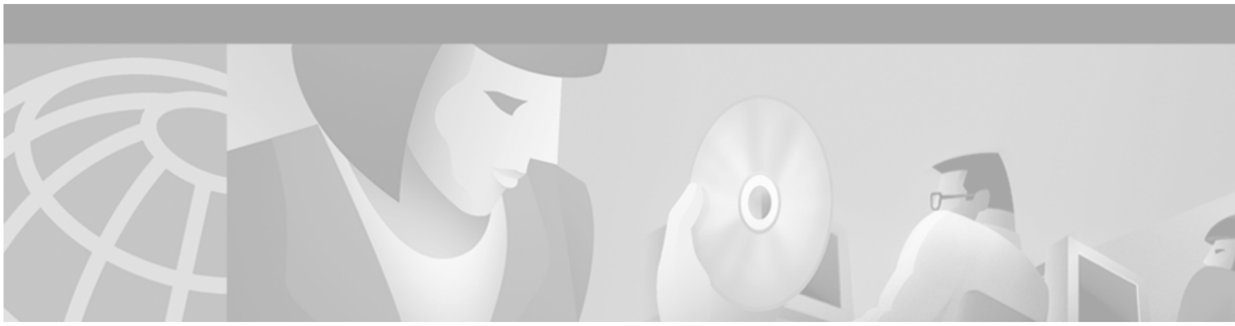

Interworking Signaling Enhancements for H.323 and SIP VoIP

Feature History

Release	Modification
12.1(3)XI	This feature was first introduced.
12.1(5)T	This feature was integrated into Cisco IOS Release 12.1(5)T
12.2(2)XA	This feature was implemented on the Cisco AS5400 and Cisco AS5350.
12.2(2)XB1	This feature was implemented on the Cisco AS5850.
12.2(11)T	This feature was integrated into the Cisco IOS Release 12.2(11)T.

This feature module describes enhancements to H.323 and Session Initiation Protocol (SIP) signaling when interworking with ISDN, T1 channel associated signaling (CAS), and E1 R2 services from the Public Switched Telephone Network (PSTN). These enhancements improve the call signaling capabilities between the Cisco VoIP gateway and the telco switch to ensure, for example, that the voice path is established (cut-through) at the appropriate point during call setup and that early alerting (ringing) does not occur.

This document includes the following sections:

Feature Overview

The Interworking Signaling Enhancements for H.323 and SIP VoIP feature enables VoIP networks to properly signal the setup and tear-down of calls, including generating in-band tones and announcements when needed at the originating or terminating switch. When a tone (for example, ringback, busy, reorder) or announcement (for example, "The number you have dialed is no longer in service") is played at the destination switch, the backward voice path from the called party to the calling party is cut-through early, so that the calling party can hear the tone or announcement. To prevent fraudulent calls, the voice path is cut-through in both directions only after the Connect message is received from the destination. The call progress indicator, which signals the availability of in-band communication, is carried end-to-end as required when interworking with ISDN and CAS protocols.

These enhancements prevent unexpected behavior such as early alerting (when an Alert message is returned immediately after a Call Proceeding message is sent), to ensure that the calling party does not hear conflicting call progress information such as a ringback tone followed by a busy tone, and does not miss hearing a tone or announcement when one should play. In addition, support for network-side ISDN and reducing the risk of speech clipping is addressed.

This feature set provides:

- End-to-end transport of Progress message with progress indicator
- Generation of in-band progress tones and announcements at appropriate switch
- Configuration of progress indicator information element (IE) at the H.323 gateway
- Cut-through of voice path at the appropriate point of a call
- Support for network-side ISDN, including disconnect with locally generated tones
- Initiation of H.245 signaling at originating gateway when Call Proceeding message is received
- Support for SIP 183 Session Progress message for Early Media cut-through

In-Band Tones and Announcements

In-band progress tones and announcements are required for PSTN services and for ISDN speech and 3.1 kHz audio services, per Bellcore and ANSI specifications. To guarantee that in-band tones and announcements are generated when required and at the appropriate switch, this feature set ensures that the progress indicator (PI) is carried end-to-end in call signaling messages between the called party and the calling party. You can also configure the progress indicator in outbound dial peers at the H.323 VoIP gateway, if necessary.

The progress indicator is an information element (IE) that signals when in-band tones and announcements are available. The progress indicator controls whether the local switch generates the appropriate tone or announcement or whether the remote switch is responsible. For example, if the terminating switch generates the ringback tone, it sends a progress indicator of 1 or 8 in the Alerting message. If the originating switch receives an Alerting message without a progress indicator, it generates the ringback tone.

The specific progress indicator that a switch sends in call messages, if any, depends on the model of the switch. To ensure that in-band communication is generated appropriately, it may be necessary in some instances to override the default behavior of the switch by manually configuring the progress indicator at the Cisco H.323 gateway.

The progress indicator is configurable in Setup messages from the outbound VoIP dial peer, typically at the originating gateway, and in Alert, Progress, and Connect messages from the outbound POTS dial peer, typically at the terminating gateway. The progress indicator is configured by using the **progress_ind** dial-peer configuration command.

Table 1 shows the progress indicator (PI) values that you can configure through Cisco IOS software on the H.323 gateway for the different types of messages.

Table 1 Configurable Progress Indicator Values for H.323 Gateways

PI	Description	Message Type
0	No progress indicator is included.	Setup
1	Call is not end-to-end ISDN; further call progress information may be available in-band.	Alert, Setup, Progress, Connect
2	Destination address is non-ISDN.	Alert, Progress, Connect
3	Origination address is non-ISDN.	Setup
8	In-band information or appropriate pattern now available.	Alert, Progress, Connect

When interworking between ISDN and non-ISDN networks:

- If the originating switch does not include a progress indicator in Setup messages, the originating gateway assumes that the originating switch is ISDN and expects the switch to generate the ringback tone. Previously, the originating gateway generated the ringback tone regardless of the PI value in the Setup message. Now, by using the **progress_ind** dial-peer configuration command, you can determine which device generates the ringback tone:
  - To enable the terminating switch to generate the ringback tone, set the progress indicator to 8 in Alert messages on the terminating gateway. The progress indicator is configured in the POTS dial peer.
  - To enable the originating gateway to generate the ringback tone, set the progress indicator to 3 in Setup messages on the originating gateway. The progress indicator is configured in the VoIP dial peer.

  Note If the terminating gateway sends an Alert message with no PI value, the originating gateway generates the ringback tone. But if the terminating gateway sends an Alert message with a PI of 1, 2, or 8, the originating gateway will not generate the ringback tone.

- The originating gateway cuts through the voice path in the backward direction when it receives a Progress or Alert message with a progress indicator of 1, 2, or 8.

Note Pure ISDN calls may use different protocols at the originating and terminating ends, for example, a call originates on ETSI and terminates on NI2. If the two protocols are not compatible end-to-end, the gateway drops all IEs from messages, including the progress indicator. Because a progress indicator is required in all Progress messages, the originating gateway inserts a progress indicator of 1 in the Progress message. To avoid dropping IEs, use the **isdn gateway-max-interworking** global configuration command to prevent the gateway from checking protocol compatibility.

End-to-End Alerting

Early alerting is prevented in these ways:

- For calls terminating at an ISDN switch—The terminating gateway sends an Alert message to the originating gateway only after it receives an Alert message from the terminating switch.
- For calls terminating at a CAS switch—The terminating gateway sends a Progress message to the originating gateway, instead of an Alert message, after it receives a Setup message.

Cut-through of Voice Path

When tones and announcements are generated at the destination switch, the backward voice path from the called party to the calling party is cut-through before the tones and announcements are played. This allows announcements such as, "The number you have called has been changed" or tones for error conditions, such as network congestion, to be forwarded to the calling party. To prevent fraudulent calls, the originating gateway does not perform full cut-through until it receives a Connect message from the destination switch. Cut-through is performed as follows:

- For calls terminating at an ISDN switch—The terminating gateway performs backward cut-through when it receives an Alert or Progress message; full cut-through (both directions) when it receives a Connect message. The originating gateway performs backward cut-through when it receives a Call Proceeding message; full cut-through when it receives a Connect message.
- For calls terminating at a CAS switch—The terminating gateway performs backward cut-through after it sends a Progress message; full cut-through (both directions) when it receives an off-hook signal. The originating gateway performs backward cut-through when it receives a Progress message; full cut-through when it receives a Connect message.

Note If the originating or terminating gateway sends a Call Proceeding message, and then it receives a Call Proceeding message with a progress indicator of 1, 2, or 8, the gateway will convert this Call Proceeding message to a Progress message with a corresponding PI.

ISDN Cause Codes

The cause code is an information element (IE) that indicates why an ISDN call failed or was otherwise disconnected. When the originating gateway receives a Release Complete message, it generates the appropriate tone based on the cause code in the message.

Table 2 lists the default cause codes that the VoIP gateway sends to the switch when a call fails at the gateway, and the corresponding tones that it generates.

Table 2 Cause Codes Generated by the Cisco VoIP Gateway

Cause Code	Description	Explanation	Tone
1	Unallocated (unassigned) number	The ISDN number is not assigned to any destination equipment.	Reorder
3	No route to destination	The call was routed through an intermediate network that does not serve the destination address.	Reorder
16	Normal call clearing	Normal call clearing has occurred.	Dial
17	User busy	The called system acknowledged the connection request but was unable to accept the call because all B channels were in use.	Busy

Table 2 Cause Codes Generated by the Cisco VoIP Gateway (continued)

Cause Code	Description	Explanation	Tone
19	No answer from user (user alerted)	The destination responded to the connection request but failed to complete the connection within the prescribed time. The problem is at the remote end of the connection.	Reorder
28	Invalid number format	The connection could not be established because the destination address was presented in an unrecognizable format or because the destination address was incomplete.	Reorder
34	No circuit/channel available	The connection could not be established because no appropriate channel was available to take the call.	Reorder

For a complete list of ISDN cause codes that are generated by the switch, see Appendix B in the *Cisco IOS Debug Command Reference*, Cisco IOS Release 12.1.

Although the VoIP gateway generates the cause codes listed in Table 2 by default, there are commands introduced in previous Cisco IOS releases that can override these defaults, allowing the gateway to send different cause codes to the switch.

The following commands override the default cause codes:

- **isdn disconnect-cause**—Sends the specified cause code to the switch when a call is disconnected.
- **isdn network-failure-cause**—Sends the specified cause code to the switch when a call fails because of internal network failures.
- **isdn voice-call-failure**—Sends the specified cause code to the switch when an inbound voice call fails with no specific cause code.

When you implement these commands, the configured cause codes are sent to the switch; otherwise, the default cause codes of the voice application are sent. For a complete description of these commands, see the *Cisco IOS Dial Services Command Reference*, Cisco IOS Release 12.1.

ISDN T306 Disconnect Timer and T310 Timer

A new disconnect timer, T306, has been added to allow in-band announcements and tones to be played before a call is disconnected. It is designed for routers that are configured as an ISDN network-side switch. The T306 timer starts when the gateway receives a Disconnect message with a progress indicator of 8. The voice path is cut-through in the backward direction, and the announcement or error tone is played until the timer expires. When the timer expires, the voice application disconnects the call. You can configure this timer by using the **isdn t306** command.

The T310 timer sets a limit for a call in the Call Proceeding state. The timer starts when the router receives a Call Proceeding message and stops when the call moves to another phase, typically Alerting, Connect, or Progress. If the timer expires while the call is in the Call Proceeding state, the router releases the call. You can configure this timer by using the **isdn t310** command.

H.245 Initiation

To avoid speech clipping, H.245 capabilities are now initiated at the originating gateway at the earliest possible moment, when the originating gateway receives a Call Proceeding message from the terminating gateway. Previously, Call Proceeding messages were not passed end-to-end across the VoIP network; H.245 was initiated only after the originating gateway received an Alert message.

Overlap Dialing

To enhance overlap dialing, the Call Proceeding message is now passed transparently from the terminating switch to the originating switch, if the originating switch does not include the Sending Complete information element in the Setup message. The Call Proceeding message notifies the originating switch that the terminating switch has collected all dialed digits that are required to route the call. If the originating switch sends a Sending Complete IE, the originating gateway responds with a Call Proceeding message, and the session application drops the Call Proceeding message sent by the terminating switch.

SIP 183 Session Progress Message

SIP 183 Session Progress messages are supported, facilitating better call treatment for SIP VoIP calls when interworking with PSTN networks. The introduction of the 183 Session Progress message allows a called user agent to suppress local alerting from the calling user agent, and to play a tone or announcement during a preliminary call session, before the full SIP session is set up. This enables the calling party to be notified of the status of the call without being charged for the preliminary portion of the call. A new Session header in the 183 Session Progress message controls whether or not the called user agent plays a tone or announcement for the calling party. The 183 Session Progress message is supported by default and does not require any special configuration.

Table 3 lists ISDN and CAS messages that are sent by the switch, and the corresponding SIP messages that the gateway generates in response.

Table 3 Mapping of ISDN Messages to Outgoing SIP Messages

ISDN/CAS Messages	Outgoing SIP Messages
Setup with PI value 1 or 3	Invite without PI
Alert without PI	180 Ringing without Session Description Protocol (SDP) body
Alert with PI value 8	183 Session Progress with Session header set to Media and SDP body
Progress with PI value 1, 2, 8, or 10	183 Session Progress with Session header set to Media and SDP body
Connect with PI value 2	200 OK without PI
Disconnect with PI value 8	183 Session Progress with Session header set to Media and SDP body (if voice path is not already setup)
Disconnect without PI	Bye

Table 4 lists the SIP messages that are generated by the gateway, and the corresponding ISDN and CAS messages that the switch produces in response.

Table 4 Mapping of Incoming SIP Messages to ISDN Messages

Incoming SIP Messages	ISDN/CAS Messages
Invite	Setup with PI value 1
180 Ringing without SDP body	Alert without PI
183 Session Progress with Session header set to Media and SDP body	Progress with PI value 8

Table 4 Mapping of Incoming SIP Messages to ISDN Messages

Incoming SIP Messages	ISDN/CAS Messages
200 OK	Connect with PI value 2
Bye	Disconnect without PI

Benefits

This feature set ensures that the call signaling for VoIP services is handled properly when interworking with CAS and ISDN networks, resulting in:

- Eliminating early alerting and early ringback
- Generating in-band tones and announcements as required
- Completing bearer transmission path (cut-through) in appropriate way
- Supporting network-side ISDN including disconnect with locally generated tones
- Reducing speech clipping caused by late initiation of H.245
- Enabling SIP called user agent to play call treatment during early media session

Restrictions

- The T306 timer is supported only on routers that are configured for network-side ISDN. The following switches support network-side ISDN:
  - National ISDN
  - NET3 BRI
  - NET5
  - QSIG
- Supplementary voice services are not supported with ISDN and CAS over an H.323 network—except on the NET5 switch.
- Progress messages require a progress indicator value and only ITU-T standards are supported.
- Progress indicator 2 is not supported in Progress messages for the DMS100 switch.
- TCL 2.0 for Interactive Voice Response (IVR) supports the interworking signaling enhancements only on the Cisco AS5300. For IVR on other Cisco platforms, you must select TCL 1.0 as the session application. To use standard IVR applications with TCL 1.0, configure the application name as "session.t.old", by using the **call application voice** global configuration command. It is not necessary to do this if you are using customized scripts.
- The Cisco AS5300 sends a Connect message to the originating gateway after it receives a Setup message only when it is configured for one of the following supported switch types:
  - 5ESS
  - NET5
  - NTT
  - QSIG
  - QSIGP

- For the SS7 interconnect for voice gateways solution, the following behavior applies to Suspend and Resume messages, which are supported on NET5 and NI2+ ISDN interfaces.
  - If the ISDN interface is NET5, the Cisco AS5300 sends a Notify message with the notification indicator (NI) set to user-suspended or user-resumed.
  - If the ISDN interface is NI2+, the Cisco AS5300 sends a Suspend or Resume message to the Cisco SC2200.
  - If the Cisco SC2200 receives an ISUP Suspend or Resume, it sends an NI2+ Suspend or Resume to the Cisco AS5300.
  - Both the Cisco AS5300 and SC2200 start timers when a Suspend message is received. The Cisco AS5300 timer, T307, is configurable from 30 to 300 seconds. The Cisco SC2200 timer, T6, is not configurable and has a default of 120 seconds, if the ISUP variant Q.761 is used.

    When the Cisco AS5300 and the SC2200 receive a Resume message, the timers are stopped. If either of the timers expire, the call is released with a cause code of normal clearing.

Related Features and Technologies

These features are dependent on the interoperability of Service Provider Features for VoIP.

Related Documents

- *Cisco AS5300 Software Configuration Guide*
- *Cisco AS5800 Operations, Administration, Maintenance, and Provisioning Guide*
- *Configuring H.323 VoIP Gateway for Cisco Access Platforms*
- *Debug Command Reference*, Cisco IOS Release 12.1
- *Dial Solutions Configuration Guide: Terminal Services*, Cisco IOS Release 12.1
- *IP and IP Routing Configuration Guide*, Cisco IOS Release 12.1.
- *Multiservice Applications Configuration Guide,* Cisco IOS Release 12.1
- *Session Initiation Protocol for Voice over IP on Cisco Access Platforms*
- *Session Initiation Protocol Gateway Call Flows*
- *Software Configuration Guide for the Cisco 2600 Series and 3600 Series Routers*
- *Using Cisco 2600 and Cisco 3600 Series Routers as H.323 VoIP Gateways*
- *Voice over IP for the Cisco AS5300*
- *Voice over IP for the Cisco AS5800 Software Configuration Guide*

Supported Platforms

- Cisco 2600 series
- Cisco 3600 series
- Cisco AS5300
- Cisco AS5350
- Cisco AS5400
- Cisco AS5800
- Cisco AS5850
- Cisco 7200 series
- Cisco 7500 series
- Cisco MC3810

These features run on all platforms that support Cisco IOS Release 12.1(5)T and VoIP features.

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

No new or modified MIBs are supported by this feature.

To obtain lists of supported MIBs by platform and Cisco IOS release, and to download MIB modules, go to the Cisco MIB web site on Cisco Connection Online (CCO) at http://www.cisco.com/public/sw-center/netmgmt/cmtk/mibs.shtml

RFCs

No new or modified RFCs are supported by this feature.

Prerequisites

To use these features, you must first:

- Configure your VoIP gateways and gatekeepers. For more information about configuring VoIP for your access platform, see the *Cisco IOS Multiservice Applications Configuration Guide*, Cisco IOS Release 12.1.
- Establish a working IP network. For more information about configuring IP, see the *Cisco IOS IP and IP Routing Configuration Guide*, Cisco IOS Release 12.1.

Configuration Tasks

See the following sections for configuring these optional signaling interworking features:

- Configuring Progress Indicator in H.323 VoIP Dial Peers (Optional)
- Configuring Progress Indicator in H.323 POTS Dial Peers (Optional)
- Verifying Progress Indicator Configuration (Optional)
- Configuring ISDN T306 and T310 Timers (Optional)
- Verifying T306 Timer Configuration (Optional)
- Configuring Disconnect With PI (Optional)
- Configuring Maximum Interworking (Optional)
- Configuring Two-Way Voice Path for RTP (Optional)
- Verifying Disconnect with PI, Maximum Interworking, and Two-Way RTP (Optional)

Configuring Progress Indicator in H.323 VoIP Dial Peers

Note This configuration procedure is supported only on VoIP gateways that use the H.323 protocol; it is not supported on gateways that use SIP.

To include a specific progress indicator in Setup messages from the outbound VoIP dial peer on an H.323 gateway, perform the following tasks beginning in global configuration mode:

	Command	Purpose
Step 1	`Router(config)#` **`dial-peer voice`** *`number`* **`voip`**	Enters dial-peer configuration mode and configures a VoIP dial-peer.
Step 2	`Router(config-dial-peer)#` **`destination-pattern`** *`string`*	Specifies the telephone number for this dial peer. A call is matched to this dial peer by using this pattern.
Step 3	`Router(config-dial-peer)#` **`session protocol cisco`**	Sets the session protocol type to Cisco proprietary H.323.
Step 4	`Router(config-dial-peer)#` **`progress_ind setup enable`** *`pi-number`*	Sets the progress indicator in Setup messages.

Configuring Progress Indicator in H.323 POTS Dial Peers

Note This configuration procedure is only supported on VoIP gateways that use the H.323 protocol; it is not supported on gateways that use SIP.

To include a specific progress indicator in Alert, Progress, or Connect messages from the outbound POTS dial peer on an H.323 gateway, perform the following tasks beginning in global configuration mode:

	Command	Purpose
Step 1	`Router(config)#` **`dial-peer voice`** *`number`* **`pots`**	Enters dial-peer configuration mode and configures a POTS dial-peer.
Step 2	`Router(config-dial-peer)#` **`destination-pattern`** *`string`*	Specifies the telephone number for this dial peer. A call is matched to this dial peer using this pattern.
Step 3	`Router(config-dial-peer)#` **`progress_ind alert enable`** *`pi-number`* or `Router(config-dial-peer)#` **`progress_ind progress enable`** *`pi-number`* or `Router(config-dial-peer)#` **`progress_ind connect enable`** *`pi-number`*	Sets the progress indicator for Alert messages. Sets the progress indicator for Progress messages. Sets the progress indicator for Connect messages.

Verifying Progress Indicator Configuration

Perform the following steps in privileged EXEC mode to verify that the progress indicator is configured and operating correctly.

Step 1 Display the running configuration file with the **show running-config** command. Verify that the configuration is accurate for the progress indicator. See the "Configuration Examples" section on page 14 for a sample configuration screen.

Step 2 Enable the **debug isdn q931** command to trace the ISDN messages. Any associated progress indicator is listed along with the messages. Make sure that the progress indicator is carried end-to-end and is not dropped anywhere.

Step 3 Enable the **debug cch323 rtp** command to verify that backward cut-through and full cut-through is performed correctly based on the progress indicator.

Configuring ISDN T306 and T310 Timers

To configure the T306 and T310 timers, perform the following tasks in interface configuration mode:

	Command	Purpose
Step 1	`Router(config)# interface serial controller:timeslot`	Enters interface configuration mode for a D-channel serial interface.
Step 2	`Router(config-if)# isdn t306 msecs`	Sets the number of milliseconds that the gateway waits before clearing a call after it receives a Disconnect message with a progress indicator of 8.
Step 3	`Router(config-if)# isdn t310 msecs`	Sets the number of milliseconds that the gateway waits before clearing a call after it receives a Call Proceeding message.

Verifying T306 Timer Configuration

Perform the following steps in privileged EXEC mode to verify that the T306 timer is configured and operating correctly.

Step 1 Display the running configuration file with the **show running-config** command. Verify that the configuration is accurate for the T306 timer. See the "Configuration Examples" section on page 14 for a sample configuration screen.

Step 2 Enable the **debug isdn q931** command to trace the ISDN messages.

Step 3 Place a call to the gateway. Disconnect the call and allow the far end to play its error message until the T306 timer expires. When the timer expires, the gateway should disconnect the call.

Configuring Disconnect With PI

To enable the H.323 gateway to treat a Disconnect message with a progress indicator (PI) the same as a standard Disconnect, perform the following tasks beginning in global configuration mode:

	Command	Purpose
Step 1	`Router(config)#` **`voice-port`** *`slot/port`*	Enters interface configuration mode for a D-channel serial interface. The **voice-port** command syntax is platform-specific. For syntax information by platform, see the Cisco IOS Release 12.1 Multiservice Applications Command Reference.
Step 2	`Router(config-voiceport)#` **`disc_pi_off`**	Enables the originating H.323 gateway to disconnect a call when it receives a Disconnect message with a progress indicator (PI).

Configuring Maximum Interworking

To enable maximum interworking on the H.323 gateway, perform the following task in global configuration mode:

Command	Purpose
`Router(config)#` **`isdn gateway-max-interworking`**	Enables maximum interworking on the H.323 gateway. This prevents the gateway from dropping information elements (IEs) in call messages if the ISDN protocol on the originating switch is different than the protocol on the terminating switch.

Configuring Two-Way Voice Path for RTP

To enable a two-way voice path when the Real-Time Transport Protocol (RTP) channel is opened, perform the following task in global configuration mode:

Command	Purpose
`Router(config)#` **`voice rtp send-recv`**	Establishes a two-way voice path when the RTP channel is opened.

Verifying Disconnect with PI, Maximum Interworking, and Two-Way RTP

To verify that the configuration is accurate, display the running configuration file by using the **show running-config** command.

Troubleshooting Tips

The following table lists some potential configuration issues and their resolutions.

Symptom	Solution
Calling party does not hear ringback tone after alerting	Enable the **debug isdn q931** command to display the ISDN messages and verify the progress indicator values. Do one of the following depending on where you want to generate the ringback tone: • To generate ringback tone at originating gateway: If the Setup message from the originating switch does not contain a PI value, use the **progress_ind** dial-peer command to set the PI to 3. • To generate ringback tone at terminating switch: If the Alert message from the terminating switch does not contain a PI value, use the **progress_ind** dial-peer command to set the PI to 8 at the terminating gateway.
Gateway not responding to Connect after receiving Progress	Enable the **debug isdn q931** command to display the ISDN messages. Verify that a progress indicator is included in the Progress message and that the PI meets ITU-T standards.

Interoperability Issues with Legacy H.323 Gateways

For H.323 gateways that are running Cisco IOS Release 12.1(2)T or earlier:

- Terminating gateways generate early alerting.
- Originating gateways do not always cut-through the voice path on receipt of a Progress message.
- When performing a staged upgrade, you should upgrade the originating gateway first to Cisco IOS Release 12.1(5)T. If you upgrade the terminating gateway first, it may exhibit unexpected behavior when it receives a Progress message.

Configuration Examples

This section provides the following configuration examples:

- Progress Indicator Configuration Example
- T306/T310 Timer Configuration Example
- Disconnect PI Configuration Example
- Maximum Interworking and Two-Way RTP Configuration Example

Progress Indicator Configuration Example

```
!
dial-peer voice 3 pots
 destination-pattern 55275
 session protocol cisco
 progress_ind progress enable 1
 progress_ind connect enable 1
 port 1:0
.
.
```

T306/T310 Timer Configuration Example

```
!
interface Serial0:23
 no ip address
 no ip directed-broadcast
 encapsulation ppp
 dialer rotary-group 0
 isdn switch-type primary-5ess
 isdn incoming-voice modem
 isdn t306 60000
 isdn t310 40000
.
.
```

Disconnect PI Configuration Example

```
!
voice-port 0:D
 disc_pi_off
!
voice-port 1:D
!
voice-port 3:D
.
.
```

Maximum Interworking and Two-Way RTP Configuration Example

```
!
async-bootp dns-server 172.22.53.210
isdn switch-type primary-5ess
isdn voice-call-failure 0
isdn alert-end-to-end
isdn gateway-max-interworking
!
cns event-service server
voice call send-alert
voice rtp send-recv
.
.
```

Command Reference

This section documents new commands. All other commands used with this feature are documented in the Cisco IOS Release 12.1 command reference publications.

- **disc_pi_off**
- **isdn gateway-max-interworking**
- **isdn t306**
- **isdn t310**
- **progress_ind**
- **voice rtp send-recv**

disc_pi_off

To enable an H.323 gateway to disconnect a call when it receives a Disconnect message with a progress indicator (PI) value, use the **disc_pi_off** voice-port configuration command. To restore the default value, use the **no** form of this command.

disc_pi_off

no disc_pi_off

Syntax Description

This command has no keywords or arguments.

Defaults

The gateway does not disconnect a call when it receives a Disconnect message with a PI value.

Command Modes

Voice-port configuration

Command History

Release	Modification
12.1(5)T	This command was introduced.
12.2(2)XA	This command was implemented on the Cisco AS5400 and Cisco AS5350.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(11)T	This command was integrated into the Cisco IOS Release 12.2(11)T.

Usage Guidelines

The **disc_pi_off** voice-port command is only valid if the Disconnect with PI is received on the inbound call leg. For example, if this command is enabled on the voice port of the originating gateway, and a Disconnect with PI is received from the terminating switch, the Disconnect with PI is converted to a Disconnect. But if this command is enabled on the voice port of the terminating gateway, and a Disconnect with PI is received from the terminating switch, the Disconnect message is not converted to a standard Disconnect, because the Disconnect message is received on the outbound call leg.

Note The **disc_pi_off** voice-port command is valid only for the default session application; it does not work for interactive voice response (IVR) applications.

Examples

The following example handles a Disconnect message with a PI value the same as a standard Disconnect message for voice port 0:23:

```
voice-port 0:D
    disc_pi_off
```

Related Commands

Command	Description
isdn t306	Sets a timer for Disconnect messages.

isdn gateway-max-interworking

To prevent the H.323 gateway from checking for ISDN protocol compatibility and dropping information elements (IEs) in call messages, use the **isdn gateway-max-interworking** global configuration command. To restore the default condition, use the **no** form of this command.

isdn gateway-max-interworking

no isdn gateway-max-interworking

Syntax Description

This command has no keywords or arguments.

Defaults

The gateway checks for protocol compatibility

Command Modes

Global configuration

Command History

Release	Modification
12.1(3)XI	This command was introduced.
12.1(5)T	This command was integrated into Cisco IOS Release 12.1(5)T.
12.2(2)XA	This command was implemented on the Cisco AS5400 and Cisco AS5350.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(11)T	This command was integrated into the Cisco IOS Release 12.2(11)T.

Usage Guidelines

If the **isdn gateway-max-interworking** command is enabled on the originating H.323 gateway, the information elements (IEs) in call messages to the terminating gateway are not checked for end-to-end protocol compatibility. If this command is enabled on the terminating gateway, IEs are not checked in the reverse direction. If this command is not enabled, and the ISDN protocols are not compatible on the originating and terminating gateways, the gateway drops all IEs, including the progress indicator. The gateway then inserts a progress indicator of 1 into all Progress messages.

Examples

The following example enables maximum interworking:

```
isdn gateway-max-interworking
```

isdn t306

To set a timer for disconnect messages received by the router, use the **isdn t306** interface configuration command. To restore the default value, use the **default** or **no** form of this command.

isdn t306 *msecs*

default isdn t306

no isdn t306

Syntax Description

msecs	Number of milliseconds that the router waits before disconnecting a call after it receives a disconnect message with a progress indicator of 8. Values are 1 to 400000 ms.

Defaults

The default depends on the switch, usually from 5000 to 30000

Command Modes

Interface configuration

Command History

Release	Modification
12.1(3)XI	This command was introduced.
12.1(5)T	This command was integrated into Cisco IOS Release 12.1(5)T.
12.2(2)XA	This command was implemented on the Cisco AS5400 and Cisco AS5350.
12.2(2)XB1	This command was implemented on the Cisco AS5850 platform.
12.2(11)T	This command was integrated into the Cisco IOS Release 12.2(11)T.

Usage Guidelines

The T306 timer is designed for routers that are configured as an ISDN network-side switch. When the router receives a disconnect message with a progress indicator of 8, it disconnects the call after waiting for the specified number of ms while the in-band announcement or error tone is playing. Be sure to set the timer long enough for the announcement to be heard or the tone to be recognized. The **isdn t306** command is used only for disconnect messages with a progress indicator of 8; otherwise, the T305 timer is used. The **disable** and **no** forms of this command have the same result: the timer waits for the default number of ms before disconnecting the call.

Examples

The following example sets the T306 timer to 60000 ms for serial interface 0:23:

```
interface serial 0:23
   isdn t306 60000
```

Related Commands

Command	Description
isdn t310	Sets a timer for Call Proceeding messages.

isdn t310

To set a timer for the Call Proceeding state, use the **isdn t310** interface configuration command. To restore the default value, use the **no** form of this command.

isdn t310 *msecs*

no isdn t310

Syntax Description

msecs	Number of milliseconds that the router waits before disconnecting a call after receiving a Call Proceeding message. Values are 1 to 400000 ms.

Defaults

The default depends on the switch, usually from 5000 to 30000

Command Modes

Interface configuration

Command History

Release	Modification
12.1(3)XI	This command was introduced.
12.1(5)T	This command was integrated into Cisco IOS Release 12.1(5)T.
12.2(2)XA	This command was implemented on the Cisco AS5400 and Cisco AS5350.
12.2(2)XB1	This command was implemented on the Cisco AS5850 platform.
12.2(11)T	This command was integrated into the Cisco IOS Release 12.2(11)T.

Usage Guidelines

The T310 timer starts when the router receives a Call Proceeding message; it stops when the call exits the Call Proceeding state, typically when the call moves to Alerting, Connect, or Progress. If the timer expires while the call is in the Call Proceeding state, the router releases the call. Set the timer to match the specific characteristics of your network.

Examples

The following example sets the T310 timer to 40,000 ms for serial interface 0:23:

```
interface serial 0:23
    isdn t310 40000
```

Related Commands

Command	Description
isdn t306	Sets a timer for Disconnect messages.

progress_ind

To set a specific progress indicator (PI) in call Setup, Progress, or Connect messages from an H.323 VoIP gateway, use the **progress_ind** dial-peer configuration command. To restore the default condition, use the **no** or **disable** forms of this command.

progress_ind {**setup** | **connect** | **progress** | **alert**} {**enable** *pi-number* | **disable**}

no progress_ind {**setup** | **connect** | **progress** | **alert**}

Note This command is not supported on VoIP gateways using SIP.

Syntax Description

setup	Sets the progress indicator for Setup messages.
connect	Sets the progress indicator for Connect messages.
progress	Sets the progress indicator for Progress messages.
alert	Sets the progress indicator for Alert messages.
enable	Enables the configuration of the progress indicator.
pi-number	The progress indicator that is sent in all messages of the specified type from the outbound dial peer. For Setup messages from a VoIP dial peer, values are 0, 1, or 3. For Progress, Connect, or Alert messages from a POTS dial peer, values are 1, 2, or 8.
disable	Disables the user configuration of the progress indicator.

Defaults

The default progress indicator from the switch is not intercepted or modified.

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.1(3)XI	This command was introduced.
12.1(5)T	This command was integrated into Cisco IOS Release 12.1(5)T.
12.2(2)XA	This command was implemented on the Cisco AS5400 and Cisco AS5350.
12.2(2)XB1	This command was implemented on the Cisco AS5850 platform.
12.2(11)T	This command was integrated into the Cisco IOS Release 12.2(11)T.

Usage Guidelines

The **progress_ind** command overrides the default progress indicator that is sent by the switch. This enables you to set the progress indicator at the H.323 gateway, if necessary, to ensure the proper end-to-end signaling for VoIP calls. This command sets the progress indicator only in messages from outbound dial peers that have a set destination pattern, configured by using the **destination-pattern** command. If a message contains multiple progress indicators, the **progress_ind** command overrides only the first progress indicator in the message.

The **disable** and **no** forms of the **progress_ind** command have the same result: the call messages are not intercepted by the session application, and the default progress indicator, if any, is forwarded unmodified.

Note A progress indicator that is configured by using the **progress_ind** command will not override the default progress indicator in a Progress message, if the Progress message is sent after backward cut-through has occurred (for example, because an Alert message with a progress indicator of 8 was sent before the Progress message).

Examples

The following example sets the progress indicator to 1 in Progress and Connect messages from the number 3 POTS dial peer:

```
dial-peer voice 3 pots
 destination-pattern 55275
 progress_ind progress enable 1
 progress_ind connect enable 1
```

Related Commands

Command	Description
dial-peer voice	Enters dial-peer configuration mode and configures a VoIP or POTS dial peer.
destination-pattern	Specifies the telephone number that is used to identify the outbound dial peer for the call.

voice rtp send-recv

To establish a two-way voice path when the Real-Time Transport Protocol (RTP) channel is opened, use the **voice rtp send-recv** global configuration command. To restore the default condition, use the **no** form of this command.

voice rtp send-recv

no voice rtp send-recv

Syntax Description

This command has no keywords or arguments.

Defaults

The voice path is cut-through in only the backward direction when the RTP channel is opened.

Command Modes

Global configuration

Command History

Release	Modification
12.1(5)T	This command was introduced.
12.2(2)XA	This command was implemented on the Cisco AS5400 and Cisco AS5350.
12.2(2)XB1	This command was implemented on the Cisco AS5850 platform.
12.2(11)T	This command was integrated into the Cisco IOS Release 12.2(11)T.

Usage Guidelines

The **voice rtp send-recv** command should be enabled only when the voice path must be cut-through (established) in both the backward and forward directions before a Connect message is received from the destination switch. The **voice rtp send-recv** command affects all VoIP calls when it is enabled.

Examples

The following example enables the voice path to cut-through in both directions when the RTP channel is opened:

```
voice rtp send-recv
```

Debug Commands

This section documents the new **debug vtsp tone** command. All other commands used with this feature are documented in the Cisco IOS Release 12.1 command reference publications.

debug vtsp tone

To display debug messages showing the types of tones generated by the VoIP gateway, use the **debug vtsp tone** command. To disable the debug messages, use the **no** form of this command.

debug vtsp tone

no debug vtsp tone

Syntax Description

This command has no keywords or arguments.

Defaults

Tone generation messages are not enabled.

Command History

Release	Modification
12.1(3)XI	This command was introduced.
12.1(5)T	This command was integrated into Cisco IOS Release 12.1(5)T.
12.2(2)XA	This command was implemented on the Cisco AS5400 and Cisco AS5350.
12.2(2)XB1	This command was implemented on the Cisco AS5850 platform.
12.2(11)T	This command was integrated into the Cisco IOS Release 12.2(11)T.

Examples

The following example shows that a ringback tone was generated by the VoIP gateway:

```
Router# debug vtsp tone
*Jan  1 16:33:52.395:act_alert:Tone Ring Back generated in direction Network

*Jan  1 16:33:52.399:ISDN Se0:23:TX ->  ALERTING pd = 8  callref = 0x9816
```

Related Commands

Command	Description
debug vtsp dsp	Shows messages from the Digital Signal Processor (DSP) on the modem to the router.
debug vtsp session	Traces how the router interacts with the Digital Signal Processor (DSP), based on the signaling indications from the signaling stack and requests from the application.

Glossary

CAS—channel associated signaling. Call signaling that enables the access server to send or receive analog calls or calls on digital trunks using robbed-bit signaling.

cause code—Defined by ITU Recommendation Q.850; indicates the reason for ISDN call failure or completion.

cut-through—Completion of the bearer transmission path between the calling party and the called party.

dial peer—An addressable call endpoint. In Voice over IP (VoIP), there are two types of dial peers: POTS and VoIP.

gateway—A gateway allows H.323 terminals to communicate with non-H.323 terminals by converting protocols. A gateway is the point at which a circuit-switched call is encoded and repackaged into IP packets.

H.323—An International Telecommunication Union (ITU-T) standard that describes packet-based video, audio, and data conferencing. H.323 is an umbrella standard that describes the architecture of the conferencing system and refers to a set of other standards (H.245, H.225.0, and Q.931) to describe its actual protocol.

interworking—The mapping of call signaling messages between two different protocol suites.

ISDN—Integrated Services Digital Network. Communication protocol offered by telephone companies that permits telephone networks to carry data, voice, and other source traffic.

POTS—Plain old telephone service. Basic telephone service supplying standard single line telephones, telephone lines, and access to the PSTN.

POTS dial peer—Dial peer connected by a traditional telephony network. POTS peers point to a particular voice port on a voice network device.

progress indicator—An information element (IE) in ISDN messages that indicates when in-band communication is used.

PSTN—Public Switched Telephone Network. General term referring to the variety of telephone networks and services in place worldwide. Sometimes called plain old telephone service (POTS) .

R2—Channelized E1 signaling used in Europe, Asia, and South America. It is equivalent to channelized T1 signaling in North America.

SIP—Session Initiation Protocol. This is a protocol developed by the IETF MMUSIC Working Group as an alternative to H.323. SIP features are compliant with IETF RFC 2543, published in March 1999. SIP equips platforms to signal the setup of voice and multimedia calls over IP networks.

VoIP—Voice over IP. The ability to carry normal telephone-style voice over an IP-based Internet with POTS-like functionality, reliability, and voice quality. VoIP is a blanket term, that generally refers to the Cisco standards-based (for example H.323) approach to IP voice traffic.

VoIP dial peer—Dial peer connected by a packet network; in the case of Voice over IP, this is an IP network. VoIP peers point to specific VoIP devices.