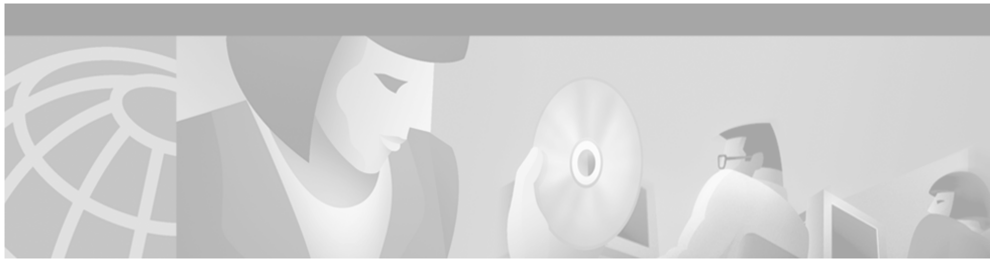


Cisco SIP IP Phone 7960 Version 1.0 Release Note

August 7, 2000

This document lists the known problems in the Cisco SIP IP phone 7960 Version 1.0 release and contains information about the Cisco SIP IP phone 7960 (hereafter referred to as the Cisco SIP IP phone) that was not included in the *Cisco SIP IP Phone 7960 Administrator Guide*.

Sections in this document include the following:

Corporate Headquarters: Cisco Systems, Inc., 170 West Tasman Drive, San Jose, CA 95134-1706 USA

Ordering the Cisco SIP IP Phone 7960 Administrator Guide

The Cisco SIP IP phone 7960 firmware is available via CCO only. Therefore, to obtain a printed copy of the *Cisco SIP IP Phone 7960 Administrator Guide*, you must either download the PDF file of the manual from CCO or order a printed and bound copy of the manual through Cisco MarketPlace.

To obtain a PDF file of the manual which you can download and print, go to: http://www.cisco.com/univercd/cc/td/doc/product/voice/c_ipphon/sip7960/admin.pdf

To order a printed and bound copy of the manual, go to Cisco MarketPlace and specify 78-10497-01.

Note To place an order for the manual through Cisco MarketPlace, you must be a registered CCO user.

Related Documentation

In addition to this release note, use the following publications to learn how to install and use the Cisco SIP IP phone:

- *Cisco SIP IP Phone 7960 Administration Guide*—Provides information for network and telephone administrators for understanding, installing, and configuring the Cisco SIP IP phone. This guide is available online at: http://www.cisco.com/univercd/cc/td/doc/product/voice/c_ipphon/sip7960/admin.pdf
- *Cisco IP Phone 7960 Getting Started Guide*—Describes how to use the phone. This guide ships with the phone and is available online at: http://www.cisco.com/univercd/cc/td/doc/product/voice/c_ipphon/ip_7960/getstart/index.htm
- *Cisco IP Phone 7960 Quick Reference Card*—Pocket-sized reference for common phone tasks. This document ships with the phone.

Known Problems in this Release

This section lists the currently known problems in the Cisco SIP IP phone 7960 Version 1.0 release.

Problem: Cisco SIP IP phone cannot parse absolute time in Expire headers (CSCdr09775)

Problem Description: The Cisco SIP IP phone cannot determine the current time on its own. Therefore, although the phone can parse Expire headers if they are based on relative time (for example, the time elapsed since a message was sent), it cannot parse Expire headers if they are based on absolute time.

Recommended Action: None.

Problem: The Cisco SIP IP phone uses half-duplex on a switch port that has been configured as full-duplex (CSCdr22751)

Problem Description: When a Cisco SIP IP phone is connected to a switch port that has been hard-coded to full duplex, the Cisco SIP IP phone displays half-duplex and the quality of the call is compromised. However, when the Cisco SIP IP phone is connected to a switch port that is configured to autosense mode, the Cisco SIP IP phone connects as 100 Mbps, full-duplex and the quality of the call is good.

Recommended Action: When connecting a Cisco SIP IP phone to a switch port, ensure that the port has been configured to autosense mode.

Problem: With a forked request, the Cisco SIP IP phone does not handle additional SIP 200 OK responses correctly (CSCdr56133)

Problem Description: When a forked request occurs, the Cisco SIP IP phone sends an ACK request to the 200 OK response of the first leg, but ignores any messages sent from the second leg. This problem causes the second call leg to be left in a suspended state until the timeout occurs.

Recommended Action: None.

Problem: The Cisco SIP IP phone cannot parse messages containing an o= SDP field greater than 2147483647 (CSCdr91126)

Problem Description: The Cisco SIP IP phone cannot parse and sends an "invalid SDP" when it receives messages containing a version parameter in the o= SDP field greater than 2147483647.

Recommended Action: When implementing SIP, do not use a version parameter in the o= SDP field greater than 2147483647.

Problem: SIP REGISTER messages do not have support for SIP 3xx Redirection responses (CSCdr93802)

Problem Description: The ability to support a 3xx Redirection response for registrations is missing. If a proxy server sends a 3xx Redirection response to a registration request, registration will fail and the phone user might not be able to place calls.

Recommended Action: None.

Problem: Daisy-chaining IP phones can cause problems

Problem Description: If you connect IP phones together in a line (daisy-chaining), a problem with one phone can affect all subsequent phones in the line. Also, the bandwidth is shared among all phones on the line.

Recommended Action: Do not connect an IP phone to another IP phone through the access port. Each IP phone should be directly connected to a switch port.

Problem: Workstation NICs cannot receive power through the network connection

Problem Description: If you are powering your phone through the network connection, you must be careful if you decide to unplug the phone's network connection and plug the cable into a workstation. Workstation NICs cannot receive power through the network connection; if power comes through the connection, the NIC can be destroyed.

Recommended Action: To prevent this, wait 10 seconds or longer after unplugging the cable from the phone. This gives the switch enough time to recognize that there is no longer a phone on the line, and to stop providing power to the cable.

Problem: LCD display issues occurring with certain types of building lighting

Problem Description: You might see Beat frequencies (scan lines) in the LCD if you are using certain types of old florescent lights in your building.

Recommended Action: Moving the phone from the lights, or replacing the lights, should resolve the problem.

Problem: Powering the IP phone from multiple sources causes different results

Problem Description: An IP phone can be powered from a wall socket, from a switch port, or from a power-source between the phone and the switch. At any given time, the phone receives power from only one source: the others are used as backup.

The phone and switch automatically determine which power source the phone uses. The phone user will experience different results based on which power source is being used by the phone if the power has to be switched to a different source:

1. If you plug a phone into a wall power socket before plugging it into the network, the phone is powered by the power cord.

2. If you then unplug the power cord, the phone resets. If the switch port is configured for 10/100MB, the switch recognizes the loss of power and brings the phone back up.

3. If the switch port is configured for 10MB only, then you must unplug the network connection, and replug it into the phone, in order for the switch to recognize the phone's loss of power.

Recommended Action: If you plugged the network connection into the phone before you plugged in the power cord, the phone receives power through the switch, and unplugging the power cord will not bring down the phone. Note that if the switch reboots, the phone will then be powered by the power cord, and unplugging the power cord results in a reset.

Admendments to the Documentation

This section contains information that has been amended or was not included in the *Cisco SIP IP Phone 7960 Administrator Guide*, the *Cisco IP Phone 7960 Getting Started Guide*, or the *Cisco IP Phone 7960 Quick Reference Card*. When applicable, the headings in this section correspond with the section titles in the documentation.

This section contains the following new or amended information:

- Supported Features, page 7
- Connecting to Power, page 7
- SIP Responses, page 8

- SIP Call Flows, page 9
- Reading the Cisco SIP IP Phone Icons, page 14

Supported Features

In addition to the features listed in the "Supported Features" section of the "Product Overview" chapter of the Cisco *SIP IP Phone 7960 Administrator Guide*, the Cisco SIP IP phone also supports network call forwarding.

Network call forwarding allows the Cisco SIP IP phone user to request forwarding service from the network (via a third party tool that enables this feature to be configured). When the call is placed to the user's phone, it is redirected to the appropriate forward destination by the SIP proxy.

Connecting to Power

In addition to the inline power sources listed in the "Connecting to Power" section of the "Product Overview" chapter of the *Cisco SIP IP Phone 7960 Administrator Guide*, the Cisco SIP IP phone can also obtain inline power from the following sources:

- 48-port 10/100 Ethernet with inline power module for the Catalyst 4006 (WS-X4148-RJ45V)
- VoIP DC Power Entry module for the Catalyst 4006 (WS-X4095-PEM)
- External -48V DC power shelf common equipment for the Catalyst 4006 with two AC-to-DC PSU's and one empty bay for redundant option (WS-X4608-2PSU) and the 110V 15A AC-to-48V DC PSU redundant option for the power shelf (WS-X4608)
- Catalyst 3524-PWR XL switch (WS-C3524-PWR-XL-EN)

SIP Responses

The compliance information for the following SIP responses has been changed from that documented in the *Cisco SIP IP Phone 7960 Administrator Guide*. The changed compliance information is as follows:

Response	Supported?	Comments
100 Trying	Yes	The Cisco SIP IP phone generates this response for an incoming INVITE. Upon receiving this response, the phone waits for a 180 Ringing, 183 Session progress, or 200 OK response.
183 Session progress	See comments	The SIP IP phone does not generate this message. Upon receiving this response, the phone provides early media cut through and then waits for a 200 OK response.
302 Moved temporarily		The Cisco SIP IP phone does not generate this response at this time. Upon receiving this response, the phone sends an INVITE containing the contact information received in the 302 Moved temporarily response.
404 Not found	Yes	The Cisco SIP IP phone generates this response if it is unable to locate the callee. Upon receiving this response, the phone notifies the user.
486 Busy here	Yes	The Cisco SIP IP phone generates this response if the called party is off hook and the call cannot be presented as a call waiting call. Upon receiving this response, the phone notifies the user and generates a busy tone.

SIP Call Flows

The documentation for the connection information (c=) SDP field in the INVITE messages of the following call flows in the *Cisco SIP IP Phone 7960 Administrator Guide* has been changed:

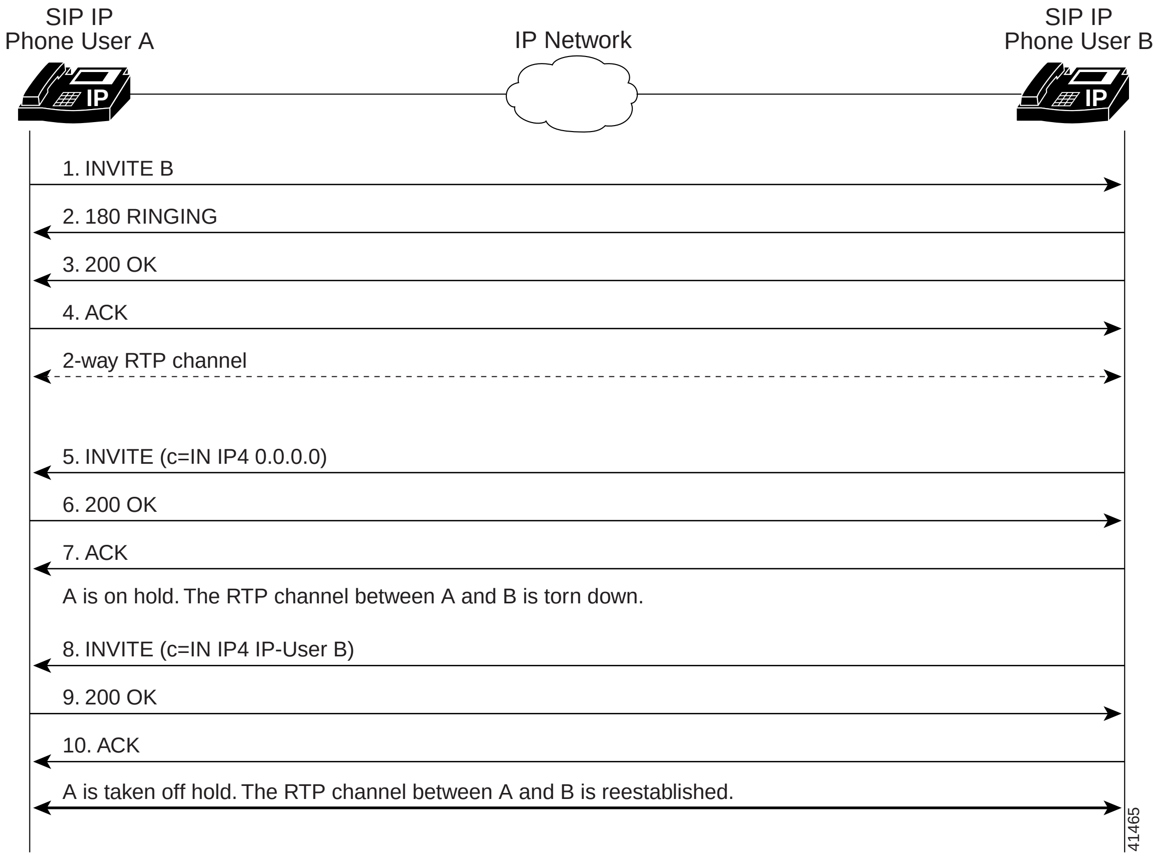
- Cisco SIP IP Phone-to-Cisco SIP IP Phone Simple Call Hold, page 9
- Cisco SIP IP Phone-to-Cisco SIP IP Phone Call Hold with Consultation, page 10
- Cisco SIP IP Phone-to-Cisco SIP IP Phone Call Waiting, page 12

Cisco SIP IP Phone-to-Cisco SIP IP Phone Simple Call Hold

As illustrated in Figure 1, to place the call on hold, IN IP4 0.0.0.0 is inserted into the c= SDP field of the INVITE message in step 5.

To reestablish the call between user A and user B, the IP address of User B is inserted into the c= SDP field of the INVITE message in Step 8. This reestablishes the 2-way RTP voice path between user A and user B.

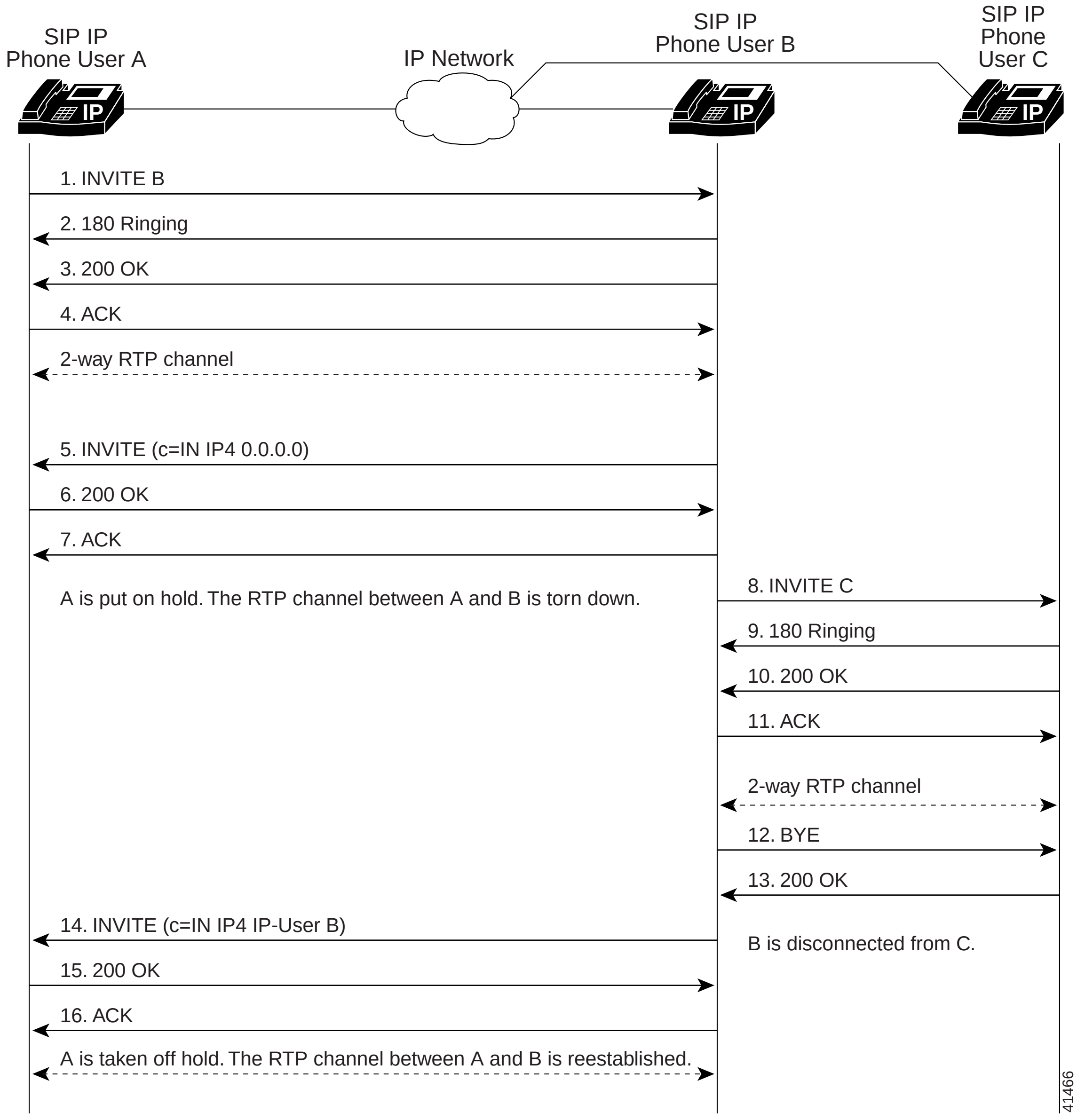
Figure 1 Cisco SIP IP Phone-to-Cisco SIP IP Phone Simple Call Hold



Cisco SIP IP Phone-to-Cisco SIP IP Phone Call Hold with Consultation

As illustrated in Figure 2, to reestablish the call between user A and user B, the IP address of User B is inserted into the c= SDP field of the INVITE message in Step 14. This reestablishes the 2-way RTP voice path between user A and user B.

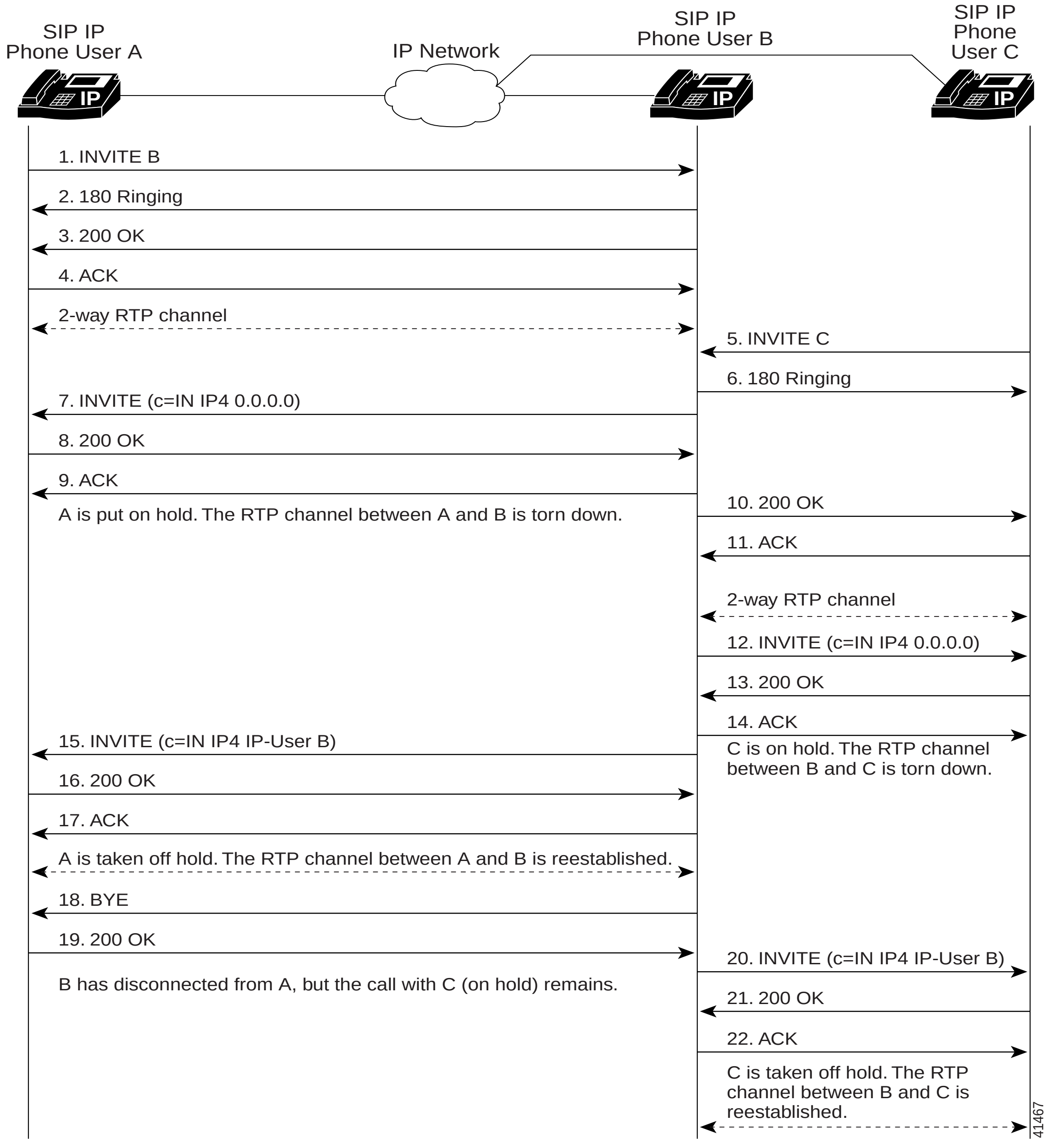
Figure 2 Cisco SIP IP Phone-to-Cisco SIP IP Phone Call Hold with Consultation



Cisco SIP IP Phone-to-Cisco SIP IP Phone Call Waiting

As illustrated in Figure 3, to reestablish the call between user A and user B, the IP address of User B is inserted into the c= SDP field of the INVITE message in Step 15. This reestablishes the 2-way RTP voice path between user A and user B.

Figure 3 Cisco SIP IP Phone-to-Cisco SIP IP Phone Call Waiting



Reading the Cisco SIP IP Phone Icons

When using the Cisco SIP IP phone, a variety of icons can display on the phone's LCD. Table 1 lists and describes each icon that you might see while using the Cisco SIP IP phone.

Table 1 Cisco SIP IP Phone User Interface Icon Meanings

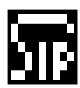
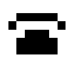
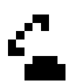
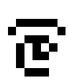
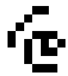
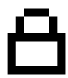
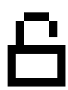
Icon	Meaning
	The Cisco IP phone 7960 that you are using is running SIP.
	The line is configured for E.164 number dialing and you can enter only numbers when placing the call. The character "x" displayed to the right of the icon indicates that registration has failed.
	The line is configured for E.164 number dialing and ready for you to place the call. When a line is configured for E.164 number dialing, you can enter only numbers when placing the call. You can change to URL dialing at any time while dialing on a line by pressing the **more** soft key and then the **URL** soft key. The character "x" displayed to the right of the icon indicates that registration has failed.
	The line is configured for URL dialing and you can enter both numbers and letters when placing the call. The character "x" displayed to the right of the icon indicates that registration has failed.

Table 1 Cisco SIP IP Phone User Interface Icon Meanings (continued)

Icon	Meaning
	The line is configured for URL dialing and ready for you to place the call. When a line is configured for URL dialing, you can enter both numbers and letters when placing the call. You can change to E.164 number dialing at any time while dialing on a line by pressing the **more** soft key and then the **Number** soft key. The character “x” displayed to the right of the icon indicates that registration has failed.
	The Cisco SIP IP phone configuration mode is locked. When the phone is locked, the phone’s network or SIP settings cannot be modified.
	The Cisco SIP IP phone configure mode is unlocked. When the phone is unlocked, the phone’s network or SIP settings can be modified.

Documented Features that are not Supported in Version 1.0

The following features, while noted as SIP features in the *Cisco IP Phone 7960 Getting Started Guide* and the *Cisco IP Phone 7960 Quick Reference Card* that shipped with the phone, are not supported in the Version 1.0 of the Cisco SIP IP phone:

- Online help
- Forwarding calls (local)
- Making calls from a directory
- Placing a conference call
- Using voice mail.

Obtaining Documentation

World Wide Web

You can access the most current Cisco documentation on the World Wide Web at http://www.cisco.com, http://www-china.cisco.com, or http://www-europe.cisco.com.

Documentation CD-ROM

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

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Cisco provides Cisco Connection Online (CCO) as a starting point for all technical assistance. Warranty or maintenance contract customers can use the Technical Assistance Center. All customers can submit technical feedback on Cisco documentation using the web, e-mail, a self-addressed stamped response card included in many printed docs, or by sending mail to Cisco.

Cisco Connection Online

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You can access CCO in the following ways:

- WWW: www.cisco.com
- Telnet: cco.cisco.com
- Modem using standard connection rates and the following terminal settings: VT100 emulation; 8 data bits; no parity; and 1 stop bit.
  - From North America, call 408 526-8070
  - From Europe, call 33 1 64 46 40 82

You can e-mail questions about using CCO to cco-team@cisco.com.

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We appreciate and value your comments.

This document is to be used in conjunction with the *Cisco SIP IP Phone 7960 Administrator Guide* publication.