Call Release Source Reporting in Gateway-Generated Call Accounting Records

Feature History

Release	Modification
12.2(13)T	This feature was introduced on the Cisco 2600 series, Cisco 3600 series, Cisco 3700 series, Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco 7200 series.

The Call Release Source Reporting in Gateway-Generated Call Accounting Records feature in Cisco IOS Release 12.2(13)T allows you to identify the source of a call release in a Voice over IP (VoIP) network. The document includes the following sections:

Feature Overview

The Call Release Source Reporting in Gateway-Generated Call Accounting Records feature allows you to identify the source of a call release for calls in a VoIP network. Each call established through the VoIP network has incoming and outgoing call legs associated with it. There are corresponding START and STOP RADIUS accounting records for each of the VoIP and plain old telephone service (POTS) call legs. One of the critical parameters in analyzing the STOP records for a particular call is the source of a call release. This release source is complementary to the associated International Telecommunication Union Telecommunication Standardization Sector (ITU-T) standard Q.850 cause code, which is shown in the call history records and vendor-specific attributes (VSAs). By following the release source on each of the STOP records, you can detect which component originated the call release.

The release source information is displayed in a VSA called “release-source” in the call accounting records (referred to as call detail records [CDRs]). This VSA “release-source” information is included in STOP records that are exchanged between the voice gateway and the RADIUS server. The release source information is also displayed in an attribute called “cCallHistoryReleaseSrc” in the call history table of the CISCO-DIAL-CONTROL-MIB. The release source values are filled in by the gateway at the time of call release. The values define whether the call was released by the calling party, the called party, or an internal or external source.

In addition to viewing the release source information on your consoles, you may also use command-line interface (CLI). Use the **show call history voice** command for output that includes a field called “ReleaseSource.”

RADIUS VSA Release Source

The RADIUS VSA provides the following information:

```
Attribute Type: 26
Vendor Type: 9 (cisco)
Vendor Attribute: 1 (Generic)
Attribute Name: release-source
Value Type: Enumeration
```

In this example, the vendor type is Cisco, and the VSA is a generic VSA.

The format of the VSA is AV pair: release-source=value, where the value portion of each attribute-value (AV) pair contains an enumeration (number) value that represents the source of the call release.

By default, non-RFC-mandatory VSAs are not included in accounting records if you do not configure the accounting template. The accounting template enables you to manage accounting records at a per-VSA level. When an accounting template is used for customizing the accounting record, the VSA name release source has to be included in the template file so that it will be included in the accounting record and sent to a RADIUS server. Refer to the following commands for more information:

- **accounting template** —Allows each dial peer to choose and send a customized accounting template to the RADIUS server.
- **acct-template—**Sends a selected group of voice accounting VSAs.
- **show call accounting template master—**Displays a list of all available VSAs in the voice accounting record.

Refer to the *RADIUS VSA Voice Implementation Guide*.

CISCO-DIAL-CONTROL-MIB

The release source information is displayed in the CISCO-DIAL-CONTROL-MIB call history table as follows:

```
cCallHistoryReleaseSrc OBJECT-TYPE
   SYNTAX INTEGER}
     callingPartyInPstn(1),
     callingPartyInVoip(2),
!and so on

MAX-ACCESS read-only
STATUS current
```

Call History Records

The **show call history voice** command displays the VSA information in the following format:

ReleaseSource=value

Release Source Values

With respect to a single network, the following release sources are possible.

Note The following numbers are assigned to the release sources listed.

1. Calling party located in the public switched telephone network (PSTN)
2. Calling party located in the VoIP network
3. Called party located in the PSTN
4. Called party located in the VoIP network
5. Internal release in a POTS leg
6. Internal release in a VoIP leg
7. Internal call-control application (for example, Tool Command Language [TCL] or Voice eXtensible Markup Language [VXML] script)
8. Internal release in VoIP authentication, authorization, and accounting (AAA)
9. CLI or Man Machine Language (MML)
10. External RADIUS server
11. External network management application
12. External call control agent (for example, a gatekeeper)

Call Direction and Release Source

The call direction is recorded at the time of call establishment. The release source is calculated at the time that the gateway receives notice of an internal or external event that triggers the release of the call. Internal events include the following:

- Socket failure in connection with the remote VoIP endpoint.
- Domain Name System (DNS) resolution failure for the session target specified in the dial peer.
- Established H.225 socket connection is lost due to an error in the socket.
- Gateway is out of retries to send a 1xx response after it has received an INVITE request. The gateway is waiting for a Provisional Acknowledgement (PRACK) response.
- Call was rejected because it matched the profile defined for the blocking of incoming calls.
- Lack of a dial peer to satisfy the match criteria for accepting or handling the call.
- Unknown call mode has been specified to set up a call.
- Insufficient digital signal processor (DSP) resources to handle the call.

External events include receipt of the following messages:

- Release complete message from the PSTN.
- Release complete message from the VoIP side.
- Disengage request (DRQ), admission reject (ARJ), or bandwidth reject (BRJ) messages from the gatekeeper.
- BYE, 4xx, 5xx, or 6xx messages from a remote user agent (UA) or Session Initiation Protocol (SIP) proxy.
- Disconnect from the RADIUS server.
- Ethernet Interface shutdown command.

The following sections show various call-disconnect scenarios and the release sources for the call legs of each scenario.

Disconnect from the PSTN Received by the Originating Gateway

Originating Gateway	Terminating Gateway
POTS Leg: Calling party located in the PSTN	VoIP Leg: Calling party located in the VoIP network
VoIP Leg: Calling party located in the PSTN	POTS Leg: Calling party located in the VoIP network
Because the gateway (either the originating gateway or the terminating gateway) did not encounter an internal release, the VSAs generated by the four call legs point to the calling party in the PSTN on the basis of the direction of the Release Complete message.	

Disconnect from the PSTN Received by the Terminating Gateway

Originating Gateway	Terminating Gateway
POTS Leg: Called party located in the VoIP network	VoIP Leg: Called party located in the PSTN
VoIP Leg: Called party located in the VoIP network	POTS Leg: Called party located in the PSTN
Because the gateway (either the originating gateway or the terminating gateway) did not encounter an internal release, the VSAs generated by the four call legs point to the called party in the PSTN on the basis of the direction of the Release Complete message.	

Disconnect Initiated by the Originating Gateway

In the following three scenarios, the call could be released because of an error in the POTS leg or the VoIP leg within the originating gateway.

1. Call Released Because of an Error in the POTS Leg

Originating Gateway	Terminating Gateway
POTS Leg: Internal error in the POTS leg	VoIP Leg: Calling party located in the VoIP network
VoIP Leg: Internal error in the POTS leg	POTS Leg: Calling party located in the VoIP network

2. Call Released Because of an Error in the VoIP Leg

Originating Gateway	Terminating Gateway
POTS Leg: Internal error in the VoIP leg	VoIP Leg: Calling party located in the VoIP network
VoIP Leg: Internal error in the VoIP leg	POTS Leg: Calling party located in the VoIP network

3. Disconnect Initiated by the Session Control Application on the Originating Gateway

Originating Gateway	Terminating Gateway
POTS Leg: Internal call control application	VoIP Leg: Calling party located in the VoIP
VoIP Leg: Internal call control application	POTS Leg: Calling party located in the VoIP

Disconnect Initiated by the Terminating Gateway

In the following two scenarios, the call could be released because of an error in the POTS leg or the VoIP leg within the terminating gateway.

1. Call Released Because of an Error in the VoIP Leg

Originating Gateway	Terminating Gateway
POTS Leg: Called party located in the VoIP network	VoIP Leg: Internal error in the VoIP leg
VoIP Leg: Called party located in the VoIP network	POTS Leg: Internal error in the VoIP leg

2. Call Released Because of an Error in the POTS Leg

Originating Gateway	Terminating Gateway
POTS Leg: Called party located in the VoIP network	VoIP Leg: Internal error in the POTS leg
VoIP Leg: Called party located in the VoIP network	POTS Leg: Internal error in the POTS leg

Benefits

- Large VoIP network deployments are easier to manage and monitor.
- Information is provided that may be used to locate and troubleshoot faulty network components, thus reducing technical assistance turnaround time and the costs of operation and maintenance.

Restrictions

The release source values from each of the call legs involved in a call must be considered in pointing to the source of the release. If the Cisco gateway is operating with a remote gateway that has a version earlier than Cisco IOS Release 12.2(13)T or with a non-Cisco gateway, you will receive only partial information about the release initiated by the remote source. That is, the ReleaseSource VSA information will be available only for the gateway that is configured with the Call Release Source Reporting in Gateway-Generated Call Accounting Records feature. The ReleaseSource information may be incomplete for identifying the exact source of the call release. However, a Cisco gateway-generated ReleaseSource would point in the direction of the call release to which it is related in the VoIP network.

Related Documents

- *Cisco IOS Voice, Video, and Fax Configuration Guide*, Release 12.2
- *RADIUS VSA Voice Implementation Guide*
- *Cisco IOS IP Configuration Guide*, Release 12.2
- *Voice—Understanding Dial Peers and Call Legs on Cisco IOS Platforms*

Supported Platforms

- Cisco 2610–2613
- Cisco 2620–2621
- Cisco 2650–2651
- Cisco 2691
- Cisco 3620
- Cisco 3631
- Cisco 3640
- Cisco 3660
- Cisco 3725
- Cisco 3745
- Cisco AS5300
- Cisco AS5350
- Cisco AS5400
- Cisco 7200 series

Determining Platform Support Through Cisco Feature Navigator

Cisco IOS software is packaged in feature sets that are supported on specific platforms. To get updated information regarding platform support for this feature, access Cisco Feature Navigator. Cisco Feature Navigator dynamically updates the list of supported platforms as new platform support is added for the feature.

Cisco Feature Navigator is a web-based tool that enables you to determine which Cisco IOS software images support a specific set of features and which features are supported in a specific Cisco IOS image. You can search by feature or release. Under the release section, you can compare releases side by side to display both the features unique to each software release and the features in common.

To access Cisco Feature Navigator, you must have an account on Cisco.com. If you have forgotten or lost your account information, send a blank e-mail to cco-locksmith@cisco.com. An automatic check will verify that your e-mail address is registered with Cisco.com. If the check is successful, account details with a new random password will be e-mailed to you. Qualified users can establish an account on Cisco.com by following the directions found at this URL:

http://www.cisco.com/register

Cisco Feature Navigator is updated regularly when major Cisco IOS software releases and technology releases occur. For the most current information, go to the Cisco Feature Navigator home page at the following URL:

http://www.cisco.com/go/fn

Availability of Cisco IOS Software Images

Platform support for particular Cisco IOS software releases is dependent on the availability of the software images for those platforms. Software images for some platforms may be deferred, delayed, or changed without prior notice. For updated information about platform support and availability of software images for each Cisco IOS software release, refer to the online release notes or, if supported, Cisco Feature Navigator.

Supported Standards, MIBs, and RFCs

Standards

No new or modified standards are supported by this feature.

MIBs

The Call Release Source Reporting in Gateway-Generated Call Accounting Records feature supports the following modified MIB:

- CISCO-DIAL-CONTROL-MIB

To locate and download MIBs for selected platforms, Cisco IOS releases, and feature sets, use Cisco MIB Locator found at the following URL:

http://tools.cisco.com/ITDIT/MIBS/servlet/index

If Cisco MIB Locator does not support the MIB information that you need, you can also obtain a list of supported MIBs and download MIBs from the Cisco MIBs page at the following URL:

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

To access Cisco MIB Locator, 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 found at this URL:

http://www.cisco.com/register

RFCs

No new or modified RFCs are supported by this feature.

Prerequisites

- Ensure that the gateway has voice functionality that is configurable for H.323 and SIP.
- Establish a working IP network. For more information about configuring IP, refer to *Cisco IOS IP Configuration Guide*, Release 12.2.
- Configure the gateway to recognize the RADIUS server.
- Configure VoIP. If VoIP is not configured, the calls will not be routed through VoIP. For more information about configuring VoIP, see *Cisco IOS Voice, Video, and Fax Configuration Guide*, Release 12.2.

Configuration Tasks

None

Troubleshooting Tips

To troubleshoot this feature, use the following commands, displaying only standard messaging (for example, Q.931, H.225, H.245, RAS, or SIP):

- To find out whether the ISDN link is up or down, use the **show isdn status** command.
- To display information about whether the ISDN link is getting the SETUP, CALLPRO, ALERT, CONNECT, and RELEASE COMPLETE messages, use the **debug isdn q931** command.
- To display information about H.225 and RAS messages exchanged between a gateway and gatekeeper, use the **debug h225 asn1** command.
- To display ASN1 contents of H.245 messages, use the **debug h245 asn1** command.
- To enable SIP-related debugging, use the **debug ccsip** commands.

Monitoring and Maintaining Call Release Source Reporting in Gateway-Generated Call Accounting Records

To display RADIUS VSA release source information through the CLI, use the following command:

Command	Purpose
`Router# show call history voice`	Displays CDR events in the call history table.

Configuration Examples

None

Command Reference

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

- **show call history voice**

show call history voice

To display call detail record (CDR) events in the call history table, use the **show call history voice** command in privileged EXEC mode.

show call history voice

Syntax Description

This command has no arguments or keywords.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.0(5)XK	This command was introduced on Cisco MC3810 multiservice access concentrators.
12.0(7)T	This command was integrated into Cisco IOS Release 12.0(7)T.
12.2(13)T	The ReleaseSource field was added to the Field Description table, and the word "record" was deleted from the command name.

Examples

The following example displays a sample of voice call history records showing a local call between two telephones attached to the same Cisco MC3810 multiservice access concentrator:

```
Router# show call history voice

ConnectionId=[0x2C7AEFDC 0x59830001 0x0 0xB0AAA3]
Media=TELE, TxDuration= 1418 ms
CallingNumber=2001
SetupTime=1157801 x 10ms
ConnectTime=1158046 x 10ms
DisconectTime=1158188 x 10ms
DisconnectText=local onhook

ConnectionId=[0x2C7AEFDC 0x59830001 0x0 0xB0AAA3]
Media=TELE, TxDuration= 1422 ms
CalledNumber=2002
SetupTime=1157802 x 10ms
ConnectTime=1158046 x 10ms
DisconectTime=1158188 x 10ms
DisconnectText=remote onhook
```

The following example from a Cisco AS5350 router displays a sample of voice call history records showing release source information:

```
Router# show call history voice

Telephony call-legs: 1
SIP call-legs: 0
H323 call-legs: 1
Total call-legs: 2

GENERIC:
SetupTime=85975291 ms
```

```
.
.
.
DisconnectCause=10
DisconnectText=normal call clearing (16)
ConnectTime=85975335
DisconnectTime=85979339
CallDuration=00:00:40
CallOrigin=1
ReleaseSource=1
.
.
.
VOIP:
ConnectionId[0x2868AD84 0x375B11D4 0x8012F7A5 0x74DE971E]
.
.
.
GENERIC:
SetupTime=85975290 ms
.
.
.
DisconnectCause=10
DisconnectText=normal call clearing (16)
ConnectTime=85975336
DisconnectTime=85979340
CallDuration=00:00:40
CallOrigin=2
ReleaseSource=1
.
.
.
TELE:
ConnectionId=[0x2868AD84 0x375B11D4 0x8012F7A5 0x74DE971E]
```

Table 1 describes significant fields in these output examples.

Table 1 show call history voice Field Descriptions

Field	Description
ConnectionID	Global call identifier for this voice call.
Media	Medium over which the call is carried. If the call is carried over the (telephone) access side, the entry is TELE. If the call is carried over the voice network side, the entry is either ATM, FR (for Frame Relay), or HDLC (for High-Level Data Link Control).
LowerIFName	Physical lower interface information. Appears only if the medium is ATM, FR, or HDLC.
TxDuration	The length of the call. Appears only if the medium is TELE.
CalledNumber	The called number.
CallingNumber	The calling number.
SetupTime	Time at which the call setup started.
ConnectTime	Time at which the call is connected.
DisconnectTime	Time at which the call is disconnected.
DisconnectText	Descriptive text that explains the reason for the disconnect.

Table 1 show call history voice Field Descriptions (continued)

Field	Description
Telephony call-legs	Total telephony call legs for which call history records are available.
SIP call-legs	Total SIP call legs for which call history records are available.
H323 call-legs	Total H.323 call legs for which call history records are available.
GENERIC	Generic or common parameters, that is, parameters that are common for Voice over IP (VoIP) and telephony call legs.
DisconnectCause	Q.931 disconnect cause code retrieved from the call control application programming interface (CCAPI). The source of the code is the disconnect location, such as a public switched telephone network (PSTN), terminating gateway, or originating gateway.
CallDuration	Duration of the call.
CallOrigin	The gateway's behavior in relation to the connection that is active for this leg.
ReleaseSource	Number value of the release source.
VOIP	VoIP call leg, either H.323 or SIP.
TELE	Telephony call leg.

Related Commands

Command	Description
show call active voice	Displays the VoIP active call table.
show dial-peer voice	Displays configuration information for dial peers.
show num-exp	Displays how the number expansions are configured in VoIP.
show voice port	Displays configuration information about a specific voice port.

Glossary

AAA—authentication, authorization, and accounting.

ARJ—admission reject message.

BRJ—bandwidth reject message.

CAC—Call Admission Control.

CCAPI—call control application programming interface.

CDR—call detail record. A CDR is a record written to a database for use in postprocessing activities. CDR files consist of several call detail blocks. These activities include many functions, but primarily are billing and network analysis.

CLI—command-line interface.

DRQ—disengage request message.

DSP—digital signal processor.

HDLC—High-Level Data Link Control. HDLC is a bit-oriented synchronous data link layer protocol developed by ISO. It is derived from Synchronous Data Link Control (SDLC). HDLC specifies a data encapsulation method on synchronous serial links using frame characters and checksums.

ITU-T—International Telecommunication Union Telecommunication Standardization Sector.

MML—Man Machine Language.

Packet of Disconnect—a RADIUS access_request packet that is intended to be used in situations where the authenticating agent server wants to disconnect the user after the session has been accepted by the RADIUS access_accept packet.

POTS—plain old telephone service. POTS is also referred to as public switched telephone network, which is a general term that refers to the variety of telephone networks and services in place worldwide.

PSTN—See POTS.

RADIUS—Remote Authentication Dial-In User Service. Database for authenticating modem and ISDN connections and for tracking connection time.

UA—user agent.

VoIP—Voice over IP. The capability to carry normal telephony-style voice over an IP-based Internet with POTS-like functionality, reliability, and voice quality. VoIP enables a router to carry voice traffic (for example, telephone calls and faxes) over an IP network. In VoIP, the digital signal processor (DSP) segments the voice signal into frames, which then are coupled in groups of two and stored in voice packets. These voice packets are transported using IP in compliance with ITU-T specification H.323.

VSA—vendor-specific attribute. A VSA is an attribute that has been implemented by a particular vendor. It uses the attribute Vendor-Specific to encapsulate the resulting attribute-vendor (AV) pair: essentially, Vendor-Specific = protocol:attribute = value.