Cisco Text Relay for Baudot Text Phones

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The Cisco Text Relay for Baudot Text Phones (Cisco Text Relay) feature implements a mechanism for reliable transport of teletype text phone (TTY) signals over VoIP calls. TTY text phones, also known as telecommunication devices for the deaf (TDD), are specialized phones that enable people who are deaf, hard of hearing, or speech-impaired to communicate over time-division multiplex (TDM) or IP networks by allowing them to type messages to one another instead of, or in addition to, talking and listening.

Cisco Text Relay is based on the proposed ITU V.151 (V.ToIP) standard. Cisco Text Relay leverages Audio/T.140, which means text characters are carried over the same Real-Time Transport Protocol (RTP) stream as voice (similar to the way dual tone multifrequency (DTMF) characters are carried in the RTP stream [RFC 2833]). Cisco Text Relay has minimal impact on bandwidth because the text characters are transported efficiently in the RTP stream. There is also a configurable redundancy option enabling TTY to run reliably in demanding network conditions.

Finding Feature Information in This Module

Your Cisco IOS software release may not support all of the features documented in this module. To reach links to specific feature documentation in this module and to see a list of the releases in which each feature is supported, use the "Feature Information for Cisco Text Relay for Baudot Text Phones" section on page 23.

Finding Support Information for Platforms and Cisco IOS Software Images

Use Cisco Feature Navigator to find information about platform support and Cisco IOS software image support. Access Cisco Feature Navigator at http://www.cisco.com/go/fn. You must have an account on Cisco.com. If you do not have an account or have forgotten your username or password, click **Cancel** at the login dialog box and follow the instructions that appear.

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Prerequisites for Cisco Text Relay for Baudot Text Phones

To enable the Cisco Text Relay for Baudot Text Phones feature, you must do the following:

- Establish a working H.323, Session Initiation Protocol (SIP), Skinny Client Control Protocol (SCCP), or Media Gateway Control Protocol (MGCP) network for voice calls.
- Ensure that you have a Cisco IOS image that supports Cisco Text Relay.
- Use text phones that support TIA-825-A, *A Frequency Shift Keyed Modem for Use on the Public Switched Telephone Network*, the basic standard defining the Baudot TTY.

Restrictions for Cisco Text Relay for Baudot Text Phones

Restrictions for Cisco Text Relay are as follows:

- The Cisco Text Relay option must be configured on both the terminating and originating gateways.
- The text relay RTP payload type configuration must match on the terminating and originating gateways in order for TTY characters to be transmitted across the VoIP network.
- Cisco Text Relay supports Baudot 45.45 and Baudot 50 TTY modulations only. Any other TTY modulation is treated as a normal voice signal.
- Cisco Text Relay does not support third-party gateways.
- The voice interface cards and platforms supported by the Cisco Text Relay for Baudot Text Phones feature in Cisco IOS releases 12.4(6)T and 12.4(4)XC are listed in Table 1.

Table 1 Cisco Text Relay for Baudot Text Phones Feature Support for Voice Interface Cards and Platforms

Cisco IOS Release	Voice Interface Cards	Platforms
12.4(6)T	VWIC or VIC cards inserted into HWIC slots	Cisco 2800 series and Cisco 3800 series.
12.4(6)T	NM-HD-1V, NM-HD-2V, NM-HD-2VE	Cisco 2600XM series, Cisco 2691, Cisco 2800 series, Cisco 3700 series, and Cisco 3800 series. **Note** The Cisco 2801 does not support network modules.
12.4(6)T	NM-HDV2, NM-HDV2-1T1/E1, NM-HDV2-2T1/E1, EVM-HD-8FXS/DID, and all associated extension modules	Cisco 2600XM series, Cisco 2691, Cisco 2800 series, Cisco 3700 series, and Cisco 3800 series. Requires PVDM2 module.

Table 1 Cisco Text Relay for Baudot Text Phones Feature Support for Voice Interface Cards and Platforms (continued)

Cisco IOS Release	Voice Interface Cards	Platforms
12.4(6)T	—	Line-side high-density analog gateways. Cisco IAD2430 and Cisco VG224.
12.4(4)XC	AS5X-FC, AS5X-PVDM2-64	Cisco AS5400XM and Cisco AS5350XM with high-density voice feature card (AS5X-FC) and digital signal processor (DSP) module (AS5X-PVDM2-64).

Information About the Cisco Text Relay for Baudot Text Phones Feature

To configure the Cisco Text Relay feature, you should understand the following concepts:

- TTY Text Phones, page 3
- Text over IP, page 3
- Cisco Text Relay, page 4

TTY Text Phones

A TTY text phone is a text communication device that allows people with hearing or speech disabilities to use the telephone. TTY text phones have neither a handshake procedure at the beginning of a call nor a carrier tone during the call. This limits the transmission speed but permits TTY tones, DTMF signals (touch tones), and voice to be intermixed on the same call. This also allows TTY calls to be put on hold and to be transferred, like voice calls.

Each TTY text character consists of a sequence of seven individual tones:

- The start tone at 1800 Hz.
- A series of five tones (at either 1400 Hz or 1800 Hz) that specifies the character.
- A stop tone at 1400 Hz. The stop tone separates one character from the next.

The first six tones are each 22 ms in duration for Baudot 45.45, 20 ms for Baudot 50, and the stop tone is 33 to 44 ms in duration. This equates to 165 ms per TTY text character, or six characters per second.

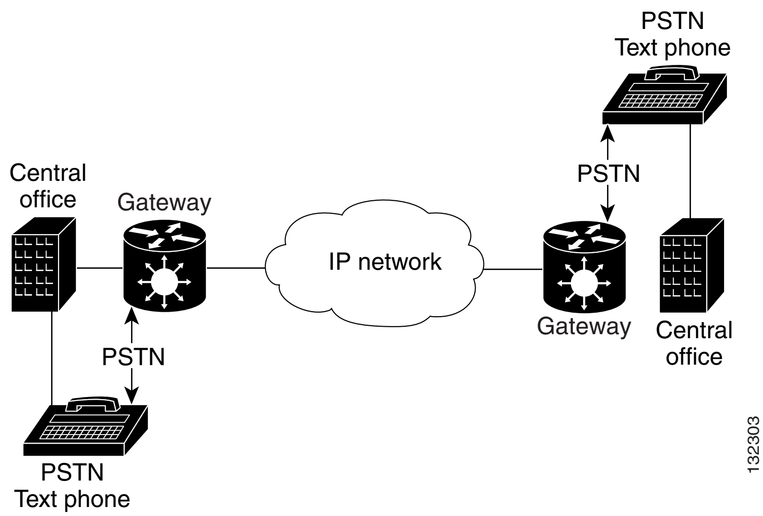
TTY text phones operate in half duplex. Users must take turns transmitting and cannot interrupt each other. The TTY text phones have a special key called go ahead, which tells the other user to type.

Text over IP

Text over IP (ToIP) is the transport of analog-modulated signals generated by TTY text phones over an IP network. In a voice network, ToIP is the transport of text characters from a legacy text phone (TTY device) connected to a public switched telephone network (PSTN) gateway through the Foreign Exchange Station (FXS) port or through an acoustic coupler.

Figure 1 shows a typical ToIP network.

Figure 1 Typical Text over IP Network



Real-time text over IP networks can transmit one character at a time bidirectionally, providing real-time communication, like voice or video systems that transport streaming media over IP.

Cisco Text Relay

Text relay is part of the voice call and is configured as an additional capability during the voice call setup. Because there is no handshake procedure when TTY text phones are used, text relay does not require a call agent or Cisco CallManager to operate in a voice network. During the call setup, Cisco IOS software instructs the DSP with the text relay parameters and modulations.

The difference between text relay and the text over G.711 audio codec is that with text relay, the tones are translated into characters. In text over G.711, the text is transported as tones. Text relay also has a better packet loss recovery rate.

Cisco Text Relay allows you to configure the text relay parameters on the gateway. Both the originating and terminating gateways must have text relay enabled for this feature to operate, and the payload types must match. Text relay calls are treated like DTMF relay, encapsulated into an RTP packet with a configured static or dynamic payload type, and sent to the other gateway. The configured text relay commands hold for the duration of the call.

Cisco Text Relay supports:

- PSTN-to-IP-to-PSTN networks, with connections from a PSTN text phone to a PSTN text phone.
- Baudot 45.45 and Baudot 50 TTY modulations.
- Internet Engineering Task Force (IETF) RFC 4351. Cisco Text Relay is partially compliant with the ITU V.151 recommendation.
- High-complexity, medium-complexity, and flex-complexity voice card configurations.

The support of all voice codecs saves bandwidth, and the redundancy level option repeats the data for redundancy and lowers the risk of packet loss, which improves the quality of the text messages.

How to Configure Cisco Text Relay for Baudot Text Phones

This section contains the following procedures:

- Configuring Cisco Text Relay Globally
- Configuring Cisco Text Relay for a Specific Dial Peer (H.323 and SIP Only)

Configuring Cisco Text Relay Globally

For MGCP configurations, Cisco Text Relay is configured globally. For H.323 and SIP configurations, Cisco Text Relay can be configured at two levels:

- Under voice service configuration mode—This configuration is the global, or system-wide, configuration that is applied to any VoIP call on the gateway. The default for voice service configuration mode is no text relay.
- Under dial peer voice configuration mode for VoIP dial peers—This configuration applies only to calls that match a specific dial peer. The default dial peer voice configuration is no text relay. See the "Configuring Cisco Text Relay for a Specific Dial Peer (H.323 and SIP Only)" section on page 6.

The two configuration tasks can be used separately or together. If both are configured, the dial peer voice configuration overrides the global configuration.

To configure Cisco Text Relay parameters globally for H.323, SIP, SCCP, and MGCP, perform the following task.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **text relay protocol cisco**
5. **text relay rtp** {[**payload-type** {*value* | **default**}] [**redundancy** *level*] | **redundancy** *level*}
6. **text relay modulation** {**baudot45.45** | **baudot50**} {**autobaud-on** | **autobaud-off**}

DETAILED STEPS

	Command or Action	Purpose

Step 1	`enable`
	
	Example:
	`Router> enable`
		Enables privileged EXEC mode.
		• Enter your password if prompted.

Step 2	`configure terminal`
	
	Example:
	`Router# configure terminal`
		Enters global configuration mode.

	Command or Action	Purpose
Step 3	**`voice service voip`** **Example:** `Router(config)# voice service voip`	Enters voice service configuration mode.
Step 4	**`text relay protocol cisco`** **Example:** `Router(conf-voi-serv)# text relay protocol cisco`	Enables text relay globally on the gateway.
Step 5	**`text relay rtp`** `{[`**`payload-type`** `{`*`value`* `|` **`default`**`}] [`**`redundancy`** *`level`*`] |` **`redundancy`** *`level`*`}` **Example:** `Router(conf-voi-serv)# text relay rtp payload-type default redundancy 2`	Configures the RTP payload type and redundancy level. • **payload-type**—Configures the RTP payload type for text packets. – The *value* range is 98 to 117 for dynamic RTP payload types. – The **default** value is 119, which is a static payload type. • **redundancy**—The redundancy level is the number of redundant text packets sent across the VoIP network. – The redundancy *level* range is 1 to 3. The default value is 2. **Note** You must configure Cisco Text Relay parameters on both the originating and terminating gateways, and the RTP payload type numbers must match. **Note** If you enable text relay and do not specify a payload type or redundancy level, the default values are used.
Step 6	**`text relay modulation`** `{`**`baudot45.45`** `|` **`baudot50`**`} {`**`autobaud-on`** `|` **`autobaud-off`**`}` **Example:** `Router(conf-voi-serv)# text relay modulation baudot50 autobaud-off`	Configures the TTY modulation on the gateway. The defaults are **baudot45.45** and **autobaud-on**. • **baudot45.45**—Configures Baudot 45.45 TTY modulation. • **baudot50**—Configures Baudot 50 TTY modulation. • **autobaud-on**— Enables the DSPs to autodetect the baud rate. • **autobaud-off**—Disables the DSP capability to autodetect the baud rate.

Configuring Cisco Text Relay for a Specific Dial Peer (H.323 and SIP Only)

To configure Cisco Text Relay for a specific dial peer for H.323 and SIP, complete the following task.

Note When Cisco Text Relay is configured for a specific dial peer, the dial peer voice configuration takes precedence over the global configuration.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **voip**
4. **text relay protocol** [**cisco** | **system**]
5. **text relay rtp** {[**payload-type** {*value* | **default**}] [**redundancy** *level*] | **redundancy** *level*}
6. **text relay modulation** {**baudot45.45** | **baudot50**} {**autobaud-on** | **autobaud-off**}

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**dial-peer voice** *tag* **voip** **Example:** `Router(config)# dial-peer voice 2000 voip`	Enters dial peer voice configuration mode.
Step 4	**text relay protocol** [**cisco** | **system**] **Example:** `Router(config-dial-peer)# text relay protocol cisco`	Sets the text relay protocol on this dial peer. • **cisco**—Uses the Cisco proprietary protocol. • **system—**Uses the global configuration settings. **Note** You must set the text relay protocol at the dial-peer level before you make configuration changes at the dial-peer level.

	Command or Action	Purpose
Step 5	**text relay rtp** {[**payload-type** {*value* | **default**}] [**redundancy** *level*] | **redundancy** *level*} **Example:** `Router(config-dial-peer)# text relay rtp payload-type default redundancy 2`	(Optional) Configures the RTP payload type and redundancy level. • **payload-type**—Configures the RTP payload type for text packets. – The *value* range is 98 to 117 for dynamic RTP payload types. – The **default** value is 119, which is a static payload type. • **redundancy**—The redundancy level is the number of redundant text packets sent across the VoIP network. – The redundancy *level* range is 1 to 3. The default is 2. **Note** If you enable text relay and do not specify a payload type or redundancy level, the default values are used.
Step 6	**text relay modulation** {**baudot45.45** | **baudot50**} {**autobaud-on** | **autobaud-off**} **Example:** `Router(config-dial-peer)# text relay modulation baudot50 autobaud-off`	Configures the TTY modulation for the dial peer. The defaults are **baudot45.45** and **autobaud-on**. • **baudot45.45**—Configures Baudot 45.45 TTY modulation. • **baudot50**—Configures Baudot 50 TTY modulation. • **autobaud-on**— Enables the DSPs to autodetect the baud rate. • **autobaud-off**— Disables the DSP capability to autodetect the baud rate.

Verifying and Troubleshooting Cisco Text Relay for Baudot Text Phones

Before using **debug** or **show** commands to troubleshoot Cisco Text Relay, be sure of the following:

- You can complete a voice call.
- Cisco Text Relay is configured on both the originating and terminating gateways.
- Both the originating and terminating gateways have the same payload type number and codec parameters.

Tip Try configuring a different text relay RTP payload type. Another application might be using the configured payload-type.

For troubleshooting, steps 1 and 2 can be used to troubleshoot the Cisco Text Relay configuration. To obtain information about the performance of the configuration, steps 3 through 8 can be used to display information about the history, status, and statistics for the Cisco Text Relay configuration.

SUMMARY STEPS

1. **debug voip rtp session text-relay**
2. **debug voip vtsp session**
3. **show call active voice brief id** *called-number*
4. **show call history voice**

5. **show dial peer voice**
6. **show mgcp**
7. **show port operational status**
8. **show voice call status**

DETAILED STEPS

Step 1 **debug voip rtp session text-relay**

This command displays specific session debug information for text relay.

Step 2 **debug voip vtsp session**

This command displays information that 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.

Step 3 **show call active voice brief id** *called-number*

This command displays a truncated version of information for the call with the specified identifier.

Step 4 **show call history voice**

This command displays the call history table for voice calls.

Step 5 **show dial-peer voice**

This command displays information for voice dial peers.

Step 6 **show mgcp**

This command provides high-level administrative information about the values configured for MGCP parameters on the router.

Step 7 **show port operational status**

This command displays the statistics for the operational status of a specific port or port range. The port should have an associated active session when the command is executed.

Step 8 **show voice call status**

This command displays the call status for active calls on the voice ports of the router.

Configuration Examples for Cisco Text Relay for Baudot Text Phones

This section provides the following configuration examples:

- Global Settings for Cisco Text Relay: Example, page 10
- Dial-Peer Settings for Cisco Text Relay: Example, page 11

Global Settings for Cisco Text Relay: Example

In the following example, the global configuration shows that the text relay cisco protocol is enabled and that the RTP redundancy level is changed from the default to level 3. The configuration for dial-peer 2000 shows that the redundancy level setting of 1 is different from, and takes precedence over, the global setting of 3.

```
version 12.3
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname 2691
!
boot-start-marker
boot system flash:c2691-ipvoice-mz.0404
boot-end-marker
!
logging buffered 100000 debugging
enable password lab
!
no aaa new-model
!
resource manager
!
memory-size iomem 25
clock timezone PST -8
clock summer-time PDT recurring
no network-clock-participate slot 1
voice-card 1
 codec complexity high
 dspfarm
!
ip subnet-zero
ip cef
!
!
no ip dhcp use vrf connected
!
!
no ip domain lookup
no ftp-server write-enable
isdn switch-type primary-qsig
!
voice service voip
 text relay protocol cisco
 text relay rtp redundancy 3
 text relay modulation baudot50 autobaud-off
 fax protocol pass-through g711ulaw
 sip
!
controller T1 1/0
 framing sf
 linecode ami
!
controller T1 1/1
 framing sf
 linecode ami
!
interface FastEthernet0/0
 ip address 10.0.0.4 255.255.0.0
 duplex auto
```

```
 speed auto
!
interface FastEthernet0/1
 no ip address
 shutdown
 duplex auto
 speed auto
!
ip default-gateway 10.2.0.1
ip classless
ip route 0.0.0.0 0.0.0.0 10.2.0.8
!
no ip http server
!
control-plane
!
dial-peer voice 2000 voip
modem relay nse codec g711ulaw redundancy gw-controlled
text relay protocol cisco
text relay rtp redundancy 1
fax rate disable
fax protocol pass-through g711alaw
!
!
line con 0
 exec-timeout 0 0
line aux 0
line vty 0 1
 exec-timeout 0 0
 password lab
 login
line vty 2 4
 login
!
ntp clock-period 17180770
ntp server 192.168.254.253 prefer
!
end
```

Dial-Peer Settings for Cisco Text Relay: Example

The following examples shows the text relay configuration on dial-peer 2000:

```
Router# show dial-peer voice 2000

VoiceOverIpPeer2000
 peer type = voice, information type = voice,
 description = `',
 tag = 2000, destination-pattern = `',
 answer-address = `', preference=0,
 CLID Restriction = None
 CLID Network Number = `'
 CLID Second Number sent
 CLID Override RDNIS = disabled,
 source carrier-id = `', target carrier-id = `',
 source trunk-group-label = `',  target trunk-group-label = `',
 numbering Type = `unknown'
 group = 2000, Admin state is up, Operation state is down,
 incoming called-number = `', connections/maximum = 0/unlimited,
 DTMF Relay = disabled,
 modem transport = relay, nse, payload-type = 100, codec = g711ulaw,
 redundancy, gateway-controlled,
```

```
URI classes:
Incoming (Called) =
Incoming (Calling) =
Destination =
huntstop = disabled,
in bound application associated: 'DEFAULT'
out bound application associated: ''
dnis-map =
permission :both
incoming COR list:maximum capability
outgoing COR list:minimum requirement
Translation profile (Incoming):
Translation profile (Outgoing):
incoming call blocking:
translation-profile = `'
disconnect-cause = `no-service'
advertise 0x40 capacity_update_timer 25 addrFamily 4 oldAddrFamily 4
type = voip, session-target = `',
technology prefix:
settle-call = disabled
ip media DSCP = ef, ip signaling DSCP = af31,
ip video rsvp-none DSCP = af41,ip video rsvp-pass DSCP = af41
ip video rsvp-fail DSCP = af41,
UDP checksum = disabled,
session-protocol = cisco, session-transport = system,
req-qos = best-effort, acc-qos = best-effort,
req-qos video = best-effort, acc-qos video = best-effort,
req-qos audio def bandwidth = 64, req-qos audio max bandwidth = 0,
req-qos video def bandwidth = 384, req-qos video max bandwidth = 0,
RTP dynamic payload-type values: NTE = 101
Cisco: NSE=100, fax=96, fax-ack=97, dtmf=121, fax-relay=122
CAS=123, TTY=119, ClearChan=125, PCM switch over u-law=0,A-law=8
RTP comfort noise payload-type = 19
fax rate = disable,   payload size =  20 bytes
fax protocol = pass-through
fax-relay ecm enable
Fax Relay SG3-to-G3 Enabled (by system configuration)
fax NSF = 0xAD0051 (default)
codec = g729r8,   payload size =  20 bytes,
text relay = enabled
tty protocol = cisco
tty rtp payload-type = 119 redundancy = 1
tty mod = baudot50 autobaud-off
Media Setting = flow-through (global)
Expect factor = 10, Icpif = 20,
Playout Mode is set to adaptive,
Initial 60 ms, Max 250 ms
Playout-delay Minimum mode is set to default, value 40 ms
Fax nominal 300 ms
Max Redirects = 1, signaling-type = cas,
VAD = enabled, Poor QOV Trap = disabled,
Source Interface = NONE
voice class sip url = system,
voice class sip rel1xx = system,
redirect ip2ip = disabled
probe disabled,
voice class perm tag = `'
Time elapsed since last clearing of voice call statistics never
Connect Time = 0, Charged Units = 0,
Successful Calls = 0, Failed Calls = 0, Incomplete Calls = 0
Accepted Calls = 0, Refused Calls = 0,
Last Disconnect Cause is "",
Last Disconnect Text is "",
Last Setup Time = 0.
```

Additional References

The following sections provide references related to the Cisco Text Relay feature.

Related Documents

Related Topic	Document Title
Secure TTY (Secure RTP)	*Media and Signaling Authentication and Encryption Feature for Cisco IOS MGCP Gateways*
Cisco IOS command references	• *Cisco IOS Debug Command Reference* • *Cisco IOS Voice Command Reference*
Cisco MGCP functionality	• *MGCP 1.0 Including NCS 1.0 and TGCP 1.0 Profiles* • *Cisco IOS MGCP and Related Protocols Configuration Guide*
Cisco SIP functionality	• *Session Initiation Protocol (SIP) for VoIP* • *Session Initiation Protocol Gateway Call Flows*

Standards

Standard	Title
T.140	*Protocol for Multimedia Application Text Conversation*
TIA-825-A	*A Frequency Shift Keyed Modem for Use on the Public Switched Telephone Network*
V.17	*A 2-Wire Modem for Facsimile Applications With Rates up to 14400 bps*
V.18	*Operational and Interworking Requirements for DCEs Operating in the Text Telephone Mode*
V.34	*A Modem Operating at Data Signalling Rates of up to 33600 bps for Use on the General Switched Telephone Network and on Leased Point-to-Point 2-Wire Telephone-Type Circuits*
V.90	*A Digital Modem and Analog Modem Pair for Use on the Public Switched Telephone Network (PSTN) at Data Signalling Rates of up to 56000 bps Downstream and up to 33600 bps Upstream*
V.151	*Procedures for the End-to-End Connection of Analog Text Telephones Over an IP Network* **Note** Cisco Text Relay is partially compliant to this draft standard.

MIB

MIBs	MIBs Link
No new or modified MIBs are supported by this feature, and support for existing MIBs has not been modified by this feature.	To locate and download MIBs for selected platforms, Cisco IOS releases, and feature sets, use Cisco MIB Locator found at the following URL: http://www.cisco.com/go/mibs

RFCs

RFC	Title
RFC 2198	*RTP Payload for Redundant Audio Data*
RFC 2793bis	*RTP Payload for Text Conversation*
RFC 4351	*RTP Payload for Text Conversation Interleaved in an Audio Stream*

Technical Assistance

Description	Link
The Cisco Technical Support website contains thousands of pages of searchable technical content, including links to products, technologies, solutions, technical tips, and tools. Registered Cisco.com users can log in from this page to access even more content.	http://www.cisco.com/techsupport

Command Reference

This section documents modified commands only.

- **debug voip rtp**
- **text relay modulation**
- **text relay protocol**
- **text relay rtp**

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** [**conference** | **dtmf-relay** | **event** | **multicast** | **named-event** [*payload-type*] | **nse** | **text-relay**] | **packet** [**callid** *id-number packet-number* | **remote-ip** *ip-address* **remote-port** *port-number packet-number*]}

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.
conference	(Optional) Provides debug information for conference packets.
dtmf-relay	(Optional) Provides debug information for dual-tone multifrequency (DTMF) packets.
event	(Optional) Enables VoIP RTP session generic event debugging trace.
multicast	(Optional) Provides debug information for multicast packets.
named-event	(Optional) Provides debug information for named telephony event (NTE) packets.
nse	(Optional) Provides debug information for named signaling events (NSEs).
text-relay	(Optional) Provides debug information for text-relay packets.
packet	Enables VoIP RTP packet debugging trace.
callid *id-number packet-number*	(Optional) Provides debug information for a specific call ID number (obtained by using the **show voip rtp connections** command). The *packet-number* argument specifies the number of packets to trace so that the display is not flooded.
remote-ip *ip-address* **remote-port** *port-number packet-number*	(Optional) Provides debug information for a remote IP address and RTP port number. The *packet-number* argument specifies the number of packets to trace so that the display is not flooded.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.2(2)XB	This command was introduced.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T.
12.4(4)XC	This command was implemented on the Cisco AS5300, Cisco AS5400, and Cisco AS5850.
12.2(15)T	This command was implemented on the Cisco 1751 and Cisco 1760.
12.4(6)T	The **text-relay** keyword was added.

Usage Guidelines

This command severely impacts performance and should be used only for single-call debug capture.

Examples

The following example shows debugging 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
```

The following example shows debugging output for the **debug voip rtp session text-relay** command:

```
Router# debug voip rtp session text-relay

Pt:119    Evt:0     4     247   37    128    Cnt:F7 4B  <Snd>>>
```

Related Commands

Command	Description
text relay protocol	Configures the system-wide protocol type for text packets transmitted between gateways.
text relay rtp	Configures the RTP payload type and redundancy level.

text relay modulation

To configure the teletype text phone (TTY) modulation used on the gateway for Cisco text relay for Baudot text phones, use the **text relay modulation** command in dial peer voice configuration mode or voice service configuration mode. To disable text relay modulation, use the **no** form of this command.

text relay modulation {**baudot45.45** | **baudot50**} {**autobaud-on** | **autobaud-off**}

no text relay modulation

Syntax Description

baudot45.45	Configures baudot 45.45 TTY modulation. This is the default baud rate.
baudot50	Configures baudot 50 TTY modulation.
autobaud-on	Enables the digital signal processors (DSPs) to autodetect the baud rate. This is the default autobaud setting.
autobaud-off	Disables the DSP capability to autodetect the baud rate.

Command Default

The TTY modulation is **baudot45.45 autobaud-on**.

Command Modes

Dial peer voice configuration
Voice service configuration

Command History

Release	Modification
12.4(6)T	This command was introduced.

Usage Guidelines

You must select a baud rate and enable or disable the autobaud functionality on the DSP.

Use this command in voice service configuration mode to set the TTY modulation globally. A global configuration is the system-wide configuration that is applied to any VoIP call on the gateway.

Use this command in dial peer voice configuration mode to set the TTY modulation for calls that match a specific dial peer. The dial peer voice configuration takes precedence over the global configuration.

Examples

The following example shows how to globally set the text relay TTY modulation to Baudot 50:

```
Router(config)# voice service voip
Router(config-voi-serv)# text relay modulation baudot50 autobaud-off
```

The following example shows how to set the text relay TTY modulation to Baudot 50 for calls that match a specific dial peer:

```
Router(config)# dial-peer voice 2000 voip
Router(config-dial-peer)# text relay modulation baudot50 autobaud-off
```

Related Commands

Command	Description
text relay protocol	Configures the system-wide protocol type for text packets transmitted between gateways.
text relay rtp	Configures the RTP payload type and redundancy level.

text relay protocol

To enable Cisco text relay for Baudot text phones, use the **text relay protocol** command in dial peer voice configuration mode or voice service configuration mode. To disable text relay capabilities, use the **no** form of this command.

text relay protocol [**cisco** | **system**]

no text relay protocol

Syntax Description

cisco	(Optional) Uses the Cisco proprietary text relay protocol.
system	(Optional; dial peer voice configuration only) Uses the global configuration settings.

Command Default

The text relay protocol is disabled.

Command Modes

Dial-peer configuration
Voice-service configuration

Command History

Release	Modification
12.4(6)T	This command was introduced.

Usage Guidelines

Use this command in voice-service configuration mode to enable text relay globally for H.323, Session Initiation Protocol (SIP), Skinny Client Control Protocol (SCCP), and Media Gateway Control Protocol (MGCP). A global configuration is the system-wide configuration that is applied to any VoIP call on the gateway.

Use this command in dial peer voice configuration mode to enable text relay for calls that match a specific dial peer. The dial peer voice configuration takes precedence over the global configuration.

Examples

The following example shows how to enable text relay for all VoIP calls on the gateway:

```
Router(config)# voice service voip
Router(config-voi-serv)# text relay protocol cisco
```

The following example shows how to enable text relay for calls that match a specific dial peer:

```
Router(config)# dial-peer voice 2000 voip
Router(config-dial-peer)# text relay protocol cisco
```

Related Commands

Command	Description
text relay modulation	Configures the TTY modulation on the gateway.
text relay rtp	Configures the RTP payload type and redundancy level.

text relay rtp

To configure the Real-Time Transport Protocol (RTP) payload type and redundancy level for Cisco text relay for Baudot text phones, use the **text relay rtp** command in dial peer voice configuration mode or voice service configuration mode. To disable the text relay RTP payload type and redundancy level, use the **no** form of this command.

text relay rtp {[**payload-type** {*value* | **default**}] [**redundancy** *level*] | **redundancy** *level*}

no text relay rtp

Syntax Description

payload-type {*value* | **default**}	The RTP payload is the data transported by RTP in a packet. • The *value* range is 98 to 117 for dynamic RTP payload types. • The **default** value is 119, which is a static payload type.
redundancy *level*	Use the redundancy option to repeat data for redundancy and to lower the risk of packet loss. The redundancy level is the number of redundant text packets sent across the VoIP network. The range is 1 to 3. The default value is 2.

Command Default

Text relay RTP is disabled.

Command Modes

Dial peer voice configuration
Voice service configuration

Command History

Release	Modification
12.4(6)T	This command was introduced.

Usage Guidelines

When using the **text relay rtp** command, you can either configure the payload-type, or the redundancy level, or both.

- Use this command in voice service configuration mode to set the RTP payload type and redundancy level globally for H.323, Session Initiation Protocol (SIP), Skinny Client Control Protocol (SCCP), and Media Gateway Control Protocol (MGCP). A global configuration is the system-wide configuration that is applied to any VoIP call on the gateway.
- Use this command in dial-peer configuration mode to set the RTP payload type and redundancy level for calls that match a specific dial peer. The dial peer voice configuration takes precedence over the global configuration.

Examples

The following example shows how to globally configure text relay RTP payload type 117 and redundancy level 2:

```
Router(config)# voice service voip
Router(config-voi-serv)# text relay rtp payload-type 117 redundancy 2
```

The following example shows how to configure the default text relay RTP payload type and redundancy level 1 for calls that match a specific dial peer:

```
Router(config)# dial-peer voice 2000 voip
Router(config-dial-peer)# text relay rtp payload-type default redundancy 1
```

Related Commands

Command	Description
text relay modulation	Configures the TTY modulation on the gateway.
text relay protocol	Configures the system-wide protocol type for text packets transmitted between gateways.

Feature Information for Cisco Text Relay for Baudot Text Phones

Table 2 lists the release history for this feature.

Cisco IOS software images are specific to a Cisco IOS software release, a feature set, and a platform. Use Cisco Feature Navigator to find information about platform support and Cisco IOS software image support. Access Cisco Feature Navigator at http://www.cisco.com/go/fn. You must have an account on Cisco.com. If you do not have an account or have forgotten your username or password, click **Cancel** at the login dialog box and follow the instructions that appear.

Note Table 2 lists only the Cisco IOS software release that introduced support for a given feature in a given Cisco IOS software release train. Unless noted otherwise, subsequent releases of that Cisco IOS software release train also support that feature.

Table 2 Feature Information for Cisco Text Relay for Baudot Text Phones

Feature Name	Releases	Feature Information
Cisco Text Relay for Baudot Text Phones	12.4(6)T 12.4(4)XC	The Cisco Text Relay for Baudot Text Phones (Cisco Text Relay) feature implements a mechanism for reliable transport of teletype text phone (TTY) signals over VoIP calls. TTY phones, also known as telecommunication devices for the deaf (TDD), are specialized phones that enable people who are deaf, hard of hearing, or speech-impaired to communicate over time-division multiplex (TDM) or IP networks by allowing them to type messages to one another instead of, or in addition to, talking and listening. Cisco Text Relay is based on the proposed ITU V.151 (V.ToIP) standard. Cisco Text Relay leverages Audio/T.140, which means text characters are carried over the same Real-Time Transport Protocol (RTP) stream as voice (similar to the way dual tone multifrequency (DTMF) characters are carried in the RTP stream [RFC 2833]). Cisco Text Relay has minimal impact on bandwidth because the text characters are transported efficiently in the RTP stream. There is also a configurable redundancy option enabling TTY to run reliably in demanding network conditions. This feature was integrated into Cisco IOS Release 12.4(4)XC and support was added for the Cisco AS5400XM and Cisco AS5350XM.

Glossary

ANS tone—answer tone. Also called CED tone. A 2100-Hz tone used to disable echo suppression for data transmission.

ANSam—A modified answer tone sent by V.8-capable equipment.

CM—call menu signal. A V.8 signal transmitted from the *call* DCE to indicate modulation modes available in the call DCE. CM consists of a repeating sequence of bits, modulated using V.21 low-band channel. See joint menu signal (JM).

DSMP—distributed stream media processor. The software component that controls the DSP on behalf of the media service provider that handles conference calls. The DSMP also manages packet transmission and reception in fast switching.

HPI—host port interface. The interface used to communicate with TI DSPs.

JM—joint menu signal. A V.8 signal transmitted from the *answer* DCE to indicate modulation modes available jointly in the call and answer DCEs. JM consists of a repeating sequence of bits, modulated using V.21 high-band channel. See call menu signal (CM).

TDD—telecommunication device for the deaf. Phones that enable those who are deaf, hard of hearing, or speech-impaired to communicate over TDM. See TTY.

text relay—Text payload that is encapsulated into RTP packets with a negotiated payload type and sent over an IP network as part of a voice call. Text relay is controlled by the voice gateways.

ToIP—Text over IP. The transport of analog modulated signals generated by PSTN text telephones over an IP Network.

TTY—text telephones or teletype phones. Phones that enable those who are deaf, hard of hearing, or speech-impaired to communicate over TDM. See TDD.

Note Refer to *Internetworking Terms and Acronyms* for terms not included in this glossary.