

SIP-to-SIP Extended Feature Functionality for Session Border Controllers

The SIP-to-SIP Extended Feature Functionality for Session Border Controllers (SBCs) enables the SIP-to-SIP functionality to conform with RFC 3261 to interoperate with SIP User Agents (UAs). The SIP-to-SIP Extended Feature Functionality includes:

- Call Admission Control (based on CPU, memory, and total calls)
- Delayed Media Call
- ENUM support
- Configuring SIP Error Message Pass Through
- Interoperability with Cisco Unified Communications Manager 5.0 and BroadSoft
- Lawful Intercept
- Media Inactivity
- Modem Passthrough over VoIP, on page 2
- TCP and UDP interworking
- Tcl scripts with SIP NOTIFY VoiceXML with SIP-to-SIP
- Transport Layer Security (TLS)

- Feature Information for SIP-to-SIP Extended Feature Functionality for Session Border Controllers, on page 1
- Modem Passthrough over VoIP, on page 2

Feature Information for SIP-to-SIP Extended Feature Functionality for Session Border Controllers

The following table provides release information about the feature or features described in this module. This table lists only the software release that introduced support for a given feature in a given software release train. Unless noted otherwise, subsequent releases of that software release train also support that feature.

Use Cisco Feature Navigator to find information about platform support and Cisco software image support. To access Cisco Feature Navigator, go to https://cfnng.cisco.com/. An account on Cisco.com is not required.

Table 1: Feature Information for Configuring SIP-to-SIP Extended Feature Functionality for Session Border Controllers

Feature Name	Releases	Feature Information
SIP-to-SIP Extended Feature Functionality for Session Border Controllers	Baseline Functionality	The SIP-to-SIP Extended Feature Functionality for Session Border Controllers (SBCs) enables the SIP-to-SIP functionality to conform with RFC 3261 to interoperate with SIP User Agents (UAs). The following commands were introduced or modified: **modem passthrough (dial-peer)**; **modem passthrough (voice-service)**; **show call active voice voice**; **show call history voice** ; **show dial-peer voice**; **voice service**

Modem Passthrough over VoIP

The Modem Passthrough over VoIP feature provides the transport of modem signals through a packet network by using pulse code modulation (PCM) encoded packets.

Prerequisites for the Modem Passthrough over VoIP Feature

- VoIP enabled network.
- Cisco IOS release version must run on the gateways for the Modem Passthrough over VoIP feature to work.
- 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.

Restrictions for the Modem Passthrough over VoIP Feature

Cisco Unified Border Element (Enterprise)

- If call started as g729, upon modem tone (2100Hz) detection both the outgoing gateway (OGW) and the trunking gateway (TGW) will genearate NSE packets towards peer side and up speed to g711 as Cisco UBE(Enterprise) passes these packets to the peer side.

Note That OGW and TGW display the new codec, but the Cisco UBE (Enterprise) continues to show the original codec g729 in the show commands.

Information about Configuring Modem Passthrough over VoIP

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 How to Configure Modem Passthrough over VoIP, on page 4 section 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

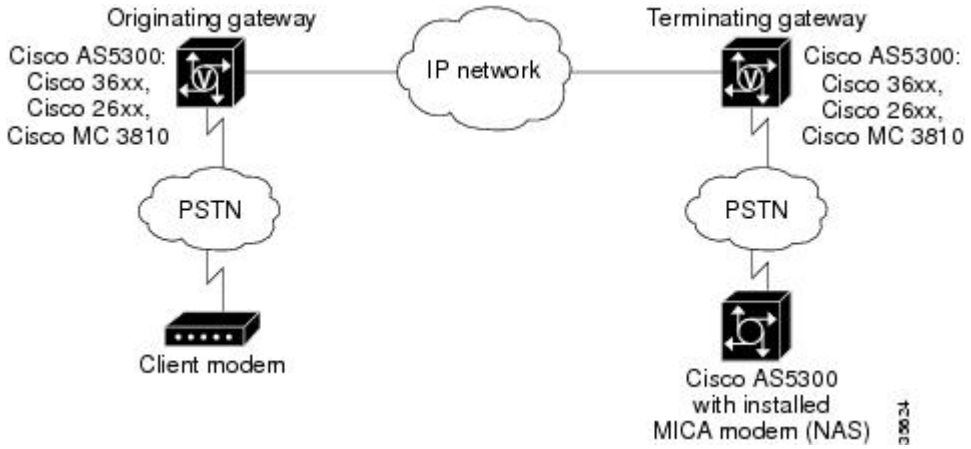
When redundancy is enabled, 10-ms sample-sized packets are sent. When redundancy is disabled, 20-ms sample-sized packets are sent.

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.

The figure below illustrates the connection from the client modem to a MICA technologies modem network access server (NAS).

Figure 1: Modem Passthrough Connection



How to Configure Modem Passthrough over VoIP

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

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

Tip 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

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.

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.

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.

To configure the Modem Passthrough over VoIP feature for all the connections of a gateway, use the following commands beginning in global configuration mode:

SUMMARY STEPS

1. **enable**
2. **voice service voip**
3. **modem passthrough nse** [**payload-type** *number*] **codec** {**g711ulaw** | **g711alaw**} [**redundancy**] [**maximum-sessions** *value*]
4. **exit**
5. **exit**

DETAILED STEPS

Procedure

	Command or Action	Purpose
Step 1	**enable** **Example:** `Device> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**voice service voip** **Example:** `Device(config)# voice service voip`	Enters voice-service configuration mode. Configures voice service for all the connections for the gateways.
Step 3	**modem passthrough nse** [**payload-type** *number*] **codec** {**g711ulaw** | **g711alaw**} [**redundancy**] [**maximum-sessions** *value*] **Example:** `Device(config)# Router(conf-voi-serv)# modem passthrough nse payload-type 97 codec g711alaw redundancy maximum-sessions 3`	Configures the Modem Passthrough over VoIP feature 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 *number*defaults to 100. When the **payload-type** is 100, and you use the

	Command or Action	Purpose
		show running-configcommand, 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 4	**exit** **Example:** `Device(conf-voi-serv)# exit`	Exits voice-service configuration mode.
Step 5	**exit** **Example:** `Device(config)# exit`	Exits global configuration mode.

Configuring Modem Passthrough over VoIP for a Specific Dial Peer

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:

SUMMARY STEPS

1. **enable**
2. **dial-peer voice** *number* **voip**
3. **modem passthrough** {**system** | **nse** [**payload-type** *number*] **codec** {**g711ulaw** | **g711alaw**}[**redundancy**]}
4. **exit**
5. **exit**

DETAILED STEPS

Procedure

	Command or Action	Purpose
Step 1	**enable** **Example:** `Device> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**dial-peer voice** *number* **voip** **Example:** `Device(config)# dial-peer voice 5 voip`	Enters dial-peer configuration mode. Configures a specific dial peer in dial-peer configuration mode.
Step 3	**modem passthrough** {**system** | **nse** [**payload-type** *number*] **codec** {**g711ulaw** | **g711alaw**}[**redundancy**]} **Example:** `Device(config-dial-peer)# modem passthrough nse payload-type 97 codec 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 *number*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 4	**exit** **Example:** `Device(config-dial-peer)# exit`	Exits dial-peer configuration mode and returns to the global configuration mode.
Step 5	**exit** **Example:** `Device(config)# exit`	Exits global configuration mode.

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.

Verifying Modem Passthrough over VoIP

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

SUMMARY STEPS

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

DETAILED STEPS

Procedure

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.

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
`Device# show call active voice brief`	Displays information for the active call table or displays the voice call history table. The brief option displays a truncated version of either option.
`Device# show dial-peer voice 15 summary`	Displays configuration information for dial peers. The *number* argument specifies a specific dial peer from 1 to 32767. The summary option displays a summary of all dial peers.

Configuration Examples

The following is sample configuration for the Modem Passthrough over VoIP feature:

```
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
```