Data Sheet

Cisco 2900, 3900, and 4000 Series Integrated Services Router Interoperability with Cisco Unified Communications Manager

Cisco® 2900, 3900, and 4000 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 2900, 3900, and 4000 (4300 and 4400) Series Integrated Services Routers can 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.

These platforms provide a highly flexible and scalable solution for small and medium-sized branch and regional offices. These platforms 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 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 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 in "secondary" mode where Cisco Unified Communications Manager takes the "primary" role. The gateway configuration and dial-plan configuration are centrally managed from Cisco Unified Communications Manager, which generates an XML config file that is downloaded by the gateway to autoconfigure.

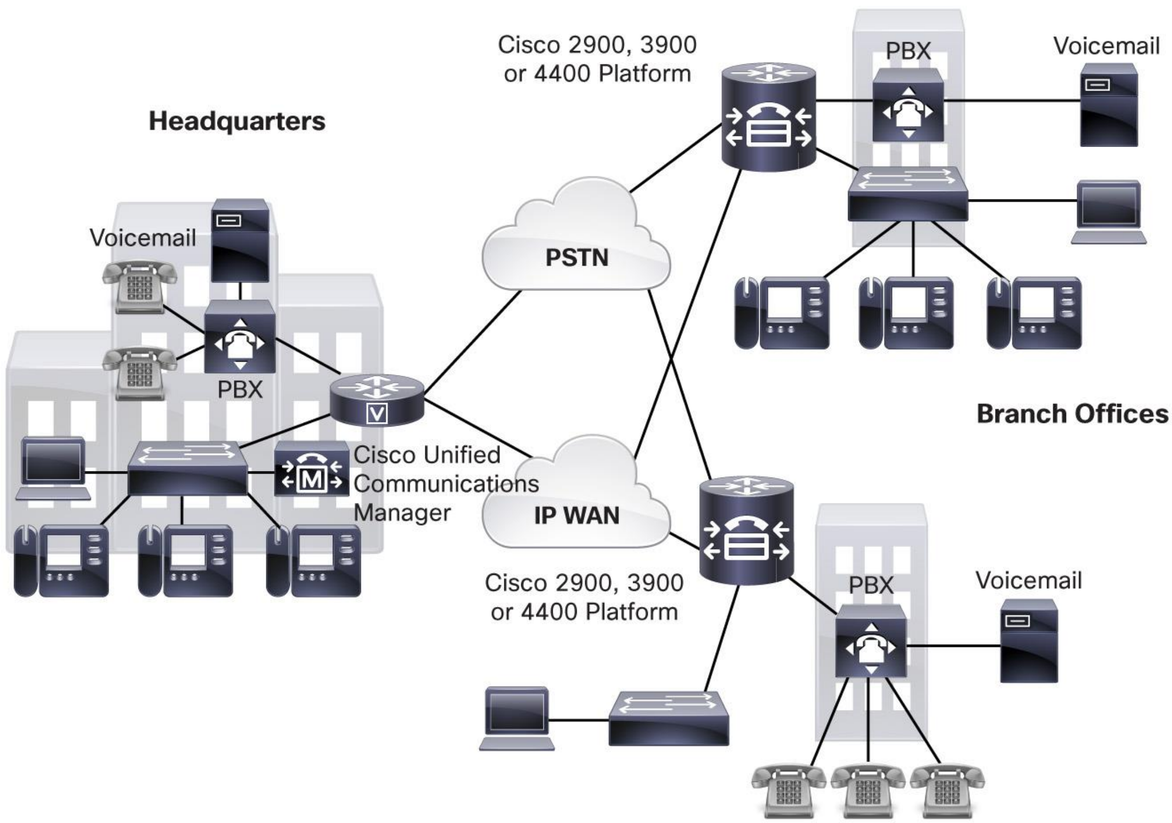
IP Telephony Phased Migration

The Cisco 2900, 3900, and 4000 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

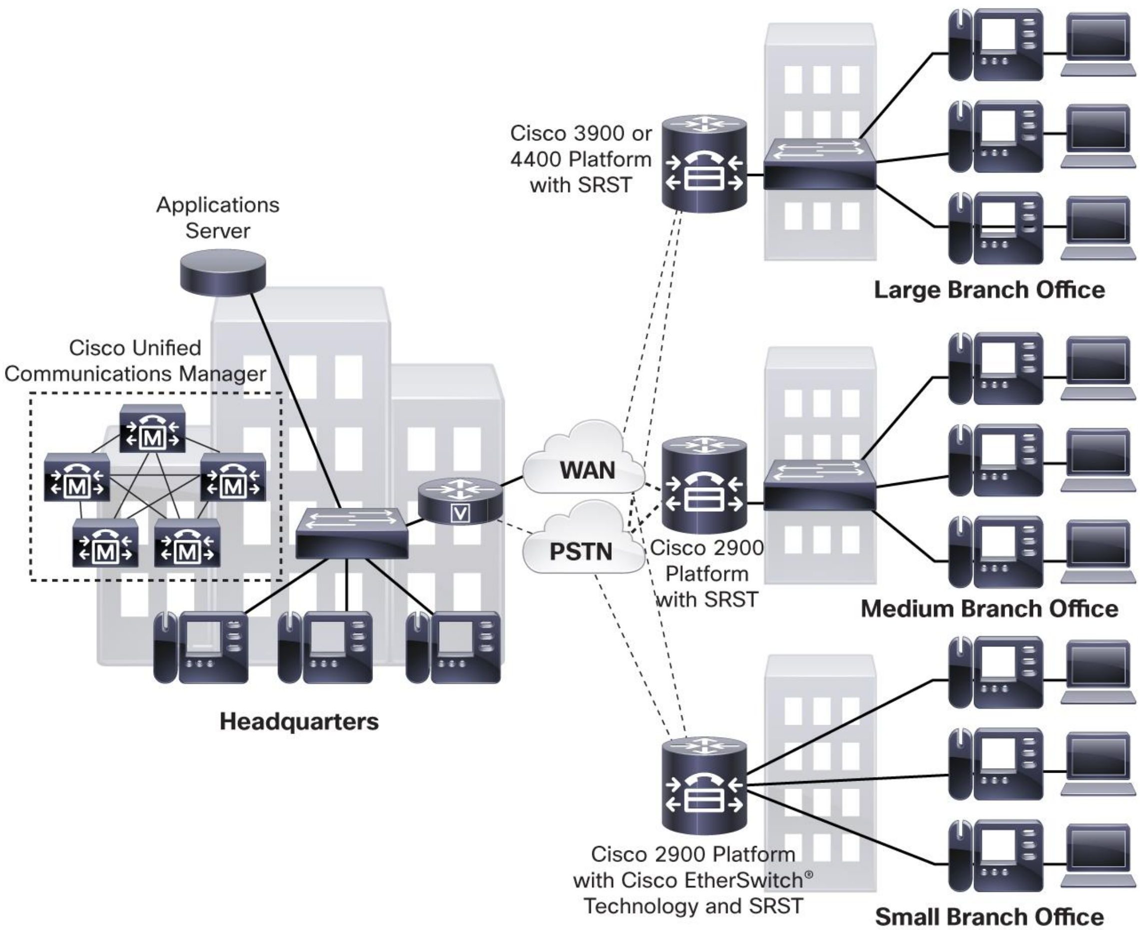
As enterprises extend their IP telephony deployments from central sites to remote offices, an important consideration is the ability to cost-effectively provide failover capability at remote branch offices. However, the size and number of these small-office sites preclude most enterprises from deploying 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 2900, 3900, and 4000 Series unified communications routers. Cisco Unified Communications offers two types of call-processing survivability: Survivable Remote Site Telephony (SRST), which helps ensure survivability of basic and most critical IP telephony services, and Enhanced Survivable Remote Site Telephony, which survives basic IP telephony services as well as advanced Unified Communications supplementary services. For more information about SRST, go to http://www.cisco.com/go/srst.

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 40,000 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 2 summarizes roadmap unified communications features of these routers that are independent of Cisco Unified Communications Manager.

Please also refer to the Cisco 4000 Series Integrated Services Routers release notes for unified communications features added in different releases.

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

Cisco 2900 and 3900 Series ISRs			Cisco 4300 and 4400 Series ISRs				
SIP	**MGCP**	**H.323**	**SIP**	**MGCP**	**H.323**	**Feature**	**Benefits**
FXS/FXO							
Y	Y[1]	Y	Y	Y	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	Y	N	Y	Analog E&M (wink, immediate, and delay) interfaces	These interfaces make direct connection to a PBX possible.
Y	Y	Y	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	Y	Y	Y	Analog direct inward dialing (DID)	Analog DID enables connection to the PSTN with DID operation.
Y	N	N	Road map	N	N	DSAPP 3-way conference	This feature enables an end user already engaged in a stable two-party call to add a third party to the conversation.
Y	N	Y	N	N	N	FXO tone answer supervision	This feature facilitates the use of tones to signal answering a call and the start of a call detail record (CDR).
Y	Y	Y	N	N	N	FXO disconnect supervision	This feature makes battery reversal or tones available for use to disconnect FXO calls.

Cisco 2900 and 3900 Series ISRs			Cisco 4300 and 4400 Series ISRs				
Y	N	Y	N	N	N	Analog Centralized Automated Message Accounting (CAMA)	Analog CAMA facilitates analog PSTN connection for E-911 support.
BRI							
Y	Y	Y	Y	Y	Y	BRI Q.931 user side (NET3)	This feature enables connection to the PSTN.
Y	N	Y	Y	Y	Y	BRI Q.931 network side (NET3)	This feature enables connection to a PBX.
Y	Y	Y	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]	Y	Y	Y	BRI Q.SIG forward, transfer, and conference	These services enable connection to a PBX or key system.
T1/E1							
N	Y[4]	N	N	Y	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	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	Y	Y	Y	T1-CAS E&M (delay dial) interfaces	These interfaces facilitate connection to a PBX, key system, or PSTN.
Y	N	Y	3.15	N	3.15	T1-CAS feature group D[5]	This feature is used to connect to a PBX or PSTN. (EANA and E&M)
Y	N	Y	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	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	Y	N	Y	E1 CAS	E1 CAS enables connection to a PBX or PSTN.
Y	N	Y	Y	N	Y	E1 MelCAS	E1 MelCAS facilitates connection to a PBX or PSTN.
Y	N	Y	3.15	N	3.15	E1 R2 (more than 30 country variants)	E1 R2 enables connection to a PBX or PSTN.
Y	Y	Y	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	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]	Y	Y	N	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.
N	Y	N	N	3.17	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	N	Y	N	N	N	Voice + Data integrated access	This feature makes the voice and serial data interfaces available on the same T1/E1.
Y	N	Y	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.
N	Y	Y	N	N	N	T1/E1 External Signaling (ext-sig)	This feature is used to enable a connection trunk for common channel signaling (TCCS) application.
ISDN							
Y	N	Y	N	N	N	ISDN video switching on gateway (drop DSPs)	This feature allows ISDN-based video conferencing systems to connect and be switched back out of the ISDN.
Y	Y	Y	Y	N	Y	Q.SIG and Q.931 Tunneling	This feature enables transparent tunneling of ISDN signaling over VoIP signaling.

Cisco 2900 and 3900 Series ISRs			Cisco 4300 and 4400 Series ISRs				
Y	N	Y	N	N	N	H.320 video gateway support	This feature integrates ISDN trunks with both voice and video traffic.
Dual-Tone Multifrequency (DTMF)							
Y	Y	Y	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.
High Availability							
Y	Y	Y	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]	Y	Y	Y	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	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. Cisco 4000 Series Integrated Services Routers do not support Secure SRST.
Y	N	Y[7]	Y	N	Y	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]	Y	Y	Y	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.
Music on Hold (MoH)							
Y	Y	Y	Y	3.15	Y	Multicast music on hold (MoH) - centralized	This feature helps the unified communications router deliver music streams from an MoH server to users on on- and off-net calls.
N	Y	N	Y	3.15	Y	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	N	Y	Y	Tone on hold	Tone indicates when a user is placed on hold.
N	Y	N	N	N	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.
Caller ID							
Y	Y	Y	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	N	N	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.
Fax/Modem							
Y	Y	Y	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

Cisco 2900 and 3900 Series ISRs			Cisco 4300 and 4400 Series ISRs				
							pass-through methods.
Y	Y[9]	Y	Y	Y	Y	T.38 standards-based fax support	This feature enables transmit T.38 fax between the PSTN and IP.
N	15.1(4) M	N	Y-16.4	Y-16.4	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).
Y	Y	Y	Y	Y	Y	Modem relay	Modem relay demodulates a modem signal at one voice gateway and passes it as packet data to another voice gateway where the signal is remodulated and sent to a receiving modem. On detection of the modem answer tone, the gateways switch into modem passthrough mode and then, if the call menu (CM) signal is detected, the two gateways switch into modem relay mode.
Y	N	N	Y	Y	Y	Modem passthrough	Modem passthrough over VoIP provides the transport of modem signals through a packet network by using pulse code modulation (PCM) encoded packets. Note: the support is specific to Cisco Prioritary NSE-based modem passthrough. We also support protocol-based modem passthrough on 2900/3900 and 4000 Series ISRs for SIP signaling only.
Y	N	Y	N	N	N	T.37 Fax On-Ramp/Off-Ramp	On-ramp faxing allows the voice gateway that handles incoming calls from a standard fax machine or the PSTN to convert a traditional Group 3 fax to an email message with a Tagged Image File Format (TIFF) attachment. Off-ramp faxing allows a voice gateway that handles calls going out from the network to a fax machine or the PSTN to convert a fax email with a TIFF attachment into a traditional fax format that can be delivered to a standard fax machine or the PSTN.
Codec							
Y	Y	Y	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.726, or G.728).
Voice Activity Detection (VAD)							
Y	Y	Y	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	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.
Miscellaneous							
Y	N	Y	Y	N	Y	Private-line automatic ringdown (PLAR)	PLAR provides a dedicated connection to another extension or an attendant.
Y	N	Y	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.
Y	Y	Y	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.
Y	N	Y	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	Y	Y[7]	Y	Y	Y	Overlap sending over voice over IP (VoIP)	This feature speeds variable-length dial strings dialing.
–	–	Y	–	–	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.

Cisco 2900 and 3900 Series ISRs			Cisco 4300 and 4400 Series ISRs				
Y	N	Y	N	N	N	VoiceXML	VoiceXML controls calls "in queue" at the gateway for call-center applications. Calls are redirected only when an agent becomes available.
Y	–	–	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.
Interactive Voice Response (IVR)							
Y	N	Y	Y	N	Y	IVR support	IVR offers automated-attendant support, voicemail support, or call routing based on service desired.
Y	N	Y	Y	N	Y	Automated Attendant	This feature uses IVR to provide automated call-answering and -forwarding services.
Security							
N	Y[1]	N	N	Y	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	N	N	N	N	Connection to Defense Switched Network (DSN)	This feature supports the U.S. Department of Defense private TDM network.
Y	N	N	Y	N	N	SIP privacy and identity	This feature enables transport of identity, both preferred (P-Preferred Identity [PPI]) and asserted (P-Asserted Identity [PAI]).
Y	Y	Y	3.17	3.15	3.17	Secure Real-Time Transport Protocol (SRTP): media authentication and encryption on unified communications routers (basic calls)	This feature enables secure gateway-to-gateway calls and secure IP phone-to-gateway calls.
Y	N	N	3.17	3.15	3.17	SRTP-Real-Time Transport Protocol (RTP) fallback operations (basic calls)	This feature enables the fallback from SRTP to RTP during capabilities negotiation at the time of call setup.
Y	Y	Y	Y	Y	Y	Signaling encryption: SIP: Transport Layer Security (TLS) or IP Security (IPsec); MGCP/H,323: IPsec	This feature encrypts signaling communication between unified communications routers and Cisco Unified Communications Manager.
IP Services							
Y	N	Y	Y	N	Y	Virtualization (Virtual Route Forwarding [VRF])	This feature supports virtual segmentation of the network using VRF.
Y	N	N	Y	N	N	IPv6	IPv6 support enables interworking with IPv6-capable networks.
Y	N	N	Y	N	N	Dynamic Host Configuration Protocol (DHCP)	DHCP enables acquisition of gateway configuration parameters from the DHCP server.
Y	Y	Y	Y	N	Y	Resource Reservation Protocol (RSVP) support	This feature helps assure high-quality voice by enabling resource reservation for call admission control.
Y	Y	Y	Y	N	Y	Authentication, authorization, and accounting (AAA)	AAA supports debit card and credit card (prepaid and postpaid calling card) applications.
Management and Monitoring							
N	Y	N	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	N	N	Y	N	N	History Info support	This feature enables support for the History Info header to transport the history information of a call.
Y	N	N	Y	N	N	Signaling health monitoring	This feature enables monitoring of the signaling connection across the signaling trunk.

[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.

[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.

[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.

[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.

Table 2. Generation 2 (G2) and 4000 Series ISRs Roadmap for Unified Communications Features Independent of Cisco Unified Communications Manager

Feature	Availability on G2 ISRs	Availability on 4000 Series ISRs	Notes
Hoot and Holler	Y	16.3	Hoot and holler networks (also known as junkyard circuits, squawk box systems, holler down circuits, and shout down circuits) provide "always on" multiuser conferences without requiring users to dial into a conference bridge.
Connection Trunk	Y	3.17	Connection Trunk mode is a permanent connection; the VoIP call is always connected independently of the plain old telephone service (POTS) port being on-hook or off-hook. Connection Trunk has statically configured endpoints and does not require a user to dial to connect calls. It also allows supplemental call signaling, such as hookflash or point-to-point hoot-and-holler, to be passed over the IP network between the two telephony devices.
Land Mobile Radio (LMR)	Y	3.17	Land Mobile Radio (LMR) is a radio operation system that allows many end users belonging to same group to communicate with each other by using handsets that can select one of several preset radio frequencies. LMR multicast-group support is in the roadmap.
Trunk Group	Y	3.11	Trunk group supports a collection of TDM voice interfaces to be defined as a dial-peer routing endpoint. During the outgoing call setup, an idle b-channel is selected from a trunk group to setup an outgoing TDM call. Trunk group maintains its b-channel active and inactive in real time. The following T1/E1 interfaces are supported: ISDN PRI interface CAS (ds0-group) interface

Unified Communications Router with Cisco Unified Communications Manager Minimum System Requirements

Tables 3 through 7 give system requirements for the unified communications routers.

Table 3. Cisco Unified Communications Routers with Cisco Unified Communications Manager Minimum System Requirements Using SIP for 2900 and 3900 ISR

Platform	TDM Protocol or Feature	Minimum Cisco IOS or Cisco IOS XE Software Release*	Minimum Cisco Unified Communications Manager Release
2900 and 3900 ISRs	Analog (FXS and FXO)	12.4(6)T	5.0
2900 and 3900 ISRs	BRI	12.4(6)T	5.0
2900 and 3900 ISRs	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 4. Cisco Unified Communications Routers with Cisco Unified Communications Manager Minimum System Requirements Using H.323 for 2900 and 3900 ISR

Platform	TDM Protocol or Feature	Minimum Cisco IOS or Cisco IOS XE Software Release*	Minimum Cisco Unified Communications Manager Release
2900 and 3900 ISRs	Analog (FXS and FXO)	12.2(1)M	3.0(5a)
2900 and 3900 ISRs	BRI	12.2(1)M	3.0(5a)
2900 and 3900 ISRs	T1 CAS and T1/E1 PRI	12.1(2)T	3.0(5a)
2900 and 3900 ISRs	T1/E1 QSIG	12.1(2)T	3.0(5a)
2900 and 3900 ISRs	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.

Table 5. Cisco Unified Communications Routers with Cisco Unified Communications Manager Minimum System Requirements Using MGCP for 2900 and 3900 ISR

Active Platforms	Interface Part Number	TDM Protocol or Feature	Minimum Cisco IOS or Cisco IOS XE Software Release	Minimum Cisco Unified Communications Manager Release
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,[1] and MLPP[2]	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,[1] MLPP,[2] 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,[1] MLPP,[2] and channel groups	15.1.3T	7.1.5 or 8.0.2
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,[1] and MLPP[2]	15.0.1M	6.1.5, 7.1.3, or 8.0

Active Platforms	Interface Part Number	TDM Protocol or Feature	Minimum Cisco IOS or Cisco IOS XE Software Release	Minimum Cisco Unified Communications Manager Release
	VWIC3-1MFT-T1/E1, VWIC3-2MFT-T1/E1, VWIC3-1MFT-G703, and VWIC3-2MFT-G703	T1 CAS, T1/E1 PRI, QSIG,[1] MLPP,[2] and channel groups	15.0.1M3	7.1.5 or 8.0.2
	VWIC3-4MFT-T1/E1	T1 CAS, T1/E1 PRI, QSIG,[1] MLPP,[2] 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,[1] and MLPP[2]	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,[1] MLPP,[2] and channel groups	15.0.1M3	7.1.5 or 8.0.2
	VWIC3-4MFT-T1/E1	T1 CAS, T1/E1 PRI, QSIG,[1], MLPP,[2] and channel groups	15.1.3T	7.1.5 or 8.0.2

Table 6. Cisco Unified Communications Routers with Cisco Unified Communications Manager Minimum System Requirements Using SIP/H.323/MGCP for 4300 and 4400 series

Active Platforms	Interface Part Number	TDM Protocol or Feature	Minimum Cisco IOS or Cisco IOS XE Software Release	Minimum Cisco Unified Communications Manager Release
Cisco 4300 Series (4351, 4331, 4321)	NIM-1MFT-T1/E1 NIM-2MFT-T1/E1 NIM-4MFT-T1/E1 NIM-8MFT-T1/E1	T1 CAS, T1/E1 PRI, QSIG[1] MLPP,[2] and 2 channel groups	XE 3.14	9.1.2, 9.1.2.su2, 10.5.2, 11.0
	NIM-1CE1T1-PRI NIM-2CE1T1-PRI NIM-8CE1T1-PRI	T1 CAS, T1/E1 PRI, QSIG,[1] MLPP,[2] and 24 channel-groups	XE 3.14	9.1.2, 9.1.2.su2, 10.5.2, 11.0
	NIM-2FXO NIM-4FXO NIM-2FXS NIM-4FXS NIM-2FXS/4FXO	Analog FXS and FXO	XE 3.16	9.1.2, 9.1.2.su2, 10.5.2, 11.0
	NIM-2BRI-NT/TE NIM-4BRI-NT/TE	BRI (NT and TE)	XE 3.14	9.1.2, 9.1.2.su2, 10.5.2, 11.0
	SM-X-NIM-ADPTR (exclude 4321)	SM Adapter for NIM	XE 3.14	9.1.2, 9.1.2.su2, 10.5.2, 11.0
Cisco 4431	NIM-1MFT-T1/E1 NIM-2MFT-T1/E1 NIM-4MFT-T1/E1 NIM-8MFT-T1/E1	T1 CAS, T1/E1 PRI, QSIG,[1] MLPP,[2] and 2 channel groups	XE 3.13	9.1.2, 9.1.2.su2, 10.0.1, 10.5, 10.5.2, 11.0

Active Platforms	Interface Part Number	TDM Protocol or Feature	Minimum Cisco IOS or Cisco IOS XE Software Release	Minimum Cisco Unified Communications Manager Release
	NIM-1CE1T1-PRI NIM-2CE1T1-PRI NIM-8CE1T1-PRI	T1 CAS, T1/E1 PRI, QSIG,[1] MLPP,[2] and 24 channel-groups	XE 3.13	9.1.2, 9.1.2.su2, 10.0.1, 10.5, 10.5.2, 11.0
	NIM-2FXO NIM-4FXO NIM-2FXS NIM-4FXS NIM-2FXS/4FXO	Analog FXS and FXO	XE 3.16	9.1.2, 9.1.2.su2, 10.0.1, 10.5, 10.5.2, 11.0
	NIM-2BRI-NT/TE NIM-4BRI-NT/TE	BRI (NT and TE)	XE 3.14	9.1.2, 10.5.2, 11.0
Cisco 4451	NIM-1MFT-T1/E1 NIM-2MFT-T1/E1 NIM-4MFT-T1/E1 NIM-8MFT-T1/E1	T1 CAS, T1/E1 PRI, QSIG,[1] MLPP,[2] and 2 channel groups	XE 3.12	9.1.2, 9.1.2.su2, 10.0, 10.0.1,10.5, 10.5.2, 11.0
	NIM-1CE1T1-PRI NIM-2CE1T1-PRI NIM-8CE1T1-PRI	T1 CAS, T1/E1 PRI, QSIG,[1] MLPP,[2] and 24 channel-groups	XE 3.12	9.1.2, 9.1.2.su2, 10.0, 10.0.1,10.5, 10.5.2, 11.0
	NIM-2FXO NIM-4FXO NIM-2FXS NIM-4FXS NIM-2FXS/4FXO	Analog FXS and FXO	XE3.13.4, XE3.14.3, XE3.15.1, XE3.16 and above (XE3.13.4 is not available for NIM-2FXS/4FXO)	9.1.2, 9.1.2.su2, 10.0.1, 10.5, 10.5.2, 11.0
	NIM-2BRI-NT/TE NIM-4BRI-NT/TE	BRI (NT and TE)	XE3.14.3, XE3.15.1, XE3.16 and above	9.1.2, 10.5.2, 11.0
	SM-X-NIM-ADPTR	SM Adapter for NIM	XE 3.14	9.1.2, 10.5.2, 11.0

[*] Non-Facility Associated Signaling (NFAS) on one single NIM module is supported on ISR 4000 Series. NFAS across multiple NIM modules is not supported.

Table 7. Cisco Unified Communications Routers with Cisco Unified Communications Manager Minimum System Requirements for Conferencing, Transcoding, and Media Termination Point (Voice only)

Active Platforms	Interface Part Numbers	TDM Protocol or Feature	Minimum Cisco IOS Software Release	Minimum Cisco Unified Communications Manager Release
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 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

Active Platforms	Interface Part Numbers	TDM Protocol or Feature	Minimum Cisco IOS Software Release	Minimum Cisco Unified Communications Manager Release
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
Cisco 4300 and 4400 Series	Onboard or on-card PVDM4 DSPs	Conferencing, transcoding, MTP, and RFC 2833	XE 3.10	9.1.2 and above

Voice Performance

Tables 8 through 10 give information about connectivity and CPU performance on the unified communications routers.

Table 8. Maximum Physical DS-0 Connectivity on the Cisco Unified Communications Routers*

Cisco router	2901	2911	2921	2951	3925	3945	3925E	3945E	4321	4331	4351	4431	4451
FXS	16	40	40	64	64	112	60	108	8	8	20**	12	20**
FXO and CAMA	16	28	28	40	40	64	36	60	8	8	20**	12	20**
E&M	8	12	12	16	16	24	14	22					
Analog DID	16	32	32	48	48	80	44	76	8	8	20**	12	20**
BRI ports	8	16	16	24	24	40	22	38					
T1/E1 ports	8	20	20	24	24	32	20	28	8	16	40**	24	40**
T1 channels	192	480	480	576	576	768	480	672	192	368	920**	576	960**
E1 channels	240	600	600	720	720	960	600	840	240	480	1200**	720	1200**

* 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.

** When using a SM to NIM adapter card for each Service Module slot to turn it into an additional NIM.

Table 9. CPU Performance on the 4300 and 4400 Series Unified Communications Routers*

Cisco ISR Platform	4321	4331	4351	4431	4451
VoIP Performance: Maximum # of Simultaneous Calls (not exceeding 75% platform CPU usage)					
Cisco Unified Border Element	100	500	1000	3000	6000
Standalone Unified Communications Router[1]					
No encryption	100	500	1000	3000	6000
SIP TLS with SRTP	100	500	1000	3000	6000
H.323 Signaling in IPsec with SRTP	100	500	1000	3000	6000
H.323 Signaling and media in IPsec	100	500	1000	3000	6000
WAN Edge Gateway[2]					
No encryption	100	500	1000	3000	6000
SIP TLS with SRTP	100	500	1000	3000	6000
H.323 Signaling in IPsec with SRTP	100	500	1000	3000	6000
H.323 Signaling and media in IPsec	100	500	1000	3000	6000

Cisco ISR Platform	4321	4331	4351	4431	4451
WAN Edge Gateway with Compressed Real-Time Protocol (CRTP)[3]					
No encryption	100	500	1000	3000	6000
SIP TLS with SRTP	100	500	1000	3000	6000
H.323 Signaling in IPsec with SRTP	100	500	1000	3000	6000
H.323 Signaling and media in IPsec	100	500	1000	3000	6000
VoIP Performance: (not exceeding 75-percent CPU)					
Maximum Number of Calls per Second	4	8	10	15	40
Software MTP Session Performance	250	600	1000	1500	3000

[1] Gigabit Ethernet or Fast Ethernet egress; no quality-of-service (QoS) features; voice traffic only

[2] T1/E1 or High-Speed Serial Interface (HSSI) serial egress; some QoS features; voice and small amount of data traffic

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

Table 10. CPU Performance on the 2900 and 3900 Series Unified Communications Routers[*]

Cisco ISR Platform	2901	2911	2921	2951	3925	3945	3925E	3945E
VoIP Performance: Maximum Number of Simultaneous Calls (not exceeding 75% platform CPU usage)								
CUBE	100	200	400	600	800	950	2100	2500
Standalone Unified Communications Router[1]								
No encryption	100	150	240	400	720	960	600	840
SIP TLS with SRTP	100	150	240	400	720	880	600	840
H.323 Signaling in IPsec with SRTP	100	150	240	400	720	780	600	840
H.323 Signaling and media in IPsec	100	150	195	325	360	385	600	840
WAN Edge Gateway[2]								
No encryption	100	150	240	400	610	650	600	840
SIP TLS with SRTP	100	150	240	400	600	645	600	840
H.323 Signaling in IPsec with SRTP	100	150	240	400	530	565	600	840
H.323 Signaling and media in IPsec	100	125	145	235	265	285	600	840
WAN Edge Gateway with Compressed Real-Time Protocol (CRTP)[3]								
No encryption	100	150	240	400	510	550	600	840
SIP TLS with SRTP	100	150	240	400	500	540	600	840
H.323 Signaling in IPsec with SRTP	100	150	240	400	445	475	600	840
H.323 Signaling and media in IPsec	95	105	120	200	220	240	600	840
VoIP Performance: Maximum Number of Calls per Second (not exceeding 75-percent CPU)								
	1	1.5	2	3	10	15	30	35

[1] Gigabit Ethernet or Fast Ethernet egress; no quality-of-service (QoS) features; voice traffic only

[2] T1/E1 or High-Speed Serial Interface (HSSI) serial egress; some QoS features; voice and small amount of data traffic

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

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 for Cisco 2900 and 3900 ISRs. For the testing of the Cisco 4400, Cisco IOS XE Software Release 3.10 was used. 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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