Call Preservation for H.323 VoIP Calls

Revised: December 12, 2005

History for Call Preservation for H.323 VoIP Calls Feature

Release	Cisco CallManager Express (Cisco CME) Version	Modification
12.4(4)XC	4.0	This feature was introduced.

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Prerequisites for Call Preservation for H.323 VoIP Calls

- For bidirectional silence detection, Cisco IOS gateways with 5510 digital signal processors (DSPs) are needed.
- It is recommended that media inactivity detection be configured so that preserved calls are torn down after conversations are over. Two available media inactivity detection features are discussed in the "How to Configure Call Preservation for H.323 VoIP Calls" section on page 4. They are Real-time Transport Protocol (RTP) and RTP Control Protocol (RTCP) inactivity detection and bidirectional silence detection. For more information about media inactivity detection, see the "Configuring Media Inactive Call Detection" chapter in the *Cisco IOS Tcl IVR and VoiceXML Application Guide —12.3(14)T and Later*.

Restrictions for Call Preservation for H.323 VoIP Calls

Call preservation for H.323 VoIP calls does not support the following:

- Calls in transient call states.
- Calls in for which a H.225.0 connection has not occurred,
- Calls on which supplementary services are in progress. For example, when one of the parties is on hold.
- Calls that involve a media resource located across a WAN, such as conference resources.
- Calls where the two parties are registered to different CallManager clusters.

Information About Call Preservation for H.323 VoIP Calls

This feature expands Cisco Survivable Remote Site Telephony's (Cisco SRST's) call preservation functionality to include the following types of failures and connections:

Failure Types

- WAN failures that include WAN links flapping or degraded WAN links
- Cisco CallManager software failure, such as when the ccm.exe service crashes on a Cisco CallManager server.
- LAN connectivity failure, except when a failure occurs at the local branch

Connection Types

- Calls between two Cisco CallManager controlled endpoints
  - During Cisco CallManager reloads
  - When a Transmission Control Protocol (TCP) connection between one or both endpoints and Cisco CallManager used for signaling H.225.0 or H.245 messages is lost or flapping
  - Between endpoints that are registered to different Cisco CallManagers in a cluster and the TCP connection between the two Cisco CallManagers is lost
  - Between IP phones and the PSTN at the same site

- Calls between Cisco IOS gateway and an endpoint controlled by a softswitch where the signaling (H.225.0, H.245 or both) flows between the gateway and the softswitch and media flows between the gateway and the endpoint.
  - When the softswitch reloads.
  - When the H.225.0 or H.245 TCP connection between the gateway and the softswitch is lost, and the softswitch does not clear the call on the endpoint
  - When the H.225.0 or H.245 TCP connection between softswitch and the endpoint is lost, and the soft-switch does not clear the call on the gateway
- Call flows that involve a Cisco IP in IP (IPIP) gateway running in media flow-around mode that reload or lose connection with the rest of the network

Note that after the media is preserved, the call is torn down later when either one of the parties hangs up or media inactivity is detected. In cases where there is a machine-generated media stream, such as music streaming from a media server, the media inactivity detection will not work and the call may hang. Cisco CallManager addresses such conditions by indicating to the gateway that such calls should not be preserved, but third-party devices or IPIP gateways would not do this.

Flapping is defined for this feature as the repeated and temporary loss of IP connectivity that can be caused by WAN or LAN failures. H.323 VoIP calls between a Cisco IOS gateway and Cisco CallManager may be torn down when flapping occurs. When Cisco CallManager detects that the TCP connection is lost, it clears the call and closes the TCP sockets used for the call by sending a TCP FIN, without sending an “H.225.0 Release Complete” or “H.245 End Session” message. This is called quiet clearing. The TCP FIN sent from the Cisco CallManager could reach the gateway if the network comes up for a short duration, and the gateway will tear the call down. Even if the TCP FIN does not reach the gateway, the TCP keepalives sent from the gateway could reach Cisco CallManager when the network comes up. Cisco CallManager will send TCP RST messages in response to the keepalives as it has already closed the TCP connection. The gateway will tear down H.323 calls if it receives the RST message.

Configuration of the Call Preservation for H.323 VoIP Calls feature involves configuring the **call preserve** command and enabling the “Allow Peer to Preserve H.323 Calls” parameter from Cisco CallManager’s Service Parameters window.

The **call preserve** command causes the gateway to ignore socket closure or socket errors on H.225.0 or H.245 connections for active calls, allowing the socket to be closed without tearing down calls using those connections.

The Cisco CallManager’s “Allow Peer to Preserve H.323 Calls” parameter preserves the following:

- Active H323 calls with quiet clear triggered by the other half of the call
- Active H323 calls with TCP socket closed on the H.323 end before the H.225 or H.245 release signal is received
- Active H323 calls with a signal distribution layer (SDL) link that is out of service and detected on the H323 end

Management

Call preservation may be reported through Syslog, which optionally can be obtained through a simple network management protocol (SNMP) trap. New syslog messages are printed when call preservation is applied. An SNMP trap can be configured on this syslog message, so you can be notified when call preservation occurs on a gateway.

Preservation information is displayed through the **show h323 calls preserved** command. The following is an example of the command’s output:

```
CallID = 11EC , Calling Number = , Called Number = 3210000 ,
RemoteSignallingIPAddress=9.13.0.26 , RemoteSignallingPort=49760 ,
RemoteMediaIPAddress=9.13.0.11 , RemoteMediaPort=17910 , Preserved Duration = 262 , Total
Duration = 562 , H225 FD = -1 , H245 FD = -1
```

The previous example represents one preserved call. One such display is provided per preserved call. The **show h323 calls preserved** displays active calls only. No history is output.

To obtain additional information about a call, you can also use the **show call active voice** command. Calls can be cleared with the **clear call voice causecode** command.

How to Configure Call Preservation for H.323 VoIP Calls

The tasks for configuring Call Preservation for H.323 VoIP Calls include the following:

- Configure Cisco IOS Gateway, page 4
- Configure Cisco CallManager, page 9

Configure Cisco IOS Gateway

The **call preserve** command activates H.323 VoIP call preservation. RTP and RTCP inactivity detection and bidirectional silence detection can be used with this feature. Note that voice activity detection (VAD) must be set to off if you are using RTP and RTCP inactivity detection. VAD may be set to on, for bidirectional silence detection. For configuration examples, see the "RTP and RTCP Inactivity Detection Configuration Example" section on page 11 and "Bidirectional Silence Detection Enable Example" section on page 11.

When bidirectional silence and RTP and RTCP inactivity detection are configured, they are enabled for all calls by default. To enable them for H.323 VoIP preserved calls only, you must use the **call preserve** command's **limit-media-detection** keyword.

H.323 VoIP call preservation can be applied to all calls and to dial peers. The required steps are described in the following sections:

- Configure H.323 VoIP Call Preservation for All Calls, page 4
- Configure H.323 VoIP Call Preservation for a Dial Peer, page 6

Configure H.323 VoIP Call Preservation for All Calls

The following describes how to configure H.323 VoIP call preservation for all calls.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **h323**
5. **call preserve** [**limit-media-detection**]

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	**voice service voip** **Example:** `Router (config)# voice service voip`	Enters voice-service configuration mode.
Step 4	**h323** **Example:** `Router (config-voi-serv)# h323`	Enables the H.323 voice service configuration commands.
Step 5	**call preserve** [**limit-media-detection**] **Example:** `Router (config-voi-h323)# call preserve`	Enables the preservation of H.323 VoIP calls. • **limit-media-detection**—Limits RTP and RTCP inactivity detection and bidirectional silence detection (if configured) to H.323 VoIP preserved calls only.
Step 6	**exit** **Example:** `Router# exit`	Exits H.323 configuration mode.
Step 7	**exit** **Example:** `Router# exit`	Exist voice service voip configuration mode.

Examples

The following configuration example enables H.323 VoIP call preservation for all calls.

```
voice service voip
 h323
  call preserve
```

The following configuration example enables H.323 VoIP call preservation and limits RTP and RTCP inactivity detection and bidirectional silence detection (if configured) to preserved calls only:

```
voice service voip
 h323
  call preserve limit-media-detection
```

Configure H.323 VoIP Call Preservation for a Dial Peer

The following describes how to configure H.323 VoIP call preservation for a dial peer.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-class h323** *tag*
4. **call preserve** [**limit-media-detection**]
5. **exit**
6. **dial-peer voice** *tag* **voip**
7. **voice-class h323** *tag*
8. **exit**

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	**voice-class h323** *tag* **Example:** `Router (config)# voice-class h323 4`	Assigns an H.323 voice class to a VoIP dial peer. • tag—Unique number to identify the voice class. Range is from 1 to 10000.
Step 4	**call preserve** [**limit-media-detection**] **Example:** `Router (config-class)# call preserve`	Enables the preservation of H.323 VoIP calls. • **limit-media-detection**—Limits RTP and RTCP inactivity detection and bidirectional silence detection (if configured) to H.323 VoIP preserved calls only.
Step 5	**exit** **Example:** `Router (config)# exit`	Exits H.323 voice class configuration mode.
Step 6	**dial-peer voice** *tag* **voip** **Example:** `Router (config)# dial-peer voice 1 voip`	Defines a particular dial peer.

	Command or Action	Purpose
Step 7	**voice-class h323** *tag* **Example:** `Router (config-dial-peer)# voice-class h323 4`	Assigns an H.323 voice class to a VoIP dial peer. • tag—Unique number to identify the voice class. Range is from 1 to 10000.
Step 8	**exit** **Example:** `Router# exit`	Exits dial-peer voice configuration mode.

Examples

The following configuration example enables H.323 VoIP call preservation for dial peer 1.

```
voice-class h323 4
 call preserve
dial-peer voice 1 voip
 voice-class h323 4
```

Troubleshooting Tips

- Enable the **voice iec syslog** command in global configuration mode to display the reason that a call has disconnected after call preservation. The following is an example of the **voice iec syslog** command output line that display this information:

```
Nov 29 12:39:55.167: %VOICE_IEC-3-GW: H323: Internal Error (Socket error):
```

- Calls on hold are not preserved and a non-standard message with “callPreserveIE FALSE” is sent in the notify message. Use the **debug h225 asn** command for debug. The following is example output:

```
Router# debug h225 asn
H.225 ASN1 Messages debugging is on
3725-GW1#
*May 3 15:57:27.920: H225.0 INCOMING ENCODE BUFFER::=
28501900060008914A00040000D2D6D6D87EB11D02000000090D194410A00100110140B50000120A80A480
04000101000100
*May 3 15:57:27.920:
*May 3 15:57:27.920: H225.0 INCOMING PDU ::=
value H323_UserInformation ::=
{
h323-uu-pdu
{
h323-message-body notify :
{
protocolIdentifier { 0 0 8 2250 0 4 }
callIdentifier
{
guid '00D2D6D6D87EB11D02000000090D1944'H
}
}
h245Tunneling FALSE
nonStandardControl
{
{
nonStandardIdentifier h221NonStandard :
{
t35CountryCode 181
t35Extension 0
```

```
manufacturerCode 18
}
data '80A48004000101000100'H
}
}
}
}
*May 3 15:57:27.924: H225 NONSTD INCOMING ENCODE BUFFER::= 80A48004000101000100
*May 3 15:57:27.924:
*May 3 15:57:27.924: H225 NONSTD INCOMING PDU ::=
value H323_UU_NonStdInfo ::=
{
callMgrParam
{
interclusterVersion 1
enterpriseID {}
}
callPreserveParam
{
callPreserveIE FALSE
}
}
```

When the call is resumed, “callPreserve” is again set to True as shown in the following output example:

```
Router# debug h225 asn
*May 3 15:57:32.676: H225.0 INCOMING ENCODE BUFFER::=
28501900060008914A00040000D2D6D6D87EB11D02000000090D194410A001001B0140B50000121480A680
04000101000943004C0580323030300140
*May 3 15:57:32.676:
*May 3 15:57:32.676: H225.0 INCOMING PDU ::=
value H323_UserInformation ::=
{
h323-uu-pdu
{
h323-message-body notify :
{
protocolIdentifier { 0 0 8 2250 0 4 }
callIdentifier
{
guid '00D2D6D6D87EB11D02000000090D1944'H
}
}
h245Tunneling FALSE
nonStandardControl
{
{
nonStandardIdentifier h221NonStandard :
{
t35CountryCode 181
t35Extension 0
manufacturerCode 18
}
data '80A6800400010100094300 4C0580323030300140'H
}
}
}
}
*May 3 15:57:32.680: H225 NONSTD INCOMING ENCODE BUFFER::=
80A680040001010009430 04C0580323030300140
*May 3 15:57:32.680:
*May 3 15:57:32.680: H225 NONSTD INCOMING PDU ::=
value H323_UU_NonStdInfo ::=
```

```
{
callMgrParam
{
interclusterVersion 1
enterpriseID {}
}
callSignallingParam
{
connectedNumber '4C058032303030'H
}
callPreserveParam
{
callPreserveIE TRUE
}
}
```

- Use the **debug cch323 all** command after call setup to see if call is going into preserved state. Note that this command generates verbose output, and a console message is printed for every preserved call. In the following output, the relevant information appears in boldface:

```
Router# debug cch323 all
(CCH323-6-CALL_PRESERVED).
Nov 29 12:39:55.167: //-1/xxxxxxxxxxxx/H323/cch323_ct_main: SOCK 3 Event 0x1
Nov 29 12:39:55.167: //31/A9E0FB268017/H323/cch323_h225_handle_conn_loss:
cch323_h225_handle_conn_loss Call not torn down despite H.225.0 socket error: socket
error status = 1, ccb status = 403760899, fd = 3, pre-V3 = 0
Nov 29 12:39:55.167: %CCH323-6-CALL_PRESERVED: cch323_h225_handle_conn_loss: H.323
call preserved due to socket closure or error, Call Id = 4593, fd = 3
Nov 29 12:39:55.167: %VOICE_IEC-3-GW: H323: Internal Error (Socket error):
IEC=1.1.186.5.7.6 on callID 31 GUID=A9E0FB26600B11DA8017000653455072
Nov 29 12:39:55.167: //-1/xxxxxxxxxxxx/H323/h323_set_release_source_for_peer:
ownCallId[31], src[6]
Nov 29 12:39:55.167: //-1/xxxxxxxxxxxx/H323/h323_gw_clean_send_blocked_watch: fd 3
Nov 29 12:39:55.167: //-1/xxxxxxxxxxxx/H323/cch323_cleanup_xport: hashDestroy for
TcpFDTbl
```

- The following are additional debug commands can be used to troubleshoot the problems associated with H.323 VoIP call preservation:
  - **debug h225 asn1**
  - **debug h225 q931**
  - **debug h245 asn1**

What to Do Next

Configure Cisco CallManager.

Configure Cisco CallManager

This section describes how to configure Cisco CallManager for the call preservation of H.323 VoIP calls.

Procedure

Step 1 Choose **Service** > **Service Parameters**.

Step 2 From the Service menu select Cisco CallManager.

Step 3 Click **Advanced**.

Step 4 Scroll to the Clusterwide Parameter (Device — H.323) section.

Step 5 Set the "Allow Peer to Preserve H.323 Calls" parameter to True.

Step 6 At the top of the screen click **Update**.

Configuration Examples for Call Preservation for H.323 VoIP Calls

The configuration examples in this section include the following:

- H.323 VoIP Call Preservation for All Calls Example
- H.323 VoIP Call Preservation for a Dial Peer Example
- H.323 Call Preservation for RTP and RTCP and Silence Detection Example
- RTP and RTCP Inactivity Detection Configuration Example
- Bidirectional Silence Detection Enable Example

H.323 VoIP Call Preservation for All Calls Example

The following configuration example enables H.323 VoIP call preservation for all calls:

```
voice service voip
 h323
  call preserve
```

H.323 VoIP Call Preservation for a Dial Peer Example

The following configuration example enables H.323 VoIP call preservation for one dial peer:

```
voice class h323 4
 call preserve

dial-peer voice 1
 voice class h323 4
```

H.323 Call Preservation for RTP and RTCP and Silence Detection Example

The following configuration example enables H.323 VoIP call preservation and limits RTP and RTCP inactivity detection and bidirectional silence detection (if configured) to H.323 VoIP preserved calls only:

```
voice service voip
 h323
  call preserve limit-media-detection
```

RTP and RTCP Inactivity Detection Configuration Example

The following configuration example can be used to enable RTP and RTCP inactivity detection for dial peers. Note that for call preservation VAD must be set to off (**no vad** command):

```
dial-peer voice 10 voip
 no vad
gateway
 timer receive-rtcp 4
ip rtcp report-interval 60
```

Bidirectional Silence Detection Enable Example

The following configuration example enables bidirectional silence detection:

```
gateway
 timer media-inactive 5
ip rtcp report interval
```

Additional References

The following sections provide references related to Call Preservation for H.323 VoIP Calls.

Related Documents

Related Topic	Document Title
Cisco CallManager	Cisco CallManager Documentation
Cisco SRST	Cisco SRST Documentation
Syslog	Syslog Documentation

Standards

Standard	Title
No new or modified standards are supported by this feature, and support for existing standards has not been modified by this feature.	—

MIBs

MIB	MIBs Link
Cisco SRST SNMP 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://www.cisco.com/go/mibs

RFCs

RFC	Title
No new or modified RFCs are supported by this feature, and support for existing RFCs has not been modified by this feature.	—

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Description	Link
The Cisco Technical Support & Documentation 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 new commands only.

call preserve

To enable the preservation of H.323 VoIP calls, use the **call preserve** command in h323, voice-class h323, and voice service voip configuration modes. To reset to the default, use the **no** form of this command.

call preserve [**limit-media-detection**]

no call preserve [**limit-media-detection**]

Syntax Description

limit-media-detection	Limits RTP and RTCP inactivity detection and bidirectional silence detection (if configured) to H.323 VoIP preserved calls only.

Command Default

H.323 VoIP call preservation is disabled.

Command Modes

h323, voice-class h323, or voice service voip

Command History

Cisco IOS Release	Version	Modification
12.4(4)XC	Cisco SRST 4.0	This command was introduced.

Usage Guidelines

The **call preserve** command activates H.323 VoIP call preservation for following types of failures and connections:

Failure Types

- WAN failures that include WAN links flapping or degraded WAN links
- Cisco CallManager software failure, such as when the ccm.exe service crashes on a Cisco CallManager server.
- LAN connectivity failure, except when a failure occurs at the local branch

Connection Types

- Calls between two Cisco CallManager controlled endpoints
  - During Cisco CallManager reloads
  - When a Transmission Control Protocol (TCP) connection between one or both endpoints and Cisco CallManager used for signaling H.225.0 or H.245 messages is lost or flapping
  - Between endpoints that are registered to different Cisco CallManagers in a cluster and the TCP connection between the two Cisco CallManagers is lost
  - Between IP phones and the PSTN at the same site

- Calls between Cisco IOS gateway and an endpoint controlled by a softswitch where the signaling (H.225.0, H.245 or both) flows between the gateway and the softswitch and media flows between the gateway and the endpoint.
  - When the softswitch reloads.
  - When the H.225.0 or H.245 TCP connection between the gateway and the softswitch is lost, and the softswitch does not clear the call on the endpoint
  - When the H.225.0 or H.245 TCP connection between softswitch and the endpoint is lost, and the soft-switch does not clear the call on the gateway
- Call flows that involve a Cisco IP in IP (IPIP) gateway running in media flow-around mode that reload or lose connection with the rest of the network

When bidirectional silence and RTP and RTCP inactivity detection are configured, they are enabled for all calls by default. To enable them for H.323 VoIP preserved calls only, you must use the **call preserve** command's **limit-media-detection** keyword.

H.323 VoIP call preservation can be applied globally to all calls and to a dial peer.

Examples

The following example enables H.323 VoIP call preservation for all calls.

```
voice service voip
 h323
  call preserve
```

The following configuration example enables H.323 VoIP call preservation for dial peer 1.

```
voice-class h323 4
 call preserve
dial-peer voice 1 voip
 voice-class h323 4
```

The following example enables H.323 VoIP call preservation and enables RTP and RTCP inactivity detection and bidirectional silence detection for preserved calls only:

```
voice service voip
 h323
  call preserve limit-media-detection
```

The following example enables RTP and RTCP inactivity detection. Note that for H.323 VoIP call preservation VAD must be set to off (**no vad** command).

```
dial-peer voice 10 voip
 no vad
gateway
 timer receive-rtcp
ip rtcp report-interval
```

The following configuration example enables bidirectional silence detection:

```
gateway
 timer media-inactive
ip rtcp report interval
```

Related Commands

Command	Description
h323	Enables the H.323 voice service configuration commands.
show h323 calls preserved	Displays data about active H.323 VoIP preserved calls.
voice-class h323	Assigns an H.323 voice class to a VoIP dial peer.
voice service voip	Enters voice-service configuration mode

show h323 calls preserved

To display data about active H.323 VoIP preserved calls, use the **show h323 calls preserved** command in user EXEC or privileged EXEC mode.

show h323 calls preserved

Command Modes

User EXEC
Privileged EXEC

Command History

Cisco IOS Release	Version	Modification
12.4(4)XC	Cisco SRST 4.0	This command was introduced.

Usage Guidelines

The **show h323 calls preserved** command displays data per preserved call. Only active calls are displayed; preserved call history is not.

If translation rules are configured, the value displayed in the "Calling Number" field may have been translated by a gateway. Gateways handle called number values as the numbers to which calls are routed.

The "CallID" field displays the shorter form of the 16-octet, globally-unique connection ID that is allocated for each call leg. The show call active voice brief command also displays a shorter form of the CallID value (part of the third octet and the fourth octet). The longer form of the CallID value is output by the **show call active voice** command.

The CallID value can be used to refer to a call leg associated with the CallID when issuing other voice commands on the gateway, such as the **show voice call status** command and the **clear call voice** command.

An output value of -1 displayed in the "H225 FD" or "H245 FD" field denotes that the call was preserved due to an error detected on the H.225.0 connection. The actual H.225.0 socket file descriptor used for this call can be found from the syslog message that was output when this call was preserved.

To obtain more information about a call, you can also use the **show call active voice** command. Calls can be cleared with the **clear call voice causecode** command.

Examples

The following is sample output from the **show h323 calls preserved** command where one active call is preserved:

```
Router# show h323 calls preserved

CallID = 11EC , Calling Number = , Called Number = 3210000 ,
RemoteSignallingIPAddress=9.13.0.26 , RemoteSignallingPort=49760 ,
RemoteMediaIPAddress=9.13.0.11 , RemoteMediaPort=17910 , Preserved Duration = 262 , Total
Duration = 562 , H225 FD = -1 , H245 FD = -1
```

Table 1 provides an alphabetical listing of the fields displayed in the output of the **show h323 calls preserved** command and a description of each field.

Table 1 show h323 calls preserved Field Descriptions

Field	Description
Called Number	The phone number entered by the caller.
CallID	The shortened name for connection ID displayed in the **show call active voice brief** command.
H225 FD	The file descriptor number of the H.225.0 TCP socket.
H245 FD	The file descriptor number of the H.245 TCP socket.
Preserved Duration	The time in seconds that the call has been preserved.
RemoteMediaIPAddress	The remote media IP address.
RemoteMediaPort	The remote media IP address.
RemoteSignallingIPAddress	The remote signaling IP address.
RemoteSignallingPort	The remote signaling port.
Total Duration	The time in seconds of the phone call.

Related Commands

Command	Description
call preserve	Enables the preservation of H.323 VoIP calls.
clear call voice	Clears one or more voice calls detected as inactive because there is no RTP or RTCP activity.
show call active voice	Displays call information for voice calls in progress.
show voice call	Displays the call status for voice ports on the Cisco router.

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This product includes software developed by the OpenSSL Project for use in the OpenSSL Toolkit (http://www.openssl.org/).

This product includes cryptographic software written by Eric Young (eay@cryptsoft.com).

This product includes software written by Tim Hudson (tjh@cryptsoft.com).

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The implementation was written so as to conform with Netscapes SSL.

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