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# Release Notes for Cisco Small Business IP Phone SPA50X and SPA30X Firmware Version 7.4.9a and 7.4.9c

**October 27, 2011**

These Release Notes describe the updates and fixes in versions 7.4.9a and 7.4.9c of the Cisco Small Business IP Phone SPA50X and SPA30X firmware.

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# Changes Since Cisco Small Business IP Phone SPA50X and SPA30X Firmware Version 7.4.9c

## Open Issues in Firmware Version 7.4.9c

Identifier	Summary
CSCtt44838 (SIP)	<p>SIP Server Failback is not Consistently Using the Primary DNS SVR Record</p> <p>The device does not consistently send SIP Request (REGISTER/INVITE) messages to the primary DNS SVR record. In the event of a failback, the device sends the SIP packet to the device in the second DNS server record.</p>
CSCtq69449 (SPCP)	<p>Attendant Console Hold Button Blinks the Wrong Color</p> <p>When a call is placed on hold, the line LED blinks red. It should blink green.</p>

## Resolved Issues in Firmware Version 7.4.9c—Phones Used With a SIP Call Control System

Identifier	Summary
CSCtt01306	<p>Cannot Perform Attended Transfer by using Speed Dial</p> <p>When the primary line is configured for shared call appearance, a blind transfer by using a speed dial causes the phone to become unresponsive.</p>
CSCtt05227	<p>Intermittent One-way Audio</p> <p>Real-time Transport Protocol (RTP) is seen in both directions, but there is no audio on the phone if G722 is listed before G711.</p>

Identifier	Summary
CSCtt18392	<p><b>Some Name Combinations Cause a Failure to Dial</b></p> <p>An XML parsing error might be caused by some first and last name combinations. When this occurs on a monitored BroadWorks BLF URL, the domain name of the monitored user is incorrect and any Speed-Dial attempts to that user fail.</p>
CSCts58141	<p><b>Music-on-Hold Prevents Voice Communication</b></p> <p>Broadworks phone users in a conference call lose voice communications if one of the users puts the call on hold. The Music On Hold (MoH) audio is heard on the non-held conferenced users.</p>
CSCts62234	<p><b>Blind Transfer with Bridge Line Appearance (BLA) Fails</b></p> <p>The blind transfer fails to connect the original caller to the target user. The original caller remains on hold when the target user answers the blind transfer.</p>
CSCtq62587	<p><b>Reduced probability of false DTMF RFC2833 events without keys being pressed.</b></p> <p>MSP (Broadworks/Broadsoft), RTP DTMF out-of-band events are being sent from the phone without DTMF keys being pressed causing a screeching sound to be sent to the PSTN caller. DTMF functions normally when the keys are pressed, but these “auto-generated” DTMF tones appear as a screeching sound. Audio is lost after screeching sound.</p>
CSCts64549 and CSCts03608	<p><b>When a Second Call is Received, Phone Becomes Unresponsive with STUN</b></p> <p>If the phone is using Session Traversal Utilities for NAT (STUN), there is a call in progress, and a second caller attempts to call the line, the original call continues and can be completed, but the user cannot use any of the phone buttons or access the menu until the phone is rebooted.</p>

# Changes in Cisco Small Business IP Phone SPA50X and SPA30X Firmware Version 7.4.9a

## Resolved Issues in Firmware Version 7.4.9a—Phones Used With a SIP Call Control System

Identifier	Summary
CSCtn24281	In a Metaswitch application, the Invite message does not contain the plus (+) character as required by the e164 message capability.
CSCtn76560	The BLF Status is not updated correctly when SIP NOTIFY messages are retransmitted.
CSCto09246	In a call-waiting scenario, after the user ends the active call, the phone does not display information about missed calls.
CSCto80487	The ringback tone is played after receiving the SIP 181 message.
CSCto81067	When using the SPA500S, the Busy Lamp Field (BLF) timer option (Retry-After) does not work as expected.
CSCto91430	The Refresh SUBSCRIBE message is sent to the CONTACT header instead of the proxy address.
CSCto95213	The Notify message does not match the To/From fields correctly per section 3.3.4 of rfc3265.
CSCto00274	Pressing the <b>cancel</b> softkey during a transfer plays the ringback tone and the call is placed on hold.
CSCto09246	Added support for the attended transfer call option for SPA50x with SPA500S.
CSCto14672	In certain timing conditions, a bye message is sent before the ack for <b>200 ok</b> on an invite message.
CSCto17505	In the Localization dictionary for Canadian French, the FIN D'APPEL softkey is not displayed correctly.

Identifier	Summary
CSCto31938	The XML FQDN resolution of the GPP parameter resolves to %2E instead of a period.
CSCto94942	With the share line configured, a call on hold cannot resume from a different phone.
CSCto97805	Added support for Directory User ID URIs over 39 characters in length.
CSCtq27158	The phone running firmware version 7.4.8 resubscribes to the CONTACT header address, and not the proxy address.
CSCtq27851	With Metaswitch, when assigning a softkey to play voice messages, the softkey does not respond after playing a message.
CSCtq35864	In some cases, the BroadWorks directory search causes the phone to restart.
CSCtq50596	Added support for SIP signaling messages containing special characters such as the plus (+) character.
CSCtq58001	Added BLF string parsing to prevent the duplication of the <b>sip:</b> tag in the URL in some Broadworks versions.
CSCtq85752	Unable to pick up a second incoming call using the <b>gpickup</b> or <b>pickup</b> soft keys.
CSCtq95659	Added inventory management type, length, and value descriptions (TLVs) to Link Layer Discovery Protocol-Media Endpoint Devices (LLDP-MED).
CSCtr39782	In languages other than English, if a softkey label is over 8 characters, the phone converts the string to English.
CSCtr39899	In the Spanish language dictionary, the Spanish translation for the <b>New Personal Dir Entry</b> and <b>Search Personal Directory</b> labels are too long to fit into the window; the last few characters of the labels are concatenated on the display.
CSCtr65293	Upon boot-up, a SPA50x attempts synchronization to a Web Provisioning server by using a Profile Rule. If a connection failure (ICMP) or 502 response is received, the phone does not retry.

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Identifier	Summary
CSCtr51701	In rare conditions with media transcoding, the phone receives 200 OK with G.722 in the SDP, but the phone continues to send RTP packets in G.711.
CSCtr95363	If the duration period for the dial tone is set to * in the script, after playing a reorder tone (the user goes off-hook and dials an invalid number), instead of playing continuously subsequent dial tones play for the duration period set for the reorder tone.

### Resolved Issues in Firmware Version 7.4.9a—Phones Used With an SPCP Call Control System

Identifier	Summary
CSCtn63391	UC500 application VoiceViewExpress: Unable to listen to multiple messages in a single session.
CSCto91170	The user must manually select the line key when dialing from a directory.

### Updates in Firmware Version 7.4.9a—Phones Used With a SIP Call Control System

#### Speed Dial Transfer with a Shared Line Causes the Phone to Lock (CSCts40009)

A shared FXO is assigned to a dedicated button. If the line is opened by pressing this button, then trying to use a speed dial number, the phone locks up.

Workaround: Power cycle the unit.

#### 3-way Conference MOH and Voice Mixing (CSCts58141)

Phones should bridge Music On Hold (MoH) audio for the non-held conferenced user during a 3-Way conference. Currently the SPA50x user receives MoH from BroadWorks Media Server. This audio is bridged to UserA. UserA and SPA50x should have 2-way audio plus MoH.

### Example Call Scenario:

- Setup:
  1. MoH is enabled at the BroadWorks Group Level, playing for all Hold calls.
  2. Conference Bridge is NULL on SPA50x Extension/Line.
- Activity
  1. SPA50x user calls BroadWorks UserA.
  2. UserA answers. Validate 2-way audio.
  3. SPA50x user presses "Conf" Soft-Key and dials BroadWorks UserB.
  4. UserA is placed on Hold. Validate MoH is heard.
  5. UserB answers call. Validate 2-way audio.
  6. SPA50x user presses "Conf" Soft-Key to bridge all parties. Validate 3-way audio.
  7. UserB places call on Hold.

### **Blind Transfer with Bridged Line Appearance (Sylantro) Fails (CSCts62234)**

Limited information in the database.

### **False DTMF RFC2833 Events (CSCtq62587)**

On a SPA508 - MSP (Broadworks/Broadsoft), RTP DTMF out-of-band events are being sent from the phone without DTMF keys being pressed causing a screeching sound to be sent to the PSTN caller. DTMF functions normally when the keys are pressed, but these "auto-generated" DTMF tones appear as a screeching sound. Audio is lost after screeching sound.

In some cases, changing a "DTMF Relay" option on adtran devices fixed the issue.

There is a possible "Talk Off" (when a human voice is read as a DTMF digit). The option changed the timer from 30ms to 80ms. Another fix was to use G.711 in-band, but this not feasible for WAN utilization. (Customers must buy extra T1s).

One report indicated that changing to RFC 2833 (AVT) DTMF transport only mode temporarily resolved the issue, but the same source is now reporting the issue persists.

### CSCts03608: Phone with STUN Freezes While Receiving a Second Incoming Call (CSCts64549)

SPA504G rings and the call is answered. A new call to the same number from a different phone and SPA504G flashes the second line button showing the new incoming call freezes the phone.

Further testing caused the phone to lock up on the third call. Other reports indicated the condition occurs on the fourth call.

### Updates in Firmware Version 7.4.9a—Phones Used With a SIP Call Control System

#### Report Configuration Deltas

A new option has been added to the Report Rule to trigger the reporting of configuration changes (deltas) to the server since the last resync, reboot, or upgrade.

The syntax of this option is:

Report Rule: [--delta] *URL*

Where *URL* is the path to where the report is stored on the server.

For example, to store delta configuration changes in a file with a name like SPA504G\_<MAC>\_<serial#>.xml, do one of the following:

- On the phone's Web GUI, set the **Report Rule** field on the **Configuration Profile** page (Voice tab > Provisioning tab > Configuration Profile) to:

**[--delta] http://reportTargetServer/reportPath/\$PN\_\$MA\_\$SN.xml**

- Add the following to your provisioning file:

```
<Report_Rule ua="na">[ --delta ]
http://reportTargetServer/reportPath/$PN_$MA_$SN.xml
</Report_Rule>
```

#### Capability to Configure DND and CFWD on a Per Line Basis (Applicable to Broadsoft)

Enable Do Not Disturb (DND) and Call Forwarding (CFWD) on a per line basis by using the new **Feature Key Sync** parameter, which has been added to all extension tabs. For any registered extension, you can enable device feature key synchronization by setting the **Feature Key Sync** to **Yes**.



Limitations:

- SPA301/SPA501—The softkey and phone menus are not available.
- SPA509—Lines 9–12 cannot be set by using the softkeys or menus.

### Support for Basic ACD Functions

To support basic Automatic Call Distribution (ACD), a new parameter, **Broadsoft ACD**, is added to the Web GUI under the **Call Feature Settings** section for each extension. The supported values are **Yes** and **No** (default).

If you set **Broadsoft ACD** to **Yes**, the phone sends a Subscribe message according to the Broadsoft specification.

If you set **Broadsoft ACD** to **No**, the phone might send out a Subscribe message because another feature is using ACD, but the phone ignores any Notify message from the Broadsoft server related to ACD.

Add the following to the configuration file to configure this feature on line 1:

```
<Broadsoft_ACD_1_ ua="na">Yes</Broadsoft_ACD_1_>
```

Limitations:

- SPA301/SPA501—ACD is not supported. The ACD Login and Status keys are not visible.
- SPA509—Lines 9–12 cannot be used as an ACD Agent since the Lines cannot be selected for Login/Logout and Agent status.

### Support for Distinctive Call Waiting Tone

Support for Distinctive Ring based on the Alert-Info header is extended to support the Distinctive Call Waiting Tone.

When the phone is off-hook on a call, the call waiting tone plays. The distinctive call waiting tone is generated based on the phone call waiting, tone frequency value, tone gain value, and the cadence value of the matched ring tone.

The cadence value, following the `c=` tag, of the matched ring tone must be an integer from 1 to 9 that specifies the ring cadence under the **Regional** tab of the Web GUI.

If there is no matching ring tone name, or an invalid cadence value is specified, the configured Call Waiting Tone is used.

### Reboot Reasons Stored in the Phone's Status XML File and Viewable from the LCD Screen and Web GUI

The phone now stores the last reboot/refresh reasons. When the phone is reset to factory defaults, this information is deleted.

The following is a list of the supported reboot/refresh reasons:

Reason	Description
DHCP Failed	A DHCP error caused a reboot when the lease expired, or when the renewal or verification failed.
Upgrade	An upgrade operation caused a reboot (regardless whether the upgrade completed or failed).
Provisioning	Changes made to parameter values by using the phone LCD or Web GUI, or a resync caused a reboot.
SIP Triggered	A SIP request caused a reboot.
Link Down	The link to the network went down causing a reboot.
VLAN Changed	The VLAN was changed causing a reboot.
RC	A remote customization caused a reboot.
User Triggered	The user manually triggered a warm reboot.
Software Req	A remote server triggered a warm reboot.
System <i>n</i>	System events (for example, running out of resources) triggered a warm reboot.
IP Changed	The phone IP address was changed triggering a warm reboot.

You can view the reboot history from the phone Web GUI, the phone LCD screen, and the phone SPA Status Dump file (<http://phoneIP/status.xml> or <http://phoneIP/admin/status.xml>).

#### *Viewing the Reboot History on the Web GUI*

(**Info > System Information > Reboot History**). On the **Reboot History** page, 5 fields were added (**Reboot Reason 1** (most recent reboot), **Reboot Reason 2**, **Reboot Reason 3**, **Reboot Reason 4**, and **Reboot Reason 5**).

Each field, if applicable, displays the reason for the reboot and a time stamp indicating when the reboot took place as in the following examples:

```
Reboot Reason 1: Provisioning(06/22/2011 13:29:33)
Reboot Reason 2: Upgrade(06/22/2011 13:01:43)
Reboot Reason 3: Provisioning(06/22/2011 10:40:12)
```

The reboot history is displayed in reverse chronological order, with the reasons for the latest reboot displayed in the **Reboot Reason 1** field.

### *Viewing the Reboot History on the Phone's LCD Screen*

A new menu, **Reboot History**, was added under the **Setup menu**. On the **Reboot History** Page, the 5 reboot entries are displayed in reverse chronological order, just like the Web GUI.

### *Viewing the Reboot History in the SPA Status Dump File*

The reboot history is stored in the SPA Status Dump file ([http://<phone\\_IP\\_address>/admin/status.xml](http://<phone_IP_address>/admin/status.xml)). In this file, tags **Reboot\_Reason\_1** to **Reboot\_Reason\_5** store the reboot history, as shown in this example:

```
<Reboot_History><Reboot_String/>
<Reboot_Reason_1>Provisioning(06/13/2011 14:03:43)</Reboot_Reason_1>
<Reboot_Reason_2>Provisioning(06/13/2011 13:58:15)</Reboot_Reason_2>
<Reboot_Reason_3>Provisioning(06/13/2011 12:08:58)</Reboot_Reason_3>
<Reboot_Reason_4>Provisioning(05/26/2011 15:26:49)</Reboot_Reason_4>
<Reboot_Reason_5>System 4(05/24/2011 10:20:06)</Reboot_Reason_5>
<Reboot_History/>
```

The Web GUI and the LCD screen get the reboot history from these tags.

## **SIP Publish Signaling Improvements**

The SPA Phone has been updated to resend the SIP PUBLISH message with the voice quality report once per 5xx response with a valid Retry-After header.

A valid time value in seconds is a positive integer from 0 to 65536. A SIP message with a Retry-After time value of 0 is treated as a “500 Server Internal Error” message.

A time value less than 0 is ignored.

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The following is a summary of the 5xx messages with Retry-After header that the phone supports:

5xx SIP responses	Description
500 Server Internal Error	Unexpected server condition that prevents fulfillment of the request.
503 Service Unavailable	Server is unavailable due to a temporary overload or maintenance.

### Support for Full SIP URI for SIP Publish

The **Voice Quality Report Address** parameter supports a full SIP URI. Examples of valid addresses are:

- collector@domain.com
- 123.collect@123.123.123.123:5555
- 5678@domain.com:5656

For example to configure for extension 1, edit the phone's configuration file as follows:

```
<Voice_Quality_Report_Address_1_ ua="na">collector@domain.com
</Voice_Quality_Report_Address_1_>
```

or

```
<Voice_Quality_Report_Address_1_ ua="na">123.collect@123.123.123.123:5555
</Voice_Quality_Report_Address_1_>
```

or

```
<Voice_Quality_Report_Address_1_ ua="na">5678@domain.com:5656
</Voice_Quality_Report_Address_1_>
```

### Audio Indication for Call Pickup Event

A new parameter, **Call Pickup Audio Notification**, was added under the **Attendant Console** section.

By default, this parameter is set to **No**. When set to **Yes**, the phone plays the Call Pickup tone when there are incoming calls to any of the lines that the user is monitoring with the Call Pickup function.

Use the following in your configuration file:

```
<Call_Pickup_Audio_Notification ua="na">Yes
</Call_Pickup_Audio_Notification>
```

This feature appears as follows in the phone configuration file:

```
<Call_Pickup_Tone ua="na">440@-10;30(.3/9.7/1)</Call_Pickup_Tone>
```

## User Definable Authentication Realm

A new parameter, **Reversed Authentication Realm**, was added at the extension level under the **Subscriber Information** section. The default value is empty; the proxy address is used as the authentication realm.

The parameter for extension 1 appears as follows in the phone configuration file:

```
<Reversed_Auth_Realm_1_ ua="na"></Reversed_Auth_Realm_1_>
```

To use a different authentication realm, enter the IP address to use in the **Reversed Authentication Realm** field.

## Accepting User Input When Screen Saver is Active

You can now dial a number when the screen saver is active and your first key entry will be accepted.

If the phone LCD is displaying the Home screen and the screen saver becomes active, and if there are no active calls, these events generated by user input are passed to the **Home** screen:

- Numeric keys
- Line keys
- Speaker key
- Headset key
- Mail Box key
- Handset off hook

All other key events are not passed.

### Handling of Failed Park/Transfer Response from the Server

The firmware has been updated so that when the phone receives a NOTIFY sipfrag message, the phone handles it immediately.

To enable this function:

- Phone's Web GUI

Set **Keep Referee When REFER Failed (SIP > SIP Parameters)** to **Yes**.

- Configuration file

Enable this feature as follows in the phone configuration file:

```
<Keep_Referee_When_REFER_Failed ua="na">Yes
</Keep_Referee_When_REFER_Failed>
```

### BroadSoft Directory Enhancements

The current setting menu for the Broadsoft directory feature includes fields to change the directory name, the directory type, its host server, the user ID, and the password.

These settings appear as follows in the phone's configuration file:

```
<!-- Broadsoft Settings -->
<Directory_Enable ua="na">No</Directory_Enable>
<XSI_Host_Server ua="na"></XSI_Host_Server>
<Directory_Name ua="na"></Directory_Name>
<Directory_Type ua="na">Enterprise</Directory_Type>
<!-- options: Enterprise/Group/Personal -->
<Directory_UserID ua="na"></Directory_UserID>
<Directory_Password ua="na"></Directory_Password>
```

To improve security for the Broadsoft directory feature, the SPA phone firmware was updated to place access restrictions on the host server and directory name entry fields.

Added six ringtones (SIP Mode Only).

Field	Access Restriction
Dir. Name	Admin password required (if set)
Host Server	Admin password required (if set)
Type	None

Field	Access Restriction
User ID	None
Password	None

Added two ringtones to the phone LCD GUI:

Ring #	GUI Label	Value of the w Parameter
11	Pulse	w=5
12	Du-dut	w=6

Two device ringtones are available to users. The ringtones appear in the phone configuration file:

```
<Ring11 ua="na">n=Pulse;w=5;c=1</Ring11>
<Ring12 ua="na">n=Du-dut;w=6;c=1</Ring12>
```

In addition, four user-configurable ring tones were added:

GUI Label	Value of the w Parameter
Warble	w=7
Low	w=8
Floor	w=9
Reverb	w=10

These four ringtones must be configured by using the phone Web GUI.

To configure/provision these ringtones using the Web GUI, go to the **Voice > Phone** page, and, in the **Ring Tone** section, modify the **n** and **w** parameters in four of the 12 ring fields (Ring1 to Ring12). Set the **n** parameter to the label of the ringtone you want displayed by the GUI. Set the **w** parameter equal to the ringtone parameter **w** value listed in the table above.

For example, to replace the ringtone in Ring1 with the Warble ringtone, change the value of the **Ring1** field to **n=warble;w=7;c=1** or configure as follows in the phone's configuration file:

```
<Ring1 ua="na">n=warble;w=7;c=1</Ring1>
```

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### Support for Display Diversion Info

This parameter controls the Diversion information on the phone display. When it is set to `Yes`, the phone screen displays the Diversion header information, if it exists, in the INVITE message. Otherwise, the Diversion header is not presented to the user.

The parameter is found under the SIP tab. The device displays the message `Display Diversion Info Y or N`. The default is `N`.

## Related Information

Support	
Cisco Small Business Support Community	<a href="http://www.cisco.com/go/smallbizsupport">www.cisco.com/go/smallbizsupport</a>
Cisco Small Business Support and Resources	<a href="http://www.cisco.com/go/smallbizhelp">www.cisco.com/go/smallbizhelp</a>
Phone Support Contacts	<a href="http://www.cisco.com/en/US/support/tsd_cisco_small_business_support_center_contacts.html">www.cisco.com/en/US/support/tsd_cisco_small_business_support_center_contacts.html</a>
Cisco Small Business Firmware Downloads	<a href="http://www.cisco.com/go/smallbizfirmware">www.cisco.com/go/smallbizfirmware</a> Select a link to download firmware for Cisco Small Business Products. No login is required. Downloads for all other Cisco Small Business products, including Network Storage Systems, are available in the Download area on Cisco.com at <a href="http://www.cisco.com/go/software">www.cisco.com/go/software</a> (registration/login required).



<b>Product Documentation</b>	
Cisco Small Business SPA50X	<a href="http://www.cisco.com/go/spa500phones">www.cisco.com/go/spa500phones</a>
<b>Cisco Small Business</b>	
Cisco Partner Central for Small Business (Partner Login Required)	<a href="http://www.cisco.com/web/partners/sell/smb">www.cisco.com/web/partners/sell/smb</a>
Cisco Small Business Home	<a href="http://www.cisco.com/smb">www.cisco.com/smb</a>

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