



Interworking Signaling Enhancements for H.323 VoIP

Feature History

Release	Modification
12.1(3)XI	This feature was introduced.
12.1(5)XM2	Support was added for the Cisco AS5350 and Cisco AS5400 universal gateways.

This feature module describes enhancements to H.323 and Session Initiation Protocol (SIP) signaling when interworking with ISDN, T1 channel associated signaling (CAS), or E1 R2 Public Switched Telephone Network (PSTN) services. 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 completed (cut-through) at the appropriate point in call setup and that early alerting (ringing) does not occur.

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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 are also allowed to 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 destination switch generates the ringback tone, it sends a progress indicator of 1 or 8 in the Alerting message. If the origination 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 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 values that you can configure through Cisco IOS on the H.323 gateway for the different types of messages.

Table 1 Configurable Progress Indicator (PI) 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.	Setup, Progress, Connect
2	Destination address is non-ISDN.	Progress, Connect
3	Origination address is non-ISDN.	Setup
8	In-band information or appropriate pattern now available.	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 gateway generated the ringback tone regardless of the PI value in the Setup message. To enable the originating gateway to generate the ringback tone, use the **progress_ind** dial-peer configuration command to set the progress indicator to 3 in Setup messages in the VoIP dial peer.
- 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

Calls that are pure ISDN may use different protocols at the originating and terminating ends. For example, a call may come in on ETSI and go out on N12. If the two protocols are not compatible end-to-end, all IEs, including the progress indicator, are dropped from messages. Because a progress indicator is required in all Progress messages, the originating gateway inserts a progress indicator of 1 in the Progress message.

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.

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
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, when 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 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

Table 3 Mapping of ISDN Messages to Outgoing SIP Messages (continued)

ISDN/CAS Messages	Outgoing SIP Messages
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
Progress without PI	No SIP message is generated
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
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 slow 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 Universal Access Server. For IVR on other Cisco platforms, you must select TCL 1.0 as the session application. To use TCL 1.0, configure the application name with a “.t.old” suffix, by using the **call application voice** global configuration command.
- 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

Related Features and Technologies

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

Related Documents

- *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
- *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*
- *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 7200 series
- Cisco 7500 series
- Cisco AS5300
- Cisco AS5350
- Cisco AS5400
- Cisco AS5800

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

RFCs

No new or modified RFCs are supported by this feature.

Prerequisites

The Cisco AS5350 and Cisco AS5400 do not support the Mica Modem Card, Microcom Modem Card, or VoIP Feature Card. Voice and modem functions are provided by the Universal Port Dial Feature card running SPE firmware. See the *Cisco AS5350 Universal Gateway Card Installation Guide* and the *Cisco AS5400 Universal Gateway Card Installation Guide* for more information. All references to the Cisco AS5300 in this document apply to the Cisco AS5350 and Cisco AS5400 platforms with the following exceptions:

- Use the Universal Port Dial Feature Card instead of the Mica or Microcom modem cards.
- Use SPE firmware instead of portware version 6.7.7.
- Run Cisco IOS Release 12.1(5)XM2 software for VoIP functionality.

Other 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)
- Configuring ISDN T306 and T310 Timers (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 in dial-peer configuration mode:

	Command	Purpose
Step 1	Router# config term	Enters global configuration mode.
Step 2	Router(config)# dial-peer voice <i>number</i> voip	Enters dial-peer configuration mode and configures a VoIP dial-peer.
Step 3	Router(config-dial)# destination-pattern <i>string</i>	Specifies the telephone number for this dial peer. A call is matched to this dial peer by using this pattern.
Step 4	Router(config-dial)# session protocol cisco	Sets the session protocol type to Cisco proprietary H.323.
Step 5	Router(config-dial)# progress_ind setup enable <i>pi-number</i>	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 Progress or Connect messages from the outbound POTS dial peer on an H.323 gateway, perform the following tasks in dial-peer configuration mode:

	Command	Purpose
Step 1	Router# config term	Enters global configuration mode.
Step 2	Router(config)# dial-peer voice <i>number</i> pots	Enters dial-peer configuration mode and configures a POTS dial-peer.
Step 3	Router(config-dial)# destination-pattern <i>string</i>	Specifies the telephone number for this dial peer. A call is matched to this dial peer using this pattern.
Step 4	Router(config-dial)# progress_ind progress enable <i>pi-number</i> or Router(config-dial)# progress_ind connect enable <i>pi-number</i>	Sets the progress indicator for Progress messages. Sets the progress indicator for Connect messages.

Verifying Progress Indicator Configuration

Perform the following steps 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 11 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 term	Enters global configuration mode.
Step 2	Router(config)# interface serial <i>controller:timeslot</i>	Enters interface configuration mode for a D-channel serial interface.
Step 3	Router(config-if)# isdn t306 <i>msecs</i>	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 4	Router(config-if)# isdn t310 <i>msecs</i>	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 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 11 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.
-

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 debug isdn q931 command to display the ISDN messages. Verify the progress indicator value in the Setup message from the originating switch. If the Setup message does not contain a PI value, use the progress_ind dial-peer command to set the PI to 3.
Gateway not responding to Connect after receiving Progress	Enable 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.

Configuration Examples

This section provides the following configuration examples:

- Progress Indicator Configuration Example
- T306/T310 Timer 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  
  
!  
.  
.
```

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.

- **isdn negotiate-bchan**
- **isdn t306**
- **isdn t310**
- **progress_ind**

isdn negotiate-bchan

To enable the router to accept a B-channel that is different from the B-channel requested in the outgoing call setup message, use the **isdn negotiate-bchan** interface configuration command. To restore the default condition, use the **no** form of this command.

isdn negotiate-bchan

no isdn negotiate-bchan

Syntax Description This command has no keywords or arguments.

Defaults The router will not accept a different B-channel.

Command Modes Interface configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.
12.1(5)XM2	The command was introduced for the Cisco AS5350 and CiscoAS5400.

Usage Guidelines The **isdn negotiate-bchan** command is supported only for NET5 and N12 switches.

Examples

The following example enables an alternate B-channel for serial interface 0:23:

```
interface serial0:23
 isdn negotiate-bchan
```

Related Commands

Command	Description
dial peer	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.

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	<i>msecs</i>	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 through 400,000 ms.
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Defaults The default depends on the switch, usually from 5,000 to 30,000 ms.

Command Modes Interface configuration

Command History	Release	Modification
	12.1(3)XI	This command was introduced.
	12.1(5)XM2	The command was introduced for the Cisco AS5350 and CiscoAS5400.

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	<i>msecs</i>	Number of milliseconds that the router waits before disconnecting a call after receiving a Call Proceeding message. Values are 1 through 400,000 ms.
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Defaults	The default depends on the switch, usually from 5,000 to 30,000 ms.
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Command Modes	Interface configuration
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Command History	Release	Modification
	12.1(3)XI	This command was introduced.
	12.1(5)XM2	The command was introduced for the Cisco AS5350 and CiscoAS5400.

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.
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Examples	The following example sets the T310 timer to 40,000 ms for serial interface 0:23:
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```
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 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} {enable pi-number | disable}
```

```
no progress_ind {setup | connect | progress}
```



Note This command is not supported on VoIP gateways that use SIP.

Syntax Description		
setup		Sets the progress indicator for setup messages.
progress		Sets the progress indicator for progress messages.
connect		Sets the progress indicator for connect messages.
enable		Enables the configuration of the progress indicator.
<i>pi-number</i>		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 or Connect 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)XM2	The command was introduced for the Cisco AS5350 and CiscoAS5400.

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 If you configure a progress indicator value for Progress messages, and 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), then the progress indicator configured through the **progress_ind** command does not override the progress indicator in 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	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.

Debug Commands

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

- **debug vtsp tone**

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)XM2	The command was introduced for the Cisco AS5350 and CiscoAS5400.

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.

