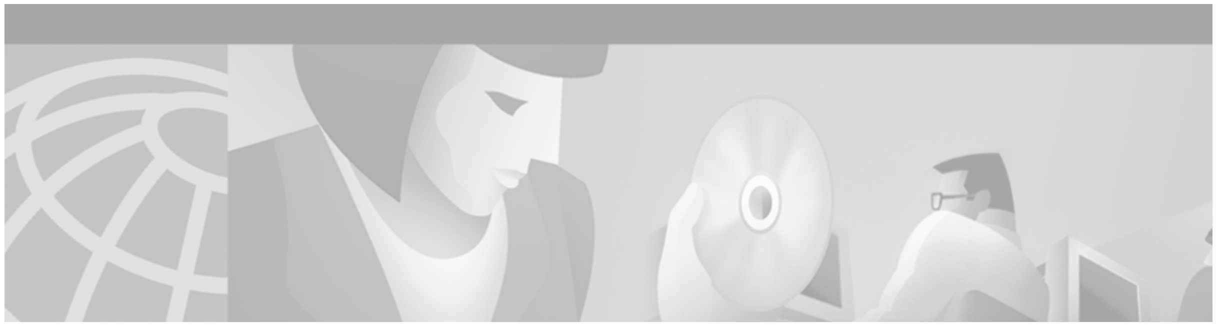

Release Notes for Cisco SIP IP Phone 7940/7960 Release 2.2

November 6, 2001

Contents

This document lists the known problems in the Cisco SIP IP Phone 7940/7960 Version 2.2 and contains information about the Cisco SIP IP Phone 7940/7960 (hereafter referred to as the Cisco SIP IP phone) that was not included in the Cisco SIP IP phone documentation.

This document includes the following sections:

New and Changed Information

For detailed information about each new feature and a list of all the Cisco SIP IP phone features, refer to the Version 2.2 of the *Cisco SIP IP Phone Administrator Guide* at the following URL:

http://www.cisco.com/univercd/cc/td/doc/product/voice/c_ipphon/sip7960/sipadm22/index.htm

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The following new features have been added to the Cisco SIP IP phone Version 2.2:

REFER Support

The Cisco SIP IP Phone uses the SIP REFER method to initiate a transfer. The REFER method allows for the initial parties to be reconnected upon a failed attended transfer attempt. REFER is used if the target endpoint supports the method; otherwise, the phone will failover to use the BYE/Also method of transfer in earlier releases.

There are no configuration changes associated with transfer. The REFER method is used as the default. The BYE/Also method of transfer is accepted also if received by the phone instead of a REFER.

7940 Support

The Cisco SIP IP Phone software recognizes the phone model (7960 or 7940) it is booting and provides the correct 6-line (7960) or 2-line (7940) support. The Cisco SIP IP Phone 7940 data sheet is located at the following URL:

http://www.cisco.com/warp/public/cc/pd/tlhw/prodlit/7940_ds.htm

OPTIONS Support

When an OPTIONS request is received, the Cisco SIP IP Phone responds with a list of phone methods and parameters supported. The Cisco SIP IP Phone does not generate an OPTIONS request.

Configurable VoIP Control Port

The listen port for SIP messages is configurable. When the NAT enable flag is used in conjunction with the VoIP Control Port, the packets are sourced from this port rather than from an ampherol port. See the voip_control_port parameter descriptions in the section, “Modifying the Default SIP Configuration File” in Chapter 3, “Managing Cisco SIP IP Phones,” at the following URL:

http://www.cisco.com/univercd/cc/td/doc/product/voice/c_ipphon/sip7960/sipadm22/index.htm

Configurable RTP Media Ports

The Cisco SIP IP Phone allows for the start and end RTP media ports to be configured. See the start_media_port and end_media_port descriptions in the section, “Modifying the Default SIP Configuration File” in Chapter 3, “Managing Cisco SIP IP Phones,” at the following URL:

http://www.cisco.com/univercd/cc/td/doc/product/voice/c_ipphon/sip7960/sipadm22/index.htm

NAT Support

When network address translation (NAT) is enabled, the Cisco SIP IP Phone provides support for SIP messages to traverse NAT/Firewall networks. The Contact and Via headers are modified to reflect the NAT parameters. The Cisco SIP IP Phone can also enable NAT received processing. See the nat_enable, nat_address, and nat_received_processing parameters in the section, “Modifying the Default SIP Configuration File” in Chapter 3, “Managing Cisco SIP IP Phones,” at the following URL:

http://www.cisco.com/univercd/cc/td/doc/product/voice/c_ipphon/sip7960/sipadm22/index.htm

Outbound Proxy Support

When an Outbound Proxy is configured, all SIP requests are sent to the Outbound Proxy Server instead of the configured Proxy Address. The Cisco SIP IP Phone does not have to register with the Outbound Proxy. All interactions, such as authentication, with the Outbound Proxy are treated the same as the interactions with the primary proxy. See the outbound_proxy and outbound_proxy_port descriptions in the section, "Modifying the Default SIP Configuration File" in Chapter 3, "Managing Cisco SIP IP Phones," at the following URL:

http://www.cisco.com/univercd/cc/td/doc/product/voice/c_ipphon/sip7960/sipadm22/index.htm

Caveats

This section describes caveats, or known problems, that are open in this release. It also lists closed caveats in this release.

Open Caveats - Release 2.2

- **CSCds27516—**When the network media is manually configured, the inline power does not work when connected to a Catalyst 3500 switch.

  Problem Description: When the network media is manually configured, the inline power support does not work when the Cisco SIP IP phone is connected to a Catalyst 3500 switch.

  Recommended Action: When connecting to a Catalyst 3500 switch, configure the phone to automatically negotiate the network media type by selecting Auto for the Network Media Type parameter located in the Network Configuration menu.

- **CSCds35841:** When in overview mode, the Cisco SIP IP phone soft keys do not work.

  Problem Description: Pressing a line button during a call displays the overview screen on which there is located a Redial and NewCall soft key. However, these soft keys are ignored by the phone if pressed.

  Recommended Action: Return to the call screen (wait 8 seconds for the call screen to reappear or press the line button again).

- **CSCds64602:** Caller cannot terminate a call transferred back by the Callee.

  Problem Description: The Cisco SIP phone does not properly handle the following call scenario:

  - Phone A calls phone B.
  - Phone B performs a call transfer with consultation back to phone A.
  - Phone B's call hangs up correctly, however, the phone A's call has no audio and requires several on and off hooks to terminate the call.

  Recommended Action: Press the speaker button or go off hook and back on hook several times to terminate the call.

- **CSCdt89255:** 7960 SDP Codec negotiation issue causes one-way voice.

  Problem Description: The 7960 SIP IP Phone SDP codec negotiation can cause one-way voice when other endpoint is using asynchronous codec support.

  Recommended Action: Ensure other clients support single codec. The 7960 SIP IP Phone does not support asynchronous codec.

- **CSCdu35450:** proxyN_port of UNPROVISIONED does not default to 5060.

  Problem Description: The proxyN_port parameter in the SIP 7960 IP Phone does not default to 5060. It defaults to 0 instead.

  Recommended Action: Set the proxyN_port parameter to 5060, rather than "" or "UNPROVISIONED".

- **CSCdu43127:** DSP Timeouts with multiple instances of DTMF and speakerphone.

  Problem Description: On rare occasions when using speakerphone, G729a codec, VAD enabled, and DTMF, the phone encounters a DSP timeout which affects only the active call. Any subsequent calls work unless the above four factors are used again. The DSP timeout occurs only when all four of the above factors happen simultaneously, and affects only the active call.

  This issue is also a side effect of excessive debug use.

  Recommended Actions:

  - Use the Handset instead of the speakerphone, or
  - Use the default codec G711ulaw, or
  - Use the default of VAD as disabled.

- **CSCdu43128:** Invalid SRV in maddr of Contact causes hung midcall INVITE.

  Problem Description: After a basic call is established and a mid-call INVITE is sent to the phone, with an invalid SRV entry in the Contact: header, the phone leaves the call hung. The SIP messaging is correct in that the mid-call INVITE gets rejected with a 500 Internal Server error, but the display still shows a connected call.

  Recommended Action: Press the EndCall softkey or hang up the phone.

- **CSCdu68098:** Requested-By header in BYE message missing for Transfer.

  Problem Description: When using the BYE/Also method of transfer the BYE message does not include the Requested-By header.

  Recommended Action: There is no workaround. However, the primary method for Transfer is now REFER so the BYE/Also method is used only when an endpoint cannot support REFER.

- **CSCdu68091**: No support for configurable action tag in REGISTER Contact.

  Problem Description: When the phone sends a REGISTER message it does not attach an action= tag to the Contact header. This can lead to mismatched registrations when another client REGISTERs with the same user ID because the 7960/7940 is always treated as action=none.

  Recommended Action: Configure the other client to have action=none to avoid mismatched registrations.

- **CSCdv33556:** INVITEs with multiple valid m= lines get rejected with 400 Bad Request.

  Problem Description: When the phone receives an INVITE with multiple valid m= SDP lines (for example, m=application and m=audio), a 400 BAD Request is sent in response.

  Recommended Action: None.

- **CSCdv26487**: CANCEL needs to handle Proxy challenges similar to BYE.

  Problem Description: There is no support for a CANCEL with credentials. If a CANCEL gets challenged by the Proxy, a new CANCEL with Proxy-Authorization credentials will not be sent.

  Recommended Action: None; there is no workaround.

Resolved Caveats - Release 2.2

- **CSCdu49096**—Support for initial Hold INVITE for 3pcc
- **CSCdu58538**—Mid-Call INVITE with no SDP does not use ACK SDP for media
- **CSCdu63337**—callerid_blocking does not work for option 2
- **CSCdv06288**—Phone shows in 486 with dead air/silence, no busy tone
- **CSCdv03772**—Route Header not formed properly when Contact or RR has transport=tls
- **CSCdv00847**—Phone generates duplicate Call-ID over a period of days
- **CSCdv16236**—7960 fails to DHCP because DHCPOFFER gets rejected
- **CSCdu80654**—Challenged re-INVITEs fail due to failed tag check
- **CSCdv09848**—MWI does not work according to latest spec
- **CSCdv05085**—Via Header parsing should be case-insensitive
- **CSCdv43533**—Via header without a port is causing responses sent to 0.0.0.0
- **CSCdv47780**—ACK with Held SDP causes 3pcc interop problems
- **CSCdu47899**—Phone sends a fixed digit duration
- **CSCdu40212**—Line name with a leading + causes all calls to fail with 404
- **CSCdt11124**—MWI Lamp not illuminated for NOTIFY's on lines 2-6
- **CSCdu58632**—Parsed Port from SRV entry not being used
- **CSCdu68083**—Remove Anonymous Call block when using Emergency Route
- **CSCdu01841**—Need support for Diversion Header
- **CSCdv51518**—Via header mapping is wrong for 487 Request Terminated
- **CSCdv51568**—Phone ignores the other-params in Record-Route
- **CSCdu59730**—Support for md5-sess authentication
- **CSCdu79396**—Support for auth-int authentication

Related Documentation

- *Cisco SIP IP Phone Administrator Guide, Version 2.2*
- *Cisco SIP IP Phone 7940/7960 User Guide*
- *Getting Started with the Cisco IP Phone 7960/7940*
- *Cisco IP Phone 7940/7960 Quick Reference Card*—Pocket-sized reference for common phone tasks. This document ships with the phone.

Obtaining Documentation

The following sections explain how to obtain documentation from Cisco Systems.

World Wide Web

You can access the most current Cisco documentation on the World Wide Web at the following URL:

http://www.cisco.com

Translated documentation is available at the following URL:

http://www.cisco.com/public/countries_languages.shtml

Documentation CD-ROM

Cisco documentation and additional literature are available in a Cisco Documentation CD-ROM package, which is shipped with your product. The Documentation CD-ROM is updated monthly and may be more current than printed documentation. The CD-ROM package is available as a single unit or through an annual subscription.

Ordering Documentation

Cisco documentation is available in the following ways:

- Registered Cisco Direct Customers can order Cisco product documentation from the Networking Products MarketPlace:

  http://www.cisco.com/cgi-bin/order/order_root.pl

- Registered Cisco.com users can order the Documentation CD-ROM through the online Subscription Store:

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

- Nonregistered Cisco.com users can order documentation through a local account representative by calling Cisco corporate headquarters (California, USA) at 408 526-7208 or, elsewhere in North America, by calling 800 553-NETS (6387).

Documentation Feedback

If you are reading Cisco product documentation on Cisco.com, you can submit technical comments electronically. Click **Leave Feedback** at the bottom of the Cisco Documentation home page. After you complete the form, print it out and fax it to Cisco at 408 527-0730.

You can e-mail your comments to bug-doc@cisco.com.

To submit your comments by mail, use the response card behind the front cover of your document, or write to the following address:

Cisco Systems
Attn: Document Resource Connection
170 West Tasman Drive
San Jose, CA 95134-9883

We appreciate your comments.

Obtaining Technical Assistance

Cisco provides Cisco.com as a starting point for all technical assistance. Customers and partners can obtain documentation, troubleshooting tips, and sample configurations from online tools by using the Cisco Technical Assistance Center (TAC) Web Site. Cisco.com registered users have complete access to the technical support resources on the Cisco TAC Web Site.

Cisco.com

Cisco.com is the foundation of a suite of interactive, networked services that provides immediate, open access to Cisco information, networking solutions, services, programs, and resources at any time, from anywhere in the world.

Cisco.com is a highly integrated Internet application and a powerful, easy-to-use tool that provides a broad range of features and services to help you to

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- Resolve technical issues with online support
- Download and test software packages
- Order Cisco learning materials and merchandise
- Register for online skill assessment, training, and certification programs

You can self-register on Cisco.com to obtain customized information and service. To access Cisco.com, go to the following URL:

http://www.cisco.com

Technical Assistance Center

The Cisco TAC is available to all customers who need technical assistance with a Cisco product, technology, or solution. Two types of support are available through the Cisco TAC: the Cisco TAC Web Site and the Cisco TAC Escalation Center.

Inquiries to Cisco TAC are categorized according to the urgency of the issue:

- Priority level 4 (P4)—You need information or assistance concerning Cisco product capabilities, product installation, or basic product configuration.
- Priority level 3 (P3)—Your network performance is degraded. Network functionality is noticeably impaired, but most business operations continue.

- Priority level 2 (P2)—Your production network is severely degraded, affecting significant aspects of business operations. No workaround is available.
- Priority level 1 (P1)—Your production network is down, and a critical impact to business operations will occur if service is not restored quickly. No workaround is available.

Which Cisco TAC resource you choose is based on the priority of the problem and the conditions of service contracts, when applicable.

Cisco TAC Web Site

The Cisco TAC Web Site allows you to resolve P3 and P4 issues yourself, saving both cost and time. The site provides around-the-clock access to online tools, knowledge bases, and software. To access the Cisco TAC Web Site, go to the following URL:

http://www.cisco.com/tac

All customers, partners, and resellers who have a valid Cisco services contract have complete access to the technical support resources on the Cisco TAC Web Site. The Cisco TAC Web Site requires a Cisco.com login ID and password. If you have a valid service contract but do not have a login ID or password, go to the following URL to register:

http://www.cisco.com/register/

If you cannot resolve your technical issues by using the Cisco TAC Web Site, and you are a Cisco.com registered user, you can open a case online by using the TAC Case Open tool at the following URL:

http://www.cisco.com/tac/caseopen

If you have Internet access, it is recommended that you open P3 and P4 cases through the Cisco TAC Web Site.

Cisco TAC Escalation Center

The Cisco TAC Escalation Center addresses issues that are classified as priority level 1 or priority level 2; these classifications are assigned when severe network degradation significantly impacts business operations. When you contact the TAC Escalation Center with a P1 or P2 problem, a Cisco TAC engineer will automatically open a case.

To obtain a directory of toll-free Cisco TAC telephone numbers for your country, go to the following URL:

http://www.cisco.com/warp/public/687/Directory/DirTAC.shtml

Before calling, please check with your network operations center to determine the level of Cisco support services to which your company is entitled; for example, SMARTnet, SMARTnet Onsite, or Network Supported Accounts (NSA). In addition, please have available your service agreement number and your product serial number.

This document is to be used in conjunction with the documents listed in the “Related Documentation” section.