Cisco 1700, 2600, 2800, 3700, and 3800 Series Voice Gateway Router Interoperability with Cisco Unified CallManager

Cisco® 1700, 2600, 2800, 3700, and 3800 series integrated access routers from Cisco Systems® can be deployed as voice gateway routers as part of the Cisco IP Communications solution. New and existing deployments can benefit by using Cisco 1700, 2600, 2800, 3700, and 3800 series routers as voice gateways with Cisco Unified CallManager.

Cisco 1700 (including Cisco 1751 and 1750 routers), 2600, 2800, 3700, and 3800 series voice gateway routers communicate directly with Cisco Unified CallManager, allowing for the deployment of IP telephony solutions that are ideal for large enterprises and service providers that offer managed network services. The Cisco 1700, 2600, 2800, 3700, and 3800 series voice gateway routers provide a highly flexible and scalable solution for small and medium-sized branches and regional offices.

The Cisco 1700, 2600, 2800, 3700, and 3800 series voice gateway routers support the widest 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). These voice gateway routers can be configured to support from 2 to 540 voice channels.

As enterprises seek to deploy an expanding list of IP telephony applications and services, Cisco 1700, 2600, 2800, 3700, and 3800 series voice gateway routers—interoperating with Cisco Unified CallManager—provide a solution that will grow with their changing needs.

Interoperability Using SIP, H.323, or MGCP

The Cisco 1700, 2600, 2800, 3700, and 3800 series voice gateway routers can communicate with the Cisco Unified CallManager using Session Initiation Protocol (SIP), H.323, or Media Gateway Control Protocol (MGCP):

- In SIP and H.323 mode, the Cisco 1700, 2600, 2800, 3700, and 3800 series voice gateway routers communicate with Cisco Unified CallManager as an intelligent gateway device.
- In MGCP mode, these routers operate as a stateless client, giving Cisco Unified CallManager full control. Dial plans are configured centrally in Cisco Unified CallManager. Voice gateway routers can then be automatically configured by a download of Extensible Markup Language (XML) files.

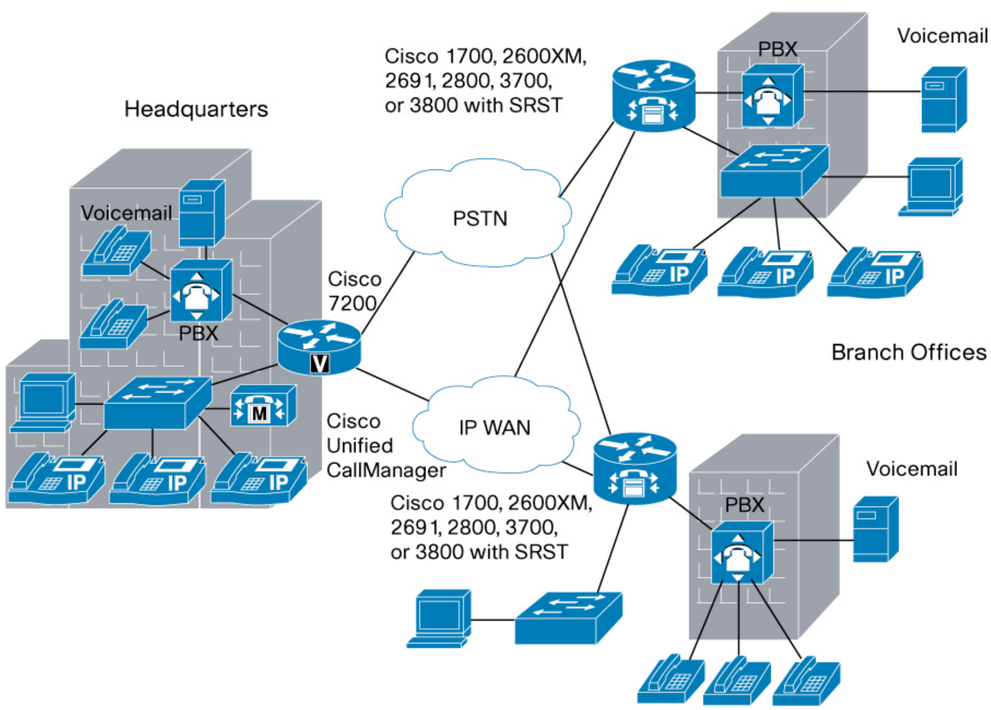
IP Telephony Phased Migration

The Cisco 1700, 2600, 2800, 3700, and 3800 series voice gateway routers help users immediately deploy an end-to-end IP telephony 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, customers can use these voice gateway routers to interconnect older PBXs over the packet infrastructure and still maintain PSTN (off-net) connectivity through their circuit-switched PBXs. Later, customers can migrate PSTN (off-net) connectivity to the voice gateway routers and start to incorporate IP phones at larger sites (Figure 1). After all sites are running IP telephony, users can begin deploying IP-based applications such as IP unified messaging, personal assistants, and extension mobility.

The Cisco 1700, 2600, 2800, 3700, and 3800 series voice gateway routers are an ideal solution for circuit-switched PBX and PSTN access within a Cisco Unified CallManager-based IP telephony architecture.

Figure 1. IP Telephony Phased Migration—Migrate Circuit-Switched PSTN and PBX Connectivity to Voice Gateway Routers



As companies seek to deploy IP telephony 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 Cisco 1700, 2600, 2800, 3700, and 3800 series voice gateway routers work in concert with the Cisco Unified CallManager, deployed in either a distributed or centralized call-processing model, to provide the IP telephony 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 sites to take advantage of new applications. An architecture in which a Cisco Unified CallManager 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 CallManager 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

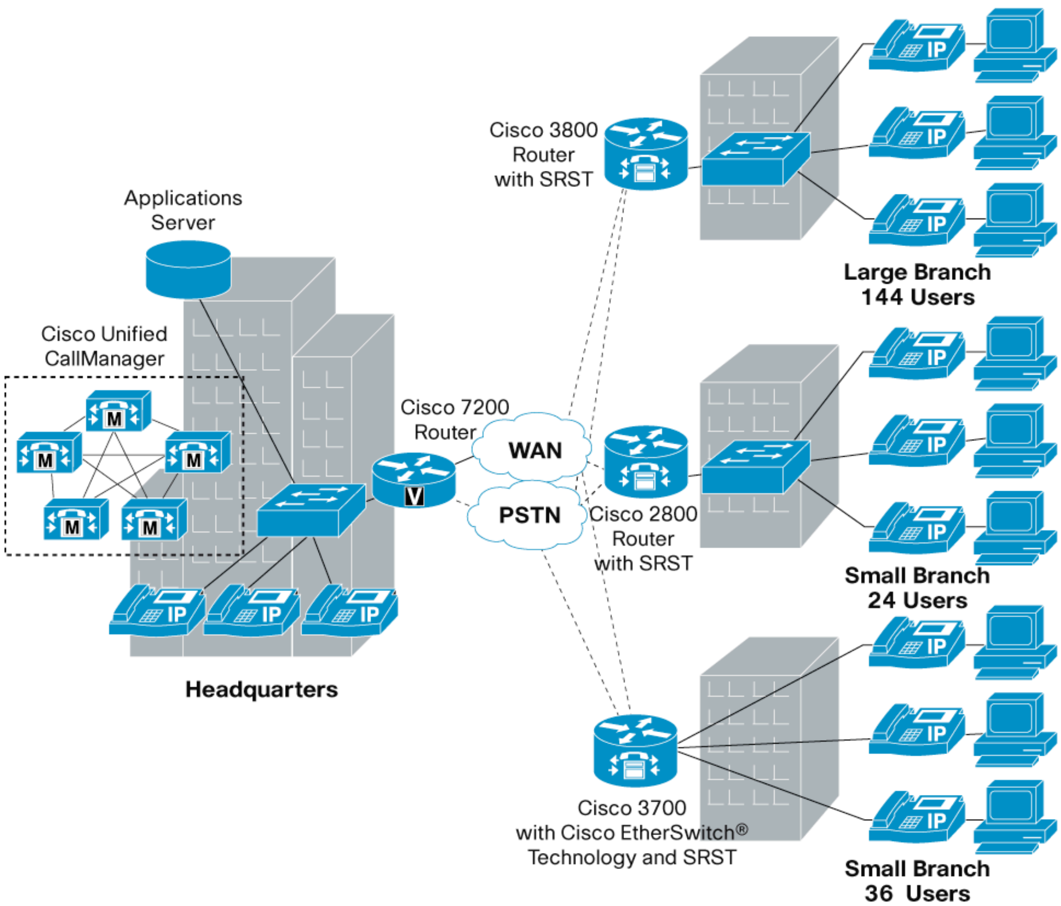
As enterprises extend their IP telephony deployments from central sites to remote offices, an important consideration is the ability to cost-effectively provide failover capability at remote branch offices. However, the size and number of these small-office sites preclude most enterprises from deploying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

Cisco Unified CallManager 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 1700, 2600, 2800, 3700, and 3800 series voice gateway routers.

SRST software automatically detects a connectivity failure between Cisco Unified CallManager and IP phones at the branch office. Using the Cisco Simple Network Automated Provisioning (SNAP) capability, SRST initiates a process to automatically configure the Cisco 1700, 2600, 2800, 3700, and 3800 series voice gateway 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 CallManager, the system automatically shifts call-processing functions back to the primary Cisco Unified CallManager cluster. Configuration for this capability is performed only once in the Cisco Unified CallManager at the central site (Figure 2).

Figure 2. Centralized Cisco Unified CallManager Deployment with SRST



Cisco Voice Gateway Router Features and Benefits

Simple Administration

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

Availability

- Provides for Cisco Unified CallManager redundancy; if a primary host Cisco Unified CallManager fails, call control fails over to the next available Cisco Unified CallManager server
- Branch survivability using SRST when connection to the Cisco Unified CallManager 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 CallManager clustering

Investment Protection

- Provides a modular platform design with a growing list of more than 90 interface combinations
- Allows users to increase voice capacity while taking advantage of their existing investments in Cisco 1700, 2600, 2800, 3700, and 3800 series routers

Voice Gateway Router With Cisco Unified Callmanager Feature Summary

Table 1 summarizes the features of the voice gateway routers with Cisco Unified CallManager.

Table 1. Voice Gateway Router with Cisco Unified CallManager 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 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 helps enable 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 helps enable connection to PSTN.
Y	N	Y	BRI Q.931 network side (NET3)	This feature helps enable connection to 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 help enable connection to a PBX or key system.
N	Y[4]	N	T1 E&M hookflash	Used to transfer a call from TDM 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 helps enable connection to a PBX or PSTN.

[1] Supports loop-start signaling only.
[2] Requires Cisco IOS Software Release 12.4(15)XZ or later and Cisco Unified CallManager 6.1 or later.
[3] This feature is supported between gateways in the absence of Cisco Unified CallManager.
[4] Requires Cisco IOS Software release 12.4(4)T or later and Cisco Unified CallManager 4.2 or later.
[5] T1-CAS feature group D is not supported on the Cisco 1700 Series voice gateway 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 helps enable 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	N3	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 voice gateway routers by making them configurable on the Cisco Unified CallManager. Configuration information is automatically downloaded at startup and after any configuration change.
Y	Y	Y	Cisco Unified CallManager failover redundancy	When the voice gateway router loses contact with the primary Cisco Unified CallManager, the gateway uses the next available Cisco Unified CallManager.
Y	Y	Y[7]	Cisco Unified CallManager call preservation during failover	Existing calls are preserved during a failover to the next available Cisco Unified CallManager. Calls are also preserved upon restoration of the primary host Cisco Unified CallManager.
Y	Y	Y	SRST and gateway fallback	When contact with the Cisco Unified CallManager 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	Y7	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 CallManager cluster and gateway fallback. Calls are also preserved upon restoration of the Cisco Unified CallManager connection.
Y	Y	Y7	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 CallManager cluster and gateway fallback. Calls are also preserved upon restoration of the Cisco Unified CallManager connection.
Y	Y	Y	Multicast music on hold (MoH) – centralized	This feature helps the voice gateway 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 voice gateway 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.

[6] Support is for forward, transfer, and conference. Message waiting indicator is from SIP to QSIG (not the reverse) and requires 12.4(11)T. Calling and connected ID are not supported.

[7] Requires Cisco Unified CallManager 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
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 CallManager for specification of the time between beeps.
Y	Y	Y	Caller ID support[8]	This feature helps the voice gateway router send the caller ID of a caller for display: • In MGCP mode, to and from IP phone, FXS, T1/E1 PRI; FXO to IP phone, not viceversa (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 viceversa.
N	Y	Y	Malicious call 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 assures 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 Group III fax 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 helps enable 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	Users can choose to transmit voice across their networks 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.723.1, G.726, and 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.
N/A	N/A	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	N/A	N/A	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.

[8] Requires Cisco IOS Software Release 12.4(15)XZ or later.
[9] Requires Cisco Unified CallManager 4.2(3). Not yet supported in Cisco Unified CallManager 5.0.

SIP	MGCP	H.323	Feature	Benefits
Y	N	Y	Interactive-voice-response (IVR) support	IVR offers automated-attendant support, voicemail support, or call routing based on service desired.
Y	N	Y	Automated attendant (AA)	This feature uses IVR to provide automated call-answering and -forwarding services.
Y	N	Y	Voice XML (VXML)	VXML controls calls “in queue” at the gateway for call center applications. Calls are redirected only when an agent becomes available.
N	Y	Y7	Overlap sending over voice over IP (VoIP)	This feature speeds variable-length dial strings dialing.
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 less 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	Users can change the numbering plan on the gateway before the call goes out over the PSTN.
N	Y	N	Billing granularity to DS-0 channel level on Cisco Unified CallManager CDR	This feature provides increased granularity on time-division multiplexing (TDM) usage down to the individual channel.
Y	Y	N	Name display on PRI using FACILITY IE (caller name [CNAM])	This feature provides caller name display on IP phones for PSTN calls.
N	Y[10]	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[11]	N	Connection to Defense Switched Network (DSN)	This feature supports the U.S. Department of Defense private TDM network.
Y[12]	Y[13]	Y[14]	SRTP: Media authentication and encryption on VGWs	Enables secure gateway to gateway calls and secure IP phone to gateway calls
Y[15]	Y[16]	Y[17]	Signaling encryption SIP: TLS, MGCP/H323: IPSEC	Encrypts signaling communication between voice gateways and CUCM

Voice Gateway Router with Cisco Unified Callmanager Minimum System Requirements

Tables 2 through 5 give system requirements for the voice gateway routers.

[10] 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.
[11] Requires Cisco IOS Software Release 12.4(2)T or later.
[12] Cisco IOS Software Release 12.4(15)T or later, Cisco Unified CallManager 5.0 (line-side) or later, Cisco Unified CallManager trunk-side support currently not available.
[13] Cisco IOS Software Release 12.4(3) or later, Cisco Unified CallManager 4.1 or later.
[14] Cisco IOS Software Release 12.4(6)T2 or later, Cisco Unified CallManager 5.0 or later.
[15] Cisco IOS Software Release 12.4(6)T, Cisco Unified CallManager 5.0 or later.
[16] Cisco IOS Software Release 12.4(3), Cisco Unified CallManager 4.1 or later.
[17] Cisco IOS Software Release 12.4(6)T1, Cisco Unified CallManager 5.0 or later.

Table 2. Voice Gateway Router with Cisco Unified CallManager Minimum System Requirements Using SIP

TDM Protocol or Feature	Minimum Cisco IOS Software Release*	Minimum Cisco Unified CallManager 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 chart shows when a Cisco IOS Software particular interface type was first tested with Cisco Unified CallManager. It does not document when individual network modules, advanced integration modules (AIMs), 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 CallManager does not need to know which network module, AIM, or platform is used. Hence, when Cisco Unified CallManager supports a particular protocol or feature, this support is sufficient for operation.

Table 3. Voice Gateway Router with Cisco Unified CallManager Minimum System Requirements Using H.323

TDM Protocol or Feature	Minimum Cisco IOS Software Release*	Minimum Cisco Unified CallManager 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 chart shows when a particular interface type is first supported in Cisco IOS Software. It does not document when individual network modules, AIMs, 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 CallManager does not need to know which network module, AIM, or platform is used. Hence, when Cisco Unified CallManager supports a particular protocol or feature, this support is sufficient for operation.

Table 4. Voice Gateway Router with Cisco Unified CallManager Minimum System Requirements Using MGCP

Active Platforms	Interface Part Number	TDM Protocol or Feature	Minimum Cisco IOS Software Release	Minimum Cisco Unified CallManager Release
Cisco 1751 and 1760	VIC2-2FXS and VIC2-2FXO	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
	VIC2-4FXO	Analog FXO	12.3(5) or 12.3(4)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
	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 2801	VIC2-2FXS and VIC2-2FXO	Analog FXS and FXO	12.3.8T4	3.3.5, 4.0.2a SR1, or 4.1.2 SR1
	VIC-4FXS/DID	Analog FXS	12.3.8T4	3.3.5, 4.0.2a SR1, or 4.1.2 SR1

Active Platforms	Interface Part Number	TDM Protocol or Feature	Minimum Cisco IOS Software Release	Minimum Cisco Unified CallManager Release
	VIC2-4FXO	Analog 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
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	BRI	12.4.2T	4.1.3 SR1
	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

Active Platforms	Interface Part Number	TDM Protocol or Feature	Minimum Cisco IOS Software Release	Minimum Cisco Unified CallManager Release
	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	BRI	12.4.2T	4.1.3 SR1
	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 2811, 2821, and 2851	EVM-HD-8FXS/DID and EVM-HD-8FXS/DID with EM-HDA-8FXS	Analog FXS and FXO	12.3.8T4	3.3.5, 4.0.2a SR1, or 4.1.2
	EVM-HD-8FXS/DID with 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
	NM-HD	BRI	12.4.2T	4.1.3 SR1
	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
	VIC2-2BRI-NT/TE	BRI	12.4.2T	4.1.3 SR1
	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
Cisco 3725 and 3745	NM-1V/2V	Analog FXS and FXO	12.2.8T	3.2(2c)spA

Active Platforms	Interface Part Number	TDM Protocol or Feature	Minimum Cisco IOS Software Release	Minimum Cisco Unified CallManager Release
	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	BRI	12.4.2T	4.1.3 SR1
	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
Cisco 3825 and 3845	EVM-HD-8FXS/DID, and EVM-HD-8FXS/DID with EM-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
	NM-HD	BRI	12.4.2T	4.1.3 SR1
	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
	VIC2-2BRI-NT/TE	BRI	12.4.2T	4.1.3 SR1
	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

Active Platforms	Interface Part Number	TDM Protocol or Feature	Minimum Cisco IOS Software Release	Minimum Cisco Unified CallManager Release
	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

**QSIG supplementary requires Cisco Unified CallManager 4.0 or later. QSIG basic services were first introduced with Cisco CallManager 3.3 and Cisco IOS Software Release 12.2.11T.
***MLPP requires Cisco Unified CallManager 4.0.2 or later.

Table 5. Voice Gateway Router with Cisco Unified CallManager 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 CallManager 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 2801	Onboard PVDM2 DSPs	Conferencing, transcoding, and media termination point (MTP)	12.3.11T	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 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 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
Cisco 3825 and 3845	Onboard PVDM2 DSPs	Conferencing and transcoding	12.3.11T	3.3.5, 4.0.2a SR1, or 4.1.2

Active Platforms	Interface Part Numbers	TDM Protocol or Feature	Minimum Cisco IOS Software Release	Minimum Cisco Unified CallManager Release
	Onboard PVDM2 DSPs	MTP	12.3.11T	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	12.3.11T	3.3.5, 4.0.2a SR1, or 4.1.2
	NM-HD-1V/2V/2VE and NM-HDV2	MTP	12.3.11T	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

Voice Performance

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

Table 6. Maximum Physical DS-0 Connectivity on the Cisco 1700, 2600, 2800, 3700, and 3800 Voice Gateway Routers*

	Cisco 1751 and 1760	Cisco 2801	Cisco 2610 XM	Cisco 2620 XM	Cisco 2650 XM	Cisco 2811	Cisco 2821	Cisco 2851	Cisco 2691	Cisco 3725	Cisco 3745	Cisco 3825	Cisco 3845
FXS	Cisco 1751: 12 Cisco 1760: 16	16	12	12	12	28	52	52	12	24	48	52	88
FXO and CAMA	Cisco 1751: 12 Cisco 1760: 16	16	8	8	8	24	36	36	8	16	32	36	56
E&M	Cisco 1751: 6 Cisco 1760: 8	8	4	4	4	12	12	12	4	8	16	16	24
Analog DID	Cisco 1751: 6 Cisco 1760: 8	8	8	8	8	24	32	32	8	16	32	32	48
BRI ports	Cisco 1751: 6 Cisco 1760: 8	8	4	4	4	12	20	20	4	8	16	20	32
T1/E1 ports	1	1	5	5	5	12	12	12	6	10	18	16	24
T1 channels	24	24	120	120	120	288	288	288	144	240	432	384	576
E1 channels	30	30	150	150	150	360	360	360	180	300	540	480	720

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

Table 7. CPU Performance on the Cisco 1700, 2600, 2800, 3700, and 3800 Series Voice Gateway Routers*

	Cisco 1751 and 1760	Cisco 2801	Cisco 2610 XM	Cisco 2620 XM	Cisco 2650 XM	Cisco 2811	Cisco 2821	Cisco 2851	Cisco 2691	Cisco 3725	Cisco 3745	Cisco 3825	Cisco 3845
VoIP Performance: Maximum Simultaneous Calls (not exceeding 75-percent platform CPU usage)													
Standalone Voice Gateway[18]													
No encryption	32	32	32	38	50	70	112	170	130	180	290	340	450
SRTP with signaling in IP Security (IPsec)	–	32	24	30	40	60	96	140	106	155	250	290	380
SRTP with signaling and media in IPsec	–	32	12	16	22	34	52	80	60	85	135	160	210
WAN Edge Gateway[19]													
No encryption	32	32	18	20	35	48	88	150	100	140	220	290	330
SRTP with signaling in IPsec	–	32	14	16	29	41	80	130	82	120	190	240	280
SRTP with signaling and media in IPsec	–	32	8	9	16	22	44	70	47	65	102	130	150
WAN Edge Gateway with Compressed Real-Time Protocol (CRTP)[20]													
No encryption	24	26	12	16	24	35	61	120	82	102	170	250	270
SRTP with signaling in IPsec	–	22	10	14	20	31	51	100	69	90	140	210	230
SRTP with signaling and media in IPsec	–	14	5	7	10	17	28	60	39	48	77	110	130
VoIP Performance: Maximum Calls per Second (not exceeding 75-percent CPU)													
	0.1	0.5	0.1	0.25	0.5	0.7	0.8	1	2	1.5	4	3	7

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.

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

Test release is Cisco IOS Software Release 12.4.1. This document contains general numbers as a guide to the approximate performance of the voice gateways. 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 you are encouraged to do proof-of-concept testing for more specific performance numbers for a specific scenario, traffic pattern, or release.

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