Cisco IOS Software Release 12.2(4)XM

Introduction

This product bulletin describes the software features introduced in Cisco IOS® Software Release 12.2(4)XM for the Cisco 1700 Modular Access Router.

New Features

The Cisco IOS 12.2(4)XM introduces SIP and MGCP signaling protocols and supplementary services, call admission control capabilities, T.37 fax relay, modem passthrough, TCL and RTSP enhanced IVR features on the Cisco 1751 modular access router. The support of these features enhances the Cisco 1700 capabilities as a feature-rich voice customer premises equipment (CPE) router with multiple signaling protocol support for small branch offices and small to medium-sized business markets. It also ties it well with other voice gateway offerings from Cisco to help customers build well-engineered, reliable, end-to-end multiservice New World networks.

In this release, G.728 codec is supported on the Cisco 1750, 1751, and Cisco 1760; and IEEE 802.1p is supported on the Cisco 1710, Cisco 1751 and Cisco 1760. The following features are supported on the Cisco 1751; support for the following features on the Cisco 1760 will be available in a later release:

- Call Admission Control (CAC) for voice-over-IP calls
- Diffserv for the Voice Signaling Traffic
- Test Command Language (TCL) Scripting
- Real Time Streaming Protocol (RTSP) Enhanced IVR
- T.37 Store and Forward Fax Relay
- Session Initiation Protocol (SIP) and Supplementary Services
- Media Gateway Control Protocol (MGCP) and Supplementary Services
- Modem Pass-Through over VoIP

IEEE 802.1p Support on the Cisco 1710, Cisco 1751 and Cisco 1760

IEEE 802.1P and 802.1Q are vital components in designing Cisco AVVID networks. Through 802.1Q trunking support, WAN routers can trunk the separate virtual LANs (VLANs) from an Ethernet switch. Ethernet switches use separate VLANs for voice and data. User Priority bits in the 802.1P portion of the 802.1Q standard header provide prioritization in Ethernet switches. Cisco 1751 and Cisco 1760 already support 802.1Q VLAN trunking. In this release, IEEE 802.1P support is introduced on the Cisco 1700 Series, so that Cisco 1700 can be positioned as qualified CPE routers for Cisco AVVID.

G.728 Codec Support

The G.728 codec uses the LD-CELP algorithm to perform speech coding at 16 Kbps. G.728 codec is based on Low Delay-Code Excited Linear Prediction (LD-CELP) speech codecs, which are normally used at low bit rates. It uses backward CELP; this means that filter coefficients are derived from previously reconstructed speech, which is available at both encoder and decoder. The encoder need not buffer a large segment of the input speech; a buffering delay of G.728 is much less than that of most CELP codecs. G.728 is supported on the Cisco 1750, 1751, and 1760 in the Cisco IOS 12.2(4)XM release.

Call Admission Control for Voice over IP calls

Call Admission Control addresses the requirements for basic call admission control, including Resource Unavailable Signaling. This implementation is ideal for service provider H.323 packet telephony networks; toll bypass, international wholesale, and tandem network offload applications. This feature provides the functionality in H.323V2, SIP, SGCP, and MGCP to reject a call if the RSVP reservation fails or local resources unavailable. Local resources refer to CPU utilization, total available memory, and total current active calls. In addition, with the CAC feature, on-net calls can fall back to PSTN if resource is unavailable on the IP network. The Cisco 1751 in this release supports this feature, and support on the Cisco 1760 will be available in the future.

Diffserv for the Voice Signaling Traffic

This feature enables prioritization of voice signaling traffic using differentiated services. The Cisco 1751 in this release supports this feature, and support on the Cisco 1760 will be available in a later release.

TCL Scripting

In Cisco IOS Software Release 12.2(4)XM, TCL scripting is introduced to the Cisco 1751. Test Command Language (TCL) Scripting provides a flexible scripting language capability to modify the Interactive Voice Response (IVR) call flow. Cisco 1751 is now enabled with Cisco IVR 2.0, which is also called scalable IVR.

Real-Time Streaming Protocol (RTSP) enhanced Interactive Voice Response

Running IVR on the VoIP gateway can limit router performance. It can limit, for example, the number of audio files and size of files supported in the flash file system, and the memory required to display and change prompts. The Real-Time Streaming Protocol (RTSP) enhanced IVR helps network administrators avoid these limitations by running the IVR on a server that is external to the gateway.

RTSP is an application-level protocol for control over delivery of data with real-time properties. It provides an extensible framework to enable controlled, on-demand delivery of real-time data, such as audio and video, using TCP or UDP.

In this release, the Cisco 1750 supports RTSP-enhanced IVR, and support on the Cisco 1760 will be available in a later release.

T.37 Store and Forward Fax Relay

In this release, the Cisco 1751 supports store-and-forward fax, single number for subscriber voice mail and fax access, and real-time fax fallback to store-and-forward fax. Support on the Cisco 1760 will be available in a later release. These features enable service providers and enterprise/managed service customers to offer value-added fax services within a unified communication solution. The real time fax relay T.38 has been supported on the Cisco 1750, 1751, and 1760 since Cisco IOS Software Release 12.2(3).

Session Initiation Protocol and Supplementary Services

SIP, defined by the IETF, is a text-based protocol, similar to HTTP and SMTP, for initiating interactive communication sessions between users. Such sessions include voice, video, chat, interactive games, and virtual reality. As an alternative to H.323, SIP is rapidly gaining support in the industry as the next standard protocol for voice communications. The fundamental objective of SIP is to develop a signaling protocol that is much less complex than H.323, while providing equivalent or greater functionality.

In this release, the Cisco 1751 supports the following SIP features:

- Session Initiation Protocol—RFC2543
- DTMF Relay for SIP calls using Named Telephone Events (NTE)
- T.38 Fax Relay over SIP

Session Initiation Protocol (SIP) for VoIP Enhancements

- SIP midcall changes of SDP sessions using an INVITE with changed SDP
- Enhanced SIP stack to support new routing related headers (Also, Contact, Expires, Max-Forwards, Record-Route, Requested-By, Route, Timestamp, Unsupported)
- Support for call hold and call transfer features.
- Support for 180/183/184 Session Progress Messages with inband alerting
- Enhanced PSTN to SIP Event/Cause code mapping
- UDP socket layer enhancements to support connected socket, enabling ICMP network notifications to be used to identify unreachable destinations rather relying only on SIP timers
- Interoperating with Amteva UCS Messaging platform
- IPSec support in SIP security

ISDN Progress Indicator support for SIP using 183 Session Progress

- Enhancements to allow VoIP calls using SIP to provide in-band call treatment such as ring back tones, announcements when inter-working with ISDN and CAS PSTN networks

SIP Diversion Header Implementation for Redirecting Number

- Call Control Diversion header for SIP messaging. Call control redirection is defined as a redirection of a call based on a subscriber service or feature such as call forwarding or call deflection. Call control redirecting information is typically used for unified messaging and voice mail services to identify a message recipient. It also adds the ability to handle 3xx response messages if they are received after 18x responses during a SIP call setup.

SIP Gateway support for Third Party Call Control

- Support for third-party call control by supporting INVITE without SDP body
- Support for FQDN in SDP
- Enhanced codec negotiation
- Enhancements in SDP parser/builder on SIP gateways

SIP User Agent MIB

- RFC2543 (SIP) MIB—CISCO-SIP-UA-MIB; allows configuring SIP timers and counters

SIP Gateway support of RSVP and "tel" URL

- Support interaction with forked requests to support find me/follow me services
- New Headers—Content-Disposition, Supported, Unsupported, RSeq, RSck

- Implement PRACK method
- Support retransmission mechanisms for 1XX responses and PRACK
- Support 183 Session Progress with Content-Disposition header set to QoS
- Support SDP enhancements for QoS attribute
- Support COMET method
- Support synchronization with the Cisco IOS QoS Module
- Account special cases for delayed media and midcall media changes
- Parser support for TEL URLs
- Implement TEL URL generation for outgoing INVITES

SIP Intra-Gateway Hair-Pinning

- Call routing capability, in which an incoming call on a specific gateway is signaled through the IP network and back out on the same gateway
- Can also be a call signaled from a line (such as a phone) to the IP network and back out to a line on the same access gateway

Gateway Support for Bind Command

- Currently both SIP signaling and SIP media path use the same IP address as the source address. This address is always provided by the IP layer and is the "best local address". Bind command will allow the SIP SPI to pickup the address as configured in the CLI by the user for the interface.

RFC2782 Compliance for DNS SRV

- Implement the RFC 2782 (_sip._udp.proxy.Cisco.com) style query mechanism. It also provides for backward compatibility with RFC 2052 (sip.udp.proxy.Cisco.com) by introducing a CLI. The CLI will allow the user to initiate a RFC 2052 style SRV query instead of the default RFC 2782.

Configurable PSTN Cause Code to SIP Response

- The CLI allows users to perform configuration of the cause-mapping table depending on the requirement. Any SIP status code can be mapped to a PSTN cause code or vice versa.

Call Transfer Capabilities Using the Refer Method

- SIP Call Transfer using the Refer Method in accordance with draft-ietf-sip-cc-transfer-04 (Blind Transfer, Attended Transfer)

SIP INVITE Request with Malformed Via Header

- Enables a response to an INVITE, which has a malformed Via field
- The response is sent to the source IP address and default port of 5060
- The response is a "400 response" with a Reason-Phrase of "Malformed Via Field"
- The response for the malformed Via field is counted in CISCO-SIP-UA-MIB by the existing object cSipStatsClientBadRequestOuts

NAT Support for SIP

- Cisco IOS NAT between VoIP solutions based on the SIP protocol

Media Gateway Control Protocol and Supplementary Features

The Media Gateway Control Protocol (MGCP) is a device control protocol developed within the IETF to allow call agents or Softswitches to control media gateways in a packet telephony system. The protocol is currently in version 1.0 as an RFC. In this master/slave model of the protocol the gateways are expected to execute commands sent by the Call Agents. MGCP describes a call control architecture where intelligence of the call control is outside the gateways and handled by external call control elements (call agents). Service Providers prefer this dumb gateway/intelligent central control model.

In this release, the Cisco 1751 supports the following MGCP features:

MGCP 1.0

- Adds support for MGCP protocol v1.0 RFC 2705
- Provides support for the multiple dialects of the MGCP protocol on the Cisco IOS Symphony based voice gateway. Maintains backward compatibility with MGCP 0.1 and SGCP 1.1 and SGCP 1.5.

MGCP based Fax (T.38) and DTMF Relay

MGCP based IETF DTMF Relay and Fax (T.38) Relay Features

MGCP Basic CLASS and Operator Services

Provides xGCP infrastructure for basic CLASS feature support on Cisco IOS gateways. In addition, it provides support for three-way calling and platform enhancements to, business gateways and residential gateways for xGCP.

- Three-way calling on RGW
- Caller ID (Caller ID Type 1) on RGW
- Caller ID with call waiting (Caller ID Type 2) on RGW
- Various tones/rings
- Distinctive call waiting tone
- Off-hook warning tone
- Message waiting indicator tone
- Three-cycle ring cadence for distinctive power ring
- Visual Message Waiting Indicator (VMWI)
- xGCP support for the following codecs: G.711, G.723, G.726, G.729,G.729a, and G.728

MGCP VoIP Call Admission Control

- Local CAC (also called System CAC)—the ability to refuse calls based on the availability of local gateway call processing resources such as CPU utilization, memory, and so on.
- Synchronization with Reservation Protocol (RSVP)—the ability to reserve bandwidth across IP networks using RSVP and report to the call agent the results of the reservation request
- Network Congestion Detection—the ability to refuse calls based on the measured level of congestion on the IP network such as the Use of Response Time Reporter (RTR) to ascertain the congestion on the network and refuse/accept calls based on this

Modem Pass-Through over Voice over IP

- Enables modem pass-through over VoIP based on H.323 with no call agent. Modem pass-through is the transport of modem signals through a packet network using PCM encoded packets

Detailed Information

For more detailed information about features with Cisco IOS Software Release 12.2(4)XM, reference the following document:

Find release Notes for Cisco 1700 Series platforms for Cisco IOS Software Release 12.2(4) XM at: http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122relnt/1700/rn1700xm.htm

Maintenance Support

Maintenance for Cisco IOS 12.2(4)XM features on the Cisco 1700 Series will be available on future 12.2X special releases until the code is incorporated into the fourth maintenance software release of 12.2T.

Life Span

12.2(4)XM will be sold until the fourth maintenance release of Cisco IOS Software Release 12.2T.

Product Support Policy

Cisco IOS Software Release 12.2(4)XM is supported by the Cisco Systems Product Support Policy found at: http://www.cisco.com/warp/customer/437/27.html

Supported Hardware Platforms

Cisco IOS Software Release 12.2(4)XM supports the Cisco 1750, Cisco 1751, and Cisco 1760 modular access routers.

Product Numbers and Descriptions

Product Numbers and Descriptions are given below.

Images are available for downloading from Cisco.com on December 17, 2001.

Platform	Software Product Description	Software Image	Product Code	Min. Flash Memory	Min. DRAM Memory
1750 /1751/1760	Cisco 1700 IOS IP/ADSL/ IPX/AT/IBM/Voice/FW/ IDS Plus IPSec 56	c1700-bk8no3r2sv3y7-mz	S17Q7HVK8-12204XM S17Q7HVK8-12204XM =	32MB	64MB
1751/1760	Cisco 1700 IOS IP/ADSL/ IPX/AT/IBM/VOX/FW/IDS Plus IPSec 56	c1700-bk8no3r2sv8y7-mz	S17Q7V8K8-12204XM S17Q7V8K8-12204XM =	32MB	64MB
1720 /1750 /175/1760	Cisco 1700 IOS IP/ADSL/ IPX/AT/IBM/FW/IDS Plus IPSec 56	c1700-bk8no3r2sy7-mz	S17Q7HK8-12204XM S17Q7HK8-12204XM =	16MB	48MB
1750 /1751/1760	Cisco 1700 IOS IP/ADSL/ IPX/AT/IBM/Voice/FW/ IDS PlusIPSec3DES	c1700-bk9no3r2sv3y7-mz	S17Q7HVK9-12204XM S17Q7HVK9-12204XM =	32MB	64MB
1751/1760	Cisco 1700 IOS IP/ADSL/ IPX/AT/IBM/VOX/FW/IDS Plus IPSec 3DES	c1700-bk9no3r2sv8y7-mz	S17Q7V8K9-12204XM S17Q7V8K9-12204XM =	32MB	64MB
1720 /1750 /1751/1760	Cisco 1700 IOS IP/ADSL/ IPX/AT/IBM/FW/IDS Plus IPSec 3DES	c1700-bk9no3r2sy7-mz	S17Q7HK9-12204XM S17Q7HK9-12204XM =	16MB	48MB
1720 /1750 /1751/1760	Cisco 1700 IOS IP/ADSL/ IPX/AT/IBM Plus	c1700-bnr2sy7-mz	S17Q7P-12204XM S17Q7P-12204XM =	16MB	48MB

Platform	Software Product Description	Software Image	Product Code	Min. Flash Memory	Min. DRAM Memory
1720 /1750 /1751/1760	Cisco 1700 IOS IP/IPX/ AT/IBM	c1700-bnr2y-mz	S17Q-12204XM S17Q-12204XM =	8MB	32MB
1750 /1751/1760	Cisco 1700 IOS IP/ADSL/ Voice/FW/IDS Plus IPSec 56	c1700-k8o3sv3y7-mz	S17C7HVK8-12204XM S17C7HVK8-12204XM =	16MB	48MB
1751/1760	Cisco 1700 IOS IP/ADSL/ VOX/FW/IDS Plus IPSec 56	c1700-k8o3sv8y7-mz	S17C7V8K8-12204XM S17C7V8K8-12204XM =	16MB	64MB
1720 /1750 /1751/1760	Cisco 1700 IOS IP/ADSL/ FW/IDS Plus IPSec 56	c1700-k8o3sy7-mz	S17C7HK8-12204XM S17C7HK8-12204XM =	16MB	48MB
1750 /1751/1760	Cisco 1700 IOS IP/ADSL/ Voice Plus IPSec 56	c1700-k8sv3y7-mz	S17C7VK8-12204XM S17C7VK8-12204XM =	16MB	48MB
1751/1760	Cisco 1700 IOS IP/ADSL/ VOX Plus IPSec 56	c1700-k8sv8y7-mz	S17C7V8P-12204XM S17C7V8P-12204XM=	16MB	64MB
1720 /1750 /1751/1760	Cisco 1700 IOS IP/ADSL Plus IPSec 56	c1700-k8sy7-mz	S17C7K8-12204XM S17C7K8-12204XM =	16MB	48MB
1750 /1751/1760	Cisco 1700 IOS IP/ADSL/ Voice/FW/IDS Plus IPSec 3DES	c1700-k9o3sv3y7-mz	S17C7HVK9-12204XM S17C7HVK9-12204XM =	16MB	48MB
1751/1760	Cisco 1700 IOS IP/ADSL/ VOX/FW/IDS Plus IPSec 3DES	c1700-k9o3sv8y7-mz	S17C7V8K9-12204XM S17C7V8K9-12204XM =	16MB	64MB
1720 /1750 /1751/1760	Cisco 1700 IOS IP/ADSL/ FW/IDS Plus IPSec 3DES	c1700-k9o3sy7-mz	S17C7HK9-12204XM S17C7HK9-12204XM =	16MB	48MB
1750 /1751/1760	Cisco 1700 IOS IP/ADSL/ Voice Plus IPSec 3DES	c1700-k9sv3y7-mz	S17C7VK9-12204XM S17C7VK9-12204XM =	16MB	48MB
1751/1760	Cisco 1700 IOS IP/ADSL/ VOX Plus IPSec 3DES	c1700-k9sv8y7-mz	S17CV8K9-12204XM S17CV8K9-12204XM =	16MB	64MB
1720 /1750 /1751/1760	Cisco 1700 IOS IP/ADSL Plus IPSec 3DES	c1700-k9sy7-mz	S17C7K9-12204XM S17C7K9-12204XM =	16MB	48MB
1750 /1751/1760	Cisco 1700 IOS IP/ADSL/ IPX/Voice/FW/IDS Plus	c1700-no3sv3y7-mz	S17B7HPV-12204XM S17B7HPV-12204XM =	16MB	48MB
1751/1760	Cisco 1700 IOS IP/ADSL/ IPX/VOX/FW/IDS Plus	c1700-no3sv8y7-mz	S17B7HPV8-12204XM S17B7HPV8-12204XM =	16MB	64MB
1720 /1750 /1751/1760	Cisco 1700 IOS IP/ADSL/ IPX/FW/IDS Plus	c1700-no3sy7-mz	S17B7HP-12204XM\ S17B7HP-12204XM =	16MB	48MB
1720 /1750 /1751/1760	Cisco 1700 IOS IP/IPX	c1700-ny-mz	S17B-12204XM S17B-12204XM =	8MB	32MB
1750 /175/1760	Cisco 1700 IOS IP/ADSL/ Voice/FW/IDS Plus	c1700-o3sv3y7-mz	S17C7HV-12204XM S17C7HV-12204XM =	16MB	48MB
1751/1760	Cisco 1700 IOS IP/ADSL/ VOX/FW/IDS Plus	c1700-o3sv8y7-mz	S17C7HV8-12204XM S17C7HV8-12204XM =	16MB	64MB
1720 /1750 /1751/1760	Cisco 1700 IOS IP/FW/ IDS	c1700-o3y-mz	S17CH-12204XM S17CH-12204XM =	8MB	24MB

Platform	Software Product Description	Software Image	Product Code	Min. Flash Memory	Min. DRAM Memory
1750 /1751/1760	Cisco 1700 IOS IP/Voice Plus	c1700-sv3y-mz	S17CVP-12204XM S17CVP-12204XM =	16MB	48MB
1750 /1751/1760	Cisco 1700 IOS IP/ADSL/ Voice Plus	c1700-sv3y7-mz	S17C7VP-12204XM S17C7VP-12204XM=	16MB	48MB
1750 /1751/1760	Cisco 1700 IOS IP/VOX Plus	c1700-sv8y-mz	S17CV8P-12204XM S17CV8P-12204XM =	16MB	48MB
1751/1760	Cisco 1700 IOS IP/ADSL/ VOX Plus	c1700-sv8y7-mz	S17C7V8P-12204XM S17C7V8P-12204XM =	16MB	64MB
1720 /1750 /1751 /1760	Cisco 1700 IOS IP/ADSL Plus	c1700-sy7-mz	S17C7P-12204XM S17C7P-12204XM =	8MB	48MB
1720 /1750 /1751 / 1760	Cisco 1700 IOS IP	c1700-y-mz	S17C-12204XM S17C-12204XM =	8MB	32MB
1720 /1750 /1751/1760	Cisco 1700 IOS IP/ADSL	c1700-y7-mz	S17C7-12204XM S17C7-12204XM =	8MB	32MB
1710	Cisco 1710 IOS IP/IPX/ AT/IBM/FW/IDS Plus IPSec 3DES	c1710-bk9no3r2sy-mz	S1710HK9-12204XM S1710HK9-12204XM=	16MB	48MB
1710	Cisco 1710 IOS IP/FW/ IDS Plus IPSec 3DES	c1710-k9o3sy-mz	S171CHKZ-12204XM S171CHKZ-12204XM=	8MB	32MB

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