



Modem Passthrough over Voice over IP

This feature module describes the Modem Passthrough over Voice over IP (VoIP) feature on Cisco AS5300 universal access server gateways and includes information on the new feature in the following sections:

- Feature Overview, page 1
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- Monitoring and Maintaining Modem Passthrough over VoIP, page 8
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Feature Overview

The Modem Passthrough over VoIP feature provides the transport of modem signals through a packet network by using pulse code modulation (PCM) encoded packets. This feature is available on the Cisco AS5300 Universal Access Server for Cisco IOS Release 12.1(3)T.

The Modem Passthrough over VoIP feature performs the following functions:

- Represses processing functions like compression, echo cancellation, high-pass filter, and voice activity detection (VAD).
- Issues redundant packets to protect against random packet drops.
- Provides static jitter buffers of 200 milliseconds to protect against clock skew.
- Discriminates modem signals from voice and fax signals, indicating the detection of the modem signal across the connection, and placing the connection in a state that transports the signal across the network with the least amount of distortion.
- Reliably maintains a modem connection across the packet network for a long duration under *normal* network conditions.

For further details, the functions of the Modem Passthrough over VoIP feature are described in the following sections.

Modem Tone Detection

The gateway is able to detect modems at speeds up to V.90.

Passthrough Switchover

When the gateway detects a data modem, both the originating gateway and the terminating gateway roll over to G.711. The roll over to G.711 disables the high-pass filter, disables echo cancellation, and disables VAD. At the end of the modem call, the voice ports revert to the prior configuration and the digital signal processor (DSP) goes back to the state before switchover. You can configure the codec by selecting the **g711alaw** or **g711ulaw** option of the **codec** command.

See also the “Configuration Tasks” section later in this document.

Controlled Redundancy

You can enable payload redundancy so that the Modem Passthrough over VoIP switchover causes the gateway to emit redundant packets.

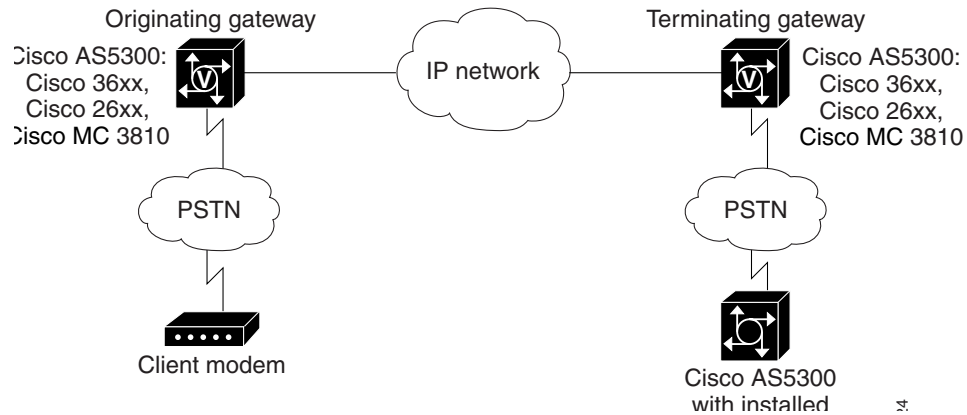
Packet Size

When redundancy is enabled, 10-ms sample-sized packets are sent. When redundancy is disabled, 20-ms sample-sized packets are sent.

Clock Slip Buffer Management

When the gateway detects a data modem, both the originating gateway and the terminating gateway switch from dynamic jitter buffers to static jitter buffers of 200-ms depth. The switch from dynamic to static is to compensate for Public Switched Telephone Network (PSTN) clocking differences at the originating gateway and the terminating gateway. At the conclusion of the modem call, the voice ports revert to dynamic jitter buffers.

Figure 1 illustrates the connection from the client modem to a MICA technologies modem network access server (NAS).

Figure 1 Modem Passthrough Connection

Benefits

The Modem Passthrough over VoIP feature offers the following benefits:

- Detects modem tones
- Passes modem signals over the WAN
- Performs proper switchover to pass modem traffic on a bearer channel
- Detects modems at speeds up to V.90

Restrictions

Cisco IOS Release 12.1(3)T must run on the gateways for the Modem Passthrough over VoIP feature to work.

Related Features and Technologies

- VoIP. For a complete overview of VoIP, refer to the “Configuring Voice over IP” chapter in *Cisco IOS Multiservice Applications Configuration Guide* for Cisco IOS Release 12.1.
- Service Assurance Agent (SAA).

Related Documents

The following documents provide additional platform-specific or hardware-related information to help implement VoIP:

- *Cisco IOS Multiservice Applications Configuration Guide*
- *Cisco IOS Multiservice Applications Command Reference*
- *Installing Voice over IP Feature Cards in Cisco AS5300 Universal Access Servers*
- *Cisco AS5300 Universal Access Server Software Configuration Guide*

Supported Platforms

- Cisco AS5300 universal access server gateways

Supported Standards, MIBs, and RFCs

Standards

ITU-T G.711

MIBs

None

To obtain lists of MIBs supported by platform and Cisco IOS release and to download MIB modules, go to the Cisco MIB web site on Cisco Connection Online (CCO).

RFCs

- RFC 2198—*RTP Payload for Redundant Audio Data*, September 1997, Perkins
- RFC 2833—*RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, February 2000, H. Schulzrinne, Scott Petrack
- RFC 1889—*A Transport Protocol for Real-Time Applications, Audio-Video Transport Working Group*, January 1996, H. Schulzrinne, S. Casner, R. Frederick, and V. Jacobson

Prerequisites

- VoIP enabled network.
- Network suitability to pass modem traffic. The key attributes are packet loss, delay, and jitter. These characteristics of the network can be determined by using the Cisco IOS feature Service Assurance Agent.

Configuration Tasks

By default, modem passthrough over VoIP capability and redundancy are disabled.



Tips

You need to configure modem passthrough in both the originating gateway and the terminating gateway for the Modem Passthrough over VoIP feature to operate. If you configure only one of the gateways in a pair, the modem call will not connect successfully.

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.

See the following sections for the Modem Passthrough over VoIP feature. The two configuration tasks can configure separately or together. If both are configured, the dial-peer configuration takes precedence over the global configuration. Consequently, a call matching a particular dial-peer will first try to apply the modem passthrough configuration on the dial-peer. Then, if a specific dial-peer is not configured, the router will use the global configuration:

- Configuring Modem Passthrough over VoIP Globally, page 5
- Configuring Modem Passthrough over VoIP for a Specific Dial Peer, page 6

Configuring Modem Passthrough over VoIP Globally

For the Modem Passthrough over VoIP feature to operate, you need to configure modem passthrough in both the originating gateway and the terminating gateway so that the modem call matches a voip dial-peer on the gateway.

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 the Modem Passthrough over VoIP feature for all the connections of a Cisco AS5300 universal access server gateway, use the following commands beginning in global configuration mode:

	Command	Purpose
Step 1	Router(config)# voice service voip	Enters voice-service configuration mode. Configures voice service for all the connections for the gateways.
Step 2	Router(conf-voi-serv)# modem passthrough nse [payload-type <i>number</i>] codec { g711ulaw g711alaw } [redundancy] [maximum-sessions <i>value</i>]	Configures the Modem Passthrough over VoIP feature for the Cisco AS5300 universal access server gateways. The default behavior is no modem passthrough . The payload type is an optional parameter for the nse keyword. Use the same payload-type number for both the originating gateway and the terminating gateway. The payload-type number can be set from 96 to 119. If you do not specify the payload-type number , the <i>number</i> defaults to 100. When the payload-type is 100, and you use the show running-config command, the payload-type parameter does not appear. Use the same codec type for both the originating gateway and the terminating gateway. g711ulaw codec is required for T1, and g711alaw codec is required for E1. The redundancy keyword is an optional parameter for sending redundant packets for modem traffic. The maximum-sessions keyword is an optional parameter for the redundancy keyword. This parameter determines the maximum simultaneous modem passthrough sessions with redundancy .
Step 3	Router(conf-voi-serv)# exit	Exits voice-service configuration mode.
Step 4	Router(config)# exit	Exits global configuration mode.

Configuring Modem Passthrough over VoIP for a Specific Dial Peer

You can configure the Modem Passthrough over VoIP feature on a specific dial peer in two ways, as follows:

- Globally in the voice-service configuration mode
- Individually in the dial-peer configuration mode on a specific dial peer

The default behavior for the voice-service configuration mode is **no modem passthrough**. This default behavior implies that modem passthrough is disabled for all dial peers on the gateway by default.

To enable Modem Passthrough on the VoIP dial peers on both the originating and terminating gateway, configure modem passthrough globally or explicitly on the dial peer.

For modem passthrough to operate, you must define VoIP dial peers on both gateways to match the call, for example, by using a destination pattern or an incoming called number. The modem passthrough parameters associated with those dial peers then will apply to the call.

**Note**

When modem passthrough is configured individually for a specific dial peer, that configuration for the specific dial peer takes precedence over the global configuration.

To configure the Modem Passthrough over VoIP feature for a specific dial peer, use 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. Configures a specific dial peer in dial-peer configuration mode.
Step 2	Router(config-dial-peer)# modem passthrough { system nse [payload-type <i>number</i>] codec { g711ulaw g711alaw } [redundancy]}	Configures the Modem Passthrough over VoIP feature for a specific dial peer. The default behavior for the Modem Passthrough for VoIP feature in dial-peer configuration mode is modem passthrough system . As required, the gateway defaults to no modem passthrough . When the system keyword is enabled, the following parameters are not available: nse , payload-type , codec , and redundancy . Instead the values from the global configuration are used. The payload type is an optional parameter for the nse keyword. Use the same payload-type number for both the originating gateway and the terminating gateway. The payload-type number can be set from 96 to 119. If you do not specify the payload-type number , the <i>number</i> defaults to 100. When the payload-type is 100, and you use the show running-config command, the payload-type parameter does not appear. Use the same codec type for both the originating gateway and the terminating gateway. g711ulaw codec is required for T1, and g711alaw codec is required for E1. The redundancy keyword is an optional parameter for sending redundant packets for modem traffic.
Step 3	Router(config-dial-peer)# exit	Exits dial-peer configuration mode and returns to the global configuration mode.
Step 4	Router(config)# exit	Exits global configuration mode.

Verifying Modem Passthrough over VoIP

To verify that the Modem Passthrough over VoIP feature is enabled, perform the following steps:

-
- Step 1** Enter the **show run** command to verify the configuration.
- Step 2** Enter the **show dial-peer voice** command to verify that Modem Passthrough over VoIP is enabled.
-

Troubleshooting Tips

To troubleshoot the Modem Passthrough over VoIP feature, perform the following steps:

- Make sure that you can make a voice call.
- Make sure that Modem Passthrough over VoIP is configured on both the originating gateway and the terminating gateway.
- Make sure that both the originating gateway and the terminating gateway have the same named signaling event (NSE) **payload-type number**.
- Make sure that both the originating gateway and the terminating gateway have the same **maximum-sessions value** when the two gateways are configured in the voice-service configuration mode.
- Use the **debug vtsp dsp** and **debug vtsp session** commands to debug a problem.

Monitoring and Maintaining Modem Passthrough over VoIP

To monitor and maintain the Modem Passthrough over VoIP feature, use the following commands in privileged EXEC mode:

Command	Purpose
Router# show call active { <i>voice</i> <i>fax</i> } [<i>brief</i>]	Displays information for the active call table or displays the voice call history table. The <i>brief</i> option displays a truncated version of either option.
Router# show dial-peer voice [<i>number</i> <i>summary</i>]	Displays configuration information for dial peers. The <i>number</i> argument specifies a specific dial peer from 1 to 32767. The <i>summary</i> option displays a summary of all dial peers.

Configuration Examples

The following is sample configuration for the Modem Passthrough over VoIP feature for the Cisco AS5300 universal access servers:

```
version 12.1
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
voice service voip
  modem passthrough nse codec g711ulaw redundancy maximum-session 5
!
!
resource-pool disable
!
!
!
!
!
ip subnet-zero
ip ftp source-interface Ethernet0
ip ftp username lab
ip ftp password lab
no ip domain-lookup
!
isdn switch-type primary-5ess
cns event-service server
!
!
!
!
mta receive maximum-recipients 0
!
!
controller T1 0
  framing esf
  clock source line primary
  linecode b8zs
  pri-group timeslots 1-24
!
controller T1 1
  shutdown
  clock source line secondary 1
!
controller T1 2
  shutdown
!
controller T1 3
  shutdown
!
!
!
interface Ethernet0
  ip address 1.1.2.2 255.0.0.0
  no ip route-cache
  no ip mroute-cache
!
interface Serial0:23
  no ip address
  encapsulation ppp
  ip mroute-cache
```

```
no logging event link-status
isdn switch-type primary-5ess
isdn incoming-voice modem
no peer default ip address
no fair-queue
no cdp enable
no ppp lcp fast-start
!
interface FastEthernet0
ip address 26.0.0.1 255.0.0.0
no ip route-cache
no ip mroute-cache
load-interval 30
duplex full
speed auto
no cdp enable
!
ip classless
ip route 17.18.0.0 255.255.0.0 1.1.1.1
no ip http server
!
!
!
!
voice-port 0:D
!
dial-peer voice 1 pots
incoming called-number 55511..
destination-pattern 020..
direct-inward-dial
port 0:D
prefix 020
!
dial-peer voice 2 voip
incoming called-number 020..
destination-pattern 55511..
modem passthrough nse codec g711ulaw redundancy
session target ipv4:26.0.0.2
!
!
line con 0
exec-timeout 0 0
transport input none
line aux 0
line vty 0 4
login
!
!
end
```

Command Reference

This section documents new and modified commands. All other commands used with this feature are documented in the *Cisco IOS Multiservice Applications Command Reference* for Cisco IOS Release 12.1.

**Note**

The modified commands are marked by asterisks.

- **modem passthrough (dial-peer)**
- **modem passthrough (voice-service)**
- **show call active ***
- **show call history ***
- **show dial-peer voice ***
- **voice service voip**

modem passthrough (dial-peer)

To configure Modem Passthrough over VoIP for a specific dial peer, use the **modem passthrough** dial-peer configuration command. To use the global default for a specific dial peer, use the **modem passthrough system** command. To disable modem passthrough for a specific dial peer, use the **no modem passthrough** command.

```
modem passthrough {system | nse [payload-type number] codec {g711ulaw | g711alaw}
[redundancy]}
```

```
no modem passthrough
```

Syntax Description

nse	Named signaling event.
payload-type	(Optional) NSE payload-type.
<i>number</i>	(Optional) The value of the payload type (96–119).
codec	Voice compression for speech or audio signals. Codec selections for upspeed. The upspeed method is the method used to dynamically change the codec type and speed to meet network conditions. This means that you might move to a faster codec when you have both voice and data calls, and then slow down when there's only voice traffic.
g711ulaw	Codec G.711 U-law 64000 bits per second for T1.
g711alaw	Codec G.711 a-law 64000 bits per second for E1.
redundancy	(Optional) Packet redundancy (RFC2198) for modem traffic.
system	Defaults to the global configuration.

Defaults

Defining **system** as the method in dial peer points to the voice service VoIP configuration default and is intended to simplify gateway configuration. The default is **modem passthrough system**. As required, the gateway defaults to **no modem passthrough**.

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.1(3)T	This command was introduced for Cisco AS5300.

Usage Guidelines

Use the **modem passthrough** dial-peer configuration command to configure modem passthrough over VoIP for a specific dial peer. The payload type is an optional parameter for the **nse** keyword. Use the same **payload-type number** for both the originating gateway and the terminating gateway. The **payload-type number** can be set from 96 to 119. If you do not specify the **payload-type number**, the *number* defaults to 100.

Use the same codec type for both the originating gateway and the terminating gateway. **g711ulaw** codec is required for T1, and **g711alaw** codec is required for E1.

The **redundancy** keyword is an optional parameter for sending redundant packets for modem traffic.

When the **system** keyword is enabled, the following parameters are not available: **nse**, **payload-type**, **codec**, and **redundancy**. The **system** keyword overrides the configuration for the dial-peer and the values from the global configuration are used.

Examples

The following example shows how Modem Passthrough over VoIP is configured for a specific dial peer in dial-peer configuration mode:

```
Router(config-dial-peer)# modem passthrough nse codec g711ulaw redundancy
```

Related Commands

Command	Description
dial-peer voice	Enters dial-peer configuration mode.

modem passthrough (voice-service)

To configure Modem Passthrough over VoIP for the Cisco AS5300 universal access server gateway, use the **modem passthrough** voice-service configuration command. To disable modem passthrough, use the **no** form of this command.

```
modem passthrough nse [payload-type number] codec {g711ulaw | g711alaw}
[redundancy] [maximum-sessions value]
```

```
no modem passthrough
```

Syntax Description		
nse		Named signaling events.
payload-type		(Optional) NSE payload type.
<i>number</i>		(Optional) The value of the payload type. The number can be from 96 to 119.
codec		Codec selections for upspeed. The upspeed method is the method used to dynamically change the codec type and speed to meet network conditions. This means that you might move to a faster codec when you have both voice and data calls, and then slow down when there's only voice traffic.
g711ulaw		Codec G.711 U-law 64000 bits per second for T1.
g711alaw		Codec G.711 a-law 64000 bits per second for E1.
redundancy		(Optional) Packet redundancy (RFC2198) for modem traffic.
maximum-sessions		(Optional) Maximum number of simultaneous modem passthrough sessions.
<i>value</i>		(Optional) The number of simultaneous modem passthrough sessions. The minimum value is 1 and the maximum value is 26. The default is 16 calls.

Defaults	
	Disabled

Command Modes	
	Voice-service configuration mode

Command History	Release	Modification
	12.1(3)T	This command was introduced for Cisco AS5300.

Usage Guidelines

Use the **modem passthrough** command to configure Modem Passthrough over VoIP for the Cisco AS5300 universal access server gateway. The default behavior for the **voice service voip** command is **no modem passthrough**.

The payload type is an optional parameter for the **nse** keyword. Use the same **payload-type number** for both the originating gateway and the terminating gateway. The **payload-type number** can be set from 96 to 119. If you do not specify the **payload-type number**, the *number* defaults to 100.

Use the same codec type for both the originating gateway and the terminating gateway. **g711ulaw** codec is required for T1, and **g711alaw** codec is required for E1.

The **redundancy** keyword is an optional parameter for sending redundant packets for modem traffic. The **maximum-sessions** keyword is an optional parameter for the **modem passthrough** command. This parameter determines the maximum number of simultaneous modem passthrough sessions. The recommended *value* for the **maximum-sessions** is 16. The value can be set from 1 to 26.

Examples

The following example shows modem passthrough configuration in voice-service configuration mode for NSE payload type 101 using codec G.711:

```
Router(conf-voi-serv)# modem passthrough nse payload-type 101 codec g711ulaw redundancy
maximum-session 1
```

Related Commands

Command	Description
voice service voip	Enters voice-service configuration mode and specifies the voice encapsulation type.

show call active

To show active call information for a call in progress, use the **show call active** privileged EXEC command.

show call active { **voice** | **fax** } [**brief**]

Syntax Description

voice	Specifies that the active call table displays voice call information.
fax	Specifies that the active call table displays fax call information.
brief	(Optional) Displays a truncated version.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3(1)T	This command was introduced.
12.0(4)XJ	This command was modified for Store and Forward Fax.
12.1(3)T	This command was modified for Modem Passthrough over VoIP on Cisco AS5300.

Usage Guidelines

Use the **show call active** privileged EXEC command to display the contents of the active call table. If you use the **voice** keyword, the active call table displays information about all the voice calls currently connected through the router or access server.

Examples

The following is sample output from the **show call active voice** command updated with the modem passthrough output:

```
Router# show call active voice
```

```
GENERIC:
SetupTime=104443 ms
Index=1
PeerAddress=50110
PeerSubAddress=
PeerId=100
PeerIfIndex=104
LogicalIfIndex=10
ConnectTime=104964
CallDuration=00:02:43
CallState=4
CallOrigin=2
ChargedUnits=0
InfoType=2
TransmitPackets=15720
```

```
TransmitBytes=2362904
ReceivePackets=15670
ReceiveBytes=2737904
TELE:
ConnectionId=[0x4B091A27 0x3EDD0003 0x0 0xFEFD4]
TxDuration=155310 ms
VoiceTxDuration=155310 ms
FaxTxDuration=0 ms
CoderTypeRate=g711ulaw
NoiseLevel=-75
ACOMLevel=11
OutSignalLevel=-13
InSignalLevel=-22
InfoActivity=2
ERLLevel=27
SessionTarget=
ImgPages=0
  GENERIC:
SetupTime=104648 ms
Index=1
PeerAddress=55240
PeerSubAddress=
PeerId=2
PeerIfIndex=105
LogicalIfIndex=0
ConnectTime=104964
CallDuration=00:02:47
CallState=4
CallOrigin=1
ChargedUnits=0
InfoType=2
TransmitPackets=16026
TransmitBytes=2608248
ReceivePackets=16075
ReceiveBytes=2609164
VOIP:
ConnectionId[0x4B091A27 0x3EDD0003 0x0 0xFEFD4]
RemoteIPAddress=1.14.82.14
RemoteUDPPort=18202
RoundTripDelay=2 ms
SelectedQoS=best-effort
tx_DtmfRelay=inband-voice
FastConnect=TRUE

SessionProtocol=cisco
SessionTarget=ipv4:1.14.82.14
OnTimeRvPlayout=40
GapFillWithSilence=0 ms
GapFillWithPrediction=0 ms
GapFillWithInterpolation=0 ms
GapFillWithRedundancy=0 ms
HiWaterPlayoutDelay=67 ms
LoWaterPlayoutDelay=67 ms
ReceiveDelay=67 ms
LostPackets=0 ms
EarlyPackets=0 ms
LatePackets=0 ms
VAD = enabled
CoderTypeRate=g729r8
CodecBytes=20
SignalingType=cas
```

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 = 157sec
Time between Buffer Drain/Fills = Min 0sec Max 0sec

```

Router# **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>
  sig:<on/off> <codec> (payload size)
  ATM <protocol> [int vpi/vci cid] vad:<y/n> dtmf:<y/n> seq:<y/n>
  sig:<on/off> <codec> (payload size)
  Tele <int>: tx:<tot>/<v>/<fax>ms <codec> noise:<l> acom:<l> i/o:<l>/<l> dBm

3      : 104443hs.1 +521 pid:100 Answer 50110 active
  dur 00:03:28 tx:20151/3036404 rx:20102/3517936
  Tele 0:D:1: tx:199630/199630/0ms g711ulaw noise:-75 acom:11 i/o:-22/-13 dBm

3      : 104648hs.1 +316 pid:2 Originate 55240 active
  dur 00:03:28 tx:20102/3276712 rx:20151/3277628
  IP 1.14.82.14:18202 rtt:3ms pl:40/0ms lost:0/0/0 delay:67/67/67ms g729r8
MODEMPASS nse buf:0/0 loss 0% 0/0 last 195s dur:0/0s

```

Table 1 provides an alphabetical listing of the **show call active** command fields and a description of each field.

Table 1 *show call active Field Descriptions*

Field	Description
ACOM Level	Current ACOM level for this call. ACOM is the combined loss achieved by the echo canceller, which is the sum of the Echo Return Loss, Echo Return Loss Enhancement, and nonlinear processing loss for the call.
Buffer Drain Events	Total number of jitter buffer drain events.
Buffer Fill Events	Total number of jitter buffer fill events.
CallDuration	Length of the call in hours, minutes and seconds, hh:mm:ss.
CallOrigin	Call origin: answer or originate.
CallState	Current state of the call.
ChargedUnits	Total number of charging units applying to this peer since system startup. The unit of measure for this field is in hundredths of seconds.
CodecBytes	Payload size in bytes for the codec used.
CoderTypeRate	Negotiated coder rate. This value specifies the send rate of voice/fax compression to its associated call leg for this call.

Table 1 show call active Field Descriptions (continued)

Field	Description
ConnectionId	Global call identifier for this gateway call.
ConnectTime	Time when the call was connected.
Consecutive-packets-lost Events	Total number of consecutive (two-or-more) packet loss events.
Corrected packet-loss Events	Total number of packet loss events that were corrected using the RFC 2198 method.
Dial-Peer	Tag of the dial peer sending this call.
ERLLevel	Current Echo Return Loss (ERL) level for this call.
FaxTxDuration	Duration of fax transmission from this peer to voice gateway for this call. You can derive the Fax Utilization Rate by dividing the FaxTxDuration value by the TxDuration value.
GapFillWithInterpolation	Duration of the voice signal played out with a signal synthesized from parameters, or samples of data preceding and following in time because voice data was lost or not received in time from the voice gateway for this call.
GapFillWithRedundancy	Duration of the voice signal played out with a signal synthesized from redundancy parameters available because voice data was lost or not received in time from the voice gateway for this call.
GapFillWithPrediction	Duration of the voice signal played out with signal synthesized from parameters, or samples of data preceding in time because voice data was lost or not received in time from the voice gateway for this call. Examples of such pullout are frame-eraser or frame-concealment strategies in G.729 and G.723.1 compression algorithms.
GapFillWithSilence	Duration of voice signal replaced with silence because voice data was lost or not received in time for this call.
HiWaterPayoutDelay	High-water mark Voice Payout FIFO Delay during this call.
Index	Dial peer identification number.
InfoActivity	Active information transfer activity state for this call.
InfoType	Information type for this call, for example, voice or fax.
InSignalLevel	Active input signal level from the telephony interface used by this call.
Last Buffer Drain/Fill Event	Time since the last jitter buffer drain/fill event, in seconds.
LogicalIfIndex	Index number of the logical interface for this call.
LoWaterPayoutDelay	Low water mark Voice Payout FIFO Delay during this call.
Modem passthrough signaling method in use	Indicates that this is a modem passthrough call and named signaling events (NSEs)—also called <i>telephone-events</i> in RFC 2833—are used for signalling codec upspeed. The upspeed method is the method used to dynamically change the codec type and speed to meet network conditions. This means that you might move to a faster codec when you have both voice and data calls, and then slow down when there's only voice traffic.

Table 1 *show call active Field Descriptions (continued)*

Field	Description
NoiseLevel	Active noise level for this call.
OnTimeRvPayout	Duration of voice playout from data received on time for this call. Derive the Total Voice Playout Duration for Active Voice by adding the OnTimeRvPayout value to the GapFill values.
OutSignalLevel	Active output signal level to telephony interface used by this call.
PeerAddress	Destination pattern or number associated with this peer.
PeerId	ID value of the peer table entry to which this call was made.
PeerIfIndex	Voice port index number for this peer. For ISDN media, this would be the index number of the B channel used for this call.
PeerSubAddress	Subaddress when this call is connected.
Percent Packet Loss	Total percent packet loss.
ReceiveBytes	Number of bytes received by the peer during this call.
ReceiveDelay	Average Playout FIFO Delay plus the Decoder Delay during this voice call.
ReceivePackets	Number of packets received by this peer during this call.
RemoteIPAddress	Remote system IP address for the VoIP call.
RemoteUDPPort	Remote system UDP listener port to which voice packets are sent.
RoundTripDelay	Voice packet round trip delay between the local and remote system on the IP backbone for this call.
SelectedQoS	Selected RSVP quality of service (QoS) for this call.
SessionProtocol	Session protocol used for an Internet call between the local and remote router through the IP backbone.
SessionTarget	Session target of the peer used for this call.
SetupTime	Value of the system UpTime when the call associated with this entry was started.
SignalingType	Signaling type for this call, for example, channel-associated signaling (CAS) or common-channel signaling (CCS).
Time between Buffer Drain/Fills	Minimum and maximum durations between jitter buffer drain/fill events, in seconds.
TransmitBytes	Number of bytes sent by this peer during this call.
TransmitPackets	Number of packets sent by this peer during this call.
TxDuration	Duration of transmit path open from this peer to the voice gateway for this call.
VAD	Whether voice activation detection (VAD) was enabled for this call.
VoiceTxDuration	Duration of voice transmission from this peer to the voice gateway for this call. Derive the Voice Utilization Rate by dividing the VoiceTxDuration value by the TxDuration value.

Related Commands

Command	Description
show call history voice	Displays the VoIP call history table.
show dial-peer voice	Displays configuration information for dial peers.
show num-exp	Displays how the number expansions are configured in VoIP.
show voice port	Displays configuration information about a specific voice port.

show call history

To display the fax call history table for a fax transmission, use the **show call history** privileged EXEC command.

show call history {**voice** | **fax**}[**last number** | **brief**]

Syntax Description

voice	Specifies that the call history table shows voice call information.
last number	(Optional) Displays the last calls connected, where the number of calls that appear is defined by the <i>number</i> argument. Valid values are from 1 to 2147483647.
brief	(Optional) Displays a truncated version of the call history table.

Defaults

No default behavior or values.

Command Modes

Privileged EXEC

Command History

Release	Modification
11.3(1)T	This command was introduced.
12.0(4)XJ	This command was modified for Store and Forward Fax.
12.1(3)T	This command was modified for Modem Passthrough over Voice over IP on Cisco AS5300 universal access servers.

Usage Guidelines

Use the **show call history voice** privileged EXEC command to display the voice call history table. The call history table contains a listing of all calls connected through this router in descending time order since VoIP was enabled. You can display subsets of the call history table by using specific keywords. To display the last calls connected through this router, use the **last** keyword, and define the number of calls you want to see with the *number* argument. To display a truncated version of the call history table, use the **brief** keyword.

Examples

The following is sample output from the **show call history** command updated with the Modem Passthrough over Voice over IP feature:

```
Router# show call history voice
```

```
GENERIC:
SetupTime=104648 ms
Index=1
PeerAddress=55240
PeerSubAddress=
PeerId=2
PeerIfIndex=105
LogicalIfIndex=0
DisconnectCause=10
```

```
DisconnectText=normal call clearing.
ConnectTime=104964
DisconnectTime=143329
CallDuration=00:06:23
CallOrigin=1
ChargedUnits=0
InfoType=speech
TransmitPackets=37668
TransmitBytes=6157536
ReceivePackets=37717
ReceiveBytes=6158452
VOIP:
ConnectionId[0x4B091A27 0x3EDD0003 0x0 0xFEFD4]
RemoteIPAddress=1.14.82.14
RemoteUDPPort=18202
RoundTripDelay=2 ms
SelectedQoS=best-effort
tx_DtmfRelay=inband-voice
FastConnect=TRUE

SessionProtocol=cisco
SessionTarget=ipv4:1.14.82.14
OnTimeRvPlayout=40
GapFillWithSilence=0 ms
GapFillWithPrediction=0 ms
GapFillWithInterpolation=0 ms
GapFillWithRedundancy=0 ms
HiWaterPlayoutDelay=67 ms
LoWaterPlayoutDelay=67 ms
ReceiveDelay=67 ms
LostPackets=0 ms
EarlyPackets=0 ms
LatePackets=0 ms
VAD = enabled
CoderTypeRate=g729r8
CodecBytes=20
cvVoIPCallHistoryIcpif=0
SignalingType=cas

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 = 373sec
Time between Buffer Drain/Fills = Min 0sec Max 0sec

GENERIC:
SetupTime=104443 ms
Index=2
PeerAddress=50110
PeerSubAddress=
PeerId=100
PeerIfIndex=104
LogicalIfIndex=10
DisconnectCause=10
DisconnectText=normal call clearing.
ConnectTime=104964
DisconnectTime=143330
CallDuration=00:06:23
CallOrigin=2
ChargedUnits=0
```

```

InfoType=speech
TransmitPackets=37717
TransmitBytes=5706436
ReceivePackets=37668
ReceiveBytes=6609552
TELE:
ConnectionId=[0x4B091A27 0x3EDD0003 0x0 0xFEFD4]
TxDuration=375300 ms
VoiceTxDuration=375300 ms
FaxTxDuration=0 ms
CoderTypeRate=g711ulaw
NoiseLevel=-75
ACOMLevel=11
SessionTarget=
ImgPages=0

```

Router# **show call history voice brief**

```

<ID>: <start>hs.<index> +<connect> +<disc> pid:<peer_id> <direction> <addr>
dur hh:mm:ss tx:<packets>/<bytes> rx:<packets>/<bytes> <disc-cause>(<text>)
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>
sig:<on/off> <codec> (payload size)
ATM <protocol> [int vpi/vci cid] vad:<y/n> dtmf:<y/n> seq:<y/n>
sig:<on/off> <codec> (payload size)
Telephony <int>: tx:<tot>/<voice>/<fax>ms <codec> noise:<lvl>dBm acom:<lvl>dBm

```

Table 2 provides an alphabetical listing of the **show call history** command fields and a description of each field.

Table 2 *show call history Field Descriptions*

Field	Description
ACOMLevel	Current ACOM level for this call. ACOM is the combined loss achieved by the echo canceller, which is the sum of the Echo Return Loss, Echo Return Loss Enhancement, and nonlinear processing loss for the call.
Buffer Drain Events	Total number of jitter buffer drain events.
Buffer Fill Events	Total number of jitter buffer fill events.
CallDuration	Length of the call in hours, minutes and seconds, hh:mm:ss.
CallOrigin	Call origin: answer or originate.
ChargedUnits	Total number of charging units applying to this peer since system startup. The unit of measure for this field is in hundredths of seconds.
CodecBytes	Payload size in bytes for the codec used.
CoderTypeRate	Negotiated coder rate. This value specifies the send rate of voice/fax compression to its associated call leg for this call.
ConnectionID	Global call identifier for the gateway call.
ConnectTime	Time when this call was connected.
Consecutive-packets-lost Events	Total number of consecutive (two-or-more) packet loss events.

Table 2 *show call history Field Descriptions (continued)*

Field	Description
Corrected packet-loss Events	Total number of packet loss events that were corrected using the RFC 2198 method.
DisconnectCause	Description explaining why this call was disconnected.
DisconnectText	Descriptive text explaining the disconnect reason.
DisconnectTime	Time when this call was disconnected.
FaxTxDuration	Duration of fax transmission from this peer to voice gateway for this call. You can derive the Fax Utilization Rate by dividing the FaxTxDuration value by the TxDuration value.
GapFillWithInterpolation	Duration of the voice signal played out with a signal synthesized from parameters or samples of data preceding and following in time because voice data was lost or not received in time from the voice gateway for this call.
GapFillWithRedundancy	Duration of the voice signal played out with a signal synthesized from redundancy parameters available because voice data was lost or not received in time from the voice gateway for this call.
GapFillWithSilence	Duration of a voice signal replaced with silence because voice data was lost or not received in time for this call.
GapFillWithPrediction	Duration of a voice signal played out with a signal synthesized from parameters, or samples of data preceding in time because voice data was lost or not received in time from the voice gateway for this call.
HiWaterPayoutDelay	High water mark Voice Payout FIFO Delay during this voice call.
Index	Dial peer identification number.
InfoType	Information type for this call, for example, voice or fax.
Last Buffer Drain/Fill Event	Time since the last jitter buffer drain/fill event, in seconds.
LogicalIfIndex	Index number of the logical voice port for this call.
LoWaterPayoutDelay	Low-water mark Voice Payout FIFO Delay during this voice call.
Modem passthrough signaling method is nse	Indicates that this is a modem passthrough call and named signaling events (NSEs)—also called <i>telephone-events</i> in RFC 2833—are used for signalling codec upspeed. The upspeed method is the method used to dynamically change the codec type and speed to meet network conditions. This means that you might move to a faster codec when you have both voice and data calls, and then slow down when there's only voice traffic.
NoiseLevel	Average noise level for this call.
OnTimeRvPayout	Duration of voice payout from data received on time for this call. Derive the Total Voice Payout Duration for Active Voice by adding the OnTimeRvPayout value to the GapFill values.
Percent Packet Loss	Total percent packet loss.
PeerAddress	Destination pattern or number associated with this peer.
PeerId	ID value of the peer entry table to which this call was made.

Table 2 *show call history Field Descriptions (continued)*

Field	Description
PeerIfIndex	Voice port index number for this peer. For ISDN media, this would be the index number of the B channel used for this call.
PeerSubAddress	Subaddress where this call is connected.
ReceiveBytes	Number of bytes received by the peer during this call.
ReceiveDelay	Average Playout FIFO Delay plus the Decoder Delay during this voice call.
ReceivePackets	Number of packets received by this peer during this call.
RemoteIPAddress	Remote system IP address for this call.
RemoteUDPPort	Remote system UDP listener port to which voice packets are sent.
RoundTripDelay	Voice packet round trip delay between the local and remote system on the IP backbone for this call.
SelectedQoS	Selected RSVP QoS for this call.
Session Protocol	Session protocol used for an Internet call between the local and remote router through the IP backbone.
Session Target	Session target of the peer used for this call.
SetUpTime	Value of the system UpTime when the call associated with this entry was started.
SignalingType	Signaling type for this call, for example, channel-associated signaling (CAS) or common-channel signaling (CCS).
Time between Buffer Drain/Fills	Minimum and maximum durations between jitter buffer drain/fill events, in seconds.
TransmitBytes	Number of bytes sent by this peer during this call.
TransmitPackets	Number of packets sent by this peer during this call.
TxDuration	Duration of the transmit path open from this peer to the voice gateway for this call.
VAD	Whether voice activation detection (VAD) was enabled for this call.
VoiceTxDuration	Duration of voice transmission from this peer to the voice gateway for this call. Derive the Voice Utilization Rate by dividing the VoiceTxDuration value by the TxDuration value.

Related Commands

Command	Description
show call active	Displays active call information for a call in progress.

show dial-peer voice

To display configuration information for dial peers, use the **show dial-peer voice** privileged EXEC command.

show dial-peer voice [*number* | **summary**]

Syntax Description	
<i>number</i>	(Optional) A specific dial peer. By using this option, you can see configuration information for a single dial peer identified by the <i>number</i> argument. Valid entries are any integers that identify a specific dial peer from 1 to 32767.
summary	(Optional) Displays a summary of all voice dial peers.

Defaults No default behavior or values.

Command Modes Privileged EXEC

Command History	Release	Modification
	11.3(1)T	This command was introduced.
	11.3(1)MA	The summary keyword was added for the Cisco MC3810 series.
	12.0(3)XG	This command was modified to support Voice over Frame Relay for the Cisco 2600 series and Cisco 3600 series routers.
	12.0(4)T	Support was added for VoFR for the Cisco 7200 series routers.
	12.1(3)T	This command was modified for Modem Passthrough over Voice over IP on the Cisco AS5300.

Usage Guidelines Use the **show dial-peer voice** privileged EXEC command to display the configuration for all VoIP and POTS dial peers configured for the router. To show configuration information for only one specific dial peer, use the *number* argument to identify the dial peer.

Examples The following is sample output from the **show dial-peer voice** command for a POTS dial peer updated with the Modem Passthrough over Voice over IP feature:

```
Router# show dial-peer voice

VoiceEncapPeer100
  information type = voice,
  tag = 100, destination-pattern = `55250',
  answer-address = `', preference=0,
  numbering Type = `unknown'
  group = 100, Admin state is up, Operation state is up,
  incoming called-number = `', connections/maximum = 0/unlimited,
  DTMF Relay = disabled,
  huntstop = disabled,
```

```

application associated:
type = pots, prefix = `5250',
forward-digits default
session-target = `', voice-port = `0:D',
direct-inward-dial = disabled,
digit_strip = enabled,

register E.164 number with GK = TRUE
Connect Time = 0, Charged Units = 0,
Successful Calls = 0, Failed Calls = 0,
Accepted Calls = 0, Refused Calls = 0,
Last Disconnect Cause is "",
Last Disconnect Text is "",
Last Setup Time = 0.
VoiceOverIpPeer2
information type = voice,
tag = 2, destination-pattern = `5240',
answer-address = `', preference=0,
numbering Type = `unknown'
group = 2, Admin state is up, Operation state is up,
incoming called-number = `', connections/maximum = 0/unlimited,
DTMF Relay = disabled,
modem passthrough = nse, payload type = 117, codec = g711ulaw, redundancy,
huntstop = disabled,
application associated:
type = voip, session-target = `ipv4:1.14.82.14',
technology prefix:
settle-call = disabled
ip precedence = 0, UDP checksum = disabled,
session-protocol = cisco, session-transport = udp, req-qos = best-effort,
acc-qos = best-effort,
fax-rate = disable, payload size = 20 bytes
codec = g729r8, payload size = 20 bytes,
Expect factor = 10, Icpif = 30,
Playout: Mode adaptive,
Expect factor = 10,
Max Redirects = 1, Icpif = 30,signaling-type = cas,

```

Table 3 provides an alphabetical listing of the **show dial-peer voice** command field and a description of each field.

Table 3 *show dial-peer voice Field Descriptions*

Field	Description
Accepted Calls	Number of calls from this peer accepted since system startup.
acc-qos	Lowest acceptable quality of service configured for calls for this peer.
Admin state	Administrative state of this peer.
answer-address	Answer address configured for this dial peer.
Charged Units	Total number of charging units applying to this peer since system startup. The unit of measure for this field is in hundredths of seconds.
codec	Default voice coder rate of speech for this peer.
Connect Time	Accumulated connect time to the peer since system startup for both incoming and outgoing calls. The unit of measure for this field is in hundredths of seconds.
dest-pat	Destination pattern (telephone number) for this peer.

Table 3 show dial-peer voice Field Descriptions (continued)

Field	Description
DTMF Relay	Indicates whether or not dual-tone multifrequency (DTMF) relay has been enabled, by using the dtmf-relay command, for this dial peer.
Expect factor	User-requested Expectation Factor of voice quality for calls through this peer.
Failed Calls	Number of failed call attempts to this peer since system startup.
group	Group number associated with this peer.
huntstop	Indicates whether dial-peer hunting has been turned on, by using the huntstop command, for this dial peer.
Icpif	Configured Calculated Planning Impairment Factor (ICPIF) value for calls sent by a dial peer.
incall-number	Full E.164 telephone number to be used to identify the dial peer.
incoming called-number	Indicates the incoming called number if it has been set by using the incoming-called number command.
information type	Information type for this call, for example, voice or fax.
Last Disconnect Cause	Encoded network cause associated with the last call. This value will be updated whenever a call is started or cleared and depends on the interface type and session protocol being used on this interface.
Last Disconnect Text	ASCII text describing the reason for the last call termination.
Last Setup Time	Value of the System Up Time when the last call to this peer was started.
Modem passthrough	Modem passthrough signalling method is named signaling event (NSE).
Operation state	Operational state of this peer.
Payload type	NSE payload type.
Permission	Configured permission level for this peer.
Poor QOV Trap	Whether Poor Quality of Voice trap messages have been enabled or disabled.
Redundancy	Packet redundancy (RFC 2198) for modem traffic.
Refused Calls	Number of calls from this peer refused since system startup.
req-qos	Configured requested quality of service for calls for this dial peer.
session-target	Session target of this peer.
sess-proto	Session protocol to be used for Internet calls between local and remote routers through the IP backbone.
Successful Calls	Number of completed calls to this peer.
tag	Unique dial peer ID number.
VAD	Whether voice activation detection (VAD) is enabled for this dial peer.

Related Commands

Command	Description
show call active voice	Displays the VoIP active call table.
show call history voice	Displays the VoIP call history table.

Command	Description
show num-exp	Displays how the number expansions are configured in VoIP.
show voice port	Displays configuration information about a specific voice port.

voice service voip

To enter voice-service configuration mode and specify the voice encapsulation type, use the **voice service** global configuration command.

voice service voip

Syntax Description	voip	Specifies VoIP encapsulation.
---------------------------	-------------	-------------------------------

Defaults No default behavior or values.

Command Modes Global configuration

Command History	Release	Modification
	12.1(1)XA	This command was introduced for Voice over ATM on the Cisco MC3810 series.
	12.1(2)T	This command was implemented for VoIP in Cisco IOS Release 12.1(2)T on the Cisco MC3810.
	12.1(3)T	This command was implemented in Cisco IOS Release 12.1(3)T for VoIP on the Cisco AS5300 universal access servers.

Usage Guidelines Use the **voice service** command to switch to voice-service configuration mode from global configuration mode and to specify a voice encapsulation type. Use the **exit** command to exit the voice-service configuration mode and return to the global configuration mode.

Examples The following example shows how to access voice-service configuration mode and specify VoIP voice encapsulation beginning in global configuration mode:

```
Router(config)# voice service voip
```

```
Router(conf-voi-serv)#
```

Related Commands	Command	Description
	modem passthrough	Configures Modem Passthrough over VoIP.

Glossary

DSP—digital signal processor.

DSPWare—The firmware running on the DSP coprocessor.

Modem Passthrough—The transport of modem signals through a packet network by using PCM encoded packets.

NAS—network access server.

NSE—named signaling event.

PCM—pulse code modulation.

PSTN—Public Switched Telephone Network.

SAA—Service Assurance Agent.

VAD—voice activity detection.