Cisco SIP IP Phone 7960 Version 2.1 Release Note

June, 2001

This document lists the known problems in the Cisco SIP IP Phone 7960 Version 2.1 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 documentation.

Sections in this document include the following:

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Ordering the Cisco SIP IP Phone 7960 Administrator Guide

The Cisco SIP IP phone 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 *Cisco SIP IP Phone 7960 Administrator Guide, Version 2.1*, which you can download and print, go to: http://www.cisco.com/univercd/cc/td/doc/product/voice/c_ipphon/sip7960/admin2.pdf

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/admin2.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 Version 2.1.

Problem: Cisco SIP IP phone crashes with an async break or PC reboot (CSCds20969)

Problem Description: The Cisco SIP IP phone might lock up if a break is sent to the phone via the RS-232 port. The phone might also lock up if the PC to which the RS-232 port is attached is rebooted.

Recommended Action: Unplug and replug the power cord on the phone.

Problem: When the network media is manually configured, the inline power does not work when connected to a Catalyst 3500 switch (CSCds27516)

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.

Problem: When in overview mode, the Cisco SIP IP phone soft keys do not work (CSCds35841)

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).

Problem: Caller cannot terminate a call transferred back by the Callee (CSCds64602)

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.

Problem: Message Waiting Indication (MWI) for Cisco uOne Messaging System with the Cisco SIP IP phone does not work after a phone reboot (CSCdt11124))

Problem Description: The MWI on the Cisco SIP IP phone is not retained following a phone power cycle or reboot because the Unsolicited Notify is a single event that occurs at the time a message is deposited or retrieved.

Recommended Action: None. If a system upgrade or other action that requires a phone reboot is to be performed, notify the end users of this condition.

Problem: 7960 SDP Codec negotiation issue causes one-way voice (CSCdt89255)

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.

Problem: proxyN_port of UNPROVISIONED does not default to 5060 (CSCdu35450)

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".

Problem: DSP Timeouts with multiple instances of DTMF and speakerphone (CSCdu43127)

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

Problem: Invalid SRV in maddr of Contact causes hung midcall INVITE (CSCdu43128)

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 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.

Problem: Line name with a leading "+" causes all calls to fail with a 404 (CSCdu40212)

Problem Description: When the phone has a lineN_name with a leading +, such as +1234567, all calls with or without the + are rejected with a "404 Not Found".

Recommended Action: Set the lineN_name so that it does not include the leading +, i.e. 1234567

Problem: An Initial INVITE with c=0.0.0.0 gets rejected with 500 (CSCdu49096)

Problem Description: The phone rejects initial INVITEs that contain a 'hold' SDP with a 500 Internal Server Error. This scenario would be used for 3rd party Call Control.

Recommended Action: None.

New Features in This Release

For detailed information about each new feature and a list of all the Cisco SIP IP phone features, refer to the Version 2.1 of the *Cisco SIP IP Phone 7960 Administrator Guide*.

The following new features have been added to the Cisco SIP IP phone Version 2.1:

- User-defined proxy routing

  The “Route” attribute of the Template tag in the dial plan template file can be used to indicate which proxy (default, emergency, FQDN) that the call should be initially routed to. For example, to configure an emergency proxy, specify value of the “Route” attribute as “emergency”.

- Backup SIP proxy

  When the primary proxy does not respond to the INVITE message sent by the Cisco SIP IP Phone after the configured number of retries, the Cisco SIP IP Phone sends the INVITE to the backup proxy. This is independent from which proxy is defined in the “Route” attribute in the dial plan template used.

  The Cisco SIP IP Phone does not have to register with the backup proxy. All interactions, such as authentication challenges, with the backup proxy is treated the same as the interactions with the primary proxy.

  The backup proxy is only used with new INVITE messages. Once the backup proxy is used, it is active for the duration of the call.

  The location of the backup SIP proxy can be defined as an IP address in the default configuration file. See proxy_backup and proxy_backup_port parameters in Modifying the Default SIP Configuration File in Chapter 3, “Managing Cisco SIP IP Phones”.

- Emergency SIP proxy

  An optional emergency SIP proxy can be configured with the “Route” attribute of the Template tag in the Dial Plan template file. See “Support of user-defined proxy routing”.

  When an emergency SIP proxy is configured and a call is initiated, the phone generates an INVITE message to the address specified in the proxy_emergency parameter. The emergency proxy is used for the call duration.

The location of the emergency proxy can be defined as an IP address in the default configuration file. See proxy_emergency and proxy_emergency_port parameters in Modifying the Default SIP Configuration File in Chapter 3, “Managing Cisco SIP IP Phones”.

- Support of DNS SRV

  DNS SRV is the Domain Name Server RR used to locate servers for a given service.

  SIP on Cisco’s SIP IP Phones use DNS SRV query to determine the IP address of the SIP Proxy or the Redirect Server. The query string generated is in compliance with RFC2782, and prepends the protocol label with an underscore “_”; as in “_protocol._transport.”. The addition of the underscore reduces the risk of the same name being used for unrelated purposes.

  Also in compliance with RFC 2782 and the draft-ietf-sip-srv-01 spec. is that the system can remember multiple IP addresses and use them properly. In the draft-ietf-sip-srv-01 spec, it is assumed that all proxies returned for the SRV record are equivalent such that the phone can register with any of the proxies and initiate a call using any other proxy.

- Configurable VAD

  VAD can be enabled or disabled with enable_vad parameter. Value 0 for disable, and value 1 for enable. See enable_vad parameter in Modifying the Default SIP Configuration File in Chapter 3, “Managing Cisco SIP IP Phones”.

- Three-way conferencing

  Three-way conferencing supports one phone conferencing with two other phones by providing mixing on the initiating phone. To set up a 3-way conference call, see documentation on Making Conference Calls in “Getting Started with the Cisco IP Phone 7960”.

  Limitations:

  – Bridge node cannot hang up a single leaf node. When bridge node hangs up, both leaf nodes get torn down. All conferencing is stopped.

    Workaround: Have one leaf node hang up to leave other leg of the call connected.

  – When a two-way call is up and active, and a call waiting call comes in. The phone that gets the call has no way to conference in the new call-waiting call.

Workaround: Hang up the call-waiting call and then conference in the person who just called.

– Unable to initiate a transfer (attended or unattended) from Bridge node.

  Workaround: None. Transfer ability is disabled on the bridge node.

– DTMF does not work on Bridge node of 3-way call

  Workaround: None. DTMF capability is disabled on the bridge node.

– Three-way call is only able to use codecs g711ulaw or g711alaw

  Workaround: None. An attempt to force a different codec will result in the conference not being established.

– When pressing hold on bridge node the two leaf nodes cannot talk.

  Workaround: Use the Mute button instead of hold for leaf nodes to still be able to communicate.

– Unable to conference in a third party while the second call leg is ringing

  Workaround: Wait for the third party to answer the second call leg prior to conferencing in the first call leg.

- Distinctive Alerting

  If the INVITE message contains an Alert-Info header, distinctive ringing is invoked, Format of the header is “Alert-info: x”. “x” can be any number. This header is only received by the phone which does not generate this header.

  Distinctive ringing is supported when the phone is idle or during a call. In the idle mode, the phone rings with a different cadence. The selected ringing type plays twice with a short pause in between. In call-waiting mode, two short beeps are generated instead of one long beep.

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* or *Getting Started with the Cisco IP Phone 7960/7940*. When applicable, the headings in this section correspond with the section titles in the documentation.

Customizing Phone Settings

After adjusting the phone ringer type and volume settings, the phone must be left untouched for 20 seconds before the settings are saved to Flash. Once the settings are saved to Flash, they will be saved across a reboot. If the phone is not allowed to remain idle for 20 seconds after the ringer type and volume settings have been saved, the settings will not be saved across a phone reboot.

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

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Obtaining Technical Assistance

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Cisco Connection Online

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- 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.