# Release Notes for Cisco Small Business SPA5XX and SPA30X IP Phones Firmware Version 7.5.1a

**April 13, 2012**

These Release Notes describe the updates and fixes in versions 7.5.1a of the Cisco Small Business SPA5XX and SPA30X IP Phones firmware.

## Contents

This document includes the following topics:

- **Cisco Small Business SPA5XX and SPA30X IP Phones Firmware Changes for Version 7.5.1a**

- **Related Information**

## Cisco Small Business SPA5XX and SPA30X IP Phones Firmware Changes for Version 7.5.1a

### New Features in Firmware Version 7.5.1a

#### Configuring Call Appearance Per Line (SIP)

This feature controls the number of call appearances that can occur on each line. The range is from **2** (default) to **10**. The characteristics are:

- Not supported on Cisco SPA501G or SPA301 phones.

- Supported only on phones operating in SIP mode.

- When the maximum numbers of calls per phone is reached, the phone does not allow you to make a new call and rejects incoming calls.

- In version 7.4.8a, only private call appearances are supported.

- In versions 7.5.1a and higher private and shared call appearances are supported.

To change the number call appearances allowed per line:

**STEP 1** Navigate to **Admin Login** > **advanced** > **Voice** > **Phone.**

**STEP 2** Under **Miscellaneous Line Key Settings** section set **Line ID Mapping** to **Horizontal First**.

**STEP 3** Set the **Call Appearance Per Line** to the number of calls per line to allow.

**STEP 4** Set Line Navigation to **Per Call**.

**STEP 5** Under the **Programmable Softkeys** section, set Programmable Softkey Enable to **Yes**.

### Reregister upon Failover or Recovery

Cisco SPA IP phones re-register to the new proxy upon a failover or recovery event. (Previously the phones would only re-register to the new proxy when the registration timer expired.)

### Display HD Voice Icon When a G.722 Call is Established

Cisco SPA IP phones display a HD Logo next to the call state. This allows a caller to clearly differentiate between a wide-band call and a narrow-band call.

### Add Ten Programmable Softkeys

Ten programmable soft keys have been added, increasing the number of programmable softkeys on Cisco SPA IP phones to a total of sixteen.

### Digest Authentication is Supported in the Profile and Report Rule

Digest Authentication based on the username and password is defined as part of profile rule and a report rule. The syntax is:

```
[--uid $SA]
[--pwd $SB]
```

In the following example, the phone uses this username and password when it is challenged by the server:

```
[--uid slee --pwd 1234] http://download.com/spacfg.xml
```

### user=phone Syntax in a SIP Message

When a telephone URL is converted to a SIP URL and the telephone number is represented by the user portion of the URI, the SIP URL includes the optional `:user=phone` parameter (RFC3261). For example:

```
To: sip:+12325551234@example.com;user=phone
```

The requirement adds a new parameter under the respective Line SIP Settings to add:

```
User=phone ( Yes/No )
```

The default value is **No**.

### Failover and Recover on a 503 Error Message

Cisco SPA IP phones perform a failover and recovery upon receiving a 503 error message in a SIP request. (Previously, the phones only performed a failover and recovery after the request time-out (no response) expired.)

### BroadSoft Advanced Call Park and Recall Support

BroadSoft Call Park and Recall improves the calling party's experience by:

- Allowing front-office personnel to distinguish between a call that reverted back to the front office and a new call.

- Provides additional routing and call handling options that improve front-office efficiency when the receptionist is unavailable or busy.

- A visual indication is provided to users when a call is parked against their extension.

For detailed signaling requirements, see the *Broadsoft/Cisco Partner Configuration Guide* or the *Cisco Small Business SPA IP Phones Administration Guide*.

### Compliance with SIP Publish RFC6035

The SIP Publish implementation complies with RFC6035. The updates include:

- SDP Media loopback sequence number.

- RTCP report loss rate.

- SDP Mismatch for packet interval.

### Scrollable Softkeys

System administrators can add a softkey titled **More**. Selecting **More** moves to the next set of softkeys.

### BroadSoft Call Agent Unavailable Reason Code

The user can enter an *unavailable reason code*.

### Speed Dial Wait, Pause, and DTMF

Speed dial entries can include *wait* (**X**) and *pause* (**,**) parameters that allow manual and automatic DTMF (Dual-tone multi-frequency) signal transmission.

Syntax:

```
{Dial_String}[ ][,|X][DTMF_string][,|X][DTMF_string]
```

Where:

*Dial_String*—The number that the user is trying to reach. For example, 8537777 or 14088537777.

(space)—A dial termination that defines or delimits the end of the dial string. The space is mandatory. If the phone encounters an X or a comma (,) before the space, the characters are treated as part of dial string.

, (comma)—A 2-second pause that is inserted for each comma in the string.

**X** (wait)—Awaits user input and acknowledgement

- The "*Waiting for more digits input. When done, press OK to continue*" message displays.

- The user manually enters DTMF signals by using the dial pad.

- The user selects **OK**, acknowledging the transmission of the manual entry is complete.

- The phone sends any DTMF signals defined by *DTMF_string*.

- The phone executes the next parameter. If there are no more parameters in the dial string to execute, the phone exits to the main screen.

The wait does not timeout; the wait prompt window is not dismissed until user confirms the wait prompt or the call is ended either by the user or ended by the remote device.

*DTMF_string*—The DTMF signals that a user is sends to a remote device after the call is connected. The phone cannot send signals other than valid DTMF signals.

**Example**

```
18887225555 ,,5552X2222
```

A speed dial entry triggers the phone to dial 18887225555. The space indicates the end of the dial string. The phone waits 4 seconds (2 commas), and then sends the DTMF signals **5552**.

A message is displayed, prompting the user to manually dial DTMF signals. When the user has finished dialing the DTMF signals, the user selects **OK** to confirm the manual DTMF signal transmission is complete. The phone sends the DTMF signals **2222**.

**Usage Guidelines**

A user can transmit DTMF signals anytime, as long as the call is connected.

The length of the string, including the Xs or commas (,), that can be supported is limited to the length of a speed-dial entry, dial screen entry, directory entry, and so forth.

If the screen displayed is not a call appearance screen and a wait is initiated, the phone displays the home screen and prompts the user to input more DTMF signals. If this occurs while the user is editing an entry, the edits might be lost.

If only the first part of a dial string matches a dial plan, when the call is dialed, the portion of the dial string that does not match the dial string is ignored.  For example:

```
85377776666 ,,1,23
```

If **8537777** matches a dial plan, the characters **6666** are ignored. The phone waits 4 seconds before sending DTMf 1. It then wait 2 seconds and sends DTMF 23.

When logging the call, the phone only logs the dial string; the DTMF strings are not logged.

Valid DTMF signals are 0-9, *, or #. All other characters are ignored.

**Limitations**

When the call is connected and immediately transferred, the phone might not be able to process the DTMF signals, depending on the duration the call is connected before it is transferred, for example, any internal call through a Cisco UC320. As soon as the call is connected, it is transferred and the DTMF signals are ignored in the Cisco UC320.

Because the Cisco SPA301 and Cisco SPA501G do not have IP phone screens, the wait (**X**) is not supported.

This feature is not supported in when the phone is in SPCP mode.

### Chunk Encoding for HTTP 1.1

Cisco SPA IP phones support HTTP Version 1.0, HTTP Version 1.1, and Chunk Encoding when HTTP Version 1.1 is the negotiated transport protocol.

### Root CA Certificate Download

Cisco SPA IP phones support root certificates–of–authority (CAs) embedded in the firmware and they will authenticate third–party signed certificates, such as those provided by Verisign, Cybertrust, and so forth.

In the provisioning tab of the phone web user interface, a new root CA can be added to the list of root CAs.

## Resolved Issues in Firmware Version 7.5.1a

| Identifier | Summary |
|------------|---------|
| CSCtr29162 | Daylight savings parameter is not properly updating the device time. |
| CSCtr57748 | During a mid-call codec change, although the G.722 codec is an available codec, it might not be acknowledged. |
| CSCts76553 | When using LDAP (Lightweight Directory Access Protocol) to select a contact and with the headset is off hook, the phone might become unresponsive. |
| CSCts80572 | Broadwork directory lookups are using an incorrect syntax in the HTTP Get request. |

| Identifier | Summary |
| --- | --- |
| CSCtt19948 | The SRTP (Secure Real-time Transport Protocol) does not properly handle a key change during a call, resulting in one-way audio. |
| CSCtt24774 | Switching between languages multiple times causes the phone to become unresponsive. |
| CSCtr21825 | When the `dns srv auto prefix` is set to **yes**, the phone does not do a proper DNS A record (return an IP address). |
| CSCtr77257 | Improper handling of Invite with Replace (non-matching dialog). |
| CSCtu39616 | Phones repeat DNS queries on static DNS SRV configurations. |
| CSCtu59498 | When multiple DHCP offers are received simultaneously, the phone does not accept the offers. |
| CSCtv00323 | Spaces used in Broadsoft directory searches are seen as delimiters and truncate the http_get URL. |
| CSCtw58740 | Phones using G.726-32 codec might fail negotiation. |
| CSCtw62255 | Phones might become unresponsive during a Sylantro server failover. |
| CSCtw75484 | The **To** header in a SIP info message does not include the **To** tag. |
| CSCtw78674 | Configuring the Call Appearances/Line setting to >2 and provisioning a Line/Ext as shared line, causes the IP phone screen to indicate that the Conference and Transfer softkey functions are not available for the line. |
| CSCtu22571 (SPCP) | The call waiting tone might interrupt a current call. |
| CSCtx43650 | In response to a 403 message, the MWI subscription does not properly handle the timer value. |
| CSCtx69119 | The **Personal** directory queries display differently on the Cisco SPA5XX phones than they do on the Cisco SPA525G or Cisco SPA525G2 phones. |

| Identifier | Summary |
|---|---|
| CSCtx80989 | Phones did not respond to `pick` or `pickup` on PSK. Avoid special handling if it is speeddial from the SD key. |
| CSCtx81224 | Cisco SPA501G and Cisco SPA301 phones can be factory reset by using the IVR help menu. |
| CSCtx99354 | The phone was giving precedence to the name in the personal directory over the name in the caller ID. |

# Related Information

| Support | |
|---|---|
| Cisco Small Business Support Community | www.cisco.com/go/smallbizsupport |
| Cisco Small Business Support and Resources | www.cisco.com/go/smallbizhelp |
| Phone Support Contacts | www.cisco.com/en/US/support/ tsd_cisco_small_business _support_center_contacts.html |
| Cisco Small Business Firmware Downloads | www.cisco.com/go/smallbizfirmware<br><br>Select a link to download firmware for Cisco Small Business Products. No login is required. |

| Product Documentation | |
|---|---|
| Cisco Small Business SPA50X | www.cisco.com/go/spa500phones |
| Regulatory Compliance and Safety Information for the Cisco SPA300 Series and Cisco SPA500 Series IP Phones | www.cisco.com/en/US/docs/voice_ip_comm/ csbpipp/ip_phones/regulatory_compliance/guide/ rcsi_500_series.pdf |
| **Cisco Small Business** | |
| Cisco Partner Central for Small Business (Partner Login Required) | www.cisco.com/web/partners/sell/smb |
| Cisco Small Business Home | www.cisco.com/smb |

78-20689-01