature centralizes and automates the configuration process for MGCP unified communications routers by making them configurable on the Cisco Unified Communications Manager. Configuration information is automatically downloaded at startup and after any configuration change.
Y	Y	Y	Cisco Unified Communications Manager failover redundancy	When the unified communications router loses contact with the primary Cisco Unified Communications Manager, the gateway uses the next available Cisco Unified Communications Manager.
Y	Y	Y[7]	Cisco Unified Communications Manager call preservation during failover	Existing calls are preserved during a failover to the next available Cisco Unified Communications Manager. Calls are also preserved upon restoration of the primary host Cisco Unified Communications Manager.
Y	Y	Y	SRST and gateway fallback	When contact with the Cisco Unified Communications Manager cluster is lost, SRST provides basic call handling for the IP phones. Gateway fallback provides support for PSTN telephony interfaces on the branch-office router for the duration of the loss.
Y	N	Y[7]	Call preservation for existing BRI and PRI calls during gateway fallback and recovery	Existing calls are preserved during a loss of connection to the Cisco Unified Communications Manager cluster and gateway fallback. Calls are also preserved upon restoration of the Cisco Unified Communications Manager connection.
Y	Y	Y[7]	Call preservation for existing T1/E1 (CAS) and analog calls during gateway fallback and recovery	Existing calls are preserved during a loss of connection to the Cisco Unified Communications Manager cluster and gateway fallback. Calls are also preserved upon restoration of the Cisco Unified Communications Manager connection.
Y	Y	Y	Multicast music on hold (MoH) - centralized	This feature helps the unified communications router deliver music streams from a MoH server to users on on- and off-net calls.
N	Y	N	Multicast MoH - distributed	This feature helps the unified communications router deliver music streams to users through the router-embedded MoH server to on- and off-net calls.
N	Y	Y	Tone on hold	Tone indicates when a user is placed on hold.
N	Y	N	Tone-on-hold timer tuning	Tone on hold is generated locally in the gateway for play to the PSTN. Tone-on-hold timer tuning allows the use of service parameter settings in Cisco Unified Communications Manager for specification of the time between beeps.

[6] Support is for forward, transfer, and conference; message-waiting indicator is from SIP to QSIG (not the reverse) and requires Cisco IOS Software Release 12.4(11)T; calling and connected ID are not supported

[7] Requires Cisco Unified Communications Manager 4.1(3)SR2 or later and Cisco IOS Software Release 12.4(9)T or later; no gatekeeper support

SIP	MGCP	H.323	Feature	Benefits
Y	Y	Y	Caller ID support[8]	This feature helps the unified communications router send the caller ID of a caller for display: In MGCP mode, to and from IP phone, FXS, T1/E1 PRI; and FXO to IP phone, not conversely (caller ID currently not supported on T1-CAS). In SIP and H.323 mode, to and from IP phone, FXS, BRI, T1/E1 PRI; and from FXO to IP phone, FXS, BRI, and T1/E1 PRI, not conversely.
N	Y	Y	Malicious caller ID (MCID) over PRI	MCID over PRI facilitates malicious call notification to on-net personnel, flags the on-net call detail record (CDR), and notifies the off-net (PSTN) system (through the network interface) of the malicious nature of the call.
N	Y	N	Multilevel precedence and preemption (MLPP) for T1-PRI (backhaul) and T1-CAS (wink start only)	This feature helps assure high-ranking personnel communication to critical organizations and personnel during network stress situations. It allows priority calls for validated users to preempt lower-priority calls.
Y	Y	Y	Group III fax support	Group III fax support facilitates transmit of Group III faxes between the PSTN and IP using either fax relay or fax pass-through methods.
Y	Y[9]	Y	T.38 standards-based fax support	This feature enables transmit T.38 fax between the PSTN and IP.
Y	N	Y	Private-line automatic ringdown (PLAR)	PLAR provides a dedicated connection to another extension or an attendant.
Y	Y	Y	Standards-based codecs[10]	You can choose to transmit voice across your network as either uncompressed pulse code modulation (PCM) or compressed from 5.3 to 64 kbps using standards-based compression algorithms (G.711, G.729, G.729a/b, G.722, Internet Low Bitrate Codec [iLBC], G.723.1, G.726, or G.728).
Y	Y	Y	Voice activity detection (VAD)	VAD conserves bandwidth during a call when there is no active voice traffic to send.
Y	Y	Y	Comfort-noise generation	While using VAD, the digital signal processor (DSP) at the destination end emulates background noise from the source side, preventing the perception that a call is disconnected.
Y	N	Y	Busy out	When the WAN or LAN connection to the router is down or network conditions are such that a call cannot be admitted, this feature will “busy out” the trunk to the PBX or PSTN.
-	-	Y	H.323 ITU Version 1, 2, 3, and 4 support	These versions of H.323 use industry-standard signaling protocols for setting up calls between gateways, gatekeepers, and H.323 endpoints.
Y	-	-	SIP IETF RFC 3261 support	This feature uses industry-standard signaling protocols for setting up calls between gateways and SIP proxies or SIP Back-to-Back User Agents.
Y	Y	Y	Authentication, authorization, and accounting (AAA)	AAA supports debit card and credit card (prepaid and postpaid calling card) applications.
Y	N	Y	IVR support	IVR offers Automated-Attendant support, voicemail support, or call routing based on service desired.
Y	N	Y	Automated Attendant	This feature uses IVR to provide automated call-answering and -forwarding services.
Y	N	Y	VoiceXML	VoiceXML controls calls “in queue” at the gateway for call-center applications. Calls are redirected only when an agent becomes available.
N	Y	Y[7]	Overlap sending over voice over IP (VoIP)	This feature speeds variable-length dial strings dialing.

8 Requires Cisco IOS Software Release 12.4(20)T or later

9 Requires Cisco Unified Communications Manager 4.2(3)

10 G.722 is not supported with MGCP. G.722 requires Cisco IOS Software Release 12.4(20)T or later with Cisco Unified Communications Manager 5.0 or later. iLBC requires Cisco IOS Software Release 12.4(15)T or later with Cisco Unified Communications Manager 6.0 or later

SIP	MGCP	H.323	Feature	Benefits
Y	N	Y	Voice + Data integrated access	This feature makes the voice and serial data interfaces available on the same T1/E1.
Y	N	Y	Fractional PRI	This feature allows for use of fewer than 23/30 channels on a T1/E1. Other channels are either unused or used for data.
Y	N	Y	FXO tone answer supervision	This feature facilitates the use of tones to signal answering a call and the start of a CDR.
Y	Y	Y	FXO disconnect supervision	This feature makes battery reversal or tones available for use to disconnect FXO calls.
Y	N	Y	ISDN video switching on gateway (drop DSPs)	This feature allows ISDN-based videoconferencing systems to connect and be switched back out the ISDN.
Y	N	Y	Set numbering plan type of outgoing calls	You can change the numbering plan on the gateway before your call goes out over the PSTN.
N	Y	N	Billing granularity to DS-0 channel level on Cisco Unified Communications Manager CDR	This feature provides increased granularity on TDM usage down to the individual channel.
Y	Y	Y	Name display on PRI using FACILITY IE (caller name [CNAM])	This feature provides caller name display on IP phones for PSTN calls.
N	Y[11]	N	Secure Telephone Unit (STU) and Secure Terminal Equipment (STE) phone support	STU and STE support the U.S. Department of Defense analog and BRI secure phones.
N	Y[12]	N	Connection to Defense Switched Network (DSN)	This feature supports the U.S. Department of Defense private TDM network.
Y[13]	Y[14]	Y[15]	Secure Real-Time Transport Protocol (SRTP): Media authentication and encryption on unified communications routers	This feature enables secure gateway-to-gateway calls and secure IP phone-to-gateway calls.
Y	-	-	SRTP-Real-Time Transport Protocol (RTP) fallback operations	This feature enables the fallback from SRTP to RTP during capabilities negotiation at the time of call setup.
Y[16]	Y[17]	Y[18]	Signaling encryption SIP: Transport Layer Security (TLS), MGCP/H,323: IP Security IPsec)	This feature encrypts signaling communication between unified communications and Cisco Unified Communications Manager.
Y	N	Y	H.320 video gateway support	This feature integrates ISDN trunks with both voice and video traffic.
Y	N	Y	Virtualization (Virtual Route Forwarding [VRF])	This feature supports virtual segmentation of the network using VRF.
Y	N	N	IPv6	IPv6 support enables interworking with IPv6-capable networks.
Y	N	N	Dynamic Host Configuration Protocol (DHCP)	DHCP enables acquisition of gateway configuration parameters from the DHCP server.
Y	Y	Y	Resource Reservation Protocol (RSVP) support	This feature helps assure high-quality voice by enabling resource reservation for call admission control.
Y	-	-	History Info support	This feature enables support for the History Info header to transport the history information of a call.
Y	-	-	SIP privacy and identity	This feature enables transport of identity, both preferred (P-Preferred Identity [PPI]) and asserted (P-Asserted Identity [PAI]).
Y	-	-	Signaling health monitoring	This feature enables monitoring of the signaling connection across the signaling trunk.

[11] Requires Cisco IOS Software Release 12.3(14)T or later; BRI operations limited: single B-channel voice only; testing limited to three phones; no data call support
[12] Requires Cisco IOS Software Release 12.4(2)T or later
[13] Requires Cisco IOS Software Release 12.4(15)T or later and Cisco Unified Communications Manager 5.0 (line-side) or later; Cisco Unified Communications Manager trunk-side support currently not available
[14] Requires Cisco IOS Software Release 12.4(3) or later and Cisco Unified Communications Manager 4.1 or later
[15] Requires Cisco IOS Software Release 12.4(6)T2 or later and Cisco Unified Communications Manager 5.0 or later
[16] Requires Cisco IOS Software Release 12.4(6)T or later and Cisco Unified Communications Manager 5.0 or later
[17] Requires Cisco IOS Software Release 12.4(3) or later and Cisco Unified Communications Manager 4.1 or later
[18] Requires Cisco IOS Software Release 12.4(6)T1 and Cisco Unified Communications Manager 5.0 or later

SIP	MGCP	H.323	Feature	Benefits
Y	Y	Y	Q.SIG and Q.931 Tunneling	This feature enables transparent tunneling of ISDN signaling over VoIP signaling.
Y[19]	N	Y[19]	Ad hoc videoconference service and unified video transcoding service on Cisco Integrated Services Routers Generation 2 (ISR G2)	This feature enables ad hoc videoconferencing and unified video transcoding on the Cisco 2900 and 3900 Series Integrated Services Routers (ISRs)
N	Y[19]	N	Cisco V.150.1 Minimum Essential Requirements	This feature delivers enhancements to the voice gateways to satisfy requirements outlined in the UCR2008 specification. Specifically, support is added for the V.150.1 Minimum Essential Requirements (modem relay) and Modem over IP (MoIP) and Fax over IP (FoIP).

Unified Communications Router with Cisco Unified Communications Manager Minimum System Requirements

Tables 2 through 5 give system requirements for the unified communications routers.

Table 2. Cisco Unified Communications Routers with Cisco Unified Communications Manager Minimum System Requirements Using SIP

TDM Protocol or Feature	Minimum Cisco IOS Software Release*	Minimum Cisco Unified Communications Manager Release
Analog (FXS and FXO)	12.4(6)T	5.0
BRI	12.4(6)T	5.0
T1 CAS and T1/E1 PRI	12.4(6)T	5.0

* This table shows when a Cisco IOS Software particular interface type was first tested with Cisco Unified Communications Manager. It does not document when individual network modules (NMs), advanced integration modules (AIMs), service modules (SMs), integrated service modules (ISMs), and platforms are first supported in Cisco IOS Software. For this information refer to the data sheet for the relevant interface. Note that when using SIP, Cisco Unified Communications Manager does not need to know which NM, SM, AIM, ISM, or platform is used. Hence, when Cisco Unified Communications Manager supports a particular protocol or feature, this support is sufficient for operation.

Table 3. Cisco Unified Communications Routers with Cisco Unified Communications Manager Minimum System Requirements Using H.323

TDM Protocol or Feature	Minimum Cisco IOS Software Release*	Minimum Cisco Unified Communications Manager Release
Analog (FXS and FXO)	12.2(1)M	3.0(5a)
BRI	12.2(1)M	3.0(5a)
T1 CAS and T1/E1 PRI	12.1(2)T	3.0(5a)
T1/E1 QSIG	12.1(2)T	3.0(5a)
MCID	12.3(11)T	4.0

* This table shows when a particular interface type is first supported in Cisco IOS Software. It does not document when individual NMs, SMs, AIMs, ISMs, and platforms are first supported in Cisco IOS Software. For this information refer to the data sheet for the relevant interface. Note that in H.323 mode, Cisco Unified Communications Manager does not need to know which NM, SM, AIM, ISM, or platform is used. Hence, when Cisco Unified Communications Manager supports a particular protocol or feature, this support is sufficient for operation.

[19] Requires Cisco IOS Software Release 15.1(4)M or later and Cisco Unified Communications Manager 8.6 or later

Table 4. Cisco Unified Communications Routers with Cisco Unified Communications Manager Minimum System Requirements Using MGCP

Active Platforms	Interface Part Number	TDM Protocol or Feature	Minimum Cisco IOS Software Release	Minimum Cisco Unified Communications Manager Release
Cisco 2801	VIC2-2FXS, VIC2-2FXO, VIC-4FXS/DID, and VIC2-4FXO	Analog FXS and FXO	12.3.8T4	3.3.5, 4.0.2a SR1, or 4.1.2 SR1
	VIC2-2BRI -NT/TE	BRI	12.3.14T	4.1.2 SR1
	VWIC-1MFT-T1/E1, VWIC-2MFT-T1/E1, VWIC-2MFT-T1/E1-DI, VWIC-1MFT-G703, and VWIC-2MFT-G703	T1 CAS E&M, T1/E1 PRI, and QSIG (basic)	12.3.8T4	3.3.5, 4.0.2a SR1, or 4.1.2 SR1
	VWIC-1MFT-T1/E1, VWIC-2MFT-T1/E1, VWIC-2MFT-T1/E1-DI, VWIC-1MFT-G703, and VWIC-2MFT-G703	T1/E1QSIG supplementary services	12.3.14T	4.0.2a SR1 or 4.1.2 SR1
	VWIC2-1MFT-T1/E1, VWIC2-2MFT-T1/E1, VWIC2-1MFT-G703, and VWIC2-2MFT-G703	T1 CAS E&M, T1/E1 PRI, and QSIG supplementary services	12.3.14T	4.0.2a SR1 or 4.1.2 SR1
	VIC3-2FXS/DID, VIC3-2FXS-E/DID, VIC3-4FXS/DID	Analog FXS and FXO	12.4.20T	6.0.1
Cisco 2811, 2821, and 2851	EVM-HD-8FXS/DID with EM-HDA-8FXS, EM3-HDA-8FXS, EM-HDA-6FXO, or EM-HDA-3FXS/4FXO	Analog FXS and FXO	12.3.11T	3.3.5, 4.0.2a SR1, or 4.1.2
	EVM-HD-8FXS/DID with EM-4BRI-NT/TE	BRI	12.4.2T	4.1.3 SR1
	NM-HDA	Analog FXS and FXO	12.3.8T4	3.3.5, 4.0.2a SR1, or 4.1.2
	NM-HDV	T1 CAS E&M and T1/E1 PRI	12.3.8T4	3.3.5, 4.0.2a SR1, or 4.1.2
	NM-HDV	T1/E1 QSIG (basic)	12.3.8T4	3.3.5, 4.0.2a SR1, or 4.1.2
	NM-HD-1V/2V/2VE	Analog FXS and FXO, T1 CAS E&M, T1/E1 PRI, and T1/E1 QSIG (basic)	12.3.8T4	3.3.5, 4.0.2a SR1, or 4.1.2
	VIC2-2BRI-NT/TE, NM-HD-1V/2V/2VE, and NM-HDV2	BRI	12.4.2T	4.1.3 SR1
	NM-HDV2	T1 CAS E&M, T1/E1 PRI, and T1/E1 QSIG (basic)	12.3.8T4	3.3.5, 4.0.2a SR1, or 4.1.2
	VWIC-1MFT-T1/E1, VWIC-2MFT-T1/E1, VWIC-2MFT-T1/E1-DI, VWIC-1MFT-G703, and VWIC-2MFT-G703	T1/E1QSIG[**] and MLPP[***]	12.3.11T	4.0.2a SR1 or 4.1.2
	VWIC2-1MFT-T1/E1, VWIC2-2MFT-T1/E1, VWIC2-1MFT-G703, and VWIC2-2MFT-G703	T1 CAS E&M, T1/E1 PRI, QSIG[**], and MLPP[***]	12.3.14T	4.0.2a SR2 or 4.1.3
	VIC3-2FXS/DID, VIC3-2FXS-E/DID, VIC3-4FXS/DID	Analog FXS and FXO	12.4.20T	6.0.1

Active Platforms	Interface Part Number	TDM Protocol or Feature	Minimum Cisco IOS Software Release	Minimum Cisco Unified Communications Manager Release
Cisco 3825 and 3845	EVM-HD-8FXS/DID with EM-HDA-8FXS, EM3-HDA-8FXS, EM-HDA-6FXO, or EM-HDA-3FXS/4FXO	Analog FXS and FXO	12.3.11T	3.3.5, 4.0.2a SR1, or 4.1.2
	EVM-HD-8FXS/DID with EM-4BRI-NT/TE	BRI	12.4.2T	4.1.3 SR1
	NM-HDA	Analog FXS and FXO	12.3.11T	3.3.5, 4.0.2a SR1, or 4.1.2
	NM-HDV	T1 CAS E&M and T1/E1 PRI	12.3.11T	3.3.5, 4.0.2a SR1, or 4.1.2
	NM-HDV	T1/E1 QSIG (basic)	12.3.11T	3.3.5, 4.0.2a SR1, or 4.1.2
	NM-HD-1V/2V/2VE	Analog FXS and FXO, T1 CAS E&M, T1/E1 PRI, and T1/E1 QSIG (basic)	12.3.11T	3.3.5, 4.0.2a SR1, or 4.1.2
	VIC2-2BRI-NT/TE, NM-HD-1V/2V/2VE, and NM-HDV2	BRI	12.4.2T	4.1.3 SR1
	NM-HDV2	T1 CAS E&M, T1/E1 PRI, and T1/E1 QSIG (basic)	12.3.11T	3.3.5, 4.0.2a SR1, or 4.1.2
	VWIC-1MFT-T1/E1, VWIC-2MFT-T1/E1, VWIC-2MFT-T1/E1-DI, VWIC-1MFT-G703, and VWIC-2MFT-G703	T1/E1QSIG** and MLPP***	12.3.11T	4.0.2a SR1 or 4.1.2
	VWIC2-1MFT-T1/E1, VWIC2-2MFT-T1/E1, VWIC2-1MFT-G703, and VWIC2-2MFT-G703	T1 CAS E&M, T1/E1 PRI, QSIG**, and MLPP***	12.3.14T	4.0.2a SR2 or 4.1.3
	VIC3-2FXS/DID, VIC3-2FXS-E/DID, and VIC3-4FXS/DID	Analog FXS and FXO	12.4(20)T	6.0.1
Cisco 2901, 2911, 2921, and 2951	EVM-HD-8FXS/DID with EM3-HDA-8FXS, EM-HDA-6FXO, or EM-HDA-3FXS/4FXO (Cisco 2911, 2921, and 2951 only)	Analog FXS and FXO	15.0.1M	6.1.5, 7.1.3, or 8.0
	VIC3-2FXS/DID, VIC3-2FXS-E/DID, VIC3-4FXS/DID, NM-HD-1V/2V/2VE, and NM-HDV2	Analog FXS and FXO	15.0.1M	6.1.5, 7.1.3, or 8.0
	EVM-HD-8FXS/DID with EM-4BRI-NT/TE, VIC2-2BRI-NT/TE, NM-HD-1V/2V/2VE, and NM-HDV2	BRI	15.0.1M	6.1.5, 7.1.3 or 8.0
	NM-HDV2	T1 CAS E&M, T1/E1 PRI, and T1/E1 QSIG (basic)	15.0.1M	6.1.5, 7.1.3 or 8.0
	VWIC2-1MFT-T1/E1, VWIC2-2MFT-T1/E1, VWIC2-1MFT-G703, and VWIC2-2MFT-G703	T1 CAS E&M, T1/E1 PRI, QSIG**, and MLPP***	15.0.1M	6.1.5, 7.1.3 or 8.0
	VWIC3-1MFT-T1/E1, VWIC3-2MFT-T1/E1, VWIC3-1MFT-G703, and VWIC3-2MFT-G703	T1 CAS, T1/E1 PRI, QSIG**, MLPP*** and channel groups	15.0.1M3	7.1.5 or 8.0.2
	VWIC3-4MFT-T1/E1 Not supported on Cisco 2901	T1 CAS, T1/E1 PRI, QSIG**, MLPP*** and channel groups	15.1.3T	7.1.5 or 8.0.2

Active Platforms	Interface Part Number	TDM Protocol or Feature	Minimum Cisco IOS Software Release	Minimum Cisco Unified Communications Manager Release
Cisco 3925 and 3945	EVM-HD-8FXS/DID with EM3-HDA-8FXS, EM-HDA-6FXO, or EM-HDA-3FXS/4FXO	Analog FXS and FXO	15.0.1M	6.1.5, 7.1.3, or 8.0
	VIC3-2FXS/DID, VIC3-2FXS-E/DID, VIC3-4FXS/DID, NM-HD-1V/2V/2VE, and NM-HDV2	Analog FXS and FXO	15.0.1M	6.1.5, 7.1.3, or 8.0
	EVM-HD-8FXS/DID with EM-4BRI-NT/TE, VIC2-2BRI-NT/TE, NM-HD-1V/2V/2VE, and NM-HDV2	BRI	15.0.1M	6.1.5, 7.1.3, or 8.0
	NM-HDV2	T1 CAS E&M, T1/E1 PRI, and T1/E1 QSIG (basic)	15.0.1M	6.1.5, 7.1.3, or 8.0
	VWIC2-1MFT-T1/E1, VWIC2-2MFT-T1/E1, VWIC2-1MFT-G703, and VWIC2-2MFT-G703	T1 CAS E&M, T1/E1 PRI, QSIG**, and MLPP***	15.0.1M	6.1.5, 7.1.3, or 8.0
	VWIC3-1MFT-T1/E1, VWIC3-2MFT-T1/E1, VWIC3-1MFT-G703, and VWIC3-2MFT-G703	T1 CAS, T1/E1 PRI, QSIG**, MLPP*** and channel-groups	15.0.1M3	7.1.5 or 8.0.2
	VWIC3-4MFT-T1/E1	T1 CAS, T1/E1 PRI, QSIG**, MLPP*** and channel groups	15.1.3T	7.1.5 or 8.0.2
Cisco 3925E and 3945E	EVM-HD-8FXS/DID with EM3-HDA-8FXS, EM-HDA-6FXO, or EM-HDA-3FXS/4FXO	Analog FXS and FXO	15.1.1T	7.1.5 or 8.0.2
	VIC3-2FXS/DID, VIC3-2FXS-E/DID, VIC3-4FXS/DID, NM-HD-1V/2V/2VE, and NM-HDV2	Analog FXS and FXO	15.1.1T	7.1.5 or 8.0.2
	EVM-HD-8FXS/DID with EM-4BRI-NT/TE, VIC2-2BRI-NT/TE, NM-HD-1V/2V/2VE, and NM-HDV2	BRI	15.1.1T	7.1.5 or 8.0.2
	NM-HDV2	T1 CAS E&M, T1/E1 PRI, and T1/E1 QSIG (basic)	15.1.1T	7.1.5 or 8.0.2
	VWIC2-1MFT-T1/E1, VWIC2-2MFT-T1/E1, VWIC2-1MFT-G703, and VWIC2-2MFT-G703	T1 CAS E&M, T1/E1 PRI, QSIG**, and MLPP***	15.1.1T	7.1.5 or 8.0.2
	VWIC3-1MFT-T1/E1, VWIC3-2MFT-T1/E1, VWIC3-1MFT-G703, and VWIC3-2MFT-G703	T1 CAS, T1/E1 PRI, QSIG**, MLPP*** and channel-groups	15.0.1M3	7.1.5 or 8.0.2
	VWIC3-4MFT-T1/E1	T1 CAS, T1/E1 PRI, QSIG**, MLPP*** and channel-groups	15.1.3T	7.1.5 or 8.0.2

** QSIG supplementary requires Cisco Unified Communications Manager 4.0 or later. QSIG basic services were first introduced with Cisco Communications Manager 3.3 and Cisco IOS Software Release 12.2.11T.

*** MLPP requires Cisco Unified Communications Manager 4.0.2 or later.

Table 5. Cisco Unified Communications Routers with Cisco Unified Communications Manager Minimum System Requirements for Conferencing, Transcoding, and Media Termination Point

Active Platforms	Interface Part Numbers	TDM Protocol or Feature	Minimum Cisco IOS Software Release	Minimum Cisco Unified Communications Manager Release
Cisco 2801	Onboard PVDM2 DSPs	Conferencing, transcoding, and media termination point (MTP)	12.3.11T	4.0.2a SR1 or 4.1.2 SR1
Cisco 2811, 2821, and 2851	Onboard PVDM2 DSPs	Conferencing and transcoding	12.3.8T4	3.3.5, 4.0.2a SR1, or 4.1.2
	Onboard PVDM2 DSPs	MTP	12.3.8T4	4.0.2a SR1 or 4.1.2
	Onboard PVDM2 DSPs	RFC 2833 MTP	12.3.11T	4.0.2a SR1 or 4.1.2
	NM-HD-1V/2V/2VE and NM-HDV2	Conferencing and transcoding	12.3.8T4	3.3.5, 4.0.2a SR1, or 4.1.2
	NM-HD-1V/2V/2VE and NM-HDV2	MTP	12.3.8T4	4.0.2a SR1 or 4.1.2
	NM-HD-1V/2V/2VE and NM-HDV2	RFC 2833 MTP	12.3.11T	4.0.2a SR1 or 4.1.2
	NM-HDV and NM-HDV-FARM	Conferencing and transcoding	12.3.8T4	3.3.5, 4.0.2a SR1, or 4.1.2
Cisco 2901, 2911, 2921, and 2951	Onboard PVDM2 and PVDM3 DSPs	Conferencing and transcoding	15.0.1M	6.1.5, 7.1.3 or 8.0
	Onboard PVDM2 and PVDM3 DSPs	MTP and RFC 2833	15.0.1M	6.1.5, 7.1.3 or 8.0
	NM-HD-1V/2V/2VE and NM-HDV2	Conferencing and transcoding	15.0.1M	6.1.5, 7.1.3 or 8.0
	NM-HD-1V/2V/2VE and NM-HDV2	MTP and RFC 2833	15.0.1M	6.1.5, 7.1.3 or 8.0
Cisco 3825 and 3845	Onboard PVDM2 DSPs	Conferencing and transcoding	12.3.11T	3.3.5, 4.0.2a SR1, or 4.1.2
	Onboard PVDM2 DSPs	MTP and RFC 2833	12.3.11T	4.0.2a SR1 or 4.1.2
	NM-HD-1V/2V/2VE and NM-HDV2	Conferencing and transcoding	12.3.11T	3.3.5, 4.0.2a SR1, or 4.1.2
	NM-HD-1V/2V/2VE and NM-HDV2	MTP and RFC 2833	12.3.11T	4.0.2a SR1 or 4.1.2
	NM-HDV and NM-HDV-FARM	Conferencing and transcoding	12.3.8T4	3.3.5, 4.0.2a SR1, or 4.1.2
Cisco 3925 and 3945	Onboard PVDM2 and PVDM3 DSPs	Conferencing and transcoding	15.0.1M	6.1.5, 7.1.3, or 8.0
	Onboard PVDM2 and PVDM3 DSPs	MTP and RFC 2833	15.0.1M	6.1.5, 7.1.3, or 8.0
	NM-HD-1V/2V/2VE and NM-HDV2	Conferencing and transcoding	15.0.1M	6.1.5, 7.1.3, or 8.0
	NM-HD-1V/2V/2VE and NM-HDV2	MTP and RFC 2833	15.0.1M	6.1.5, 7.1.3, or 8.0
Cisco 3925E and 3945E	Onboard PVDM2 and PVDM3 DSPs	Conferencing and transcoding	15.1.1T	7.1.5 or 8.0.2
	Onboard PVDM2 and PVDM3 DSPs	MTP and RFC 2833	15.1.1T	7.1.5 or 8.0.2
	NM-HD-1V/2V/2VE and NM-HDV2	Conferencing and transcoding	15.1.1T	7.1.5 or 8.0.2
	NM-HD-1V/2V/2VE and NM-HDV2	MTP and RFC 2833	15.1.1T	7.1.5 or 8.0.2

Voice Performance

Tables 6 and 7 give information about connectivity and CPU performance, respectively, on the unified communications routers.

Table 6. Maximum Physical DS-0 Connectivity on the Cisco Unified Communications Routers[*]

	Cisco 1861	Cisco 2801	Cisco 2811	Cisco 2821	Cisco 2851	Cisco 3825	Cisco 3845	Cisco 2901	Cisco 2911	Cisco 2921	Cisco 2951	Cisco 3925	Cisco 3945	Cisco 3925E	Cisco 3945E
FXS	4	16	28	52	52	52	88	16	40	40	64	64	112	60	108
FXO and CAMA	4	16	24	36	36	36	56	16	28	28	40	40	64	36	60
E&M	-	8	12	12	12	16	24	8	12	12	16	16	24	14	22
Analog DID	-	8	24	32	32	32	48	16	32	32	48	48	80	44	76
BRI ports	2	8	12	20	20	20	32	8	16	16	24	24	40	22	38
T1/E1 ports	-	1	12	12	12	16	24	8	20	20	24	24	32	20	28
T1 channels	-	24	288	288	288	384	576	192	480	480	576	576	768	480	672
E1 channels	-	30	360	360	360	480	720	240	600	600	720	720	960	600	840

* This table contains physical connectivity numbers. You should also use CPU performance as a guide to determine how many voice calls can actually be supported on each platform.

Table 7. CPU Performance on the Cisco Unified Communications Routers[*]

	Cisco 2801	Cisco 2811	Cisco 2821	Cisco 2851	Cisco 3825	Cisco 3845	Cisco 2901	Cisco 2911	Cisco 2921	Cisco 2951	Cisco 3925	Cisco 3945	Cisco 3925E	Cisco 3945E
VoIP Performance: Maximum Number of Simultaneous Calls (not exceeding 75-percent platform CPU usage)														
Cisco Unified Border Element														
	55	110	200	225	400	500	100	200	400	600	800	950	2100	2500
Standalone Unified Communications Router[20]														
No encryption	32	70	112	170	340	450	100	150	240	400	720	960	600	840
SIP TLS with SRTP	32	65	104	160	320	420	100	150	240	400	720	880	600	840
H.323 Signaling in IPsec with SRTP	32	60	96	140	290	370	100	150	240	400	720	780	600	840
H.323 Signaling and media in IPsec	32	34	52	80	150	185	100	150	195	325	360	385	600	840
WAN Edge Gateway[21]														
No encryption	32	48	80	140	270	320	100	150	240	400	610	650	600	840
SIP TLS with SRTP	32	45	75	130	250	300	100	150	240	400	600	645	600	840

[20] Gigabit Ethernet or Fast Ethernet egress; no quality-of-service (QoS) features; voice traffic only
[21] T1/E1 or High-Speed Serial Interface (HSSI) serial egress; some QoS features; voice and small amount of data traffic

	Cisco 2801	Cisco 2811	Cisco 2821	Cisco 2851	Cisco 3825	Cisco 3845	Cisco 2901	Cisco 2911	Cisco 2921	Cisco 2951	Cisco 3925	Cisco 3945	Cisco 3925E	Cisco 3945E
H.323 Signaling in IPsec with SRTP	32	41	80	124	220	270	100	150	240	400	530	565	600	840
H.323 Signaling and media in IPsec	22	22	44	60	110	135	100	125	145	235	265	285	600	840
WAN Edge Gateway with Compressed Real-Time Protocol (CRTP)[22]														
No encryption	26	35	61	120	225	270	100	150	240	400	510	550	600	840
SIP TLS with SRTP	26	32	56	112	210	255	100	150	240	400	500	540	600	840
H.323 Signaling in IPsec with SRTP	22	31	51	100	185	225	100	150	240	400	445	475	600	840
H.323 Signaling and media in IPsec	14	17	28	50	93	113	95	105	120	200	220	240	600	840
VoIP Performance: Maximum Number of Calls per Second (not exceeding 75-percent CPU)														
	0.5	0.7	0.8	1	3	7	1	1.5	2	3	10	15	30	35

Notes:

1. All results represent G.729A or G.711 (20-ms packetization) switched H.323 calls with VAD turned off.
2. The call success rate (CSR) of all tests is 98 to 100 percent.
3. Call duration of tests is 180 seconds except for calls-per-second rate testing, where the duration is shorter.

The test release is Cisco IOS Software Release 15.0.1M. This document contains general numbers as a guide to the approximate performance of the unified communications routers. The numbers are extrapolated from a large number of disparate tests, test conditions, and traffic patterns. Several nontesting factors have also been accounted for. Therefore, actual test results will vary, and we encourage you to do proof-of-concept testing for more specific performance numbers for a specific scenario, traffic pattern, or release.

End-of-Sale Platforms

Tables 8 and 9 provide system requirements for the end-of-sale unified communications routers.

Table 8. Cisco Unified Communications Routers with Cisco Unified Communications Manager Minimum System Requirements Using MGCP

Active Platforms	Interface Part Number	TDM Protocol or Feature	Minimum Cisco IOS Software Release	Minimum Cisco Unified Communications Manager Release
Cisco 1751 and 1760	VIC2-2FXS, VIC2-2FXO, and VIC2-4FXO	Analog FXS and FXO	12.3(5) or 12.3(4)T	3.3.5, 4.0.2a SR1, or 4.1.2 SR1
	VIC-4FXS/DID	Analog FXS	12.3(2)T	3.3.5, 4.0.2a SR1, or 4.1.2 SR1
	VIC2-2BRI -NT/TE	BRI	12.3(5) or 12.3(4)T	4.1.2 SR1
	VWIC-1MFT-T1/E1, VWIC-2MFT-T1/E1, VWIC-2MFT-T1/E1-DI, VWIC-1MFT-G703, and VWIC-2MFT-G703	T1 CAS E&M, T1/E1 PRI, and QSIG (basic)	12.2(15)T	3.3.2, 3.3.5, 4.0, 4.0.2a SR1, or 4.1.2 SR1

[22] T1/E1 or HSSI serial egress; some QoS features; CRTP; voice and small amount of data traffic

Active Platforms	Interface Part Number	TDM Protocol or Feature	Minimum Cisco IOS Software Release	Minimum Cisco Unified Communications Manager Release
	VWIC-1MFT-T1/E1, VWIC-2MFT-T1/E1, VWIC-2MFT-T1/E1-DI, VWIC-1MFT-G703, and VWIC-2MFT-G703	T1/E1QSIG supplementary services	12.3(14)T	4.0.2a SR1 or 4.1.2 SR1
	VWIC2-1MFT-T1/E1, VWIC2-2MFT-T1/E1, VWIC2-1MFT-G703, and VWIC2-2MFT-G703	T1 CAS E&M, T1/E1 PRI, and QSIG supplementary services	12.3(14)T	4.0.2a SR1 or 4.1.2 SR1
Cisco 2600XM	NM-1V/2V	Analog FXS and FXO	12.2.8T	3.0(8)
	NM-1V/2V	BRI	12.3.11T	4.1
	NM-HDA	Analog FXS and FXO	12.2.8T	3.2(2c)spA
	NM-HDV	T1 CAS E&M and T1/E1 PRI	12.2.11T	3.1
	NM-HDV	T1/E1 QSIG (basic)	12.2.11T	3.3
	AIM-VOICE-30 and AIM-ATM-VOICE-30	T1 CAS E&M, T1/E1 PRI, and T1/E1 QSIG (basic)	12.2.11T	3.3(3) SR2
	NM-HD-1V/2V/2VE	Analog FXS and FXO, T1 CAS E&M, T1/E1 PRI, and T1/E1 QSIG (basic)	12.3.4T	3.3(3) SR2
	NM-HD-1V/2V/2VE and NM-HDV2	BRI	12.4.2T	4.1.3 SR1
	NM-HDV2	T1 CAS E&M, T1/E1 PRI, and T1/E1 QSIG (basic)	12.3.7T	3.3.4, 4.0.1 SR1
	VWIC-1MFT-T1/E1, VWIC-2MFT-T1/E1, VWIC-2MFT-T1/E1-DI, VWIC-1MFT-G703, and VWIC-2MFT-G703	T1 CAS E&M and T1/E1 PRI	12.2.11T	3.1
	VWIC-1MFT-T1/E1, VWIC-2MFT-T1/E1, VWIC-2MFT-T1/E1-DI, VWIC-1MFT-G703, and VWIC-2MFT-G703	T1/E1QSIG**	12.3.11T	4.0
	VWIC-1MFT-T1/E1, VWIC-2MFT-T1/E1, VWIC-2MFT-T1/E1-DI, VWIC-1MFT-G703, and VWIC-2MFT-G703	MLPP	12.3.11T	4.0.2
	VWIC2-1MFT-T1/E1, VWIC2-2MFT-T1/E1, VWIC2-1MFT-G703, and VWIC2-2MFT-G703	T1 CAS E&M, T1/E1 PRI, QSIG**, and MLPP***	12.3.14T	4.0.2a SR2 or 4.1.3
Cisco 2691	NM-1V/2V	Analog FXS and FXO	12.2.8T	3.2(2c)spA
	NM-1V/2V	BRI	12.3.11T	4.1
	NM-HDA	Analog FXS and FXO	12.2.8T	3.2(2c)spA
	NM-HDV	T1 CAS E&M and T1/E1 PRI	12.2.11T	3.2(2c)spA
	NM-HDV	T1/E1 QSIG (basic)	12.2.11T	3.3
	AIM-VOICE-30 and AIM-ATM-VOICE-30	T1 CAS E&M, T1/E1 PRI, and T1/E1 QSIG (basic)	12.2.11T	3.3(3) SR2
	NM-HD-1V/2V/2VE	Analog FXS and FXO, T1 CAS E&M, T1/E1 PRI, and T1/E1 QSIG (basic)	12.3.4T	3.3(3) SR2
	NM-HD-1V/2V/2VE and NM-HDV2	BRI	12.4.2T	4.1.3 SR1
	NM-HDV2	T1 CAS E&M, T1/E1 PRI, and T1/E1 QSIG (basic)	12.3.7T	3.3.4, 4.0.1 SR1
	VWIC-1MFT-T1/E1, VWIC-2MFT-T1/E1, VWIC-2MFT-T1/E1-DI, VWIC-1MFT-G703, and VWIC-2MFT-G703	T1 CAS E&M and T1/E1 PRI	12.2.11T	3.1

Active Platforms	Interface Part Number	TDM Protocol or Feature	Minimum Cisco IOS Software Release	Minimum Cisco Unified Communications Manager Release
	VWIC-1MFT-T1/E1, VWIC-2MFT-T1/E1, VWIC-2MFT-T1/E1-DI, VWIC-1MFT-G703, and VWIC-2MFT-G703	T1/E1QSIG[**]	12.3.11T	4.0
	VWIC-1MFT-T1/E1, VWIC-2MFT-T1/E1, VWIC-2MFT-T1/E1-DI, VWIC-1MFT-G703, and VWIC-2MFT-G703	MLPP	12.3.11T	4.0.2
	VWIC2-1MFT-T1/E1, VWIC2-2MFT-T1/E1, VWIC2-1MFT-G703, and VWIC2-2MFT-G703	T1 CAS E&M, T1/E1 PRI, QSIG[**], and MLPP[***]	12.3.14T	4.0.2a SR2 or 4.1.3
	VWIC-1MFT-T1/E1, VWIC-2MFT-T1/E1, VWIC-2MFT-T1/E1-DI, VWIC-1MFT-G703, and VWIC-2MFT-G703	T1/E1QSIG[**] and MLPP[***]	12.3.11T	4.0.2a SR1 or 4.1.2
	VWIC2-1MFT-T1/E1, VWIC2-2MFT-T1/E1, VWIC2-1MFT-G703, and VWIC2-2MFT-G703	T1 CAS E&M, T1/E1 PRI, QSIG[**], and MLPP[***]	12.3.14T	4.0.2a SR2 or 4.1.3
	VIC3-2FXS/DID, VIC3-2FXS-E/DID, and VIC3-4FXS/DID	Analog FXS and FXO	12.4.20T	7.0
Cisco 3725 and 3745	NM-1V/2V	Analog FXS and FXO	12.2.8T	3.2(2c)spA
	NM-1V/2V	BRI	12.3.11T	4.1
	NM-HDA	Analog FXS and FXO	12.2.8T	3.2(2c)spA
	NM-HDV	T1 CAS E&M and T1/E1 PRI	12.2.11T	3.2(2c)spA
	NM-HDV	T1/E1 QSIG (basic)	12.2.11T	3.3
	AIM-VOICE-30 and AIM-ATM-VOICE-30	T1 CAS E&M, T1/E1 PRI, and T1/E1 QSIG (basic)	12.2.11T	3.3(3) SR2
	NM-HD-1V/2V/2VE	Analog FXS and FXO, T1 CAS E&M, T1/E1 PRI, and T1/E1 QSIG	12.3.4T	3.3(3) SR2
	NM-HD-1V/2V/2VE and NM-HDV2	BRI	12.4.2T	4.1.3 SR1
	NM-HDV2	T1 CAS E&M, T1/E1 PRI, and T1/E1 QSIG (basic)	12.3.7T	3.3.4 or 4.0.1 SR1
	VWIC-1MFT-T1/E1, VWIC-2MFT-T1/E1, VWIC-2MFT-T1/E1-DI, VWIC-1MFT-G703, and VWIC-2MFT-G703	T1 CAS E&M and T1/E1 PRI	12.2.11T	3.1
	VWIC-1MFT-T1/E1, VWIC-2MFT-T1/E1, VWIC-2MFT-T1/E1-DI, VWIC-1MFT-G703, and VWIC-2MFT-G703	T1/E1QSIG[**]	12.3.11T	4.0
	VWIC-1MFT-T1/E1, VWIC-2MFT-T1/E1, VWIC-2MFT-T1/E1-DI, VWIC-1MFT-G703, and VWIC-2MFT-G703	MLPP	12.3.11T	4.0.2
	VWIC2-1MFT-T1/E1, VWIC2-2MFT-T1/E1, VWIC2-1MFT-G703, and VWIC2-2MFT-G703	T1 CAS E&M, T1/E1 PRI, QSIG[**], and MLPP[***]	12.3.14T	4.0.2a SR2 or 4.1.3

Table 9. Cisco Unified Communications Routers with Cisco Unified Communications Manager Minimum System Requirements for Conferencing, Transcoding, and Media Termination Point

Active Platforms	Interface Part Numbers	TDM Protocol or Feature	Minimum Cisco IOS Software Release	Minimum Cisco Unified Communications Manager Release
Cisco 1751 and 1760	Onboard PVDM-256K DSPs	Conferencing and transcoding	12.3.8T	3.3.5, 4.0.2a SR1, or 4.1.2 SR1
Cisco 2600XM	NM-HDV and NM-HDV-FARM	Conferencing and transcoding	12.2.13T	3.2(2c)
	NM-HD-1V/2V/2VE and NM-HDV2	MTP	12.3.8T	4.0.1
	NM-HD-1V/2V/2VE and NM-HDV2	Conferencing and transcoding	12.3.8T	3.3.4 or 4.0.1
	NM-HD-1V/2V/2VE and NM-HDV2	RFC 2833 MTP	12.3.11T	4.0
Cisco 2691	NM-HDV and NM-HDV-FARM	Conferencing and transcoding	12.2.13T	3.2(2c)
	NM-HD-1V/2V/2VE and NM-HDV2	Conferencing and transcoding	12.3.8T	3.3.4 or 4.0.1
	NM-HD-1V/2V/2VE and NM-HDV2	MTP	12.3.8T	4.0.1
	NM-HD-1V/2V/2VE and NM-HDV2	RFC 2833 MTP	12.3.11T	4.0
Cisco 3725 and 3745	NM-HD-1V/2V/2VE and NM-HDV2	Conferencing and transcoding	12.3.8T4	3.3.5, 4.0.2a SR1, or 4.1.2
	NM-HD-1V/2V/2VE and NM-HDV2	Conferencing and transcoding	12.3.8T	3.3.4 or 4.0.1
	NM-HD-1V/2V/2VE and NM-HDV2	MTP	12.3.8T	4.0.1
	NM-HDV and NM-HDV-FARM	RFC 2833 MTP	12.3.11T	4.0

Tables 10 and 11 provide information about connectivity and CPU performance, respectively, on the end-of-sale unified communications routers.

Table 10. Maximum Physical DS-0 Connectivity on the Cisco Unified Communications Routers*

	Cisco 1751 and 1760	Cisco 2610 XM	Cisco 2620 XM	Cisco 2650 XM	Cisco 2691	Cisco 3725	Cisco 3745
FXS	Cisco 1751: 12 Cisco 1760: 16	12	12	12	12	24	48
FXO and CAMA	Cisco 1751: 12 Cisco 1760: 16	8	8	8	8	16	32
E&M	Cisco 1751: 6 Cisco 1760: 8	4	4	4	4	8	16
Analog DID	Cisco 1751: 6 Cisco 1760: 8	8	8	8	8	16	32
BRI ports	Cisco 1751: 6 Cisco 1760: 8	4	4	4	4	8	16
T1/E1 ports	1	5	5	5	6	10	18
T1 channels	24	120	120	120	144	240	432
E1 channels	30	150	150	150	180	300	540

Table 11. CPU Performance on the Cisco Unified Communications Routers*

	Cisco 1751 and 1760	Cisco 2610 XM	Cisco 2620 XM	Cisco 2650 XM	Cisco 2691	Cisco 3725	Cisco 3745
VoIP Performance: Maximum Number of Simultaneous Calls (not exceeding 75-percent platform CPU usage)							
Standalone Unified Communications Router[23]							
No encryption	32	32	38	50	130	180	290
SRTP with signaling in IPsec	-	24	30	40	106	155	250
SRTP with signaling and media in IPsec	-	12	16	22	60	85	135
WAN Edge Gateway[24]							
No encryption	32	18	20	35	100	140	220
SRTP with signaling in IPsec	-	14	16	29	82	120	190
SRTP with signaling and media in IPsec	-	8	9	16	47	65	102
WAN Edge Gateway with Compressed Real-Time Protocol (CRTP)[25]							
No encryption	24	12	16	24	82	102	170
SRTP with signaling in IPsec	-	10	14	20	69	90	140
SRTP with signaling and media in IPsec	-	5	7	10	39	48	77
VoIP Performance: Maximum Number of Calls per Second (not exceeding 75-percent CPU)							
	0.1	0.1	0.25	0.5	2	1.5	4

Notes:

1. All results represent G.729A or G.711 (20-ms packetization) switched H.323 calls with VAD turned off.
2. The call success rate (CSR) of all tests is 98 to 100 percent.
3. Call duration of tests is 180 seconds except for calls-per-second rate testing, where the duration is shorter.

The test release is Cisco IOS Software Release 12.4M. This document contains general numbers as a guide to the approximate performance of the unified communications routers. The numbers are extrapolated from a large number of disparate tests, test conditions, and traffic patterns. Several nontesting factors have also been accounted for. Therefore, actual test results will vary, and we encourage you to do proof-of-concept testing for more specific performance numbers for a specific scenario, traffic pattern, or release.

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[23] Gigabit Ethernet or Fast Ethernet egress; no QoS features; voice traffic only
[24] T1/E1 or High-Speed Serial Interface (HSSI) serial egress; some QoS features; voice and small amount of data traffic
[25] T1/E1 or HSSI serial egress; some QoS features; CRTP; voice and small amount of data traffic

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