New Voice and Telephony Features in Cisco IOS Releases 12.3T and 12.4

This document lists new Cisco IOS voice and telephony features in Cisco IOS Releases 12.3T and 12.4, and the location in the Cisco IOS Voice Configuration Library where each feature is documented. This information is presented in two tables:

Unless otherwise specified, all features supported in Cisco IOS Release 12.3T are also supported in Cisco IOS Release 12.4.

To determine what software releases and platform support each feature uses, see

Cisco Feature Navigator.

Note For information about the full set of Cisco IOS voice features, see the entire **Cisco IOS Voice Configuration Library**—including library preface, glossary, and other documents—at http://www.cisco.com/univercd/cc/td/doc/product/software/ios124/124tcg/vcl.htm.

Finding Support Information for Platforms and Cisco IOS Software Images

Use Cisco Feature Navigator to find information about platform support and Cisco IOS software image support. Access Cisco Feature Navigator at http://www.cisco.com/go/fn. You must have an account on Cisco.com. If you do not have an account or have forgotten your username or password, click **Cancel** at the login dialog box and follow the instructions that appear.

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New Voice and Telephony Features in Cisco IOS Releases 12.3T and 12.4 in Alphabetical Order

Table 1 lists in alphabetical order new voice and telephony features in Cisco IOS Releases 12.3T and 12.4.

Table 1 New Voice Features in Cisco IOS Releases 12.3T and 12.4 in Alphabetical Order

Feature	First Supported Release	Feature Description	Where Documented
Accounting Server Connectivity Failure and Recovery Detection	12.3(4)T	Provides the scriptable option to reject new calls entering the VoIP network and tear down all existing calls on detecting connectivity failure to the method list that is associated with RADIUS-based accounting servers.	*Cisco IOS Voice Troubleshooting and Monitoring Guide*
AIM-CUE	12.3(7)T	Provides support for Cisco Unity Express voice mail and auto attendant for either Cisco CallManager or Cisco CallManager Express IP Communications networks. The AIM-CUE is supported on the Cisco 2600XM series, Cisco 2691, and Cisco 3700 series voice gateway routers on an AIM form factor.	Installing Advanced Integration Modules in Cisco 2600 Series, Cisco 3600 Series, and Cisco 3700 Series Routers
Call Failure Recovery (Rotary) on IPIPGW	12.3(11)T	This enhancement eliminates the need for identical codec capabilities for all dial peers in the rotary group, and allows the IP-to-IP gateway to restart the codec negotiation process with the originating endpoint based on the codec capabilities of the next dial peer in the rotary group.	"Configuring a Cisco Multiservice IP-to-IP Gateway" chapter in the *Cisco Multiservice IP-to-IP Gateway Application Guide*

Table 1 New Voice Features in Cisco IOS Releases 12.3T and 12.4 in Alphabetical Order (continued)

Feature	First Supported Release	Feature Description	Where Documented
Call Pickup ringing extension	12.3(11)T	The SRST **pickup** command has been introduced to enable the PickUp soft key on all Cisco IP phones, allowing an external direct inward dialing (DID) call coming into one extension to be picked up from another extension during SRST.	The **pickup** command in the *Cisco Survivable Remote Site Telephony Version 3.2 Command Reference*.
Call Routing Enhancements to the H.323 Gatekeeper and GKTMP (GK API)	12.3(7)T	Improves routing flexibility in customer networks where an external route server is used to select potential endpoints for call completion. (1) Nonblocking GKTMP (GK API): Timing changes associated with recovery processing when socket errors occur. (2) Separate DNIS for alternate endpoints: It is now possible to associate a unique DNIS with each alternate endpoint. (3) Support for "z" tag in RESPONSE xRQ: Enhances the responses a route server can provide to the H.323 gatekeeper to allow greater flexibility for combinations of gateway endpoints and gatekeepers.	*Cisco Gatekeeper External Interface Reference*
Call Statistics on Voice-Enabled Gateways	12.3(4)T	Enables the collection of voice call statistics based on user-configured time ranges. The statistics that can be collected are from the following functional areas: • RADIUS accounting • Cisco IOS generated internal error codes (IECs) • Gateway port (interface) statistics	*"Voice Performance Statistics on Cisco Gateways"* chapter in the *Cisco IOS Voice Troubleshooting and Monitoring Guide*

Table 1 New Voice Features in Cisco IOS Releases 12.3T and 12.4 in Alphabetical Order (continued)

Feature	First Supported Release	Feature Description	Where Documented
Cisco CallManager Express 3.0	12.3(4)T	Enables Cisco routers to deliver key system or hybrid PBX functionality for enterprise branch offices or small businesses. Cisco CME is ideal for customers who have data connectivity requirements and also have a need for a telephony solution in the same office. Whether offered through a service provider's managed services offering or purchased directly by a corporation, Cisco CME offers most of the core telephony features required in the small office, in addition to many advanced features not available with traditional telephony solutions. Being able to deliver IP telephony and data routing using a single converged solution allows customers to optimize their operations and maintenance costs, resulting in a very cost-effective solution that meets office needs. **Note** Before Version 3.0, Cisco CallManager Express was named Cisco IOS Telephony Services (Cisco ITS).	Cisco CallManager Express 3.0

Table 1 New Voice Features in Cisco IOS Releases 12.3T and 12.4 in Alphabetical Order (continued)

Feature	First Supported Release	Feature Description	Where Documented
Cisco CallManager Express 3.1	12.3(7)T	Introduces enhancements to allow interoperability with a mix of platforms in a WAN across an H.323 network. The mix of platforms can include earlier versions of Cisco CME, Cisco CallManager, Cisco BTS Softswitch (BTS), and Cisco PSTN Gateway (PGW), and also other Cisco IOS voice gateways. The enhancements include support of H.450.12 standards for H.450 capabilities exchange with remote H.323 endpoints, automatic detection of Cisco CallManager endpoints, H.323-to H.323 hairpin call routing, and H.323-to-H.323 routing to H.450 tandem gateways. Other enhancements introduced in this feature include Call Park, CFwdAll Soft Key Restriction Control, and enhancements to automatic line selection and ephone hunt groups. Language display localization and directory search are supported on Cisco IP Phone 7905 and Cisco IP Phone 7912. Call progress tone localization is supported on Cisco IP Phone 7902, Cisco IP Phone 7905, and Cisco IP Phone 7912. The Cisco Wireless IP Phone 7920 and Cisco IP Phone Conference Station 7936 are fully supported in the Cisco IOS CLI and Cisco CME GUI.	Cisco CallManager Express 3.1
Cisco CallManager Express 3.2	12.3(11)T	Cisco CallManager Express 3.2 adds a number of key telephony features including support for 240 phones, transcoding, and RCF 2833 DTMF support.	Cisco CallManager Express 3.2

Table 1 New Voice Features in Cisco IOS Releases 12.3T and 12.4 in Alphabetical Order (continued)

Feature	First Supported Release	Feature Description	Where Documented
Cisco VG224 24-Port Analog Phone Gateway	12.3(7)T	Enables a hybrid of using VoIP Technology (AVVID-based architectures with Cisco CallManager as call control) with TDM analog endpoints (analog phones, fax machines, analog modems). Supported on Cisco CallManager Release 3.2 or later.	Voice Gateways
COR list increase from 10 to 20	12.3(11)T	The maximum number of COR lists has been increased to 20.	The **cor** command in the *Cisco Survivable Remote Site Telephony Version 3.2 Command Reference*.
Customizable Tone Download to Cisco IOS MGCP Gateways from Cisco Call Manager	12.3(4)T	Implements the downloading of region-specific tones and the associated frequencies, amplitudes, and cadences using XML-based configuration files during gateway registration. The feature also introduces the ability to play out tones that contain up to four frequencies.	"Configuring Tone Download to MGCP Gateway" chapter in the *Cisco CallManager and Cisco IOS Interoperability Guide*
Enhanced Conferencing and Transcoding for Voice Gateway Routers	12.3(8)T	Provides conferencing, transcoding, and MTP services for Cisco voice gateways in a Cisco CallManager network. Uses DSP resources on the NM-HDV2 and NM-HD high-density digital voice/fax network modules.	"Configuring Enhanced Conferencing and Transcoding for Voice Gateway Routers" chapter in the *Cisco CallManager and Cisco IOS Interoperability Guide*
Enhanced ITU-T G.168 Echo Cancellation	12.3(4)T	Extends the existing ITU-T G.168 Echo Cancellation functionality by adding support for medium complexity CODECs.	"Configuring Echo Cancellation" chapter in the "*Cisco IOS Voic Port Configuration Guide*
Enhanced ITU-T G.168 Echo Cancellation	12.3(7)T	Includes platforms using the TI C5510 DSP.	"Configuring Echo Cancellation" chapter in the "*Cisco IOS Voic Port Configuration Guide*
ETSI Call Transfer	12.3(8)T	Provides support for European Telecommunications Standards Institute (ETSI) explicit call transfer functionality on Cisco IOS gateways.	"*Configuring Telephony Call-Redirect Features*" chapter in the *Cisco IOS TcL IVR and VoiceXML Application Guide*

Table 1 New Voice Features in Cisco IOS Releases 12.3T and 12.4 in Alphabetical Order (continued)

Feature	First Supported Release	Feature Description	Where Documented
External Music on Hold Source	12.3(11)T	Cisco SRST has been enhanced with the **moh-live** command. The **moh-live** command provides live feed MOH streams from an audio device connected to an E&M or FXO port to Cisco IP phones in SRST mode. Music from a live feed is from a fixed source and is continuously fed into the MOH playout buffer instead of being read from a flash file. Live feed MOH can also be multicast to Cisco IP phones.	*Integrating Cisco CallManager and Cisco SRST to Use Cisco SRST As a Multicast MOH Resource*
Gatekeeper Prefix Selection for Hair-Pinned Calls	12.3(11)T	Allows H.323 gatekeeper to terminate/hairpin calls from a TDM/PSTN endpoint back through the same originating gateway based on priority/zone prefix values.	*Cisco Multiservice IP-to-IP Gateway Application Guide*

Table 1 New Voice Features in Cisco IOS Releases 12.3T and 12.4 in Alphabetical Order (continued)

Feature	First Supported Release	Feature Description	Where Documented
High-Density Analog (FXO/FXS/DID) and Digital (BRI) Extension Module for Voice/Fax (EVM-HD)	12.3(11)T	The High-Density Analog (FXO/FXS/DID) and Digital (BRI) Extension Module for Voice/Fax (EVM-HD) feature delivers a higher density integrated analog/digital voice interface. The EVM-HD-8FXS/DID baseboard provides eight FXS and DID ports. This network module accesses digital signal processor (DSP) modules on the motherboard, instead of using onboard DSPs. You can increase the port density by plugging in up to two optional expansion modules in any combination: • EM-HDA-8FXS—8-port voice/fax expansion module • EM-HDA-3FXS/4FXO—7-port voice/fax expansion module • EM-HDA-6FXO—6-port voice/fax expansion module • EM-4BRI-NT/TE—4-port ISDN BRI expansion module PVDM2 DSP modules are used in combination with the EVM-HD-8FXS/DID baseboard and its expansion modules. PVDM2 modules are available separately and installed in the DSP module slots located inside the router chassis.	*Cisco IOS Voice Port Configuration Guide*
HTTP Client API for Tcl IVR	12.3(14)T	Enables Tcl IVR applications to retrieve data from or post data to an external HTTP server. Also introduces a new command-line-interface structure for configuring voice applications and support for additional Tcl 8.3.4 commands.	*HTTP Client API for Tcl IVR* *Tcl IVR 2.0 Programming Guide*

Table 1 New Voice Features in Cisco IOS Releases 12.3T and 12.4 in Alphabetical Order (continued)

Feature	**First Supported Release**	**Feature Description**	**Where Documented**
Inactive Call Detection	12.3(4)T	Detects inactive (silent) H.323 or SIP call-legs on Cisco IOS software-based gateways, and reports this situation to the TcL IVR 2.0 application (which can disconnect the call). Enables the Cisco IOS software to not automatically disconnect detected inactive calls. Inactivity is defined as no RTP/RTCP packets for a configurable length of time.	"Troubleshooting Voice Applications" chapter of the *Cisco IOS Voice Troubleshooting and Monitoring Guide*
Increase alias command from 10 to 50	12.3(11)T	The SRST **alias** command has been enhanced in the following areas: • The maximum number of **alias** commands used for creating calls to telephone numbers that are unavailable during Cisco CallManager fallback has increased to 50. • The **cfw** keyword was added, providing call forward no-answer/busy capabilities. • The *alternate-number* argument can be used in multiple **alias** commands.	The **alias** command in the *Cisco Survivable Remote Site Telephony Version 3.2 Command Reference*
Increase phones supported from 240 to 720 on Access Router	12.3(11)T	The Cisco 3845 now supports 720 phones and up to 960 ephone-dns or virtual ports.	*Cisco Survivable Remote Site Telephony (SRST) 3.2 Specifications for Cisco IOS Software Release 12.3(11)T*

Table 1 New Voice Features in Cisco IOS Releases 12.3T and 12.4 in Alphabetical Order (continued)

Feature	First Supported Release	Feature Description	Where Documented
Interoperability Enhancements to the Cisco Multiservice IP-IP Gateway	12.3(7)T	Introduces the following enhancements: (1) IP-to-IP gateway is now interoperable with the Cisco ATA-188 and Microsoft NetMeeting. (2) Tcl IVR 2.0: Full support for programmable/scripted applications with VoIP endpoints; prepaid applications are one example. (3) IP-to-IP gateway image consolidation: Combines TDM-to-IP voice gateway and IP- to-IP gateway feature sets in a single Cisco IOS image. The combined functions can run concurrently on the same hardware platform. This reduces the cost of deployment for the IP-to-IP gateway in enterprise networks.	*Cisco Multiservice IP-to-IP Gateway Application Guide*
IP Communications High-Density Digital Voice/Fax Network Module	12.3(7)T	Supports high-density digital voice and low-density analog voice connectivity along with data and integrated access connectivity. The network modules offer built-in T1/E1 VIC/VWIC) slot for FXS, FXO, E&M, Centralized Automatic Message Accounting (CAMA), direct inward dialing (DID), BRI, or E1 and T1 cards, up to a maximum of four T1/E1 ports. Supports up to 32 HDLC channels with an aggregate capacity of 2.048 Mbps.	"Configuring Digital Voice Ports" chapter in the *Cisco IOS Voice Port Configuration Guide*
IP Communications Voice/Fax Network Module	12.3(4)T	Provides the ability to directly connect the PSTN and legacy telephony equipment to Cisco 2600XM series, Cisco 2691, Cisco 3600 series, and Cisco 3700 series multiservice routers, enabling applications such as IP telephony, toll bypass, and full gateway integration. The VWICs supported by the new network modules include 2- and 4-port FXS; 2- and 4-port FXO; 2-port DID, E&M, and BRI (S/T); and 1- and 2-port T1/E1.	*Cisco IOS Voice Port Configuration Guide*

Table 1 New Voice Features in Cisco IOS Releases 12.3T and 12.4 in Alphabetical Order (continued)

Feature	First Supported Release	Feature Description	Where Documented
ISDN Calling Name Display	12.3(4)T	Provides calling name display to SIP customers on calls that originate on ISDN networks.	"Configuring SIP ISDN Support Features" chapter of the *Cisco IOS SIP Configuration Guide*
Japanese Katakana Localization	12.3(11)T	Japanese Katakana is now supported when using the SRST **user-locale** command. The new **JP** keyword is available to Cisco SRST systems running under Cisco CallManager V4.0.	The **user-locale** command in the *Cisco Survivable Remote Site Telephony Version 3.2 Command Reference.*
Lossless Compression R1, ATM Cell Switching, and BITS Clocking	12.3(7)T	Introduces a new compression technique in DSP firmware and add enhancements to Cisco IOS software that include cell switching on ATM segmentation and reassembly (SAR), and the use of an external BITS clocking source. These features enable Cisco multiservice routers to be used to transparently groom and compress traffic in a wireless service provider network and enable a service provider to optimize the bandwidth used to backhaul the traffic from a cell site to the mobile central office for more efficient use of existing T1 and E1 lines.	*Lossless Compression R1, ATM Cell Switching and BITS Clocking*
Malicious Caller Identification Invocation Support for Enterprise Networks	12.3(14)T	Extends support for MCID service in the PSTN to the Cisco 2801.	Malicious Caller Identification Invocation Support for Enterprise Networks
MCID for Cisco IOS Voice Gateways	12.3(11)T	Supports the Malicious Call Identification (MCID) supplementary service that enables Cisco CallManager 4.0 to identify the source of malicious calls.	"Configuring MCID for Cisco IOS Voice Gateways" chapter in the *Cisco CallManager and Cisco IOS Interoperability Guide.*
Media and Signaling Authentication and Encryption Feature for Cisco IOS MGCP Gateways	12.3(14)T	Delivers media and signaling authentication and encryption on the Cisco 2600XM series, Cisco 2691, Cisco 3660 series, Cisco 3700 series, and Cisco VG224. This feature enables secure gateway-to-gateway and IP-phone-to-gateway calls, and interoperates with Cisco 7970 IP phones.	*Media and Signaling Authentication and Encryption Feature for Cisco IOS MGCP Gateways*

Table 1 New Voice Features in Cisco IOS Releases 12.3T and 12.4 in Alphabetical Order (continued)

Feature	First Supported Release	Feature Description	Where Documented
MGCP Configuration Control for Setting Fax Rate	12.3(8)T	Establishes the maximum fax transmission rate for MGCP T.38 sessions. MGCP fax rate is set to the highest possible transmission speed allowed by the voice codec.	"Configuring T.38 Fax Relay" section in the *Cisco Fax Services over IP Application Guide*
MGCP Gateway Support for Cisco CallManager Network Specific Facilities	12.3(4)T	Provides an enhancement to Cisco CallManager functionality enabling users to configure the NSF ISDN information element of the route pattern. This feature is compatible with Cisco CallManager Version 3.3(2) and later.	"Configuring MGCP PRI Backhaul and T1 CAS Support for Cisco CallManager" chapter in the *Cisco CallManager and Cisco IOS Interoperability Guide*
MGCP Line Control Signaling Package support	12.3(8)T	Supports the transport of line supervision signals in the media stream using RFC 2833 event packets in PacketCable GR303-switched IP systems, using the modified **mgcp package-capability package** command. When the **lcs-package** keyword is used, the named telephony events (NTEs) associated with the LCS package are enabled automatically.	"Basic MGCP Configuration" chapter of the *Cisco IOS MGCP and Related Protocols Configuration Guide*
MGCP-Controlled Backhaul of BRI Signaling in Conjunction with Cisco CallManager	12.3(2)T	Provides MGCP service to remote-office media gateways that connect by means of ISDN BRI trunks to a centralized Cisco CallManager media-gateway controller for the purpose of call processing. D-channel signal information is backhauled to the call manager through a Transmission Control Protocol (TCP) session.	"Configuring MGCP-Controlled Backhaul of BRI Signaling in Conjunction with Cisco CallManager" chapter in the *Cisco CallManager and Cisco IOS Interoperability Guide*.

Table 1 New Voice Features in Cisco IOS Releases 12.3T and 12.4 in Alphabetical Order (continued)

Feature	First Supported Release	Feature Description	Where Documented
MLPP for Analog and BRI Endpoints on Cisco IOS Voice Gateways	12.3(14)T	Provides the capability for Cisco IOS voice gateways to present analog and basic rate interface (BRI) phones to be controlled by Cisco CallManager as though they were Cisco IP phones, enabling the following: • Line-side support for the Multilevel Precedence and Preemption (MLPP) feature • Cisco CallManager registration of analog and Basic Rate Interface (BRI) endpoints • Cisco CallManager endpoint auto configuration support • Modem pass-through support • Cisco Survivable Remote Site Telephony (SRST) support	MLPP for Analog and BRI Endpoints on Cisco IOS Voice Gateways
MLPP for Cisco IOS Voice Gateways	12.3(11)T	Supports Multilevel Precedence and Preemption (MLPP) service, allowing authorized users to preempt lower priority voice calls using Cisco CallManager version 4.0.	"Configuring MGCP Gateway Support for Cisco CallManager" chapter in the *Cisco CallManager and Cisco IOS Interoperability Guide*.
NextPort Voice Tuning and Background Noise Statistics with NextPort Dual-Filter G.168 Echo Cancellation	12.3(11)T	This feature allows you to dynamically configure voice services on the NextPort-based platforms: Cisco AS5350, Cisco AS5400, Cisco AS5400HPX, and Cisco AS5850. This feature also provides improved voice quality and statistics reporting and adds dual-filter G.168 echo canceller capability in NextPort SPE firmware (SPEware) version 10.2.2 and later with Cisco IOS Release 12.3(11)T.	"NextPort Voice Tuning and Background Noise Statistics with NextPort Dual-Filter G.168 Echo Cancellation" chapter in the *Cisco IOS Voice Port Configuration*
Option to disable H225 TCP Timer from phone to gateway to maintain calls in progress during WAN outage	12.3(11)T	To preserve existing H.323 calls on the branch in the event of an outage, disable the H.225 keepalive timer by entering the **no h225 timeout keepalive** command. This feature is supported for SRST on Cisco IOS Releases 12.3(7)T1 and higher.	"Overview of Cisco IOS SRST" chapter in the *Cisco IOS SRST Version 3.2 System Administrator Guide*

Table 1 New Voice Features in Cisco IOS Releases 12.3T and 12.4 in Alphabetical Order (continued)

Feature	First Supported Release	Feature Description	Where Documented
Out-of-Band to In-Band DTMF Relay for Cisco IOS Voice Gateways	12.3(11)T	RFC 2833 capability enabling DTMF relay communication between SIP devices and nonSIP endpoints using Cisco CallManager version 4.0.	"Configuring Enhanced Conferencing and Transcoding for Voice Gateway Routers" chapter in the *Cisco CallManager and Cisco IOS Interoperability Guide*.
Overlap Signaling Processing on H.323 Terminating Gateways	12.3(11)T	In an overlap signaling scenario, The called number in the SETUP message does not contain enough digits to match the incoming dial peer for the dial peer to select the right application. With this change, the H.323 layer determines if a partial match is detected, appends the called number with the needed digits. The new called number is checked to see if it matches any of the incoming dial-peers. If either full match or no match is returned, the call proceeds with SETUP procedure.	*Cisco Multiservice IP-to-IP Gateway Application Guide*
PCR Support for the Cisco Signaling Link Terminal	12.3(2)T	Adds support for Message Transfer Part Layer 2 (MTP2) PCR on the Cisco signaling link terminal (SLT). Preventive Cyclic Retransmission (PCR) adds basic MTP2 functionality that is used when Signaling System 7 (SS7) signaling links are transmitted over satellite connections between the Cisco SLT and a signal transfer point (STP).	PCR Support for the Cisco Signaling Link Terminal
Persistent TDM Switched Circuits	12.3(2)T	Enables the Cisco AS5850 universal gateway to connect one or more DS0s from one or more E1s or T1s to another set of E1s or T1s.	Persistent TDM Switched Circuits
PLAR (Private Line Automatic Ring-down) for Trading Turrets	12.3(4)T	Re-establishes an H.323 call automatically if one end hangs up or goes on hold and then goes off-hook again. This provides PLAR functionality used in the financial industry for trading Turret operation of PLAR.	*Cisco Hoot and Holler over IP*

Table 1 New Voice Features in Cisco IOS Releases 12.3T and 12.4 in Alphabetical Order (continued)

Feature	First Supported Release	Feature Description	Where Documented
QSIG Supplementary Features for Cisco IOS Voice Gateways	12.3(11)T	Supports Q Signaling (QSIG) over PRI backhaul interfaces on MGCP gateways to Cisco CallManager version 4.0.	"Configuring MGCP PRI Backhaul and T1 CAS Support for Cisco CallManager" chapter in the *Cisco CallManager and Cisco IOS Interoperability Guide*.
RCF 2833 DTMF Support from SCCP Devices to Cisco Unity Express	12.3(11)T	Cisco Skinny Client Control Protocol (SCCP) phones, such as those used with Cisco SRST systems, provide only out-of-band DTMF digit indications. To enable SCCP phones to send digit information to remote SIP-based IVR and voice-mail applications, Cisco SRST 3.2 and later versions provide conversion from the out-of-band SCCP digit indication to the SIP standard for DTMF relay, which is RFC 2833. You select this method in the SIP VoIP dial peer using the **tmf-relay rtp-nte** command.	*Cisco IOS Survivable Remote Site Telephony Version 3.2 System Administrator Guide*
Second-Generation 1- and 2-Port T1/E1 Multiflex Trunk Voice/WAN Interface Cards	12.3(14)T	Enables T1/E1 multiflex voice/WAN interface cards to support enhanced voice and data applications in Cisco multiservice routers. Provides the following: flexible T1 and E1 support; drop-and-insert multiplexing capability on all versions; support for a dedicated hardware echo-cancellation module; and, on 2-port cards, capability for each port to be clocked from an independent clock source.	*Second-Generation 1- and 2-Port T1/E1 Multiflex Trunk Voice/WAN Interface Cards* "Configuring Hardware Echo Cancellation" chapter in the *Voice Port Configuration Guide*.

Table 1 New Voice Features in Cisco IOS Releases 12.3T and 12.4 in Alphabetical Order (continued)

Feature	First Supported Release	Feature Description	Where Documented
Secure SRST	12.3(14)T	Secure Cisco IP phones that are located at remote sites and that are attached to gateway routers can communicate securely using the WAN with Cisco CallManager. But if the WAN link or Cisco CallManager goes down, all communication through the remote phones becomes nonsecure. To overcome this situation, gateway routers can now function in secure SRST mode, which activates when the WAN link or Cisco CallManager goes down. When the WAN link or Cisco CallManager is restored, Cisco CallManager resumes secure call-handling capabilities. Secure SRST provides new SRST security features such as authentication, integrity, and media encryption. Authentication provides assurance to one party that another party is whom it claims to be. Integrity provides assurance that the given data has not been altered between the entities. Encryption implies confidentiality; that is, that no one can read the data except the intended recipient. These security features allow privacy for SRST voice calls and protect against voice security violations and identity theft.	"Setting Up Secure SRST" chapter in the *Cisco IOS Survivable Remote Site Telephony Version 3.3 System Administrator Guide*.
Signal ISDN B-Channel ID to Enable Application Control of Voice Gateway Trunks	12.3(7)T	Enables call management applications to identify specific ISDN bearer (B) channels used during a voice gateway call for billing purposes. With the identification of the B channel, SIP and H.323 gateways can enable port-specific features such as voice recording and call transfer.	"Configuring SIP ISDN Support Features chapter of the *Cisco IOS SIP Configuration Guide* and "Configuring H.323 Gateway" chapter of the *Cisco IOS H.323 Configuration Guide*

Table 1 New Voice Features in Cisco IOS Releases 12.3T and 12.4 in Alphabetical Order (continued)

Feature	First Supported Release	Feature Description	Where Documented
SIP Audible Message-Waiting Indicator for FXS Phones	12.3(8)T	Enables an FXS port on a voice gateway to receive audible MWI in a SIP-enabled network. The FXS port on a voice gateway is an RJ-11 connector that allows connections to basic telephone service equipment.	"Configuring SIP MWI Support" chapter of the *Cisco IOS SIP Configuration Guide*
SIP Debug Output Filtering Support	12.3(4)T	Provides the ability to filter relevant SIP debugging traces of desired calls.	*Cisco IOS Voice Troubleshooting and Monitoring Guide*
SIP Gateway Support Enhancements to the bind Command	12.3(4)T	Allows configuration of the source IP address of signaling packets or both signaling and media packets using the **bind** command.	"Configuring SIP Gateway Support for the bind Command" chapter of the *Cisco IOS SIP Configuration Guide*
SIP Header Support and SUBSCRIBE and NOTIFY for External Triggers	12.3(4)T	Provides a mechanism for applications to send and receive SIP headers and to send SUBSCRIBE messages and receive NOTIFY events.	"Configuring Additional SIP Application Support" chapter of the *Cisco IOS SIP Configuration Guide*
SIP NOTIFY-Based Out-of-Band DTMF Relay Support	12.3(4)T	Supports SCCP devices through SIP originating and terminating gateway use of Cisco proprietary NOTIFY-based out-of-band DTMF relay, which can also be used by analog phones attached to analog voice ports (FXS) on a router.	"SIP NOTIFY-Based Out-of-Band DTMF Relay Support" chapter of the *Cisco IOS SIP Configuration Guide*
SIP Redirect Processing Enhancement	12.3(4)T	Allows flexibility in the handling of incoming redirect or 3xx class of responses. Redirect responses can now be enabled or disabled through the command-line interface, providing critical functionality for service providers who deploy Cisco gateways.	"Basic SIP Configuration" chapter of the *Cisco IOS SIP Configuration Guide*
SIP Register Support	12.3(4)T	Allows SIP gateways to register E.164 numbers to a SIP proxy or registrar on behalf of analog telephone voice ports (FXS), IP phone virtual voice ports (EFXS), and local SCCP phones.	"Basic SIP Configuration" chapter of the *Cisco IOS SIP Configuration Guide*

Table 1 New Voice Features in Cisco IOS Releases 12.3T and 12.4 in Alphabetical Order (continued)

Feature	First Supported Release	Feature Description	Where Documented
SIP RFC 3261 Enhancements	12.3(4)T	Provides enhanced SIP functionality on Cisco IOS gateways, as defined by RFCs 3261 and 3311, and support for the following: • Ability to receive and process SIP UPDATE requests • Initial Offer and Answer exchanges • Branch and Sent-by parameters in the Via header • Merged request detection • Loose-routing	"Basic SIP Configuration" chapter of the *Cisco IOS SIP Configuration Guide*
SIP Survivable Remote Site Telephony (SRST) Version 3.0	12.3(4)T	Describes Survivable Remote Site Telephony (SRST) functionality for Session Initiation Protocol (SIP) networks. SIP SRST provides backup to an external SIP proxy server by providing basic registrar and redirect services. These services are used by a SIP IP phone in the event of a WAN connection outage where the SIP phone is unable to communicate with its primary SIP proxy. The SIP SRST device also provides PSTN gateway access for placing and receiving PSTN calls.	SIP Survivable Remote Site Telephony (SRST)
SIP: Cisco IOS Gateway HTTP Digest Authentication and Registration	12.3(8)T	Implements authentication using the digest access on the client side of a common SIP stack. The Cisco IOS Session Initiation Protocol (SIP) gateway responds to authentication challenges from an authenticating server, proxy server, or user agent server (UAS). This feature also maintains parity between the Cisco gateways, proxy servers, and SIP phones that already support authentication.	"Configuring SIP AAA Features" chapter in the *Cisco IOS SIP Configuration Guide*

Table 1 New Voice Features in Cisco IOS Releases 12.3T and 12.4 in Alphabetical Order (continued)

Feature	First Supported Release	Feature Description	Where Documented
SIP: Cisco IOS Gateway Reason Header and Buffered Calling Name Completion	12.3(8)T	Implements support for the followingm functions: Reason Header and Buffered Calling Name Completion. The Cisco IOS SIP gateway reason header support provided on Cisco IOS gateways is defined by RFC 3326.	"Configuring SIP Message, Timer, and Response Features" chapter of the *Cisco IOS SIP Configuration Guide*
SRST: Survivable Remote Site Telephony Version 3.0	12.3(4)T	Provides Cisco CallManager with fallback support for Cisco IP phones attached to a Cisco router on your local network. Cisco SRST enables routers to provide call-handling support for Cisco IP phones when they lose connection to remote primary, secondary, or tertiary Cisco CallManager installations or when the WAN connection is down. Cisco CallManager supports Cisco IP phones at remote sites attached to Cisco multiservice routers across the WAN. Prior to Cisco SRST, when the WAN connection between a router and the Cisco CallManager failed or connectivity with Cisco CallManager was lost for some reason, Cisco IP phones on the network became unusable for the duration of the failure. Cisco SRST overcomes this problem and ensures that the Cisco IP phones offer continuous (although minimal) service by providing call-handling support for Cisco IP phones directly from the Cisco SRST router. The system automatically detects a failure and uses Simple Network Auto Provisioning (SNAP) technology to autoconfigure the branch office router to provide call processing for Cisco IP phones that are registered with the router. When the WAN link or connection to the primary Cisco CallManager is restored, call handling reverts back to the primary Cisco CallManager.	Cisco Survivable Remote Site Telephony (SRST) Version 3.0

Table 1 New Voice Features in Cisco IOS Releases 12.3T and 12.4 in Alphabetical Order (continued)

Feature	First Supported Release	Feature Description	Where Documented
Support Translation Profiles (CME and SRST)	12.3(11)T	Cisco SRST 3.2 supports translation profiles. Translation profiles allow you to group translation rules together and to associate translation rules with the following: • Called numbers • Calling numbers • Redirected called numbers	"Setting Up Call Handling" chapter in the *Cisco IOS Survivable Remote Site Telephony Version 3.2 System Administrator Guide*. Also, the **translation-profile** command in the *Cisco IOS Survivable Remote Site Telephony Version 3.2 System Administrator Guide*.
Survivable Remote Site Telephony 3.1	12.3(7)T	Provides Cisco CallManager with fallback support for Cisco IP phones attached to a Cisco router on a local network. The Cisco Wireless IP Phone 7920 and Cisco IP Phone Conference Station 7936 are fully supported.	*Cisco Survivable Remote Site Telephony (SRST) Version 3.1*
T.37 Fax Status Notification Enhancement in an MTA Environment	12.3(7)T	Provides the ability to delegate control of fax operations directly to a mail transfer agent (MTA) by configuring a T.37 fax off-ramp gateway to deliver all fax delivery errors to the fax mail originator in Simple Mail Transfer Protocol (SMTP) delivery status notification (DSN) messages with descriptive error codes.	"Configuring T.37 Store-and-Forward Fax" chapter in the *Cisco Fax Services over IP Application Guide*
T.38 Fax Relay on the Cisco Catalyst 6000 and Cisco 7600 Communication Media Module	12.3(14)T	Provides support for T.38 fax relay on the Cisco Catalyst 6000 and Cisco 7600 CMM.	"Configuring T.38 Fax Relay" chapter in the *Cisco Fax Services over IP Application Guide*.
T.38 Fax Statistics	12.3(14)T	Enables access servers with NextPort digital signal processors to gather detailed statistics about T.38 fax-relay calls. Statistics can be compiled into detailed call records for diagnostic and billing purposes.	*RADIUS VSA Voice Implementation Guide*

Table 1 New Voice Features in Cisco IOS Releases 12.3T and 12.4 in Alphabetical Order (continued)

Feature	First Supported Release	Feature Description	Where Documented
Tcl IVR 2.0 Session Interaction	12.3(4)T	Allows different instances of Tcl IVR applications (sessions) to communicate with other sessions on the same gateway and for applications to dynamically bridge call legs between different sessions. This enables different callers on the same gateway to be notified of each others' presence and to interact. You can also start a session without an active call leg so that a session can act as an application service for other sessions. This feature is useful for implementing a call-monitoring "server" application that is responsible for monitoring incoming calls and dynamically connecting selected callers.	Configuring Tcl IVR 2.0 Session Interaction chapter of the *Cisco IOS Tcl IVR and VoiceXML Applications Guide*
V.110 Support for MGCP-Dial	12.3(2)T	Provides support for V.110 encapsulation for Dial-MGCP Applications.	"Configuring NAS Package for MGCP" chapter of the *Cisco IOS MGCP and Related Protocols Configuration Guide*
Videoconferencing for the Cisco Multiservice IP-to-IP Gateway	12.3(4)T	Provides enhanced Quality of Service through RSVP synchronization with H.323 signaling protocol and DSCP packet marking.	"Configuring Multiservice IP-to-IP Videoconferencing" chapter in the *Cisco Multiservice IP-to-IP Gateway*
Voice Application HTTP Client Cookie Support	12.3(8)T	Implements HTTP cookie support for Cisco IOS VoiceXML applications.	Configuring Basic Functionality for Tcl IVR and VoiceXML Applications" chapter in the *Cisco IOS Tcl IVR and VoiceXML Application Guide.pdf*
Voice Application Monitoring and Troubleshooting Enhancements	12.3(8)T	Enables detailed monitoring of voice application instances and call legs using event logs and statistics. Records for terminated application instances and call legs are saved in history to assist in fault isolation. This comprehensive management information helps you diagnose problems in the network and identify the causes.	Monitoring and Troubleshooting Voice Applications chapter in the Cisco IOS Tcl IVR and VoiceXML Application Guide.pdf

Table 1 New Voice Features in Cisco IOS Releases 12.3T and 12.4 in Alphabetical Order (continued)

Feature	First Supported Release	Feature Description	Where Documented
Voice DSP Crash Dump Analysis	12.3(4)T	Allows Cisco IOS voice platforms using TI DSPs the ability to capture the contents of the DSP memory into a dump file if there is a DSP crash.	"Troubleshooting Digital Voice Interfaces to the IP Newtwork" chapter in th *Cisco IOS Voice Troubleshooting and Monitoring Guide*
VoiceXML Voice Store and Forward	12.3(7)T	Adds VXML capability on Cisco 2691 router and Cisco 3725 and Cisco 3745 routers.	*Cisco VoiceXML Programmer's Guide*
VoIP Alternate Path Fallback SNMP Trap	12.3(14)T	Enhances support for the PSTN Fallback feature by providing the capability to generate SNMP traps when the fallback subsystem redirects or rejects an H.323 VoIP call because a network condition fails to meet a configured threshold.	Trunk-Management Features
VoIP Debug Filtering	12.3(4)T	Allows you to filter and trace voice call debug messages based on selected filtering criteria, reducing the volume of output for more efficient troubleshooting.	*"Filtering Troubleshooting Output" chapter of the Cisco IOS Voice Troubleshooting and Monitoring Guide*
VoIP Internal Error Codes	12.3(4)T	Generates internal error codes (IECs) for gateway-detected errors that cause the gateway to release or refuse a call. IECs enhance troubleshooting for VoIP networks by helping to determine the source and reason for call termination.	"Cisco VoIP Intewrnal Error Codes" chapter in the *Cisco IOS Voice Troubleshooting and Monitoring Guide*

New Voice and Telephony Features in Cisco IOS Releases 12.3T and 12.4 Listed by First Supported Release

Table 2 lists new voice and telephony features in Cisco IOS Release 12.3T and 12.4 by the maintenance release in which each feature was added. The most recent release is listed first.

Table 2 New Voice Features in Cisco IOS Releases 12.3T and 12.4 Listed by First Supported Release

First Supported Release	Feature	Feature Description	Where Documented
12.3(14)T	HTTP Client API for Tcl IVR	Enables Tcl IVR applications to retrieve data from or post data to an external HTTP server. Also introduces a new command-line-interface structure for configuring voice applications and support for additional Tcl 8.3.4 commands.	*HTTP Client API for TcL IVR* *TcL IVR 2.0 Programming Guide*
12.3(14)T	Malicious Caller Identification Invocation Support for Enterprise Networks	Extends support for MCID service in the PSTN to the Cisco 2801.	*Malicious Caller Identification Invocation Support for Enterprise Networks*
12.3(14)T	MCID for Cisco IOS Voice Gateways	Extends support for the Malicious Call Identification (MCID) supplementary service to the Cisco 2801. This feature enables Cisco CallManager to identify the source of malicious calls.	"Configuring MCID for Cisco IOS Voice Gateways" chapter in the *Cisco CallManager and Cisco IOS Interoperability Guide*.
12.3(14)T	Media and Signaling Authentication and Encryption Feature for Cisco IOS MGCP Gateways	Delivers media and signaling authentication and encryption on the Cisco 2600XM series, Cisco 2691, Cisco 3660 series, Cisco 3700 series, and Cisco VG224. This feature enables secure gateway-to-gateway and IP-phone-to-gateway calls, and interoperates with Cisco 7970 IP phones.	*Media and Signaling Authentication and Encryption Feature for Cisco IOS MGCP Gateways*

Table 2 New Voice Features in Cisco IOS Releases 12.3T and 12.4 Listed by First Supported Release (continued)

First Supported Release	Feature	Feature Description	Where Documented
12.3(14)T	MLPP for Analog and BRI Endpoints on Cisco IOS Voice Gateways	Provides the capability for Cisco IOS voice gateways to present analog and basic rate interface (BRI) phones to be controlled by Cisco CallManager as though they were Cisco IP phones, enabling the following: • Line-side support for the Multilevel Precedence and Preemption (MLPP) feature • Cisco CallManager registration of analog and Basic Rate Interface (BRI) endpoints • Cisco CallManager endpoint auto configuration support • Modem pass-through support • Cisco Survivable Remote Site Telephony (SRST) support	*MLPP for Analog and BRI Endpoints on Cisco IOS Voice Gateways*
12.3(14)T	Second-Generation 1- and 2-Port T1/E1 Multiflex Trunk Voice/WAN Interface Cards	Enables T1/E1 multiflex voice/WAN interface cards to support enhanced voice and data applications in Cisco multiservice routers. Provides the following: flexible T1 and E1 support; drop-and-insert multiplexing capability on all versions; support for a dedicated hardware echo-cancellation module; and, on 2-port cards, capability for each port to be clocked from an independent clock source.	*Second-Generation 1- and 2-Port T1/E1 Multiflex Trunk Voice/WAN Interface Cards* "Configuring Hardware Echo Cancellation" chapter in the *Voice Port Configuration Guide*.

Table 2 New Voice Features in Cisco IOS Releases 12.3T and 12.4 Listed by First Supported Release (continued)

First Supported Release	Feature	Feature Description	Where Documented
12.3(14)T	Secure SRST	Secure Cisco IP phones that are located at remote sites and that are attached to gateway routers can communicate securely using the WAN with Cisco CallManager. But if the WAN link or Cisco CallManager goes down, all communication through the remote phones becomes nonsecure. To overcome this situation, gateway routers can now function in secure SRST mode, which activates when the WAN link or Cisco CallManager goes down. When the WAN link or Cisco CallManager is restored, Cisco CallManager resumes secure call-handling capabilities. Secure SRST provides new SRST security features such as authentication, integrity, and media encryption. Authentication provides assurance to one party that another party is whom it claims to be. Integrity provides assurance that the given data has not been altered between the entities. Encryption implies confidentiality; that is, that no one can read the data except the intended recipient. These security features allow privacy for SRST voice calls and protect against voice security violations and identity theft.	"Setting Up Secure SRST" chapter in the *Cisco IOS Survivable Remote Site Telephony Version 3.3 System Administrator Guide*.
12.3(14)T	T.38 Fax Relay on the Cisco Catalyst 6000 and Cisco 7600 Communication Media Module	Provides support for T.38 fax relay on the Cisco Catalyst 6000 and Cisco 7600 CMM.	"Configuring T.38 Fax Relay" chapter in the *Cisco Fax Services over IP Application Guide*.
12.3(14)T	T.38 Fax Statistics	Enables access servers with NextPort digital signal processors to gather detailed statistics about T.38 fax-relay calls. Statistics can be compiled into detailed call records for diagnostic and billing purposes.	*RADIUS VSA Voice Implementation Guide*

Table 2 New Voice Features in Cisco IOS Releases 12.3T and 12.4 Listed by First Supported Release (continued)

First Supported Release	Feature	Feature Description	Where Documented
12.3(14)T	VoIP Alternate Path Fallback SNMP Trap	Enhances support for the PSTN Fallback feature by providing the capability to generate SNMP traps when the fallback subsystem redirects or rejects an H.323 VoIP call because a network condition fails to meet a configured threshold.	Trunk-Management Features
12.3(11)T	Cisco CallManager Express 3.2	Cisco CallManager Express 3.2 adds a number of key telephony features including support for 240 phones, transcoding, and RFC 2833 DTMF support.	Cisco CallManager Express 3.2 index
12.3(11)T	Call Failure Recovery (Rotary) on IPIPGW	This enhancement eliminates the need for identical codec capabilities for all dial peers in the rotary group, and allows the IP-to-IP gateway to restart the codec negotiation process with the originating endpoint based on the codec capabilities of the next dial peer in the rotary group.	*Cisco Multiservice IP-to-IP Gateway Application Guide*
12.3(11)T	Call Pickup ringing extension	The SRST **pickup** command has been introduced to enable the PickUp soft key on all Cisco IP phones, allowing an external direct inward dialing (DID) call coming into one extension to be picked up from another extension during SRST.	The **pickup** command in the *Cisco Survivable Remote Site Telephony Version 3.2 Command Reference.*
12.3(11)T	COR list increase from 10 to 20	The maximum number of COR lists has been increased to 20.	The **cor** command in the *Cisco Survivable Remote Site Telephony Version 3.2 Command Reference.*
12.3(11)T	External Music on Hold Source	Cisco SRST has been enhanced with the **moh-live** command. The **moh-live** command provides live feed MOH streams from an audio device connected to an E&M or FXO port to Cisco IP phones in SRST mode. Music from a live feed is from a fixed source and is continuously fed into the MOH playout buffer instead of being read from a flash file. Live feed MOH can also be multicast to Cisco IP phones.	*Integrating Cisco CallManager and Cisco SRST to Use Cisco SRST As a Multicast MOH Resource*

Table 2 New Voice Features in Cisco IOS Releases 12.3T and 12.4 Listed by First Supported Release (continued)

First Supported Release	Feature	Feature Description	Where Documented
12.3(11)T	Gatekeeper Prefix Selection for Hair-Pinned Calls	Allows the H.323 Gatekeeper to terminate/hairpin calls from a TDM/PSTN endpoint back through the same originating gateway based on priority/zone prefix values.	*Cisco Multiservice IP-to-IP Gateway Application Guide*
12.3(11)T	High-Density Analog (FXO/FXS/DID) and Digital (BRI) Extension Module for Voice/Fax (EVM-HD)	The High-Density Analog (FXO/FXS/DID) and Digital (BRI) Extension Module for Voice/Fax (EVM-HD) feature delivers a higher density integrated analog/digital voice interface. The EVM-HD-8FXS/DID baseboard provides eight FXS and DID ports. This network module accesses digital signal processor (DSP) modules on the motherboard, instead of using onboard DSPs. You can increase the port density by plugging in up to two optional expansion modules in any combination: • EM-HDA-8FXS—8-port voice/fax expansion module • EM-HDA-3FXS/4FXO—7-port voice/fax expansion module • EM-HDA-6FXO—6-port voice/fax expansion module • EM-4BRI-NT/TE—4-port ISDN BRI expansion module PVDM2 DSP modules are used in combination with the EVM-HD-8FXS/DID baseboard and its expansion modules. PVDM2 modules are available separately and installed in the DSP module slots located inside the router chassis.	*Cisco IOS Voice Port Configuration Guide*

Table 2 New Voice Features in Cisco IOS Releases 12.3T and 12.4 Listed by First Supported Release (continued)

First Supported Release	Feature	Feature Description	Where Documented
12.3(11)T	Increase alias command from 10 to 50	The SRST **alias** command has been enhanced in the following areas: • The maximum number of **alias** commands used for creating calls to telephone numbers that are unavailable during Cisco CallManager fallback has increased to 50. • The **cfw** keyword was added, providing call forward no-answer/busy capabilities. The *alternate-number* argument can be used in multiple **alias** commands.	The **alias** command in the *Cisco Survivable Remote Site Telephony Version 3.2 Command Reference*
12.3(11)T	Increase phones supported from 240 to 720 on Access Router	The Cisco 3845 now supports 720 phones and up to 960 ephone-dns or virtual ports.	*Cisco Survivable Remote Site Telephony (SRST) 3.2 Specifications for Cisco IOS Software Release 12.3(11)T*
12.3(11)T	Japanese Katakana Localization	Japanese Katakana is now supported when using the SRST **user-locale** command. The new **JP** keyword is available to Cisco SRST systems running under Cisco CallManager V4.0.	The **user-locale** command in the *Cisco Survivable Remote Site Telephony Version 3.2 Command Reference*.
12.3(11)T	MCID for Cisco IOS Voice Gateways	Supports the Malicious Call Identification (MCID) supplementary service that enables Cisco CallManager 4.0 to identify the source of malicious calls.	"Configuring MCID for Cisco IOS Voice Gateways" chapter in the *Cisco CallManager and Cisco IOS Interoperability Guide*.
12.3(11)T	MLPP for Cisco IOS Voice Gateways	Supports Multilevel Precedence and Preemption (MLPP) service, allowing authorized users to preempt lower priority voice calls using Cisco CallManager 4.0.	"Configuring MGCP Gateway Support for Cisco CallManager" chapter in the *Cisco CallManager and Cisco IOS Interoperability Guide*.

Table 2 New Voice Features in Cisco IOS Releases 12.3T and 12.4 Listed by First Supported Release (continued)

First Supported Release	Feature	Feature Description	Where Documented
12.3(11)T	NextPort Voice Tuning and Background Noise Statistics with NextPort Dual-Filter G.168 Echo Cancellation	This feature allows you to dynamically configure voice services on the NextPort-based platforms: Cisco AS5350, Cisco AS5400, Cisco AS5400HPX, and Cisco AS5850. This feature also provides improved voice quality and statistics reporting and adds dual-filter G.168 echo canceller capability in NextPort SPE firmware (SPEware) version 10.2.2 and later with Cisco IOS Release 12.3(11)T.	"NextPort Voice Tuning and Background Noise Statistics with NextPort Dual-Filter G.168 Echo Cancellation" chapter in the *Cisco IOS Voice Port Configuration*
12.3(11)T	Option to disable H225 TCP Timer from phone to gateway to maintain calls in progress during WAN outage	To preserve existing H.323 calls on the branch in the event of an outage, disable the H.225 keepalive timer by entering the **no h225 timeout keepalive** command. This feature is supported for SRST on Cisco IOS Releases 12.3(7)T1 and higher.	"Overview of Cisco IOS SRST" chapter in the *Cisco IOS SRST Version 3.2 System Administrator Guide*
12.3(11)T	Out-of-Band to In-Band DTMF Relay for Cisco IOS Voice Gateways	RFC 2833 capability enabling DTMF relay communication between SIP devices and nonSIP endpoints using Cisco CallManager 4.0	"Configuring Enhanced Conferencing and Transcoding for Voice Gateway Routers" chapter in the *Cisco CallManager and Cisco IOS Interoperability Guide*.
12.3(11)T	Overlap Signaling Processing on H.323 Terminating Gateways	In an overlap signaling scenario, The called number in the SETUP message does not contain enough digits to match the incoming dial peer for the dial peer to select the right application. With this change, the H.323 layer determines if a partial match is detected, appends the called number with the needed digits. The new called number is checked to see if it matches any of the incoming dial-peers. If either full match or no match is returned, the call will proceeds with SETUP procedure.	*Cisco Multiservice IP-to-IP Gateway Application Guide*

Table 2 New Voice Features in Cisco IOS Releases 12.3T and 12.4 Listed by First Supported Release (continued)

First Supported Release	Feature	Feature Description	Where Documented
12.3(11)T	QSIG Supplementary Features for Cisco IOS Voice Gateways	Supports Q Signaling (QSIG) over PRI backhaul interfaces on MGCP gateways to Cisco CallManager version 4.0.	"Configuring MGCP PRI Backhaul and T1 CAS Support for Cisco CallManager" chapter in the *Cisco CallManager and Cisco IOS Interoperability Guide*.
12.3(11)T	RCF 2833 DTMF Support from SCCP Devices to Cisco Unity Express	Cisco Skinny Client Control Protocol (SCCP) phones, such as those used with Cisco SRST systems, provide only out-of-band DTMF digit indications. To enable SCCP phones to send digit information to remote SIP-based IVR and voice-mail applications, Cisco SRST 3.2 and later versions provide conversion from the out-of-band SCCP digit indication to the SIP standard for DTMF relay, which is RFC 2833. You select this method in the SIP VoIP dial peer using the **dtmf-relay rtp-nte** command.	*Cisco IOS Survivable Remote Site Telephony Version 3.2 System Administrator Guide*
12.3(11)T	Support Translation Profiles (CME and SRST)	Cisco SRST 3.2 supports translation profiles. Translation profiles allow you to group translation rules together and to associate translation rules with the following: • Called numbers • Calling numbers • Redirected called numbers	"Setting Up Call Handling" chapter in the *Cisco IOS Survivable Remote Site Telephony Version 3.2 System Administrator Guide*. Also, the **translation-profile** command in the *Cisco IOS Survivable Remote Site Telephony Version 3.2 System Administrator Guide*
12.3(8)T	Enhanced Conferencing and Transcoding for Voice Gateway Routers	Provides conferencing, transcoding, and MTP services for Cisco voice gateways in a Cisco CallManager network. Uses DSP resources on the NM-HDV2 and NM-HD high-density digital voice/fax network modules.	"Configuring Enhanced Conferencing and Transcoding for Voice Gateway Routers" chapter in the *Cisco CallManager and Cisco IOS Interoperability Guide*
12.3(8)T	ETSI Call Transfer	Provides support for European Telecommunications Standards Institute (ETSI) explicit call transfer functionality on Cisco IOS gateways.	"Configuring Telephony Call-Redirect Features" chapter in the *Cisco IOS TcL IVR and VoiceXML Application Guide*.

Table 2 New Voice Features in Cisco IOS Releases 12.3T and 12.4 Listed by First Supported Release (continued)

First Supported Release	Feature	Feature Description	Where Documented
12.3(8)T	MGCP Configuration Control for Setting Fax Rate	Establishes the maximum fax transmission rate for MGCP T.38 sessions. The MGCP fax rate is set to the highest possible transmission speed allowed by the voice codec by default.	"Configuring T.38 Fax Relay" section in the *Cisco Fax Services over IP Application Guide*
12.3(8)T	MGCP Line Control Signaling Package support	Supports the transport of line supervision signals in the media stream using RFC 2833 event packets in PacketCable GR303-switched IP systems, using the modified **mgcp package-capability package** command. When the **lcs-package** keyword is used, the named telephony events (NTEs) associated with the LCS package are enabled automatically.	"Basic MGCP Configuration" chapter of the *Cisco IOS MGCP and Related Protocols Configuration Guide*
12.3(8)T	SIP Audible Message-Waiting Indicator for FXS Phones	Enables an FXS port on a voice gateway to receive audible MWI in a SIP-enabled network. The FXS port on a voice gateway is an RJ-11 connector that allows connections to basic telephone service equipment.	"Configuring SIP MWI Support" chapter of the *Cisco IOS SIP Configuration Guide*
12.3(8)T	SIP: Cisco IOS Gateway HTTP Digest Authentication and Registration	Implements authentication using the digest access on the client side of a common SIP stack. The Cisco IOS Session Initiation Protocol (SIP) gateway responds to authentication challenges from an authenticating server, proxy server, or user agent server (UAS). This feature also maintains parity between the Cisco gateways, proxy servers, and SIP phones that already support authentication.	"Configuring SIP AAA Features" chapter in the *Cisco IOS SIP Configuration Guide*
12.3(8)T	SIP: Cisco IOS Gateway Reason Header and Buffered Calling Name Completion	Implements support for the following functions: Reason Header and Buffered Calling Name Completion. The Cisco IOS SIP gateway reason header support provided on Cisco IOS gateways is defined by RFC 3326.	"Configuring SIP Message, Timer, and Response Features" chapter of the *Cisco IOS SIP Configuration Guide*

Table 2 New Voice Features in Cisco IOS Releases 12.3T and 12.4 Listed by First Supported Release (continued)

First Supported Release	Feature	Feature Description	Where Documented
12.3(8)T	Voice Application HTTP Client Cookie Support	Implements HTTP cookie support for Cisco IOS VoiceXML applications.	Configuring Basic Functionality for Tcl IVR and VoiceXML Applications" chapter in the *Cisco IOS Tcl IVR and VoiceXML Application Guide.pdf*
12.3(8)T	Voice Application Monitoring and Troubleshooting Enhancements	Enables detailed monitoring of voice application instances and call legs using event logs and statistics. Records for terminated application instances and call legs are saved in history to assist in fault isolation. This comprehensive management information helps you diagnose problems in the network and identify the causes.	"Monitoring and Troubleshooting Voice Applications" chapter in the *Cisco IOS Tcl IVR and VoiceXML Application Guide.pdf*

Table 2 New Voice Features in Cisco IOS Releases 12.3T and 12.4 Listed by First Supported Release (continued)

First Supported Release	Feature	Feature Description	Where Documented
12.3(7)T	AIM-CUE	Provides support for Cisco Unity Express voice mail and auto attendant for either Cisco CallManager or Cisco CallManager Express IP Communications networks. The AIM-CUE is supported on the Cisco 2600XM series, Cisco 2691, and Cisco 3700 series voice gateway routers on an AIM form factor.	Installing Advanced Integration Modules in Cisco 2600 Series, Cisco 3600 Series, and Cisco 3700 Series Routers
12.3(7)T	Call Routing Enhancements to the H.323 Gatekeeper and GKTMP (GK API)	Improves routing flexibility in customer networks where an external route server is used to select potential endpoints for call completion. • Nonblocking GKTMP (GK API): Timing changes associated with recovery processing when socket errors occur. • Separate DNIS for alternate endpoints: It is now possible to associate a unique DNIS with each alternate endpoint. • Support for “z” tag in RESPONSE xRQ: Enhances the responses a route server can provide to the H.323 gatekeeper to allow greater flexibility for combinations of gateway endpoints and gatekeepers.	*Cisco Gatekeeper External Interface Reference*

Table 2 New Voice Features in Cisco IOS Releases 12.3T and 12.4 Listed by First Supported Release (continued)

First Supported Release	Feature	Feature Description	Where Documented
12.3(7)T	Cisco CallManager Express 3.1	Introduces enhancements to allow interoperability with a mix of platforms in a WAN across an H.323 network. The mix of platforms can include earlier versions of Cisco CME, Cisco CallManager, Cisco BTS Softswitch (BTS), and Cisco PSTN Gateway (PGW), and also other Cisco IOS voice gateways. The enhancements include support of H.450.12 standards for H.450 capabilities exchange with remote H.323 endpoints, automatic detection of Cisco CallManager endpoints, H.323-to H.323 hairpin call routing, and H.323-to-H.323 routing to H.450 tandem gateways. Other enhancements introduced in this feature include Call Park, CFwdAll Soft Key Restriction Control, and enhancements to automatic line selection and ephone hunt groups. Language display localization and directory search are supported on Cisco IP Phone 7905 and Cisco IP Phone 7912. Call progress tone localization is supported on Cisco IP Phone 7902, Cisco IP Phone 7905, and Cisco IP Phone 7912. The Cisco Wireless IP Phone 7920 and Cisco IP Phone Conference Station 7936 are fully supported in the Cisco IOS CLI and Cisco CME GUI.	Cisco CallManager Express 3.1
12.3(7)T	Cisco VG224 24-Port Analog Phone Gateway	Enables a hybrid of using VoIP Technology (AVVID based architectures with Cisco CallManager as call control) with TDM analog endpoints (analog phones, fax machines, analog modems). Supported on Cisco CallManager Release 3.2 or a later release.	Voice Gateways

Table 2 New Voice Features in Cisco IOS Releases 12.3T and 12.4 Listed by First Supported Release (continued)

First Supported Release	Feature	Feature Description	Where Documented
12.3(7)T	Enhanced ITU-T G.168 Echo Cancellation	Includes platforms using the TI C5510 DSP.	"Configuring Echo Cancellation" chapter in the "*Cisco IOS Voic Port Configuration Guide*
12.3(7)T	Interoperability Enhancements to the Cisco Multiservice IP-IP Gateway	Introduces the following enhancements: (1) IP-to-IP gateway is now interoperable with the Cisco ATA-188 and Microsoft NetMeeting. (2) Tcl IVR 2.0: Full support for programmable/scripted applications with VoIP endpoints; prepaid applications are one example. (3) IP-to-IP gateway image consolidation: Combines TDM-to-IP voice gateway and IP-to-IP gateway feature sets in a single Cisco IOS image. The combined functions can run concurrently on the same hardware platform. This reduces the cost of deployment for the IP-to-IP gateway in enterprise networks.	*Cisco Multiservice IP-to-IP Gateway Application Guide*
12.3(7)T	IP Communications High-Density Digital Voice/Fax Network Module	Supports high-density digital voice and low-density analog voice connectivity along with data and integrated access connectivity. The network modules offer built-in T1/E1 VIC/VWIC) slot for FXS, FXO, E&M, Centralized Automatic Message Accounting (CAMA), direct inward dialing (DID), BRI, or E1 and T1 cards, up to a maximum of four T1/E1 ports. Supports up to 32 HDLC channels with an aggregate capacity of 2.048 Mbps.	"Configuring Digital Voice Ports" chapter in the *Cisco IOS Voice Port Configuration Guide*

Table 2 New Voice Features in Cisco IOS Releases 12.3T and 12.4 Listed by First Supported Release (continued)

First Supported Release	Feature	Feature Description	Where Documented
12.3(7)T	Lossless Compression R1, ATM Cell Switching and BITS Clocking	Introduces a new compression technique in DSP firmware and add enhancements to Cisco IOS software that include cell switching on ATM segmentation and reassembly (SAR), and the use of an external BITS clocking source. These features enable Cisco multiservice routers to be used to transparently groom and compress traffic in a wireless service provider network and enable a service provider to optimize the bandwidth used to backhaul the traffic from a cell site to the mobile central office for more efficient use of existing T1 and E1 lines.	Lossless Compression R1, ATM Cell Switching and BITS Clocking
12.3(7)T	Signal ISDN B-Channel ID to Enable Application Control of Voice Gateway Trunks	Enables call management applications to identify specific ISDN bearer (B) channels used during a voice gateway call for billing purposes. With the identification of the B channel, SIP and H.323 gateways can enable port-specific features such as voice recording and call transfer.	"Configuring SIP ISDN Support Features" chapter of the *Cisco IOS SIP Configuration Guide* and "Configuring H.323 Gateway" chapter of the *Cisco IOS H.323 Configuration Guide*
12.3(7)T	Survivable Remote Site Telephony 3.1	Provides Cisco CallManager with fallback support for Cisco IP phones attached to a Cisco router on a local network. The Cisco Wireless IP Phone 7920 and Cisco IP Phone Conference Station 7936 are fully supported.	*Cisco Survivable Remote Site Telephony (SRST) Version 3.1*
12.3(7)T	T.37 Fax Status Notification Enhancement in an MTA Environment	Provides the ability to delegate control of fax operations directly to a mail transfer agent (MTA) by configuring a T.37 fax off-ramp gateway to deliver all fax delivery errors to the fax mail originator in Simple Mail Transfer Protocol (SMTP) delivery status notification (DSN) messages with descriptive error codes.	"Configuring T.37 Store-and-Forward Fax" chapter in the *Cisco Fax Services over IP Application Guide*
12.3(7)T	VoiceXML Voice Store and Forward	Adds VXML capability on Cisco 2691 router and Cisco 3725 and Cisco 3745 routers.	*Cisco VoiceXML Programmer's Guide*

Table 2 New Voice Features in Cisco IOS Releases 12.3T and 12.4 Listed by First Supported Release (continued)

First Supported Release	Feature	Feature Description	Where Documented
12.3(4)T	Accounting Server Connectivity Failure and Recovery Detection	Provides the scriptable option to reject new calls entering the VoIP network and tear down all existing calls on detecting connectivity failure to the method list that is associated with RADIUS-based accounting servers.	*Cisco IOS Voice Troubleshooting and Monitoring Guide*
12.3(4)T	Call Statistics on Voice-Enabled Gateways	Enables the collection of voice call statistics based on user-configured time ranges. The statistics that can be collected are from the following functional areas: • RADIUS accounting • Cisco IOS generated internal error codes (IECs) • Gateway port (interface) statistics	*"Voice Performance Statistics on Cisco Gateways"* chapter in the *Cisco IOS Voice Troubleshooting and Monitoring Guide*
12.3(4)T	Cisco CallManager Express 3.0	Enables Cisco routers to deliver key system or hybrid PBX functionality for enterprise branch offices or small businesses. Cisco CME is ideal for customers who have data connectivity requirements and also have a need for a telephony solution in the same office. Whether offered through a service provider's managed services offering or purchased directly by a corporation, Cisco CME offers most of the core telephony features required in the small office, in addition to many advanced features not available with traditional telephony solutions. Being able to deliver IP telephony and data routing using a single converged solution allows customers to optimize their operations and maintenance costs, resulting in a very cost-effective solution that meets office needs. **Note** Prior to Version 3.0, Cisco CallManager Express was known as Cisco IOS Telephony Services (Cisco ITS).	Cisco CallManager Express 3.0

Table 2 New Voice Features in Cisco IOS Releases 12.3T and 12.4 Listed by First Supported Release (continued)

First Supported Release	Feature	Feature Description	Where Documented
12.3(4)T	Customizable Tone Download to Cisco IOS MGCP Gateways from Cisco Call Manager	Implements the downloading of region-specific tones and the associated frequencies, amplitudes, and cadences using XML-based configuration files during gateway registration. The feature also introduces the ability to play out tones that contain up to four frequencies.	"Configuring Tone Download to MGCP Gateway" chapter in the *Cisco CallManager and Cisco IOS Interoperability Guide*
12.3(4)T	Enhanced ITU-T G.168 Echo Cancellation	Extends the existing ITU-T G.168 Echo Cancellation functionality by adding support for medium complexity CODECs.	"Configuring Echo Cancellation" chapter in the "*Cisco IOS Voic Port Configuration Guide*
12.3(4)T	Inactive Call Detection	Detects inactive (silent) H.323 or SIP call-legs on Cisco IOS software-based gateways, and reports this situation to the Tcl IVR 2.0 application (which can disconnect the call). Enables the Cisco IOS software to not automatically disconnect detected inactive calls. Inactivity is defined as no RTP/RTCP packets for a configurable length of time.	"Troubleshooting Voice Applications" chapter of the *Cisco IOS Voice Troubleshooting and Monitoring Guide*
12.3(4)T	IP Communications Voice/Fax Network Module	Provides the ability to directly connect the PSTN and legacy telephony equipment to Cisco 2600XM series, Cisco 2691, Cisco 3600 series, and Cisco 3700 series multiservice routers, enabling applications such as IP telephony, toll bypass, and full gateway integration. The VWICs supported by the new network modules include 2- and 4-port FXS; 2- and 4-port FXO; 2-port DID, E&M, and BRI (S/T); and 1- and 2-port T1/E1.	*Cisco IOS Voice Port Configuration Guide*
12.3(4)T	ISDN Calling Name Display	Provides calling name display to SIP customers on calls that originate on ISDN networks.	"Configuring SIP ISDN Support Features" chapter of the *Cisco IOS SIP Configuration Guide*

Table 2 New Voice Features in Cisco IOS Releases 12.3T and 12.4 Listed by First Supported Release (continued)

First Supported Release	Feature	Feature Description	Where Documented
12.3(4)T	MGCP Gateway Support for Cisco CallManager Network Specific Facilities	Provides an enhancement to Cisco CallManager functionality enabling users to configure the NSF ISDN information element of the route pattern. This feature is compatible with Cisco CallManager Version 3.3(2) and later.	"Configuring MGCP PRI Backhaul and T1 CAS Support for Cisco CallManager" chapter in the *Cisco CallManager and Cisco IOS Interoperability Guide*
12.3(4)T	PLAR (Private Line Automatic Ring-down) for Trading Turrets	Re-establishes an H.323 call automatically if one end hangs up or goes on hold and then goes off-hook again. This provides PLAR functionality used in the financial industry for trading Turret operation of PLAR.	*Cisco Hoot and Holler over IP*
12.3(4)T	SIP Debug Output Filtering Support	Provides the ability to filter relevant SIP debugging traces of desired calls.	*Cisco IOS Voice Troubleshooting and Monitoring Guide*
12.3(4)T	SIP Gateway Support Enhancements to the bind Command	Allows configuration of the source IP address of signaling packets or both signaling and media packets using the **bind** command.	"Configuring SIP Gateway Support for the bind Command" chapter of the *Cisco IOS SIP Configuration Guide*
12.3(4)T	SIP Header Support and SUBSCRIBE and NOTIFY for External Triggers	Provides a mechanism for applications to send and receive SIP headers and to send SUBSCRIBE messages and receive NOTIFY events.	"Configuring Additional SIP Application Support" chapter of the *Cisco IOS SIP Configuration Guide*
12.3(4)T	SIP NOTIFY-Based Out-of-Band DTMF Relay Support	Supports SCCP devices through SIP originating and terminating gateway use of Cisco proprietary NOTIFY-based out-of-band DTMF relay, which can also be used by analog phones attached to analog voice ports (FXS) on a router.	"SIP NOTIFY-Based Out-of-Band DTMF Relay Support" chapter of the *Cisco IOS SIP Configuration Guide*
12.3(4)T	SIP Redirect Processing Enhancement	Allows flexibility in the handling of incoming redirect or 3xx class of responses. Redirect responses can now be enabled or disabled through the command-line interface, providing critical functionality for service providers who deploy Cisco gateways.	"Basic SIP Configuration" chapter of the *Cisco IOS SIP Configuration Guide*

Table 2 New Voice Features in Cisco IOS Releases 12.3T and 12.4 Listed by First Supported Release (continued)

First Supported Release	Feature	Feature Description	Where Documented
12.3(4)T	SIP Register Support	Allows SIP gateways to register E.164 numbers to a SIP proxy or registrar on behalf of analog telephone voice ports (FXS), IP phone virtual voice ports (EFXS), and local SCCP phones.	"Basic SIP Configuration" chapter of the *Cisco IOS SIP Configuration Guide*
12.3(4)T	SIP RFC 3261 Enhancements	Provides enhanced SIP functionality on Cisco IOS gateways, as defined by RFCs 3261 and 3311, and support for the following: • Ability to receive and process SIP UPDATE requests • Initial Offer and Answer exchanges • Branch and Sent-by parameters in the Via header • Merged request detection • Loose-routing	"Basic SIP Configuration" chapter of the *Cisco IOS SIP Configuration Guide*
12.3(4)T	SIP Survivable Remote Site Telephony (SRST) Version 3.0	Describes Survivable Remote Site Telephony (SRST) functionality for Session Initiation Protocol (SIP) networks. SIP SRST provides backup to an external SIP proxy server by providing basic registrar and redirect services. These services are used by a SIP IP phone in the event of a WAN connection outage where the SIP phone is unable to communicate with its primary SIP proxy. The SIP SRST device also provides PSTN gateway access for placing and receiving PSTN calls.	SIP Survivable Remote Site Telephony (SRST)

Table 2 New Voice Features in Cisco IOS Releases 12.3T and 12.4 Listed by First Supported Release (continued)

First Supported Release	Feature	Feature Description	Where Documented
12.3(4)T	SRST: Survivable Remote Site Telephony Version 3.0	Provides Cisco CallManager with fallback support for Cisco IP phones attached to a Cisco router on your local network. Cisco SRST enables routers to provide call-handling support for Cisco IP phones when they lose connection to remote primary, secondary, or tertiary Cisco CallManager installations or when the WAN connection is down. Cisco CallManager supports Cisco IP phones at remote sites attached to Cisco multiservice routers across the WAN. Prior to Cisco SRST, when the WAN connection between a router and the Cisco CallManager failed or connectivity with Cisco CallManager was lost for some reason, Cisco IP phones on the network became unusable for the duration of the failure. Cisco SRST overcomes this problem and ensures that the Cisco IP phones offer continuous (although minimal) service by providing call-handling support for Cisco IP phones directly from the Cisco SRST router. The system automatically detects a failure and uses Simple Network Auto Provisioning (SNAP) technology to autoconfigure the branch office router to provide call processing for Cisco IP phones that are registered with the router. When the WAN link or connection to the primary Cisco CallManager is restored, call handling reverts back to the primary Cisco CallManager.	Cisco Survivable Remote Site Telephony (SRST) Version 3.0

Table 2 New Voice Features in Cisco IOS Releases 12.3T and 12.4 Listed by First Supported Release (continued)

First Supported Release	Feature	Feature Description	Where Documented
12.3(4)T	Tcl IVR 2.0 Session Interaction	Allows different instances of Tcl IVR applications (sessions) to communicate with other sessions on the same gateway and for applications to dynamically bridge call legs between different sessions. This enables different callers on the same gateway to be notified of each others' presence and to interact. You can also start a session without an active call leg so that a session can act as an application service for other sessions. This feature is useful for implementing a call-monitoring "server" application that is responsible for monitoring incoming calls and dynamically connecting selected callers.	"Configuring Tcl IVR 2.0 Session Interaction" chapter of the *Cisco IOS Tcl IVR and VoiceXML Applications Guide*
12.3(4)T	Videoconferencing for the Cisco Multiservice IP-to-IP Gateway	Provides enhanced Quality of Service through RSVP synchronization with H.323 signaling protocol and DSCP packet marking.	"Configuring Multiservice IP-to-IP Videoconferencing" chapter in the *Cisco Multiservice IP-to-IP Gateway*
12.3(4)T	Voice DSP Crash Dump Analysis	Allows Cisco IOS voice platforms using TI DSPs the ability to capture the contents of the DSP memory into a dump file if there is a DSP crash.	"Troubleshooting Digital Voice Interfaces to the IP Newtwork" chapter in th *Cisco IOS Voice Troubleshooting and Monitoring Guide*
12.3(4)T	VoIP Debug Filtering	Allows you to filter and trace voice call debug messages based on selected filtering criteria, reducing the volume of output for more efficient troubleshooting.	*"Filtering Troubleshooting Output" chapter of the Cisco IOS Voice Troubleshooting and Monitoring Guide*
12.3(4)T	VoIP Internal Error Codes	Generates internal error codes (IECs) for gateway-detected errors that cause the gateway to release or refuse a call. IECs enhance troubleshooting for VoIP networks by helping to determine the source and reason for call termination.	"Cisco VoIP Intewrnal Error Codes" chapter in the *Cisco IOS Voice Troubleshooting and Monitoring Guide*

Table 2 New Voice Features in Cisco IOS Releases 12.3T and 12.4 Listed by First Supported Release (continued)

First Supported Release	Feature	Feature Description	Where Documented
12.3(2)T	MGCP-Controlled Backhaul of BRI Signaling in Conjunction with Cisco CallManager	Provides MGCP service to remote-office media gateways that connect by means of ISDN BRI trunks to a centralized Cisco CallManager media-gateway controller for the purpose of call processing. D-channel signal information is backhauled to the call manager through a Transmission Control Protocol (TCP) session.	"Configuring MGCP-Controlled Backhaul of BRI Signaling in Conjunction with Cisco CallManager" chapter in the *Cisco CallManager and Cisco IOS Interoperability Guide*.
12.3(2)T	PCR Support for the Cisco Signaling Link Terminal	Adds support for Message Transfer Part Layer 2 (MTP2) PCR on the Cisco signaling link terminal (SLT). Preventive Cyclic Retransmission (PCR) adds basic MTP2 functionality that is used when Signaling System 7 (SS7) signaling links are transmitted over satellite connections between the Cisco SLT and a signal transfer point (STP).	*PCR Support for the Cisco Signaling Link Terminal*
12.3(2)T	Persistent TDM Switched Circuits	Enables the Cisco AS5850 universal gateway to connect one or more DS0s from one or more E1s or T1s to another set of E1s or T1s.	Persistent TDM Switched Circuits
12.3(2)T	V.110 Support for MGCP-Dial	Provides support for V.110 encapsulation for Dial-MGCP Applications.	"Configuring NAS Package for MGCP" chapter of the *Cisco IOS MGCP and Related Protocols Configuration Guide*