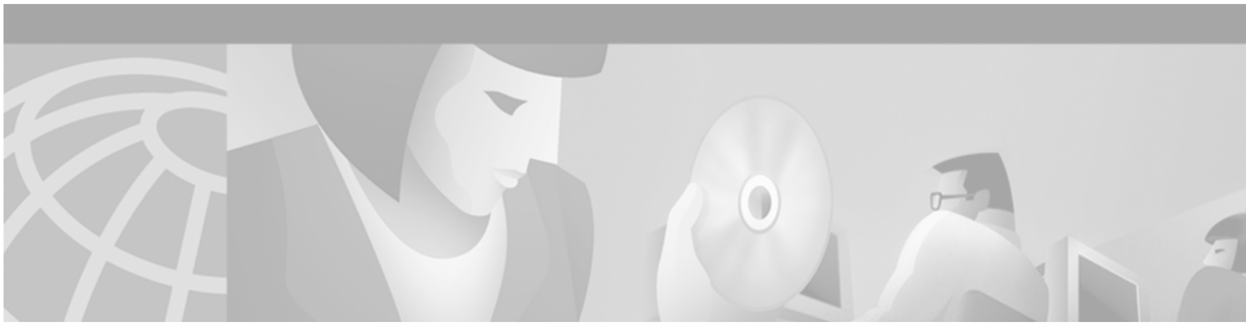

Dual Tone Multifrequency Relay for SIP Calls Using Named Telephone Events

Document Update Alert

This document was originally produced for Cisco IOS Release 12.2(11)T. This feature has been updated in subsequent releases, and more recent documentation is available.

If you are using Cisco IOS Release 12.2(11)T or higher, refer to the following section in the Configuring Additional SIP Features chapter of the *Cisco IOS SIP Configuration Guide*, Cisco IOS Voice Configuration Library, Release 12.3:

- *DTMF Relay for SIPCalls Using NTEs*

Feature History

Release	Modification
12.2(2)XB	This feature was introduced on the Cisco 2600 series, the Cisco 3600, the Cisco AS5300, and the Cisco AS5400.
12.2(2)XB1	This feature was implemented on the Cisco AS5850 platform.
12.2(8)T	This feature was integrated into Cisco IOS Release 12.2(8)T. **Note** The Cisco AS5300, Cisco AS5400, and Cisco AS5850 are not supported in this release.
12.2(11)T	This feature was integrated into Cisco IOS Release 12.2(11)T and support was added for the Cisco AS5300, Cisco AS5400, and Cisco AS5850 platforms.

This document describes the Dual Tone Multifrequency (DTMF) Relay for SIP Calls Using Named Telephone Events (NTE) feature in Cisco IOS Release 12.2(11)T. For consistency, the feature is referred to as the SIP NTE DTMF relay feature in this document.

This document includes the following sections:

Feature Overview

The SIP NTE DTMF relay feature is used for the following applications:

- Reliable DTMF Relay, page 2
- SIP Phone Support, page 3

These applications are discussed in more detail in the following sections.

Note The SIP NTE DTMF relay feature is implemented for SIP calls only on Cisco Voice-over-IP (VoIP) gateways.

Reliable DTMF Relay

The SIP NTE DTMF relay feature provides reliable digit relay between Cisco VoIP gateways when a low bandwidth codec is used. Using NTE to relay DTMF tones provides a standardized means of transporting DTMF tones in Real-Time Transport Protocol (RTP) packets according to section 3 of RFC 2833, *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, developed by the Internet Engineering Task Force (IETF) Audio/Video Transport (AVT) working group. RFC 2833 defines formats of NTE RTP packets used to transport DTMF digits, hookflash, and other telephony events between two peer endpoints.

DTMF tones are generated when a button on a touch-tone phone is pressed. When the tone is generated, it is compressed, transported to the other party, and decompressed. If a low-bandwidth codec, such as a G.729 or G.723 is used without a DTMF relay method, the tone may be distorted during compression and decompression.

With the SIP NTE DTMF relay feature, the endpoints perform per-call negotiation of the DTMF relay method. They also negotiate to determine the payload type value for the NTE RTP packets.

In a SIP call, the gateway forms a Session Description Protocol (SDP) message that indicates:

- If NTE will be used
- Which events will be sent using NTE
- NTE payload type value

The SIP NTE DTMF relay feature can relay hookflash events in the RTP stream using NTP packets.

Note The SIP NTE DTMF relay feature does not support hookflash generation for advanced features such as call waiting and conferencing.

SIP Phone Support

The SIP NTE DTMF relay feature adds SIP phone support. When SIP IP phones are running software that does not have the capability to generate DTMF tones, the phones use NTE packets to indicate DTMF digits. With the SIP NTE DTMF relay feature, Cisco VoIP gateways can communicate with SIP phones that use NTE packets to indicate DTMF digits. The Cisco VoIP gateways can relay the digits to other endpoints.

Benefits

This feature provides the following benefits:

- Reliable DTMF digit relay between Cisco VoIP gateways when low-bandwidth codecs are used
- Ability to communicate with SIP phone software that uses NTE packets to indicate DTMF digits

Restrictions

The SIP NTE DTMF relay feature is available only for SIP calls on Cisco VoIP gateways. The SIP NTE DTMF relay feature supports only hookflash relay and does not support hookflash generation for advanced features such as call waiting and conferencing.

Related Features and Technologies

- Cisco VoIP
- Cisco IVR
- Cisco IP Phones
- Cisco SIP Proxy Server

Related Documents

- *Cisco IOS Voice, Video, and Fax Configuration Guide*, Release 12.2
- *Cisco IOS Voice, Video, and Fax Command Reference*, Release 12.2
- Session Initiation Protocol Gateway Call Flows
- Session Initiation Protocol for Voice over IP on Cisco Access Platforms
- *Enhancements to the Session Initiation Protocol for VoIP on Cisco Access Platforms*
- RFC 2833, *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*
- *RTP Payload for Comfort Noise*, Internet Draft of the Internet Engineering Task Force (IETF) Audio/Video Transport (AVT) working group

Supported Platforms

Table 1 Cisco IOS Release and Platform Support for this Feature

Platform	12.2(2)XB	12.2(2)XB1	12.2(8)T	12.2(11)T
Cisco 2600 series	X	X	X	X
Cisco 3600 series	X	X	X	X
Cisco AS5300	X	X	Not supported	X
Cisco AS5400	X	X	Not supported	X
Cisco AS5850	Not supported	X	Not supported	X
Cisco 7200 series	X	X	X	X

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 quickly 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.

MIBs

- CISCO-VOICE-DIAL-CONTROL-MIB

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

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

RFCs

- RFC 2833, *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*
- *RTP Payload for Comfort Noise*, Internet Draft of the Internet Engineering Task Force (IETF) Audio/Video Transport (AVT) working group
- RFC 1890, *RTP Profile for Audio and Video Conferences with Minimal Control*

Prerequisites

Before configuring or using the SIP NTE DTMF relay feature, you must have a working VoIP network using SIP on Cisco gateways. See the documents in "Related Documents" for more information.

Configuration Tasks

See the following sections for configuration tasks for the SIP NTE DTMF relay feature. Each task in the list is identified as either optional or required.

- Configuring DTMF Relay and NTE Payload Type (required)

Configuring DTMF Relay and NTE Payload Type

To configure DTMF relay and NTE Payload Type, enter the following commands in global configuration mode:

	Command	Purpose
Step 1	`Router(config)# dial-peer voice number voip`	Enters dial-peer configuration mode and defines a remote VoIP dial peer. The *number* argument is one or more digits identifying the dial peer. Valid entries are from 1 to 2147483647. The **voip** keyword indicates a VoIP peer using voice encapsulation on the IP network.
Step 2	`Router(config-dial-peer)# session protocol sipv2`	Configures the SIP protocol on the gateway.

	Command	Purpose
Step 3	`Router(config-dial-peer)# dtmf-relay rtp-nte`	Allows DTMF relay using NTE RTP packets. DTMF tones are encoded in the NTE format and transported in the same RTP channel as the voice.
Step 4	`Router(config-dial-peer)# rtp payload-type nte number comfort-noise [13 | 19]`	The **nte** keyword chooses the type of payload in the NTE packet. Number values are 96 through 127; the default value is 101. **Note** Reserved values (for example, 96, 97, 100, 121, 122, 123, 125, 126, and 127) cannot be configured for the **rtp-nte payload** value unless they are freed by reassigning them to some other values that are within the 96 through 127 range. The **comfort-noise** keyword indicates the RTP payload type of comfort noise. The July 2001 draft entitled *RTP Payload for Comfort Noise*, from the IETF AVT working group, designates 13 as the payload type for comfort noise. Previous Cisco equipment uses 19 as the payload type for comfort noise. If you are connecting to a GW that complies with the *RTP Payload for Comfort Noise* draft, use 13. Only use 19 if you are connecting to older Cisco gateways that use DSPware before version 3.4.32.

Verifying DTMF Relay and NTE Payload Type

Enter the **show running** command to verify that DTMF relay and NTE are configured on the dial peer. For example:

```
!
dial-peer voice 1000 pots
 destination-pattern 4961234
 port 1/0/0
!
dial-peer voice 2000 voip
 application session
 destination-pattern 4965678
 session protocol sipv2
 session target ipv4:11.0.13.34
 dtmf-relay rtp-nte
! RTP payload type value = 101 (default)
!
dial-peer voice 3000 voip
 application session
 destination-pattern 2021010101
 session protocol sipv2
 session target ipv4:11.0.13.34
 dtmf-relay rtp-nte
 rtp payload-type nte 110
! RTP payload type value = 110 (user assigned)
!
```

Monitoring and Maintaining SIP NTE DTMF relay

Command	Purpose
`Router# debug voip rtp session named-event`	Turns on debugging for RTP NTEs.
`Router# show voip rtp connections`	Shows local and remote Calling ID and IP address and port information.

Configuration Examples

This section provides the following configuration examples:

- DTMF Relay using RTP-NTE Example
- RTP Using Payload Type NTE Example

DTMF Relay using RTP-NTE Example

Example 1 provides an example of DTMF relay using RTP-NTE:

Example 1 DTMF Relay Using RTP-NTE Example

```
Router(config)# dial-peer voice 62 voip
Router(config-dial-peer)# session protocol sipv2
Router(config-dial-peer)# dtmf-relay rtp-nte
```

RTP Using Payload Type NTE Example

Example 2 provides an example of RTP Using Payload Type NTE with the default value of 101:

Example 2 RTP Using Payload Type NTE

```
Router(config)# dial-peer voice 62 voip
Router(config-dial-peer)# rtp payload-type nte 101
```

Command Reference

This section documents new or modified commands. All other commands used with this feature are documented in the Cisco IOS Release 12.2 command reference publications.

New Commands

- **debug voip rtp**
- **rtp payload-type**

Modified Commands

- **dtmf-relay**

debug voip rtp

To enable debugging for Real-Time Transport Protocol (RTP) named event packets, use the **debug voip rtp** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug voip rtp {error | session [nse | multicast | conference | dtmf-relay | named-event] | packet remote-ip *ipaddress* **remote-port** *portnum packetnum* **| packet callid** *idnum packetnum***}**

no debug voip rtp

Syntax Description

error	Prints out a trace for error cases.
session	Provides all session debug information. If used with a keyword, supplies more specific debug information according to the keywords used.
nse	Provides debug information for named signaling events (NSEs).
multicast	Provides debug information for multicast packets.
conference	Provides debug information for conference packets.
dtmf-relay	Provides debug information for dual-tone multifrequency (DTMF) packets.
named-event	Provides debug information for named telephony event (NTE) packets.
packet remote-ip *ipaddress* **remote-port** *portnum packetnum*	Provides debug information for a remote IP address and port number. Using the *packetnum* argument specifies the number of packets to trace so that the display is not flooded.
packet callid *idnum packetnum*	Provides debug information for a specific call ID number (obtained by using the **show voip rtp connections** command). Using the *packetnum* argument specifies the number of packets to trace so that the display is not flooded.

Defaults

Debugging for RTP named event packets is not enabled.

Command Modes

EXEC

Command History

Release	Modification
12.2(2)XB	This command was introduced.
12.2(2)XB1	This command was implemented on the Cisco AS5850 platform.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and support was added for the Cisco AS5300, Cisco AS5400, and Cisco AS5850 platforms.

Examples

The following example illustrates the output for the **debug voip rtp session named-event** command. The example is for a gateway that sends digits 1, 2, 3, then receives digits 9,8,7. The payload type, event ID, and additional packet payload are shown in each log.

The first three packets indicate the start of the tone (initial packet and two redundant). The last three packets indicate the end of the tone (initial packet and two redundant). The packets in between are refresh packets that are sent every 50 milliseconds (without redundancy).

```
Router# debug voip rtp session named-event

00:09:29:          Pt:99     Evt:1       Pkt:03 00 00  <<<Rcv>
00:09:29:          Pt:99     Evt:1       Pkt:03 00 00  <<<Rcv>
00:09:29:          Pt:99     Evt:1       Pkt:03 00 00  <<<Rcv>
00:09:29:          Pt:99     Evt:1       Pkt:03 01 90  <<<Rcv>
00:09:29:          Pt:99     Evt:1       Pkt:03 03 20  <<<Rcv>
00:09:29:          Pt:99     Evt:1       Pkt:03 04 B0  <<<Rcv>
00:09:29:          Pt:99     Evt:1       Pkt:83 04 C8  <<<Rcv>
00:09:29:          Pt:99     Evt:1       Pkt:83 04 C8  <<<Rcv>
00:09:29:          Pt:99     Evt:1       Pkt:83 04 C8  <<<Rcv>
00:09:29:          Pt:99     Evt:2       Pkt:03 00 00  <<<Rcv>
00:09:29:          Pt:99     Evt:2       Pkt:03 00 00  <<<Rcv>
00:09:29:          Pt:99     Evt:2       Pkt:03 00 00  <<<Rcv>
00:09:29:          Pt:99     Evt:2       Pkt:03 01 90  <<<Rcv>
00:09:29:          Pt:99     Evt:2       Pkt:03 03 20  <<<Rcv>
00:09:29:          Pt:99     Evt:2       Pkt:03 04 B0  <<<Rcv>
00:09:29:          Pt:99     Evt:2       Pkt:83 05 18  <<<Rcv>
00:09:29:          Pt:99     Evt:2       Pkt:83 05 18  <<<Rcv>
00:09:29:          Pt:99     Evt:2       Pkt:83 05 18  <<<Rcv>
00:09:29:          Pt:99     Evt:3       Pkt:03 00 00  <<<Rcv>
00:09:29:          Pt:99     Evt:3       Pkt:03 00 00  <<<Rcv>
00:09:29:          Pt:99     Evt:3       Pkt:03 00 00  <<<Rcv>
00:09:30:          Pt:99     Evt:3       Pkt:03 01 90  <<<Rcv>
00:09:30:          Pt:99     Evt:3       Pkt:03 03 20  <<<Rcv>
00:09:30:          Pt:99     Evt:3       Pkt:03 04 B0  <<<Rcv>
00:09:30:          Pt:99     Evt:3       Pkt:03 06 40  <<<Rcv>
00:09:30:          Pt:99     Evt:3       Pkt:83 06 80  <<<Rcv>
00:09:30:          Pt:99     Evt:3       Pkt:83 06 80  <<<Rcv>
00:09:30:          Pt:99     Evt:3       Pkt:83 06 80  <<<Rcv>
00:09:31:  <Snd>>> Pt:99     Evt:9       Pkt:02 00 00
00:09:31:  <Snd>>> Pt:99     Evt:9       Pkt:02 00 00
00:09:31:  <Snd>>> Pt:99     Evt:9       Pkt:02 00 00
00:09:31:  <Snd>>> Pt:99     Evt:9       Pkt:02 01 90
00:09:31:  <Snd>>> Pt:99     Evt:9       Pkt:02 03 20
00:09:31:  <Snd>>> Pt:99     Evt:9       Pkt:02 04 B0
00:09:31:  <Snd>>> Pt:99     Evt:9       Pkt:02 06 40
00:09:31:  <Snd>>> Pt:99     Evt:9       Pkt:82 06 58
00:09:31:  <Snd>>> Pt:99     Evt:9       Pkt:82 06 58
00:09:31:  <Snd>>> Pt:99     Evt:9       Pkt:82 06 58
00:09:31:  <Snd>>> Pt:99     Evt:8       Pkt:02 00 00
00:09:31:  <Snd>>> Pt:99     Evt:8       Pkt:02 00 00
00:09:31:  <Snd>>> Pt:99     Evt:8       Pkt:02 00 00
00:09:31:  <Snd>>> Pt:99     Evt:8       Pkt:02 01 90
00:09:31:  <Snd>>> Pt:99     Evt:8       Pkt:02 03 20
00:09:31:  <Snd>>> Pt:99     Evt:8       Pkt:02 04 B0
00:09:31:  <Snd>>> Pt:99     Evt:8       Pkt:02 06 40
00:09:31:  <Snd>>> Pt:99     Evt:8       Pkt:82 06 90
00:09:31:  <Snd>>> Pt:99     Evt:8       Pkt:82 06 90
00:09:31:  <Snd>>> Pt:99     Evt:8       Pkt:82 06 90
00:09:31:  <Snd>>> Pt:99     Evt:7       Pkt:02 00 00
00:09:31:  <Snd>>> Pt:99     Evt:7       Pkt:02 00 00
00:09:31:  <Snd>>> Pt:99     Evt:7       Pkt:02 00 00
00:09:31:  <Snd>>> Pt:99     Evt:7       Pkt:02 01 90
00:09:31:  <Snd>>> Pt:99     Evt:7       Pkt:02 03 20
00:09:31:  <Snd>>> Pt:99     Evt:7       Pkt:02 04 B0
00:09:32:  <Snd>>> Pt:99     Evt:7       Pkt:02 06 40
00:09:32:  <Snd>>> Pt:99     Evt:7       Pkt:82 06 58
00:09:32:  <Snd>>> Pt:99     Evt:7       Pkt:82 06 58
00:09:32:  <Snd>>> Pt:99     Evt:7       Pkt:82 06 58
```

Related Commands

Command	Description
show voip rtp connections	Shows local and remote Call ID number, IP address, and port number.

dtmf-relay

To specify how an H.323 or Session Initiation Protocol (SIP) gateway relays dual tone multifrequency (DTMF) tones between telephony interfaces and an IP network, use the **dtmf-relay** command in dial-peer configuration mode. To remove all signalling options and send the DTMF tones as part of the audio stream, use the **no** form of this command.

dtmf-relay [**cisco-rtp**] [**h245-alphanumeric**] [**h245-signal**] [**rtp-nte**]

no dtmf-relay [**cisco-rtp**] [**h245-alphanumeric**] [**h245-signal**] [**rtp-nte**]

Syntax Description

cisco-rtp	Forwards DTMF tones by using Real-Time Transport Protocol (RTP) with a Cisco proprietary payload type.
h245-alphanumeric	Forwards DTMF tones by using the H.245 "alphanumeric" User Input Indication method. Supports tones 0-9, *, #, and A-D.
h245-signal	Forwards DTMF tones by using the H.245 "signal" User Input Indication method. Supports tones 0-9, *, #, and A-D.
rtp-nte	Forwards DTMF tones by using Real-Time Transport Protocol (RTP) with the Named Telephone Event (NTE) payload type.

Defaults

No default behavior or values.

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.0(1)T	This command was introduced.
12.0(2)XH	The **h245-signal** keyword was added.
12.0(5)T	This command was modified for H.323 V2.
12.2(2)XB	The **rtp-nte** keyword was added.
12.2(2)XB1	This command was implemented on the Cisco AS5850 platform.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and support was added for the Cisco AS5300, Cisco AS5400, and Cisco AS5850 platforms.

Usage Guidelines

DTMF is the tone generated when you press a digit on a touch-tone phone. This tone is compressed at one end of a call; when the tone is decompressed at the other end, it can become distorted, depending on the codec used. The DTMF relay feature transports DTMF tones generated after call establishment out of band using either a standard H.323 out-of-band method and a proprietary RTP-based mechanism, or for SIP calls, an NTE RTP packet.

The gateway only sends DTMF tones in the format you specify if the remote device supports it. If the remote device supports multiple formats, the gateway chooses the format based on the following priority:

1. **cisco-rtp** (highest priority)
2. **h245-signal**
3. **h245-alphanumeric**
4. **rtp-nte**
5. None—DTMF sent in-band

The principal advantage of the **dtmf-relay** command is that it sends DTMF tones with greater fidelity than is possible in-band for most low-bandwidth codecs, such as G.729 and G.723. Without the use of DTMF relay, calls established with low-bandwidth codecs may have trouble accessing automated DTMF-based systems, such as voice-mail, menu-based ADC/Kentrox systems, and automated banking systems.

Note The **cisco-rtp** option of the **dtmf-relay** command is a proprietary Cisco implementation and operates only between two Cisco AS5800 universal access servers running Cisco IOS Release 12.0(2)XH, or between Cisco AS5800 universal access servers or Cisco 2600 or Cisco 3600 modular access routers running Cisco IOS Release 12.0(2)XH or later releases. Otherwise, the DTMF relay feature does not function, and the gateway sends DTMF tones in-band.

Examples

The following example demonstrates use of the **dtmf-relay** command with the SIP NTE DTMF relay feature:

```
Router(config-dial-peer)# dtmf-relay rtp-nte
```

Related Commands

Command	Description
rtp payload-type	Chooses the type of payload in the RTP NTE packet.

rtp payload-type

To identify the payload type of a Real-Time Transport Protocol (RTP) packet, use the **rtp payload-type** command in dial-peer configuration mode. To remove the RTP payload type, use the **no** form of this command.

rtp payload-type {cisco-cas-payload *number*| **cisco-clear-channel** *number* | **cisco-codec-fax-ack** *number* | **cisco-codec-fax-ind** *number* | **cisco-fax-relay** *number* | **cisco-pcm-switch-over-alaw** *number* | **cisco-pcm-switch-over-ulaw** *number*| **cisco-rtp-dtmf-relay** *number* | **nte** *number* | **nse** *number*} **[comfort-noise {13 | 19}]**

no rtp payload-type nte

Syntax Description

cisco-cas-payload *number*	Identifies the payload type as Cisco CAS RTP payload. Number values are 96 through 127; the default value is 101.
cisco-clear-channel *number*	Identifies the payload type as Cisco clear channel RTP payload. Number values are 96 through 127; the default value is 101.
cisco-codec-fax-ack *number*	Identifies the payload type as Cisco codec fax acknowledge. Number values are 96 through 127; the default value is 101.
cisco-codec-fax-ind *number*	Identifies the payload type as Cisco codec fax indication. Number values are 96 through 127; the default value is 101.
cisco-fax-relay *number*	Identifies the payload type as Cisco fax relay. Number values are 96 through 127; the default value is 101.
cisco-pcm-switch-over-alaw *number*	Identifies the payload type as Cisco RTP PCM codec switch over indication (a-law). Number values are 96 through 127; the default value is 101.
cisco-pcm-switch-over-ulaw *number*	Identifies the payload type as Cisco RTP PCM codec switch over indication (u-law). Number values are 96 through 127; the default value is 101.
cisco-rtp-dtmf-relay *number*	Identifies the payload type as Cisco RTP DTMF relay. Number values are 96 through 127; the default value is 101.
nte *number*	Identifies the payload type as a Named Telephone Event (NTE). Number values are 96 through 127; the default value is 101.
nse *number*	Identifies the payload type as a Named Signaling Event (NSE). Number values are 96 through 127; the default value is 101.
comfort-noise	Indicates the RTP payload type of comfort noise. The July 2001 draft entitled *RTP Payload for Comfort Noise*, from the IETF AVT working group, designates 13 as the payload type for comfort noise. Previous Cisco equipment uses 19 as the payload type for comfort noise. If you are connecting to a GW that complies with the *RTP Payload for Comfort Noise* draft, use 13. Only use 19 if you are connecting to older Cisco gateways that use DSPware before version 3.4.32.

Defaults

The default *number* value is 101.

Command Modes

Dial-peer configuration

Command History

Release	Modification
12.2(2)T	This command was introduced.
12.2(2)XB	The **nte** and **comfort-noise** keywords were introduced.
12.2(2)XB1	This command was implemented on the Cisco AS5850 platform.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T.
12.2(11)T	This command was integrated into Cisco IOS Release 12.2(11)T and support was added for the Cisco AS5300, Cisco AS5400, and Cisco AS5850 platforms.

Usage Guidelines

Use the **rtp payload-type nte** command to identify the payload type of an RTP NTE. Use this command after the **dtmf-relay** command is used to choose the NTE method of dual tone multifrequency (DTMF) relay for a Session Initiation Protocol (SIP) call.

Examples

The following example demonstrates the use of the **rtp payload-type nte** command with the SIP NTE DTMF relay feature:

```
Router(config-dial-peer)# rtp payload-type nte 99
```

Related Commands

Command	Description
dtmf-relay	Specifies how an H.323 or SIP gateway relays DTMF tones between telephony interfaces and an IP network.

Glossary

DTMF—dual tone multifrequency. Tones that are generated when a button on a touch-tone phone is pressed. When the tone is generated, it is compressed, transported to the other party, and decompressed.

IVR—Interactive voice response. Scripts which are used to collect information from a user to process commands; for example, to retrieve voice mail. DTMF digits are entered in response to IVR scripts. In low-bandwidth compression, DTMF digits can become distorted and unrecognizable by IVR scripts.

NTE—Named Telephony Event. An event such as DTMF digits that must be encoded and transported in an RTP packet. RFC 2833 specifies the format of the RTP NTE payload.

RTP—Real-Time Transport Protocol. A protocol for transporting multimedia over IP; see RFC 1889, *RTP: A Transport Protocol for Real-Time Applications*.

SDP—Session Description Protocol. A protocol for defining information needed to establish multimedia transport over IP. SDP transmits information such as session announcement, session invitation, transport addresses, and media types. In a SIP call, SDP messages indicates if NTE will be used, which events will be sent using NTE, and the NTE payload type value. See RFC 2327, *SDP: Session Description Protocol*.

SIP—Session Initiation Protocol. A protocol for transporting multimedia that is independent of the underlying packet control layer, such as User Datagram Protocol (UDP), and is based on a client/server architecture. See RFC 2543, *SIP: Session Initiation Protocol*.