

G.Clear, GSMFR, and G.726 Codecs and Modem and Fax Passthrough

Feature History

Release	Modification
12.2(11)T	These features were introduced on the Cisco AS5350, Cisco AS5400, Cisco AS5400HPX, and Cisco AS5850 universal gateways. G. Clear was introduced only on the Cisco AS5300 universal access server.

This document describes new codec support for Clear Channel (G.Clear), GSM Full Rate (GSMFR), and G.726 (16K, 24K, and 32K). This document also describes modem and fax passthrough for the Cisco AS5400 and Cisco AS5850 universal gateways only. This document includes the following sections:

- [Feature Overview, page 1](#)
- [Supported Platforms, page 8](#)
- [Supported Standards, MIBs, and RFCs, page 9](#)
- [Prerequisites, page 9](#)
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Feature Overview

Features that are available on Cisco AS5300 universal access servers are now available on Cisco AS5350, Cisco AS5400, Cisco AS5400HPX, and Cisco AS5850 universal gateways. Also, the G.Clear codec is supported on Cisco AS5300 universal access servers.

The voice codec features that are described and configured in this document are as follows:

- G.Clear including Media Gateway Control Protocol (MGCP) and session initiation protocol (SIP)
- GSMFR

- Support for dynamic payload types
- G.726

The Modem and Fax Passthrough feature is also supported.

Codec

The term codec stands for coder-decoder. A codec is a particular method of transforming analog voice into a digital bit stream (and vice versa) and also refers to the type of compression used. Several different codecs have been developed to perform these functions, and each one is known by the number of the International Telecommunication Union Telecommunication Standardization Sector (ITU-T) standard in which it is defined. For example, two common codecs are the G.711 and the G.729 codecs.

Codecs use different algorithms to encode analog voice into digital bit streams and have different bit rates, frame sizes, and coding delays associated with them. Codecs also differ in the amount of perceived voice quality they achieve. Specialized hardware and software in the digital signal processors (DSPs) perform codec transformation and compression functions, and different DSPs may offer different selections of codecs.

Select the same type of codec at both ends of the call. For instance, if a call was coded with a G.729 codec, it must be decoded with a G.729 codec. Codec choice is configured on dial peers.

Table 1 lists the H.323, SIP, and MGCP codecs that are supported for voice.

Table 1 Voice Codec/Signaling Support Matrix

Codec	H.323	SIP	MGCP
g711ulaw	Yes	Yes	Yes
g711alaw	Yes	Yes	Yes
g729r8 ¹	Yes	Yes	Yes
g729br8 ¹	Yes	Yes	Yes
g723ar53	Yes	Yes	Yes
g723ar63	Yes	Yes	Yes
g723r53	Yes	Yes	Yes
g723r63	Yes	Yes	Yes
gsmfr	Yes	Yes	No
g726r16 ²	Yes	Yes	Yes
g726r24 ²	Yes	Yes	Yes
g726r32	Yes	Yes	Yes
clear-channel ²	Yes	Yes	Yes

1. Annex A is used in the Cisco platforms that are supported in this software release.

2. For dynamic payload types.

For more information, refer to the “Configuring Dial Plans, Dial Peers, and Digit Manipulation” chapter and the “Configuring Voice Ports” chapter in the *Cisco IOS Voice, Video, and Fax Configuration Guide*, Release 12.2.

Clear Channel (G.Clear) Codec

G.Clear guarantees bit integrity when transferring a DS-0 through a gateway server, supports the transporting of nonvoice circuit data sessions through a Voice over IP (VoIP) network, and enables the VoIP networks to transport ISDN and switched 56 circuit-switched data calls. With the availability of G.Clear, ISDN data calls that do not require bonding can be supported.

In a transit application, because it is possible to have a mix of voice and data calls, not supporting G.Clear limits the solution to voice-only calls. The end-user application is in charge of handling packet loss and error recovery. This packet loss management precludes the use of clear channel with some applications unless the IP network is carefully engineered.

In an MGCP environment, the voice gateway backhauls the public switched telephony network (PSTN) signaling channel to the call agent. The call agent examines the bearer capability and determines when a G.Clear call should be established.

**Note**

G.Clear codecs cannot be configured on a T1 channel associated signaling (CAS) trunk for incoming traffic. T1 CAS trunks use least significant bit-robbing for signaling, which causes the data to be incorrect and re-sent from high level protocols. Traffic on an incoming E1 R2 trunk can be configured.

If you configure the G.Clear codec under a VoIP dial peer, you cannot transmit and receive voice traffic with that dial peer on the voice gateway router. The G.Clear codec does not accommodate voice services like dial tone or ring-back tone.

GSM Full Rate Codec

The GSMFR codec was introduced in 1987. The GSMFR speech coder has a frame size of 20 ms and operates at a bit rate of 13 kbps. GSMFR is an RPE-LTP (Regular Pulse Excited – Linear Predictive) coder.

In order to write VoiceXML scripts that can function as the user interface for a simple voice-mail system, the network must support GSMFR codecs. The network messaging must be capable of recording a voice message and depositing the message to an external server for later retrieval.

This codec supports the Cisco infrastructure and application partner components required for service providers to deploy unified messaging applications.

Adaptive Differential PCM Voice Codec—G.726

Adaptive differential pulse code modulation (ADPCM) voice codec operates at bit rates of 16, 24, and 32 kbps. ADPCM provides the following functionality:

- Voice mail recording and playback, which is a requirement for Internet voice mail.
- Voice transport for cellular, wireless, and cable markets.
- High voice quality voice transport at 32 kbps.

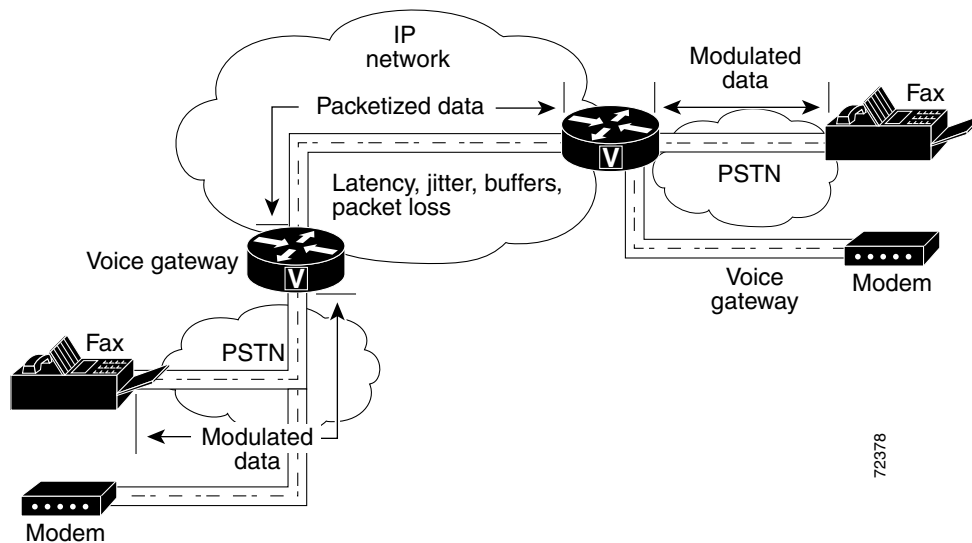
Modem and Fax Passthrough

When service providers and aggregators are implementing VoIP, they sometimes cannot separate fax or data traffic from voice traffic. These carriers that aggregate voice traffic over VoIP infrastructures require service offerings to carry fax and data as easily as voice.

On detection of the modem answer tone, the gateways switch into modem passthrough mode. With modem passthrough, the modem traffic is carried between the two gateways in real-time transport protocol (RTP) packets, using an uncompressed or lightly compressed voice codec—G.711 u-law, G.711 a-law, or Voice Band Data (VBD). Packet redundancy may be used to mitigate the effects of packet loss in the IP network. Even so, modem passthrough remains susceptible to packet loss, jitter, and latency in the IP network.

Figure 1 illustrates how modem and fax passthrough works in an IP network.

Figure 1 Modem and Fax Passthrough



The Modem Passthrough feature is also known as Voice Band Data (VBD) by the International Telecommunication Union (ITU). VBD refers to the transport of modem signals over a voice channel through a packet network with an encoding appropriate for modem signals. The minimum set of coders for VBD mode is G.711 ulaw and alaw.

For VBD mode of operation, the path between the originating and answering gateway remains in a voice configuration. The modem signals are encoded using an appropriate speech codec suitable for the task, and samples are transported across a packet network. Currently G.711 is supported.

Some system requirements for the use of VBD follow:

- Use a voice codec that passes voice band modulated signals with minimal distortion.
- Have end-to-end constant latency.
- Disable Voice Activity Detection (VAD) and Comfort Noise Generation (CNG) during the data transfer phase.
- Disable any DC removal filters that may be integral with the speech encoder used.
- Be capable of tone detection, including mid-call dual tone multifrequency (DTMF), as well insertion of tones, announcements, and voice prompts.

To use VBD, you should consider the appropriate application of:

- Echo cancellers on a VBD channel
- RFC 2198

Modem and Fax Passthrough Switchover

When the gateways detect a data modem, both the originating gateway and the terminating gateway switch to modem passthrough mode. This switchover includes the following:

- Switching to the G.711 codec
- Disabling the high pass filter
- Disabling Voice Activity Detection (VAD)
- Using special jitter buffer management algorithms
- On detection of modem phase reversal tone, disabling the echo canceler

At the end of the modem or fax call, the voice ports revert to the previous configuration and the DSPs switch back to the original voice codec.

Controlled Redundancy Using RTP (RFC 2198)

Packet loss is a persistent issue in voice applications. The disruption of speech, which is characteristic of packet loss, can be somewhat resolved with controlled redundancy and the RTP. Controlled redundancy reconstructs missing information at the receiver end from the redundant data that arrives in the transmitted packets.

Some of the requirements for a controlled redundancy are as follows:

- The packets have to carry a primary encoding and one redundant encoding.
- Because the use of variable size encodings is desirable, each encoded block in the packet must have a length indicator.
- The RTP header provides a time-stamp field that corresponds to the time of creation of the encoded data and redundant blocks of data correspond to different time intervals than the primary data. So each block of redundant encoding requires its own time stamp.

You can enable redundancy so that the modem and fax passthrough switchover causes the gateway to transmit redundant packets and redundancy can be enabled in one or both of the gateways. When only one gateway is configured, the other gateway receives the packets correctly, but does not produce redundant packets.



Note

The current Cisco implementation of RFC 2198 reflects a redundant encoding of 1X or 1 repeat of the original packet. This means that any loss scenario in which two or more consecutive packets are dropped would cause a loss of data translated into a retrain, Failure To Train (FTT), or call drop, etc. in modem and fax passthrough.

See the [“Restrictions” section on page 6](#) for more information.

Clock Slip Buffer Management

When the gateways detect a data modem, both the originating gateway and the terminating gateway switch from dynamic and adaptive buffers to static de-jitter buffers. The use of a static de-jitter buffer is required for modem passthrough because the adaptation process in a dynamic de-jitter buffer causes a retrain on the modem connection. When the modem call is concluded, the voice ports revert to dynamic jitter buffers.

In addition, the modem passthrough data management algorithm is designed to handle and compensate for clocking differences in the PSTN between the originating and terminating gateways. This additional clock-slip monitoring prevents issues that show up in long duration modem calls.

Benefits

G.Clear Codec

VoIP networks can transport ISDN and switched 56 circuit-switched data calls.

Modem and Fax Passthrough

- Detection of modem and fax tones is possible up to and including V.90 enabling the proper switchover from voice to modem or fax passthrough.
- VoIP service providers who aggregate voice traffic can pass modem and fax signals over VoIP networks.
- RFC 2198 payload redundancy improves reliability in networks with low packet loss.
- Fax passthrough can be used to interoperate with VoIP endpoints that do not support T.38 fax relay.

Restrictions

G. Clear Codec

The G.Clear support with the 64k unrestricted bandwidth is available only with MGCP signaling because the requests are handled between the call agent (CA) and the MGCP gateway. In order to use the G.Clear support with the H.323 or SIP protocols, the bearer capability in the Q.931 message must be the same as a regular voice call. The voice gateway negotiates the G.Clear codec based on the voice gateway configuration.

GSMFR Codec

See [Table 1 on page 2](#) for specific information.

G.726 Codec

See [Table 1 on page 2](#) for specific information.

Modem and Fax Passthrough

Modem and fax passthrough are required to support interoperability with Cisco and third-party devices that do not support modem or fax relay. The Cisco AS5400 and Cisco AS5850 universal gateways have the following limitations:

- Performance—To handle packet loss, redundant encoding (1X or one repeat of the original packet) is required and the amount of data transferred in each packet is doubled. The doubling of packets imposes a limitation on the total number of ports that can run modem passthrough at one time.

Starting from port 0 and grouping the next 36 consecutive ports, defined as a subsystem—there are 3 subsystems per dfc108 card on the Cisco AS5400 (3 times 36 for a total of 108 ports) and 9 subsystems on the Cisco AS5850 tetryl card (9 times 36 for a total of 324 ports)—the limitations are as follows:

- 36 10- or 20-ms modem passthrough sessions with no redundancy
- 20-ms modem passthrough sessions with redundancy
- 10-ms modem passthrough sessions with redundancy

Examples of modem passthrough sessions mixed with a high load voice session type are as follows:

- 10 10-ms modem passthrough sessions and 20 G711, no VAD sessions
- 12 10-ms modem passthrough sessions and 16 G.711, no VAD sessions

You can simply calculate that two voice sessions with no VAD equate to one modem passthrough session with redundancy. With 10-ms modem passthrough, each subsystem has a modem passthrough limit of 20 sessions. With 20-ms modem passthrough, each subsystem has a modem passthrough limit of 30 sessions. The same limitations would apply to all subsequent subsystems.

The Cisco AS5400 and Cisco AS5850 have the capability to transmit 20-ms packets and receive 10-ms packets that significantly improves performance over what can currently be handled with 10-ms in both directions. Currently, other Cisco universal gateway implementations may have an outgoing packet size limitation that imposes the use of 10-ms packets, as opposed to 20-ms (optimal setting). This restriction limits the number of ports that can run modem passthrough to 20 per subsystem (10-ms connections only).


Note

Modem and fax passthrough do not support the switch from G.Clear to G.711. If modem passthrough and the G.Clear codec are configured, the gateway will not be able to detect the modem answer tone or fax tone.

When using modem passthrough, it is recommended that the user (either client or NAS) disable V.8bis if feasible. With V.8bis enabled, connections may take an additional 4 to 10 seconds. It should be noted that if V.8bis is disabled, modems attempting K56 flex will connect at V.34 rates. Because of the V.8bis handshake occurring before codec switchover, this will be true for modem passthrough during most flex attempts regardless of whether V.8bis is enabled or disabled.

Related Documents

G.Clear, GSMFR, and G.726 Codecs

- *VoiceXML Application Guide*, Cisco IOS Release 12.2(2)XB

H.323 Dynamic Payload Types for DTMF Relay

- *Dual Tone Multifrequency Relay for SIP Calls Using Named Telephone Events*, Cisco IOS Release 12.2(8)T
- *H.323 DTMF Relay Using Named Telephone Events*, Cisco IOS Release 12.2(11)T

MGCP

- *MGCP-based DTMF Relay and T.38 Fax Relay*, Cisco IOS Release 12.2(2)XB

SIP

- *Enhanced Codec Support for SIP Using Dynamic Payloads*, Cisco IOS Release 12.2(7)XO
- *SIP T.38 Fax Relay*, Cisco IOS Release 12.2(8)T
- *Session Initiation Protocol (SIP) for VoIP*, Cisco IOS Release 12.2(8)T

Modem and Fax Passthrough

- *V.92 Modem on Hold for Cisco AS5300 Universal Access Servers*, Cisco IOS Release 12.2(2)XB
- *V.92 Quick Connect for Cisco AS5300 Universal Access Servers*, Cisco IOS Release 12.2(2)XB
- *V.92 Modem on Hold for Cisco AS5350 and Cisco AS5400 Universal Access Servers*, Cisco IOS Release 12.2(2)XB
- *V.92 Quick Connect for Cisco AS5350 and Cisco AS5400 Universal Access Servers*, Cisco IOS Release 12.2(2)XB
- *Cisco IOS Voice, Video, and Fax Configuration Guide*, Cisco IOS Release 12.2
- *Cisco IOS Voice, Video, and Fax Command Reference*, Cisco IOS Release 12.2
- *Cisco IOS Configuration Fundamentals Configuration Guide*, Cisco IOS Release 12.2
- *Cisco IOS Debug Command Reference*, Release 12.2

Supported Platforms

- Cisco AS5300 (G.Clear only)
- Cisco AS5350
- Cisco AS5400
- Cisco AS5400HPX
- Cisco AS5850

Determining Platform Support Through Cisco Feature Navigator

Cisco IOS software is packaged in feature sets that support specific platforms. To get updated information regarding platform support for this feature, access Cisco Feature Navigator. Cisco Feature Navigator dynamically updates the list of supported platforms as new platform support is added for the feature.

Cisco Feature Navigator is a web-based tool that enables you to determine which Cisco IOS software images support a specific set of features and which features are supported in a specific Cisco IOS image. You can search by feature or release. Under the release section, you can compare releases side by side to display both the features unique to each software release and the features in common.

To access Cisco Feature Navigator, you must have an account on Cisco.com. If you have forgotten or lost your account information, send a blank e-mail to cco-locksmith@cisco.com. An automatic check will verify that your e-mail address is registered with Cisco.com. If the check is successful, account details with a new random password will be e-mailed to you. Qualified users can establish an account on Cisco.com by following the directions at <http://www.cisco.com/register>.

Cisco Feature Navigator is updated regularly when major Cisco IOS software releases and technology releases occur. For the most current information, go to the Cisco Feature Navigator home page at the following URL:

<http://www.cisco.com/go/fn>

Availability of Cisco IOS Software Images

Platform support for particular Cisco IOS software releases is dependent on the availability of the software images for those platforms. Software images for some platforms may be deferred, delayed, or changed without prior notice. For updated information about platform support and availability of software images for each Cisco IOS software release, refer to the online release notes or, if supported, Cisco Feature Navigator.

Supported Standards, MIBs, and RFCs

Standards

No new or modified standards are supported by this feature.

MIBs

The new MIB supported by this feature:

- CISCO-VOICE-COMMON-DIAL-CONTROL-MIB

To obtain lists of supported MIBs by platform and Cisco IOS release, and to download MIB modules, go to the Cisco MIB website on Cisco.com at the following URL:

<http://www.cisco.com/public/sw-center/netmgmt/cmtk/mibs.shtml>

RFCs

- RFC 2198, *Payload for Redundant Audio Data*

Prerequisites

The following sections describe the prerequisites for each feature:

- [G.Clear, GSMFR, and G.726 Prerequisites, page 9](#)
- [Modem and Fax Passthrough Prerequisites, page 9](#)

G.Clear, GSMFR, and G.726 Prerequisites

For MGCP, if 64 Kb unrestricted is specified for the bearer capabilities signaled on the incoming message, the CA can influence the codec negotiation by way of the ingress or egress Create Connection (CRCX).

Modem and Fax Passthrough Prerequisites

Before configuring your universal gateway for modem passthrough, perform the following tasks:

- Establish a working VoIP-enabled network.
- Verify network suitability to pass modem traffic by determining the packet loss threshold to pass modem traffic. Packet loss and latency are two impairments that can have a dramatic effect on modem passthrough performance.

With respect to packet loss, up to 1 percent random packet loss causes little degradation of Carrier Sensitive Route (CSR) with either V.90 or V.34 as long as redundancy (1X only) is enabled. When two consecutive packets of loss occur, a retrain occurs that degrades throughput during the time that the retrain occurs. If this loss occurs during connection, it may cause a failure to connect or a lower connect rate to occur.

With less than 1 percent random packet loss, the effect is minimal. If redundancy is disabled, CSR drops significantly (perhaps as much as 40 to 80 percent), which is expected. In theory you may lose every other packet (fixed loss) and still maintain connectivity, if redundancy is enabled but networks do not have fixed loss. Certainly higher random packet loss, particularly multiple consecutive packets, causes problems with respect to negotiation of speed and retrain and call connectivity.

Delay causes problems, particularly with V.90, if total round trip delay on the end-to-end connection exceeds a certain threshold (around 400 to 425 ms). It is recommended that networks that have known delay above 60 ms (one way on the IP leg) use V.34 with a fixed connect rate of 28,800 kbps to minimize the effects of delay on CSR.

- Verify the key characteristics of the network by using the response time reporter (RTR) feature of Cisco IOS software to check the packet loss, delay, and jitter. See [“Restrictions” section on page 6](#) for more information. For more information on RTR, refer to the *Cisco IOS Configuration Fundamentals Configuration Guide*.

If you are using a gateway that does not support T.38 fax relay or if you choose to use fax passthrough instead of T.38 fax relay, follow these guidelines:

- Create a VoIP dial peer on both the originating and terminating gateway.
- Disable T.38 fax relay on these VoIP dial peers on both the originating and terminating gateway (H.323 and SIP) using the **fax rate disable** command. Use the **mgcp fax t38 inhibit** command for MGCP fax passthrough.
- Associate the destination phone number (to which the fax machine is attached) using the **incoming-called-number** command on the VoIP dial peer.

The default fax mode on all Cisco AS5400 universal gateways is T.38 fax relay. For best performance, verify that you have T.38 fax relay on both the originating and the terminating gateways. If two Cisco IOS gateways have differing transports, they do not negotiate.

A fax type (relay or passthrough) topology for what action the gateway would take depending on the gateway configuration is shown in [Table 2](#). The topology is: sending fax machine to originating gateway; originating gateway to the IP; IP to the terminating gateway; terminating gateway to the receiving fax machine.

Table 2 Supported Fax Types

Originating Gateway	Terminating Gateway	Fax Type
T.38	T.38	T.38
T.38	T.38 + modem passthrough	T.38
T.38 + modem passthrough	T.38	T.38
T.38 + modem passthrough	T.38 + modem passthrough	T.38
T.38 + modem passthrough + fax rate disable	T.38 + modem passthrough	Fax passthrough
T.38 + modem passthrough	T.38 + modem passthrough + fax rate disable	Fax passthrough

Table 2 *Supported Fax Types (continued)*

Originating Gateway	Terminating Gateway	Fax Type
T.38 + modem passthrough + fax rate disable	T.38 + modem passthrough + fax rate disable	Fax passthrough
modem passthrough + fax rate disable	T.38 + modem passthrough + fax rate disable	Fax passthrough
T.38 + modem passthrough + fax rate disable	modem passthrough + fax rate disable	Fax passthrough
modem passthrough + fax rate disable	modem passthrough + fax rate disable	Fax passthrough

The commands for the above configurations are as follows:

- T.38—**fax protocol t38 ls-redundancy 0 hs-redundancy 0**
- Modem passthrough—**modem passthrough nse codec g711ulaw**
- Fax rate disable—**fax rate disable** (enter this command in voip dial-peer configuration mode)

Configuration Tasks

See the following sections for configuration tasks for the voice codecs and modem passthrough features. Each task in the list is identified as either required or optional.

- [Configuring G.Clear Codecs for H.323 Support, page 12](#) (required)
- [Configuring G.Clear Codecs for SIP Support, page 12](#) (required)
- [Configuring Codecs for MGCP Support, page 13](#) (required)
- [Configuring Backward Compatibility for H.323 and SIP, page 13](#) (required)
- [Configuring Backward Compatibility for MGCP, page 14](#) (required)
- [Configuring Modem and Fax Passthrough, page 15](#) (required)

Configuring G.Clear Codecs for H.323 Support

To configure the codecs for H.323 support, enter the following commands beginning in global configuration mode:

	Command	Purpose
Step 1	Router(config)# dial-peer voice <i>number</i> voip	Enters dial-peer configuration mode and specifies the VoIP number.
Step 2	Router(config-dial-peer)# incoming called-number <i>number</i>	Specifies the incoming called number for G.Clear.
Step 3	Router(config-dial-peer)# destination pattern [+] <i>string</i> [T]	Specifies either the prefix or the full E.164 telephone number (depending on your dial plan) to be used for a dial peer.
Step 4	Router(config-dial-peer)# session target { ipv4 : <i>destination-address</i> }	Specifies a network-specific address for a specified VoIP dial peer for H.323. The <i>destination-address</i> argument is the IP address of the dial peer.
Step 5	Router(config-dial-peer)# codec clear-channel	Specifies the clear-channel codec. Note G.Clear codecs cannot be configured on a T1 CAS trunk for incoming traffic. Traffic on an incoming E1 R2 trunk can be configured.
Step 6	Router(config-dial-peer)# no vad	Disables voice activity detection (VAD). Note VAD is enabled by default.

Configuring G.Clear Codecs for SIP Support

To configure the codecs for SIP support, enter the following commands beginning in global configuration mode:

	Command	Purpose
Step 1	Router(config)# dial-peer voice <i>number</i> voip	Enters dial-peer configuration mode and specifies the VoIP number.
Step 2	Router(config-dial-peer)# incoming called-number <i>number</i>	Specifies the incoming called number for G.Clear.
Step 3	Router(config-dial-peer)# destination pattern [+] <i>string</i> [T]	Specifies either the prefix or the full E.164 telephone number (depending on your dial plan) to be used for a dial peer.
Step 4	Router(config-dial-peer)# session target { ipv4 : <i>destination-address</i> }	Specifies a network-specific address for a specified VoIP dial peer. The <i>destination-address</i> argument is the IP address of the dial peer.
Step 5	Router(config-dial-peer)# session protocol sipv2	Specifies a session protocol for calls between the local and remote servers using the packet network. The sipv2 keyword configures the VoIP dial peer to use IETF SIP.

	Command	Purpose
Step 6	Router(config-dial-peer)# codec clear-channel	Specifies the clear-channel codec. Note G.Clear codecs cannot be configured on a T1 CAS trunk for incoming traffic. Traffic on an incoming E1 R2 trunk can be configured.
Step 7	Router(config-dial-peer)# no vad	Disables voice activity detection (VAD). Note VAD is enabled by default.

Configuring Codecs for MGCP Support

To configure the codecs for MGCP support, enter the following command in global configuration mode:

Command	Purpose
Router(config)# mgcp codec clear-channel	Specifies the codec for G.Clear for MGCP support. The clear-channel keyword specifies Clear Channel at 64,000 bps, with medium and high complexity. Note G.Clear codecs cannot be configured on a T1 CAS trunk for incoming traffic. Traffic on an incoming E1 R2 trunk can be configured.

Configuring Backward Compatibility for H.323 and SIP

To configure backward compatibility for H.323 and SIP codecs to all previous releases, enter the following command in dial-peer configuration mode:

Command	Purpose
Router(config-dial-peer)# rtp payload-type { cisco-cas-payload cisco-clear-channel cisco-codec-fax-ack cisco-codec-fax-ind cisco-fax-relay cisco-pcm-switch-over-ulaw cisco-pcm-switch-over-alaw cisco-rtp-dtmf-relay nse nse }	Specifies the codecs for H.323 and SIP. The keywords are as follows: <ul style="list-style-type: none"> • cisco-cas-payload—Specifies the payload type for CAS RTP. • cisco-clear-channel—Specifies the payload type for Clear Channel. This keyword supports the static payload type. • cisco-codec-fax-ack—Specifies the payload type for fax acknowledgment. • cisco-codec-fax-ind—Specifies the payload type for fax indication.

Command	Purpose
	<ul style="list-style-type: none"> • cisco-fax-relay—Specifies the payload type for fax relay. • cisco-pcm-switch-over-ulaw—Specifies the payload type for upspeed to G.711 ulaw. • cisco-pcm-switch-over-alaw—Specifies the payload type for upspeed to G.711 alaw. • cisco-rtp-dtmf-relay—Specifies the payload type for DTMF relay. • nse—Specifies the payload type for Named Signaling Event (NSE). • nte—Specifies the payload type for Names Telephone Event (NTE).

Configuring Backward Compatibility for MGCP

To configure backward compatibility for MGCP to all previous releases, enter the following command in global configuration mode:

Command	Purpose
<pre>Router(config)# mgcp rtp payload-type {clear-channel cisco-pcm-switch-over-ulaw cisco-pcm-switch-over-alaw g726r16 g726r24}</pre>	<p>Specifies the codecs for MGCP. The keywords are as follows:</p> <ul style="list-style-type: none"> • clear-channel—Specifies the payload type for Clear Channel. This keyword supports the static payload type. • cisco-pcm-switch-over-ulaw—Specifies the payload type for upspeed to G.711 ulaw. • cisco-pcm-switch-over-alaw—Specifies the payload type for upspeed to G.711 alaw. • g726r16—Specifies the payload type for G.726 for 16K. • g726r24—Specifies the payload type for G.726 for 24K.

Verifying the MGCP Configuration

Use the **show mgcp** command to verify the MGCP configuration.

```
Router# show mgcp
```

```
MGCP Admin State ACTIVE, Oper State ACTIVE - Cause Code NONE
MGCP call-agent: none Initial protocol service is MGCP 0.1
MGCP block-newcalls DISABLED
MGCP send SGCP RSIP: forced/restart/graceful/disconnected DISABLED
MGCP quarantine mode process/step
MGCP quarantine of persistent events is ENABLED
MGCP dtmf-relay for VoIP disabled for all codec types
MGCP dtmf-relay for VoAAL2 disabled for all codec types
MGCP voip modem passthrough mode: NSE, codec: g711alaw, redundancy: DISABLED,
MGCP voaal2 modem passthrough disabled
MGCP voip modem relay: Disabled.
MGCP TSE payload: 100
MGCP T.38 Named Signalling Event (NSE) response timer: 200
```

```

MGCP Network (IP/AAL2) Continuity Test timer: 200
MGCP 'RTP stream loss' timer disabled
MGCP request timeout 500
MGCP maximum exponential request timeout 4000
MGCP gateway port: 2427, MGCP maximum waiting delay 3000
MGCP restart delay 0, MGCP vad ENABLED
MGCP rtrcac DISABLED
MGCP system resource check DISABLED
MGCP xpc-codec: DISABLED, MGCP persistent hookflash: DISABLED
MGCP persistent offhook: ENABLED, MGCP persistent onhook: DISABLED
MGCP persistent offhook: ENABLED, MGCP persistent onhook: DISABLED
MGCP piggyback msg DISABLED, MGCP endpoint offset DISABLED
MGCP simple-sdp DISABLED
MGCP undotted-notation DISABLED
MGCP codec type g729r8, MGCP packetization period 10
MGCP JB threshold lwm 30, MGCP JB threshold hwm 150
MGCP LAT threshold lwm 150, MGCP LAT threshold hwm 300
MGCP PL threshold lwm 1000, MGCP PL threshold hwm 10000
MGCP CL threshold lwm 1000, MGCP CL threshold hwm 10000
MGCP playout mode is adaptive 60, 4, 200 in msec
MGCP media (RTP) dscp: ef, MGCP signaling dscp: af31
MGCP default package: ms-package
MGCP supported packages: gm-package dtmf-package mf-package trunk-package
  rtp-package nas-package as-package script-package ms-package
  dt-package mo-package res-package mt-package
MGCP Digit Map matching order: shortest match
SGCP Digit Map matching order: always left-to-right
MGCP VoAAL2 ignore-lco-codec DISABLED
MGCP T.38 Fax is DISABLED
MGCP T.38 Fax ECM is DISABLED
MGCP T.38 Fax NSF Override is DISABLED
MGCP T.38 Fax Low Speed Redundancy: 0MGCP T.38 Fax High Speed Redundancy: 0
MGCP Upspeed payload type for G711ulaw: 0, G711alaw: 8
MGCP Dynamic payload type for G.726-16K codec
GCP Dynamic payload type for G.726-24K codec
MGCP Dynamic payload type for G.Clear codec

```

**Note**

Refer to the *Cisco IOS Voice, Video, and Fax Command Reference*, Release 12.2, for significant fields descriptions.

Configuring Modem and Fax Passthrough

By default, modem and fax passthrough over VoIP capability and redundancy are disabled. Redundancy can be enabled in one or both of the gateways. When only a single gateway is configured for redundancy, the other gateway receives the packets correctly but does not produce redundant packets.

Modem and fax passthrough can be configured for a specific dial peer or for a system dial peer. If modem and fax passthrough are configured for both a specific dial peer and the system dial peer, the specific dial peer configuration takes precedence over the system configuration.

Consequently, when a call matches a particular dial peer, the universal gateway first applies the configuration on the dial peer. Then, if a specific dial peer is not configured, the universal gateway will use the system configuration.

**Note**

For modem and fax passthrough to operate correctly, you must configure both the originating gateway and the terminating gateway. If you configure only one of the gateways in a pair, the modem or fax call will not connect.

Recommended Load Balance for Modem and Fax Passthrough with Voice Codecs

When redundant encoding (1X or one repeat of the original packet) is used to control packet loss, the amount of data transferred in each packet doubles. Doubling the packet payload length imposes a limitation on the total number of DSP ports that can run modem or fax passthrough at one time on a specific subsystem.

Table 3 shows the number of subsystems (a group of contiguous DSP ports) listed by the DSP feature card and a specific universal gateway.

Table 3 Allowable Subsystems in Cisco AS5000

AS5000 Series	DSP Feature Card	Subsystems Per DSP Feature Card
AS5350	AS535-DFC-60NP	2
AS5350	AS535-DFC-108NP	3
AS5400, AS5400HPX	AS54-DFC-60NP	2
AS5400, AS5400HPX	AS54-DFC-108NP	3
AS5850	AS58-1CT3/216U	6
AS5850	AS58-324UPC-CC	9

The AS535-DFC-60NP and AS54-DFC-60NP feature cards have 30 contiguous DSP ports in each subsystem. The remaining DSP feature cards have 36 contiguous DSP ports in each subsystem. Starting with port 0 and grouping the next 35 consecutive ports (36 ports total), creates one subsystem. For example, the total number of DSP ports per AS535-DFC-108NP = $36 * 3 = 108$ DSP ports.

Load balance settings apply to one subsystem. To calculate the total number of modem or fax passthrough sessions per gateway, use this calculation: sessions per subsystem * subsystems per DSP feature card * DSP feature cards.



Note

There are no modem or fax passthrough limitations when redundancy is disabled. Modem and fax passthrough limits only apply when redundancy is enabled.



Note

It is recommended that Cisco AS5000 universal gateways must be configured to transmit and receive 20-ms packets (optimal setting) or 10-ms packets. The packet setting can be symmetric, i.e. transmit and receive using the same packetization rate, or asymmetric, in which the transmit and receive packetization rates are different.

Some Cisco universal gateways transmit fax passthrough (with redundancy enabled) using 10-ms packets only. Although the Cisco AS5300 can only transmit 10-ms packets, the Cisco AS5300 has the ability to receive 20-ms packets with Cisco IOS Release 12.2(11)T.

Modem Passthrough

Table 4 and Table 5 have the allowable number of modem passthrough and voice sessions listed by payload size. The voice codec is given also. The tables may be used as a reference for load balancing modem passthrough calls when voice calls are active. It should be noted that if less than 15 percent of the overall call load is modem passthrough, or if less than a full call load is placed on the gateway, load balancing may not be required.

The default settings per subsystem are:

- 10-ms packets: 18 modem passthrough sessions
- 20-ms packets: 30 modem passthrough sessions

These default settings are based on the default codec configuration of G.729 with VAD enabled.

The recommended configuration for Cisco AS5000 series is 20 ms. A hybrid or mixed 20-ms to 10-ms gateway connection may provide a 10 to 15 percent gain in the number of sessions per subsystem improving modem passthrough performance compared to 10-ms packets in both directions.

Table 4 *Modem Passthrough Configuration of 10 ms to 10 ms*

Payload Size OGW	Payload Size TGW	Number of Modem Passthrough Sessions	Number of Voice Sessions	Voice Codec	VAD
10 ms	10 ms	20	0	None	-
10 ms	10 ms	17	19	GSMFR	Off
10 ms	10 ms	17	19	G.723r63	Off
10 ms	10 ms	17	19	G.723r53	Off
10 ms	10 ms	14	22	G.729	Off
10 ms	10 ms	14	22	G.726r16	Off
10 ms	10 ms	13	23	G.726r32	Off
10 ms	10 ms	6	30	G.711u	Off
10 ms	10 ms	18	18	G.729abr8	On
10 ms	10 ms	18	18	G.726r32	On
10 ms	10 ms	14	22	G.711u	On

Table 5 Modem Passthrough Configurations of 20 ms to 20 ms

Payload Size OGW	Payload Size TGW	Number of Modem Passthrough Sessions	Number of Voice Sessions	Voice Codec	VAD
20 ms	20 ms	30	0	None	-
20 ms	20 ms	28	8	G.711u	On
20 ms	20 ms	23	13	G.711u	Off
20 ms	20 ms	27	9	G.726r32	Off
20 ms	20 ms	27	9	G.726r24	Off
20 ms	20 ms	28	8	G.726r16	Off
20 ms	20 ms	29	7	G729r8	Off
20 ms	20 ms	29	7	GSM FR	Off

Fax Passthrough

Table 6 through Table 9 are load-balance references for fax passthrough with and without Error Correction Mode (ECM). If less than a full call load is placed on the gateway subsystems, load balancing may not be required. The recommended configuration for Cisco AS5000 universal gateways is 20 ms.

Table 6 Fax Passthrough Configurations of 10 ms to 10 ms without ECM

Payload Size OGW	Payload Size TGW	Number of Fax Passthrough Sessions	Number of Voice Sessions	Voice Codec	VAD
10 ms	10 ms	21	0	None	-
10 ms	10 ms	15	21	GSMFR	Off
10 ms	10 ms	16	20	G.723r63	Off
10 ms	10 ms	16	20	G.723r53	Off
10 ms	10 ms	14	22	G.729	Off
10 ms	10 ms	14	22	G.726r16	Off
10 ms	10 ms	11	25	G.726r32	Off
10 ms	10 ms	4	32	G.711u	Off
10 ms	10 ms	16	20	G.729ar8	On
10 ms	10 ms	14	22	G.726r32	On
10 ms	10 ms	12	24	G.711u	On

Table 7 Fax Passthrough Configurations of 10 ms to 10 ms with ECM

Payload Size OGW	Payload Size TGW	Number of Fax Passthrough Sessions	Number of Voice Sessions	Voice Codec	VAD
10 ms	10 ms	22	0	None	-
10 ms	10 ms	16	20	GSMFR	Off
10 ms	10 ms	17	19	G.723r63	Off
10 ms	10 ms	17	19	G.723r53	Off
10 ms	10 ms	15	21	G.729	Off
10 ms	10 ms	15	21	G.726r16	Off
10 ms	10 ms	14	22	G.726r32	Off
10 ms	10 ms	3	33	G.711u	Off
10 ms	10 ms	19	17	G.729ar8	On
10 ms	10 ms	17	19	G.726r32	On
10 ms	10 ms	16	20	G.711u	On

Table 8 Fax Passthrough Configuration of 20 ms to 20 ms without ECM

Payload Size OGW	Payload Size TGW	Number of Fax Passthrough Sessions	Number of Voice Sessions	Voice Codec	VAD
20 ms	20 ms	24	0	None	-
20 ms	20 ms	15	21	G.711u	On
20 ms	20 ms	8	28	G.711u	Off
20 ms	20 ms	14	22	G.726r32	Off
20 ms	20 ms	22	14	G.729ar8	On
20 ms	20 ms	20	16	G729r8	Off
20 ms	20 ms	22	14	GSM FR	Off

Table 9 Fax Passthrough Configurations of 20 ms to 20 ms with ECM

Payload Size OGW	Payload Size TGW	Number of Fax Passthrough Sessions	Number of Voice Sessions	Voice Codec	VAD
20 ms	20 ms	25	0	None	-
20 ms	20 ms	21	15	G.711u	On
20 ms	20 ms	8	28	G.711u	Off
20 ms	20 ms	17	19	G.726r32	Off
20 ms	20 ms	23	13	G.729ar8	On
20 ms	20 ms	20	16	G729r8	Off
20 ms	20 ms	22	14	GSM FR	Off

The sections that describe the modem and fax passthrough configuration tasks are as follows:

- [Configuring Modem and Fax Passthrough for H.323 and SIP Support \(Dial Peer\), page 20](#) (required)
- [Configuring Modem and Fax Passthrough for H.323 and SIP Support \(Global\), page 21](#) (required)
- [Configuring Modem and Fax Passthrough for MGCP Support, page 21](#) (required)

Configuring Modem and Fax Passthrough for H.323 and SIP Support (Dial Peer)

To configure modem and fax passthrough for H.323 and SIP support on a specific dial peer, use the following commands in dial-peer configuration mode:

	Command	Purpose
Step 1	Router(config-dial-peer)# modem passthrough nse codec {g711alaw g711ulaw} redundancy	Specifies modem passthrough for the specified codec. The keywords are as follows: <ul style="list-style-type: none"> • nse—Used to specify Named Signaling Event (NSE). • codec—Used to specify the type of codec. <ul style="list-style-type: none"> – g711alaw—Specifies the G.711 a-law codec type. – g711ulaw—Specifies the G.711 u-law codec type. • redundancy—Specifies the RFC 2198 for packet redundancy. <p>Note Use the same codec type for both the originating gateway and the terminating gateway. The G.711 u-law codec is required for T1.</p>
Step 2	Router(config-dial-peer)# fax rate disable	Enables fax passthrough.

Configuring Modem and Fax Passthrough for H.323 and SIP Support (Global)

When using the **voice service voip** and **modem passthrough nse** commands on a terminating gateway to globally set up fax or modem passthrough with NSEs, you must also ensure that each incoming call will be associated with a VoIP dial peer to retrieve the global fax or modem configuration. You associate calls with dial peers by using the **incoming called-number** command to specify a sequence of digits that incoming calls can match. You can ensure that all calls will match at least one dial peer by using the following commands:

```
Router(config)# dial-peer voice tag voip
Router(config-dial-peer)# incoming called-number .
```

To configure modem and fax passthrough for all the dial peers of a gateway, use the following commands in voice-service configuration mode:

Command	Purpose
<pre>Router(conf-voi-serv)# modem passthrough nse [payload-type number] codec {g711alaw g711ulaw} [redundancy] [maximum-sessions value]</pre>	<p>Configures modem passthrough for all dial peers of a gateway. The default is no modem passthrough.</p> <p>The keywords are as follows:</p> <ul style="list-style-type: none"> • nse—Specifies the NSE. • payload-type number—(Optional) NSE payload type. The valid range is from 96 to 119. • codec—Specified the codec for upspeed. Upspeed dynamically changes the codec and speed to meet network conditions. • g711alaw—Specifies the 64 Kbps upspeed for E1. • g711ulaw—Specifies the 64 Kbps upspeed for T1. • redundancy—(Optional) Specifies the RFC 2198 for packet redundancy. • maximum-sessions value—(Optional) Specifies the number of simultaneous modem passthrough sessions. The valid range is from 1 (minimum) to 26 (maximum). The default is 16.

Configuring Modem and Fax Passthrough for MGCP Support

You can configure modem and fax passthrough with MGCP support by entering global configuration mode. You must configure a VoIP dial peer on both the originating and terminating gateways to match the call—for example, using a destination pattern. The modem and fax passthrough parameters associated with those dial peers will then apply to the calls between them.



Note

When modem and fax passthrough is configured individually for a specific dial peer, the dial-peer configuration takes precedence over the system configuration for that specific dial peer.

To configure modem and fax passthrough with MGCP support on Cisco AS5400 universal gateways, use the following commands in global configuration mode:

	Command	Purpose
Step 1	Router(config)# mgcp modem passthrough voip codec { g711ulaw g711alaw }	Specifies the modem passthrough VoIP codec. The keywords are as follow: <ul style="list-style-type: none"> • codec—Used to specify the type of codec. <ul style="list-style-type: none"> – g711ulaw—Specifies the G.711 u-law codec type. – g711alaw—Specifies the G.711 a-law codec type. <p>Note Use the same codec type for both the originating gateway and the terminating gateway. The G.711 u-law codec is required for T1.</p>
Step 2	Router(config)# mgcp modem passthrough voip mode nse	Configures modem passthrough mode. The mode and nse keywords set the MGCP modem and fax passthrough mode to NSE.
Step 3	Router(config)# mgcp modem passthrough voip redundancy sample-duration value maximum sessions value	Specifies redundant packets for modem traffic. <ul style="list-style-type: none"> • redundancy sample-duration—When redundancy is on, all calls on the same server are affected. The valid values are 10 and 20 ms. The default is 10 ms. • maximum sessions value—Specifies the number of maximum redundancy sessions that run simultaneously on each subsystem. The valid range is from 1 (minimum) to 26 (maximum). The default is 16.
Step 4	Router(config)# mgcp fax t38 inhibit	Enables fax passthrough.

Verifying Modem and Fax Passthrough

To verify that modem and fax passthrough are enabled, use the following commands:

- **show running-config** to verify the configuration. The following is sample output of a running configuration on a Cisco AS5300 and Cisco AS5400:

Cisco AS5300 Universal Access Server

```
Router# show running-config

Building configuration...

Current configuration : 6109 bytes
!
version 12.2
no parser cache
service timestamps debug datetime msec
```

```

service timestamps log uptime
no service password-encryption
service internal
!
hostname as5300-2
!
aaa new-model
aaa authentication login h323 local
aaa session-id common
!
dial-peer cor custom
!
dial-peer voice 5321 pots
destination-pattern 5321..
direct-inward-dial
port 0:D
prefix 3
!
dial-peer voice 5311 voip
incoming called-number 5321..
destination-pattern 5311..
modem passthrough nse payload-type 119 codec g711alaw
session target ipv4:10.0.0.0
fax rate disable
fax protocol t38 ls-redundancy 0 hs-redundancy 0
!
dial-peer voice 5411 voip
incoming called-number 5321..
destination-pattern 5411..
modem passthrough nse payload-type 119 codec g711alaw
session target ipv4:10.0.0.0
codec g729r8
exit

```

Cisco AS5400 Universal Gateway

```

Router# show running-config
!
Building configuration...

Current configuration : 5943 bytes
!
version 12.2
no parser cache
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname as5400
!
aaa new-model
aaa authentication login h323 local
aaa session-id common
!
username 1234 password 0 5678
!
dial-peer cor custom
!
dial-peer voice 5311 voip
incoming called-number 5421..
destination-pattern 5311..
modem passthrough nse codec g711ulaw
session target ipv4:10.0.0.0
!

```

```

dial-peer voice 5411 voip
 destination-pattern 5411..
 modem passthrough nse codec g711ulaw
 session target ipv4:10.0.0.0
!
dial-peer voice 5421 pots
 incoming called-number 5311..
 destination-pattern 5421..
 direct-inward-dial
 port 4/0:D
 prefix 3
!
dial-peer voice 5411 voip
 codec gsmfr
 exit

```

- Use the **show dial-peer voice** command to verify that modem passthrough over VoIP is enabled. The following is sample output of a modem and fax passthrough configuration with SIP support on a Cisco AS5400 universal gateway:

```

Router# show dial-peer voice 5411
!
VoiceOverIpPeer5411
 information type = voice,
 description = '',
 tag = 5411, destination-pattern = `5411..',
 answer-address = '', preference=0,
 CLID Restriction = None
 CLID Network Number = ``
 CLID Second Number sent
 source carrier-id = '',target carrier-id = '',
 source trunk-group-label = '',target trunk-group-label = '',
 numbering Type = `unknown'
 group = 5411, Admin state is up, Operation state is up,
 incoming called-number = '', connections/maximum = 0/unlimited,
 DTMF Relay = disabled
 modem transport = passthrough, nse, payload type = 100, codec = g711ulaw,
 redundancy,
 huntstop = disabled
 in bound application associated: 'DEFAULT'
 out bound application associated: ''
 dnis-map =
 permission :both
 incoming COR list:maximum capability
 outgoing COR list:minimum requirement
 Translation profile (Incoming):
 Translation profile (Outgoing):
 incoming call blocking:
 translation-profile = ``
 disconnect-cause = `no-service'
 type = voip, session-target = `ipv4:10.0.0.0',
 technology prefix:
 settle-call = disabled
 ip media DSCP = default, ip signaling DSCP = default, UDP checksum = disabled,
 session-protocol = cisco, session-transport = system, req-qos = best-effort,
 acc-qos = best-effort,
 RTP dynamic payload type values: NTE = 101
 Cisco: NSE=100, fax=96, fax-ack=97, dtmf=121, fax-relay=122
 CAS=123, ClearChan=125, PCM switch over u-law=0,A-law=8
 fax rate = voice, payload size = 20 bytes
 fax protocol = system
 fax-relay ecm enable
 fax NSF = 0xAD0051 (default)
 codec = g729r8, payload size = 20 bytes

```

```

Expect factor = 0, Icpif = 20
Playout Mode is set to default
Initial 60 ms, Max 300 ms
Playout-delay Minimum mode is set to default, value 40 ms
Max Redirects = 1, signaling-type = ext-signal,
VAD = enabled, Poor QOV Trap = disabled,
Source Interface = NONE
voice class sip url = system,
voice class sip rellxx = system,
voice class perm tag = ``
Time elapsed since last clearing of voice call statistics never
Connect Time = 0, Charged Units = 0,
Successful Calls = 0, Failed Calls = 0, Incomplete Calls = 0
Accepted Calls = 0, Refused Calls = 0,
Last Disconnect Cause is "",
Last Disconnect Text is "",

```

**Note**

Refer to the *Cisco IOS Voice, Video, and Fax Command Reference*, Release 12.2, for significant fields descriptions.

Troubleshooting Tips for Modem and Fax Passthrough

To troubleshoot modem and fax passthrough, perform the following checks:

- Ensure that you can make a voice call.
- Ensure that modem passthrough over VoIP is configured with the same parameters on both the originating gateway and the terminating gateway.
- Ensure that the originating and terminating gateways have the same NSE payload-type number.
- Use the **debug vtsp session** command to display the voice telephony call control session. The following is sample output on a Cisco AS5400 universal gateway:

```

Router# debug vtsp session
!
!Voice telephony call control session debugging is on
!
as5400#
*Jan 1 00:41:49.275: //-1/xxxxxxxxxxxxx/VTSP:():-1:-1:-1/vtsp_do_call_setup_req: .
*Jan 1 00:41:49.275: //-1/xxxxxxxxxxxxx/VTSP:():-1:-1:-1/vtsp_allocate_cdb: ,cdb
0x64463920
*Jan 1 00:41:49.275: //8/xxxxxxxxxxxxx/VTSP:(4/0:D):-1:0:0/vtsp_insert_cdb: ,cdb
0x64463920, CallID=8
*Jan 1 00:41:49.275: //8/4E2302238023/VTSP:(4/0:D):-1:0:0/vtsp_do_call_setup_req:
calling oct3 0x0,
called oct3 0x81
*Jan 1 00:41:49.275: //8/4E2302238023/VTSP:(4/0:D):-1:0:0/vtsp_do_call_setup_req:
Call ID=32803,
guid=64463FA8
*Jan 1 00:41:49.275: //8/4E2302238023/VTSP:(4/0:D):-1:0:0/vtsp_do_call_setup_req:
type=0,
under_spec=1, name=, ds0=0, ds1=-1, echo_cancel=1, gain_control 0, auto_gain_control
0,
dual_tone_detect 0, seq_tone_detect 0, calling=, called=542100, playout
mode=0,playout_init = 0,
playout_min = 0, playout_max = 0
vtsp_do_call_setup_req: redirect DN: reason: -1
*Jan 1 00:41:49.275: digit_strip:1, pcn:542100, poa:5421..
*Jan 1 00:41:49.275: pcn:00, poa:...
*Jan 1 00:41:49.275: Final pcn:00, poa:..., dial_string:300

```

```

*Jan 1 00:41:49.275: //8/4E2302238023/VTSP:(4/0:D):-1:20480:0/vtsp_request_call:
calling oct3 0x0,
called oct3 0x81
*Jan 1 00:41:49.275: //8/4E2302238023/VTSP:(4/0:D):-1:20480:0/vtsp_request_call:
cdb->sdb->type = 5
*Jan 1 00:41:49.275:
//-1/xxxxxxxxxxxx/VTSP:():-1:-1:-1/vtsp_create_call_active_on_setup_req:
*Jan 1 00:41:49.275: vtsp_create_call_active_on_setup_req: target route label is
*Jan 1 00:41:49.275:
//8/4E2302238023/VTSP:(4/0:D):-1:20480:0/vtsp_create_call_active_on_setup_req:
*Jan 1 00:41:49.275: vtsp_create_call_active_on_setup_req : tgt carrier id
*Jan 1 00:41:49.275:
//8/4E2302238023/VTSP:(4/0:D):-1:20480:0/vtsp_create_call_active_on_setup_req:
*Jan 1 00:41:49.275: vtsp_create_call_active_on_setup_req : src carrier id
*Jan 1 00:41:49.307: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_process_event:
vtsp:[4/0:D (8),
S_SETUP_REQUEST, E_TSP_PROCEEDING]
*Jan 1 00:41:49.311:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/act_setup_pend_proceeding: .
*Jan 1 00:41:49.311:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_open_voice_and_set_params: .
*Jan 1 00:41:49.311:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_modem_proto_from_cdb:
cap_modem_proto 0
*Jan 1 00:41:49.311: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/set_playout_cdb:
playout default
*Jan 1 00:41:49.315: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_process_event:
vtsp:[4/0:D (8),
S_SETUP_REQ_PROC, E_TSP_PROGRESS]
*Jan 1 00:41:49.315:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/act_setup_pend_progress: .
*Jan 1 00:41:49.315: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_process_event:
vtsp:[4/0:D (8),
S_SETUP_REQ_PROC, E_CC_BRIDGE]
*Jan 1 00:41:49.315: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/act_bridge: .
*Jan 1 00:41:49.315: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_process_event:
vtsp:[4/0:D (8),
S_SETUP_REQ_PROC, E_CC_CAPS_IND]
*Jan 1 00:41:49.315: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/act_caps_ind: .
*Jan 1 00:41:49.315: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/act_caps_ind: RTP
PT:NTE[101],NTEtx[101],NSE[100],FaxInd[96],FaxAck[97],CiscoDTMF[121],FaxRelay[122],CAS
sig[123],ClearC
han[125],PCMu[0],PCMa[8]Codec[4],TxDynamicPayload[0], RxDynamicPayload[0]
*Jan 1 00:41:49.319: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/act_caps_ind: dtmf
relay: mode=1,
codec=1
*Jan 1 00:41:49.319: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/act_caps_ind:
passthrough:
cap_modem_proto 4, cap_modem_codec 1, cap_modem_redundancy 1, payload 100, modem_relay
0, gw-xid=0
*Jan 1 00:41:49.319: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/act_caps_ind: Encap 1,
Vad 2, Codec
0x4, CodecBytes 20,
FaxRate 2, FaxBytes 20, FaxNsf 0xAD0051
SignalType 1
DtmfRelay 1, Modem 2, SeqNumStart 0x1EAC
*Jan 1 00:41:49.319: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/act_caps_ind:
*Jan 1 00:41:49.319: FORKING Parameters are forking mask: 7,
simple_forking_codec_mask: 39039,
complex_forking_codec_mask 39039
*Jan 1 00:41:49.319: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/act_caps_ind: [
mode:0,init:60,
min:40, max:200]

```

```

*Jan 1 00:41:49.319: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_process_event:
vtsp:[4/0:D (8),
S_SETUP_REQ_PROC, E_CC_CAPS_ACK]
*Jan 1 00:41:49.319: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/act_caps_ack: .
*Jan 1 00:41:49.319: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/act_caps_ack:
passthrough:
cap_modem_proto 4, cap_modem_codec 1, cap_modem_redundancy 1, payload 100, modem_relay
0, gw-xid=0
*Jan 1 00:41:49.319: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/act_caps_ack: Named
Telephone Event
payload: rcv 101, tx 101
*Jan 1 00:41:49.319: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/act_switch_codec:
*Jan 1 00:41:49.319: DTMF Relay in act_switch_codec is 1
*Jan 1 00:41:49.319: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/set_dsp_encap_config:
*Jan 1 00:41:49.319: set_dsp_encap_config: logical ssrc 40
*Jan 1 00:41:49.319:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_modem_proto_from_cdb:
cap_modem_proto 4
*Jan 1 00:41:49.319:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_is_modem_redundancy: 1
*Jan 1 00:41:49.319: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/act_switch_codec: codec
= 16
*Jan 1 00:41:49.319: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_timer: 251219
*Jan 1 00:41:49.339: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_process_event:
vtsp:[4/0:D (8),
SP_PENDING_CODEC_SWITCH, E_DSPRM_PEND_SUCCESS]
*Jan 1 00:41:49.339: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/act_pend_codec_success:
*Jan 1 00:41:49.339: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_timer_stop: 251221
*Jan 1 00:41:49.339:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_open_voice_and_set_params: .
*Jan 1 00:41:49.339: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/set_dsp_encap_config:
*Jan 1 00:41:49.339: set_dsp_encap_config: logical ssrc 40
*Jan 1 00:41:49.339:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_modem_proto_from_cdb:
cap_modem_proto 4
*Jan 1 00:41:49.339:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_is_modem_redundancy: 1
*Jan 1 00:41:49.339: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/set_payout_cdb:
payout default
*Jan 1 00:41:49.339: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_add_fork:
*Jan 1 00:41:49.339: vtsp_add_fork
*Jan 1 00:41:49.339: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_update_fork_info:
*Jan 1 00:41:49.339: vtsp_update_fork_info: add_fork=0
*Jan 1 00:41:49.339:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_get_xmit_info_node:
*Jan 1 00:41:49.339: vtsp_get_xmit_info_node
*Jan 1 00:41:49.339: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_update_fork_info:
*Jan 1 00:41:49.339: vtsp_update_fork_info xmit func is 61173904, context is
6445BC8Cpeer_call_id:
7, stream_count: 1, update_flag 0
*Jan 1 00:41:49.339: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_update_fork_info:
*Jan 1 00:41:49.339: The stream bit-mask is 1
*Jan 1 00:41:49.339: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_update_fork_info:
*Jan 1 00:41:49.339: The stream type is 0
*Jan 1 00:41:49.339: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_update_fork_info:
*Jan 1 00:41:49.339: The logical ssrc is 64 for stream 0
*Jan 1 00:41:49.339:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_update_stream_count:
*Jan 1 00:41:49.339: g711_voice_count=0 g711_avt_count = 0
g711_voice_avt_count = 0 complex_voice_count = 1
complex_avt_count = 0 complex_voice_avt_count = 0
*Jan 1 00:41:56.139: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_process_event:
vtsp:[4/0:D (8),
S_SETUP_REQ_PROC, E_TSP_CONNECT]

```

```

*Jan 1 00:41:56.139: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/act_setup_pend_connect:
*Jan 1 00:41:56.139:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_ring_noan_timer_stop: 251901
*Jan 1 00:41:58.555: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_process_event:
vtsp:[4/0:D (8),
S_CONNECT, E_DSP_MODEM_TONE]
*Jan 1 00:41:58.555: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/act_modem_detected: .
*Jan 1 00:41:58.555:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_modem_proto_from_cdb:
cap_modem_proto 4
*Jan 1 00:41:58.555:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_modem_proto_from_cdb:
cap_modem_proto 4
*Jan 1 00:41:58.555:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_modem_proto_from_cdb:
cap_modem_proto 4
*Jan 1 00:41:58.555:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_is_modem_redundancy: 1
*Jan 1 00:41:58.555:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_accept_modem_passthrough_session:
cdb: 64463920, active sessions 0, max sessions: 16 rejected sessions till now: 0
*Jan 1 00:41:58.555:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_handle_modem_detect_nse: .
*Jan 1 00:41:58.555: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_save_modem_params:
codec 4
*Jan 1 00:41:58.555:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_modem_proto_from_cdb:
cap_modem_proto 4
*Jan 1 00:41:58.555: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/act_pcm_switchover: .
*Jan 1 00:41:58.555: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/act_pcm_switchover:
passthru_mode 4,
cdb->codec_params.modem 2, cap_modem_proto 4, echo_cancel 1, fax_relay_on 1
*Jan 1 00:41:58.555:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_modem_proto_from_cdb:
cap_modem_proto 4
*Jan 1 00:41:58.555:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_is_modem_redundancy: 1
*Jan 1 00:41:58.555: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/act_pcm_switchover:
Modem Relay is
disabled. Do not enable CM Detection
*Jan 1 00:41:58.555:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_modem_proto_from_cdb:
cap_modem_proto 4
*Jan 1 00:41:58.555:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_is_modem_redundancy: 1
*Jan 1 00:41:58.555:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_modem_proto_from_cdb:
cap_modem_proto 4
*Jan 1 00:41:58.555:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_is_modem_redundancy: 1
*Jan 1 00:41:58.555:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_modem_proto_from_cdb:
cap_modem_proto 4
*Jan 1 00:41:58.555:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_is_modem_redundancy: 1
*Jan 1 00:41:58.555: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/act_pcm_switchover:
switched over to
codec 0x1
*Jan 1 00:41:58.559:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_pcm_switchover_timer_start:
252143
*Jan 1 00:41:59.051: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_process_event:
vtsp:[4/0:D (8),
S_CONNECT, E_DSP_MODEM_TONE]

```

```

*Jan 1 00:41:59.051: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/act_modem_detected: .
*Jan 1 00:41:59.051:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_modem_proto_from_cdb:
cap_modem_proto 4
*Jan 1 00:41:59.135: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_process_event:
vtsp:[4/0:D (8),
S_CONNECT, E_DSP_MODEM_PHASE]
*Jan 1 00:41:59.135:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/act_modem_phase_detected: .
*Jan 1 00:41:59.135:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_modem_proto_from_cdb:
cap_modem_proto 4
*Jan 1 00:41:59.135:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/act_modem_phase_detected: NSE mode -
phase reversal detected
*Jan 1 00:41:59.135:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_handle_modem_phase_nse: .
*Jan 1 00:42:43.559: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_main: vtsp_main:
switchover
timer: 256643
*Jan 1 00:42:43.559: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_process_event:
vtsp:[4/0:D (8),
S_CONNECT, E_PCM_SWITCHOVER_TIMER]
!Output when the call disconnects (close and released).
as5400#
*Jan 1 00:43:07.047: %PORT-6-SM_PORT_CLEARED: All Ports Are Cleared
*Jan 1 00:43:07.051: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_process_event:
vtsp:[4/0:D (8),
S_CONNECT, E_DSPRM_SHUTDOWN]
*Jan 1 00:43:07.051: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/act_alarm: .
*Jan 1 00:43:07.051:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_ring_noan_timer_stop: 258992
*Jan 1 00:43:07.051: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_timer_stop: 258992
*Jan 1 00:43:07.055: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_process_event:
vtsp:[4/0:D (8),
S_WAIT_HOST_DISC, E_CC_BRIDGE_DROP]
*Jan 1 00:43:07.059: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/act_bdrop: .
*Jan 1 00:43:07.059:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_free_modem_passthrough_session:
cdb: 64463920, active sessions: 1
*Jan 1 00:43:07.059:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_modem_proto_from_cdb:
cap_modem_proto 4
*Jan 1 00:43:07.059:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_modem_proto_from_cdb:
cap_modem_proto 4
*Jan 1 00:43:07.059:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_is_modem_redundancy: 1
*Jan 1 00:43:07.059:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_remove_stream_node:
*Jan 1 00:43:07.059: vtsp_remove_stream_node
*Jan 1 00:43:07.059:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_get_xmit_info_node:
*Jan 1 00:43:07.059: vtsp_get_xmit_info_node
*Jan 1 00:43:07.059: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_is_record_active:
*Jan 1 00:43:07.059: vtsp_is_record_active
*Jan 1 00:43:07.059: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_process_event:
vtsp:[4/0:D (8),
S_WAIT_HOST_DISC, E_CC_DISCONNECT]
*Jan 1 00:43:07.059: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/act_get_error_stats: .
*Jan 1 00:43:07.059: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/act_stats_complete: .
*Jan 1 00:43:07.059: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_timer_stop: 258993
*Jan 1 00:43:07.059:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_ring_noan_timer_stop: 258993

```

```

*Jan 1 00:43:07.059: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_timer: 258993
*Jan 1 00:43:07.059: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_process_event:
vtsp:[4/0:D (8),
S_WAIT_RELEASE, E_TSP_DISCONNECT_CONF]
*Jan 1 00:43:07.059: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/act_wrelease_release: .
*Jan 1 00:43:07.063:
//8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_play_busy_timer_stop:
*Jan 1 00:43:07.063: vtsp_play_busy_timer_stop: 258994
*Jan 1 00:43:07.063: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_timer_stop: 258994
*Jan 1 00:43:07.063: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_do_call_history: .
*Jan 1 00:43:07.063: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_do_call_history:
*Jan 1 00:43:07.063: vtsp_do_call_history : src carrier id
*Jan 1 00:43:07.063: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_do_call_history:
*Jan 1 00:43:07.063: vtsp_do_call_history : tgt carrier id
*Jan 1 00:43:07.063: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_do_call_history:
CoderRate 5
*Jan 1 00:43:07.063: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_process_event:
vtsp:[4/0:D (8),
S_CLOSE_DSPRM, E_DSPRM_CLOSE_COMPLETE]
*Jan 1 00:43:07.063: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/act_terminate: .
*Jan 1 00:43:07.063: //8/4E2302238023/VTSP:(4/0:D):18:20480:0/vtsp_free_cdb: ,cdb
0x64463920

```



Note Refer to the *Cisco IOS Debug Command Reference*, Release 12.2, for significant fields descriptions.

Monitoring and Maintaining Modem and Fax Passthrough

To monitor and maintain modem and fax passthrough, enter the following commands in privileged EXEC mode, as needed:

Command	Purpose
Router# show call history voice [brief]	Displays the modem and fax information for the call history table. The brief keyword displays a truncated version.
Router# show call active voice [brief]	Displays the modem and fax information for the active call table. The brief keyword displays a truncated version.

The following are sample output using the **show call history voice** command (last 2 calls) and **show call active voice brief** command.

show call history voice last 2

```
as5400# show call active voice last 2
```

```

Telephony call-legs: 1
SIP call-legs: 0
H323 call-legs: 1
Total call-legs: 2
GENERIC:
SetupTime=104962 ms
Index=1
PeerAddress=
PeerSubAddress=
PeerId=5311
PeerIfIndex=237
LogicalIfIndex=0

```

```
ConnectTime=107960
CallDuration=00:00:18
CallState=4
CallOrigin=2
ChargedUnits=0
InfoType=2
TransmitPackets=1352
TransmitBytes=265786
ReceivePackets=792
ReceiveBytes=254167
VOIP:
ConnectionId[0xE6652251 0xC24011D3 0x8020CD96 0xA5651944]
IncomingConnectionId[0xE6652251 0xC24011D3 0x8020CD96 0xA5651944]
RemoteIPAddress=200.200.200.3
RemoteUDPPort=18480
RemoteSignallingIPAddress=10.0.0.0
RemoteSignallingPort=11004
RemoteMediaIPAddress=10.0.0.0
RemoteMediaPort=18480
RoundTripDelay=0 ms
SelectedQoS=best-effort
tx_DtmfRelay=inband-voice
FastConnect=TRUE
AnnexE=FALSE
Separate H245 Connection=FALSE
H245 Tunneling=TRUE
SessionProtocol=cisco
ProtocolCallId=
SessionTarget=
OnTimeRvPlayout=10000
GapFillWithSilence=0 ms
GapFillWithPrediction=0 ms
GapFillWithInterpolation=0 ms
GapFillWithRedundancy=0 ms
HiWaterPlayoutDelay=150 ms
LoWaterPlayoutDelay=100 ms
ReceiveDelay=140 ms
LostPackets=1
EarlyPackets=0
LatePackets=0
VAD = enabled
CoderTypeRate=g711ulaw
CodecBytes=20
Modem passthrough signaling method is nse:
Buffer Fill Events = 0
Buffer Drain Events = 0
Percent Packet Loss = 0
Consecutive-packets-lost Events = 0
Corrected packet-loss Events = 0
Last Buffer Drain/Fill Event = 0sec
Time between Buffer Drain/Fills = Min 0sec Max 0sec
CallerName=
CallerIDBlocked=False
OriginalCallingNumber=
OriginalCallingOctet=0x0
OriginalCalledNumber=542100
OriginalCalledOctet=0x81
OriginalRedirectCalledNumber=
OriginalRedirectCalledOctet=0x7F
GwCollectedCalledNumber=
GwReceivedCdn=542100
GwReceivedCgn=
GkProvidedE164Cdn=
GkProvidedE164Cgn=
```

```

GwFinalTranslatedCdn=
GwFinalTranslatedCgn=
TranslatedCallingNumber=
TranslatedCallingOctet=0x0
TranslatedCalledNumber=542100
TranslatedCalledOctet=0x81
TranslatedRedirectCalledNumber=
TranslatedRedirectCalledOctet=0x7F
Username=
GENERIC:
SetupTime=104965 ms
Index=1
PeerAddress=542100
PeerSubAddress=
PeerId=5421
PeerIfIndex=239
LogicalIfIndex=94
ConnectTime=107960
CallDuration=00:00:20
CallState=4
CallOrigin=1
ChargedUnits=0
InfoType=2
TransmitPackets=909
TransmitBytes=292192
ReceivePackets=1469
ReceiveBytes=303811
TELE:
ConnectionId=[0xE6652251 0xC24011D3 0x8020CD96 0xA5651944]
IncomingConnectionId=[0xE6652251 0xC24011D3 0x8020CD96 0xA5651944]
TxDuration=50245 ms
VoiceTxDuration=33000 ms
FaxTxDuration=0 ms
CoderTypeRate=g711ulaw
NoiseLevel=-123
ACOMLevel=45
OutSignalLevel=-14
InSignalLevel=-14
InfoActivity=1
ERLLevel=45
SessionTarget=
ImgPages=0
CallerName=
CallerIDBlocked=False
OriginalCallingNumber=
OriginalCallingOctet=0x0
OriginalCalledNumber=542100
OriginalCalledOctet=0x81
OriginalRedirectCalledNumber=
OriginalRedirectCalledOctet=0x7F
GwCollectedCalledNumber=
GwReceivedCdn=542100
GwReceivedCgn=
GwFinalTranslatedCdn=300
GwFinalTranslatedCgn=
TranslatedCallingNumber=
TranslatedCallingOctet=0x0
TranslatedCalledNumber=542100
TranslatedCalledOctet=0x81
TranslatedRedirectCalledNumber=
TranslatedRedirectCalledOctet=0x7FTelephony call-legs: 1
SIP call-legs: 0
H323 call-legs: 1
Total call-legs: 2

```

show call active voice briefRouter# **show call active voice brief**

```

<ID>: <start>hs.<index> +<connect> pid:<peer_id> <dir> <addr> <state>
dur hh:mm:ss tx:<packets>/<bytes> rx:<packets>/<bytes>
IP <ip>:<udp> rtt:<time>ms pl:<play>/<gap>ms lost:<lost>/<early>/<late>
delay:<last>/<min>/<max>ms <codec>
MODEMPASS <method> buf:<fills>/<drains> loss <overall%> <multipkt>/<corrected>
last <buf event time>s dur:<Min>/<Max>s
FR <protocol> [int dlci cid] vad:<y/n> dtmf:<y/n> seq:<y/n>
<codec> (payload size)
ATM <protocol> [int vpi/vci cid] vad:<y/n> dtmf:<y/n> seq:<y/n>
<codec> (payload size)
Tele <int>: tx:<tot>/<v>/<fax>ms <codec> noise:<l> acom:<l> i/o:<l>/<l> dBm
MODEMRELAY info:<rcvd>/<sent>/<resent> xid:<rcvd>/<sent> total:<rcvd>/<sent>/<drops>
speeds (bps): local <rx>/<tx> remote <rx>/<tx>
Proxy <ip>:<audio udp>,<video udp>,<tcp0>,<tcp1>,<tcp2>,<tcp3> endpt: <type>/<manf>
bw: <req>/<act> codec: <audio>/<video>
tx: <audio pkts>/<audio bytes>,<video pkts>/<video bytes>,<t120 pkts>/<t120 bytes>
rx: <audio pkts>/<audio bytes>,<video pkts>/<video bytes>,<t120 pkts>/<t120 bytes>
Telephony call-legs: 1
SIP call-legs: 0
H323 call-legs: 1
Total call-legs: 2
11F3 : 104962hs.1 +2998 pid:5311 Answer active
dur 00:00:28 tx:1868/433486 rx:1308/421867
IP 200.200.200.3:18480 rtt:2ms pl:21000/0ms lost:1/0/0 delay:145/100/150ms g711ulaw
MODEMPASS nse buf:0/0 loss 0% 0/0 last 0s dur:0/0s
11F3 : 104965hs.1 +2995 pid:5421 Originate 542100 active
dur 00:00:28 tx:1309/422192 rx:1868/433486
Tele 4/0:D (6): tx:58230/41000/0ms g711ulaw noise:-123 acom:45 i/0:-14/-14 dBm
Telephony call-legs: 1
SIP call-legs: 0
H323 call-legs: 1
Total call-legs: 2

```

**Note**

Refer to the *Cisco IOS Voice, Video, and Fax Command Reference*, Release 12.2, for significant fields descriptions.

Configuration Examples

This section provides the following configuration examples:

- [G.Clear Codec with H.323 Support Configuration Example, page 34](#)
- [G.Clear Codec with SIP Support Configuration Example, page 34](#)
- [G.Clear Codec with MGCP Support Configuration Example, page 34](#)
- [Modem Passthrough with H.323 Support Configuration Example, page 36](#)
- [Modem Passthrough with MGCP Support Configuration Example, page 37](#)
- [Fax Passthrough with H.323 Support Configuration Example, page 39](#)
- [Fax Passthrough with SIP Support Configuration Example, page 41](#)
- [Fax Passthrough with MGCP Support Configuration Example, page 42](#)

G.Clear Codec with H.323 Support Configuration Example

The following is sample output displaying G.Clear codec with H.323 support:

```
dial-peer voice 5551212 voip
  incoming called-number 5551212
  destination pattern 666...
  session target ipv4:10.10.0.0
  codec clear-channel
no vad
```

G.Clear Codec with SIP Support Configuration Example

The following is sample output displaying G.Clear codec with SIP support:

```
dial-peer voice 5551212 voip
  incoming called-number 6660000
  destination pattern 555...
  session target ipv4:10.10.0.0
  session protocol sipv2
  codec clear-channel
no vad
```

G.Clear Codec with MGCP Support Configuration Example

The following is sample output displaying G.Clear codec with MGCP support on the Cisco AS5850:

```
version 12.2
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname 5850
!
no logging buffered
no logging rate-limit
!
resource-pool disable
resource-pool call treatment resource channel-not-available
resource-pool call treatment profile busy
!
clock calendar-valid
dial-tdm-clock priority 1 trunk-slot 1 port 0
spe link-info poll voice 5
spe default-firmware spe-firmware-1
spe 2/02
  shutdown
spe 2/03 2/05
  busyout
spe 2/08
  shutdown
spe 2/09 2/11
  busyout
spe 2/14
  shutdown
spe 2/15 2/17
  busyout
ip subnet-zero
```

```
ip cef distributed
ip ftp username mgcusr
ip ftp password lab
no ip domain-lookup
ip host brios 255.255.255.255
ip host colos_tftp 10.10.0.0
ip dhcp smart-relay
!
backhaul-session-manager
  set set1 client nft
  group group1 set set1
  session group group1 10.10.0.0 6667 10.10.0.0 6667 1
!
isdn switch-type primary-net5
no voice hpi capture buffer
no voice hpi capture destination
!
mrp client session history duration 0
mrp client session history records 0
memory check-interval 3600
memory validate-checksum 7200
redundancy
  no keepalive-enable
  mode classic-split
!
controller E1 0/1
  pri-group timeslots 1-31 service mgcp
!
interface Serial0/0:15
  no ip address
  isdn switch-type primary-net5
  isdn incoming-voice modem
  isdn bind-13 backhaul set1
  no cdp enable
!
mgcp codec clear-channel packetization-period 10
mgcp vad
mgcp quarantine mode process
mgcp package-capability dtmf-package
mgcp package-capability mf-package
mgcp package-capability rtp-package
mgcp package-capability nas-package
mgcp package-capability as-package
mgcp package-capability script-package
mgcp default-package ms-package
no mgcp timer receive-rtcp
no mgcp piggyback message
mgcp fax t38 inhibit
!
mgcp profile default
mgcp profile PRI
  call-agent 10.10.0.0 service-type mgcp version 0.1
  port 0/0:15
  port 0/1:15
  port 0/2:15
  port 0/3:15
  port 0/4:15
  port 0/5:15
  port 0/6:15
  port 0/7:15
```

```

port 0/8:15
port 0/9:15
port 0/10:15
port 0/11:15
gateway
!
line con 0
  exec-timeout 0 0
  logging synchronous
line aux 0
  exec-timeout 0 0
  logging synchronous
line vty 0 4
  password lab
  no login
line 2/00 5/323
  flush-at-activation
  modem InOut
  no modem status-poll
  no modem log rs232
  transport input all
  autoselect ppp
!
end

```

Modem Passthrough with H.323 Support Configuration Example

The following is sample output displaying H.323 and SIP support on the Cisco AS5300:

```

version 12.2
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname 5850
!
no logging buffered
no logging rate-limit
!
resource-pool disable
dial-tdm-clock priority 1 trunk-slot 1 port 0
spe link-info poll voice 5
spe default-firmware spe-firmware-1
ip subnet-zero
ip cef distributed
ip ftp username mgcusr
ip ftp password lab
no ip domain lookup
ip host colos_tftp 10.10.0.0
ip host brios 255.255.255.255
ip dhcp smart-relay
!
isdn switch-type primary-net5
!
voice service voip
  h323
  modem passthrough nse codec g711alaw redundancy sample-duration 20
!
no voice hpi capture buffer
no voice hpi capture destination
!

```

```

mrpc client session history duration 0
mrpc client session history records 0
memory check-interval 3600
memory validate-checksum 7200
redundancy
  no keepalive-enable
  mode classic-split
!
controller E1 0/0
  pri-group timeslots 1-31
!
dial-peer voice 5001 pots
  incoming called-number 550
  destination-pattern 800
  direct-inward-dial
  port 0/0:D
  prefix 800
!
dial-peer voice 500 voip
  incoming called-number 800
  destination-pattern 550
  session target ipv4:10.10.0.0
  codec g726r32
!
gateway
!
line con 0
  exec-timeout 0 0
  logging synchronous
line aux 0
  exec-timeout 0 0
  logging synchronous
line vty 0 4
  password lab
  no login
line 2/00 5/323
  flush-at-activation
  no modem status-poll
  no modem log rs232

```

Modem Passthrough with MGCP Support Configuration Example

The following is sample output displaying MGCP support on the Cisco AS5850:

```

version 12.2
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname 5850
!
no logging buffered
no logging rate-limit
!
resource-pool disable
resource-pool call treatment resource channel-not-available
resource-pool call treatment profile busy
!
clock calendar-valid
dial-tdm-clock priority 1 trunk-slot 1 port 0
spe link-info poll voice 5

```

```

spe default-firmware spe-firmware-1
spe 2/02
  shutdown
spe 2/03 2/05
  busyout
spe 2/08
  shutdown
spe 2/09 2/11
  busyout
spe 2/14
  shutdown
spe 2/15 2/17
  busyout
!
ip subnet-zero
ip cef distributed
ip ftp username mgcusr
ip ftp password lab
no ip domain-lookup
ip host brios 255.255.255.255
ip host colos_tftp 10.10.0.0
ip dhcp smart-relay
!
backhaul-session-manager
  set set1 client nft
  group group1 set set1
  session group group1 10.10.0.0 6667 10.10.0.0 6667 1
!
isdn switch-type primary-net5
no voice hpi capture buffer
no voice hpi capture destination
!
mrcp client session history duration 0
mrcp client session history records 0
memory check-interval 3600
memory validate-checksum 7200
redundancy
  no keepalive-enable
  mode classic-split
!
controller E1 0/1
  pri-group timeslots 1-31 service mgcp
!
interface Serial0/0:15
  no ip address
  isdn switch-type primary-net5
  isdn incoming-voice modem
  isdn bind-13 backhaul set1
  no cdp enable
!
mgcp modem passthrough voip mode nse
mgcp modem passthrough voip codec g711alaw
mgcp codec g729r8 packetization-period 10
mgcp vad
mgcp quarantine mode process
mgcp package-capability dtmf-package
mgcp package-capability mf-package
mgcp package-capability rtp-package
mgcp package-capability nas-package
mgcp package-capability as-package
mgcp package-capability script-package
mgcp default-package ms-package
no mgcp timer receive-rtcp
no mgcp piggyback message

```

```

mgcp fax t38 inhibit
mgcp profile default
mgcp profile PRI
  call-agent 10.10.0.0 service-type mgcp version 0.1
  port 0/0:15
  port 0/1:15
  port 0/2:15
  port 0/3:15
  port 0/4:15
  port 0/5:15
  port 0/6:15
  port 0/7:15
  port 0/8:15
  port 0/9:15
  port 0/10:15
  port 0/11:15
gateway
!
line con 0
  exec-timeout 0 0
  logging synchronous
line aux 0
  exec-timeout 0 0
  logging synchronous
line vty 0 4
  password lab
  no login
line 2/00 5/323
  flush-at-activation
  modem InOut
  no modem status-poll
  no modem log rs232
  transport input all
  autoselect ppp

```

Fax Passthrough with H.323 Support Configuration Example

The following is sample output displaying H.323 support on the Cisco AS5850:

```

version 12.2
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname 5850
!
no logging buffered
no logging rate-limit
!
resource-pool disable
dial-tdm-clock priority 1 trunk-slot 1 port 0
spe link-info poll voice 5
spe default-firmware spe-firmware-1
ip subnet-zero
ip cef distributed
ip ftp username mgcusr
ip ftp password lab
no ip domain lookup
ip host colos_tftp 10.10.0.0
ip host brios 255.255.255.255
ip dhcp smart-relay
!

```

```
isdn switch-type primary-net5
!
voice service voip
  h323
  modem passthrough nse codec g711alaw redundancy sample-duration 20
!
no voice hpi capture buffer
no voice hpi capture destination
!
mrcp client session history duration 0
mrcp client session history records 0
memory check-interval 3600
memory validate-checksum 7200
redundancy
  no keepalive-enable
  mode classic-split
!
controller E1 0/0
  pri-group timeslots 1-31
!
dial-peer voice 5001 pots
  incoming called-number 550
  destination-pattern 800
  direct-inward-dial
  port 0/0:D
  prefix 800
!
dial-peer voice 500 voip
  incoming called-number 800
  destination-pattern 550
  session target ipv4:10.10.0.0
  fax rate disable
  codec g726r32
!
gateway
!
line con 0
  exec-timeout 0 0
  logging synchronous
line aux 0
  exec-timeout 0 0
  logging synchronous
line vty 0 4
  password lab
  no login
line 2/00 5/323
  flush-at-activation
  no modem status-poll
  no modem log rs232
```

Fax Passthrough with SIP Support Configuration Example

The following is sample output displaying SIP support on the Cisco AS5850:

```
version 12.2
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname 5850
!
no logging buffered
no logging rate-limit
!
resource-pool disable
dial-tdm-clock priority 1 trunk-slot 1 port 0
spe link-info poll voice 5
spe default-firmware spe-firmware-1
ip subnet-zero
ip cef distributed
ip ftp username mgcusr
ip ftp password lab
no ip domain lookup
ip host colos_tftp 10.10.0.0
ip host brios 255.255.255.255
ip dhcp smart-relay
!
isdn switch-type primary-net5
!
voice service voip
  h323
  modem passthrough nse codec g711alaw redundancy sample-duration 20
!
no voice hpi capture buffer
no voice hpi capture destination
!
mrcp client session history duration 0
mrcp client session history records 0
memory check-interval 3600
memory validate-checksum 7200
redundancy
  no keepalive-enable
  mode classic-split
!
controller E1 0/0
  pri-group timeslots 1-31
!
dial-peer voice 5001 pots
  incoming called-number 550
  destination-pattern 800
  direct-inward-dial
  port 0/0:D
  prefix 800
!
dial-peer voice 500 voip
  incoming called-number 800
  destination-pattern 550
  session target ipv4:10.10.0.0
  session protocol sipv2
  fax rate disable
  codec g726r32
!
```

```

gateway
line con 0
  exec-timeout 0 0
  logging synchronous
line aux 0
  exec-timeout 0 0
  logging synchronous
line vty 0 4
  password lab
  no login
line 2/00 5/323
  flush-at-activation
  no modem status-poll
  no modem log rs232

```

Fax Passthrough with MGCP Support Configuration Example

The following is sample output displaying MGCP support on the Cisco AS5850:

```

version 12.2
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname 5850
!
no logging buffered
no logging rate-limit
!
resource-pool disable
resource-pool call treatment resource channel-not-available
resource-pool call treatment profile busy
!
clock calendar-valid
dial-tdm-clock priority 1 trunk-slot 1 port 0
spe link-info poll voice 5
spe default-firmware spe-firmware-1
spe 2/02
  shutdown
spe 2/03 2/05
  busyout
spe 2/08
  shutdown
spe 2/09 2/11
  busyout
spe 2/14
  shutdown
spe 2/15 2/17
  busyout
!
ip subnet-zero
ip cef distributed
ip ftp username mgcusr
ip ftp password lab
no ip domain-lookup
ip host brios 255.255.255.255
ip host colos_tftp 10.10.0.0
ip dhcp smart-relay
!

```

```
backhaul-session-manager
  set set1 client nft
  group group1 set set1
  session group group1 10.10.0.0 6667 10.10.0.0 6667 1
!
isdn switch-type primary-net5
no voice hpi capture buffer
no voice hpi capture destination
!
mrcc client session history duration 0
mrcc client session history records 0
memory check-interval 3600
memory validate-checksum 7200
redundancy
  no keepalive-enable
  mode classic-split
!
controller E1 0/1
  pri-group timeslots 1-31 service mgcp
!
interface Serial0/0:15
  no ip address
  isdn switch-type primary-net5
  isdn incoming-voice modem
  isdn bind-13 backhaul set1
  no cdp enable
!
mgcp modem passthrough voip mode nse
mgcp modem passthrough voip codec g711alaw
mgcp codec g729r8 packetization-period 10
mgcp vad
mgcp quarantine mode process
mgcp package-capability dtmf-package
mgcp package-capability mf-package
mgcp package-capability rtp-package
mgcp package-capability nas-package
mgcp package-capability as-package
mgcp package-capability script-package
mgcp default-package ms-package
no mgcp timer receive-rtcp
no mgcp piggyback message
mgcp fax t38 inhibit
!
mgcp profile default
mgcp profile PRI
  call-agent 10.10.0.0 service-type mgcp version 0.1
  port 0/0:15
  port 0/1:15
  port 0/2:15
  port 0/3:15
  port 0/4:15
  port 0/5:15
  port 0/6:15
  port 0/7:15
  port 0/8:15
  port 0/9:15
  port 0/10:15
  port 0/11:15
gateway
!
line con 0
  exec-timeout 0 0
  logging synchronous
```

```
line aux 0
  exec-timeout 0 0
  logging synchronous
line vty 0 4
  password lab
  no login
line 2/00 5/323
  flush-at-activation
  modem InOut
  no modem status-poll
  no modem log rs232
  transport input all
  autoselect ppp
!
```

Command Reference

This section documents the new [mgcp rtp payload-type](#) command. All other commands used with this feature are documented in the Cisco IOS Release 12.2 command reference publications.

mgcp rtp payload-type

To specify use of the correct real-time transport protocol (RTP) payload type for backward compatibility in Media Gateway Control Protocol (MGCP) networks, use the **mgcp rtp payload-type** command in global configuration mode. To disable the backward compatibility, use the no form of this command.

Fax and Modem Codecs

```
mgcp rtp payload-type { cisco-pcm-switch-over-alaw 127 | cisco-pcm-switch-over-ulaw 126 }
```

```
no mgcp rtp payload-type { cisco-pcm-switch-over-alaw 127 | cisco-pcm-switch-over-ulaw 126 }
```

Voice Codecs

```
mgcp rtp payload-type { clear-channel | g726r16 | g726r24 } static
```

```
no mgcp rtp payload-type { clear-channel | g726r16 | g726r24 } static
```

Syntax Description

cisco-pcm-switch-over-alaw 127	Payload type for upspeed to the G.711 a-law codec.
cisco-pcm-switch-over-ulaw 126	Payload type for upspeed to the G.711 u-law codec.
clear-channel	Payload type for clear channel codec.
g726r16	Payload type for the G.726 codec at 16K.
g726r24	Payload type for the G.726 codec at 24K.
static	Static payload type.

Defaults

For fax and modem codecs, the default if this command is not used is a static RTP payload type.

For voice codecs, the default if this command is not used is a dynamic RTP payload type between 96 and 127.

Command Modes

Global configuration

Command History

Release	Modification
12.2(11)T	This command was introduced on the following platforms: Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5400HPX, and Cisco AS5850.

Usage Guidelines

Cisco IOS Release 12.2(11)T introduces a different RTP payload type negotiation for MGCP VoIP calls than was present in previous Cisco IOS images. To ensure interoperability between gateways using different Cisco IOS images, follow these guidelines:

- For fax and modem codecs—If either the originating or terminating MGCP gateway is running Cisco IOS Release 12.2(11)T or a later release and the other gateway is running a release earlier than Cisco IOS Release 12.2(11)T, use the **mgcp rtp payload-type** command on the gateway with the later release.

- For voice codecs—If you are using a Clear Channel, G.726R16, or G.726R24 codec, and either the originating or terminating MGCP gateway is running Cisco IOS Release 12.2(11)T or a later release and the other gateway is running a release earlier than Cisco IOS Release 12.2(11)T, use the **mgcp rtp payload-type** command on the gateway with the later release.

If both the originating and terminating gateways are using Cisco IOS Release 12.2(11)T or a later version, this command is not required.

Examples

The following example specifies use of dynamic RTP payload type for fax and modem calls for mu-law PCM calls in an MGCP network in which the other gateway is running a version of Cisco IOS that is earlier than Release 12.2(11)T:

```
mgcp rtp payload-type cisco-pcm-switch-over-ulaw 126
```

The following example specifies use of a static RTP payload type for a G.726R16 codec in an MGCP network in which the other gateway is running a version of Cisco IOS that is earlier than Release 12.2(11)T:

```
mgcp rtp payload-type g726r16 static
```

Related Commands

Command	Description
mgcp codec	Select the default codec type and its optional packetization period value.