



ADMINISTRATION GUIDE

Cisco SPA and Wireless IP Phone

Cisco SPA525G, SPA922, SPA942, SPA962, and
WIP310

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About This Document

This document provides information needed to administer the Cisco SPA and Wireless IP phones through the web administration interface. It covers the following SPA and Wireless IP phone models:

- SPA922
- SPA942
- SPA962
- SPA525G
- WIP310

This guide describes how to configure and administer your Cisco SPA and Wireless IP phones. It does not cover configuration and administration of the Cisco SPA9000 or other Cisco SPA components. See the links in the [“Related Documentation” section on page 277](#) for more information on these products.

Audience

This manual is for Cisco partners, value added resellers (VARs) and Internet service providers. Use this manual to configure the administrative functions of the Cisco SPA and Wireless IP Phones.

Organization

This table describes the contents of each chapter in this document.

Chapter	Title	Description
Chapter 1	This chapter.	
Chapter 2	Getting Started	Describes the first steps to administering your Cisco SPA IP or Wireless-G phones, including upgrading firmware and accessing the web administration interface.
Chapter 3	Configuring Lines and Extensions	Contains information for configuring phone lines and extensions.
Chapter 4	Customizing Cisco SPA and Wireless IP Phones	Gives information about customizing phones, including configuring call features and phone features.
Chapter 5	Configuring SIP, SPCP, and NAT	Covers configuring the phone protocol (SIP or SPCP) and NAT.
Chapter 6	Configuring Security, Quality, and Network Features	Describes configuring security, quality, and network features such as codecs, domains, and VLAN settings.
Chapter 7	Provisioning Basics	Describes how to remotely provision and upgrade the phones.
Chapter 8	Configuring Regional Parameters and Supplementary Services	Covers regional parameters, such as tones, ring patterns, and localization.
Chapter 9	Configuring the SPA932 Attendant Console	Describes configuring the SPA932 attendant console for use with the SPA962 and SPA525G.
Appendix A	Creating an LED Script	Describes creating LED scripts.
Appendix B	SPA and Wireless IP Phone Field Reference	Describes all fields for the IP phone with values and configuration information.

Chapter	Title	Description
Appendix C	Where to Go From Here	Provides URLs for related product documentation and resources available on Cisco.com.

Read Me First

The PDF Find/Search tool lets you find information quickly and easily online. You can:

- Search an individual PDF
- Search multiple PDFs at once (for example, all PDFs in a specific folder or disk drive)
- Perform advanced searches

Finding Information in PDF Files

The IP phone documents are published as PDF files. The PDF Find/Search tool within Adobe® Reader® lets you find information quickly and easily online. You can perform the following tasks:

- Search an individual PDF file.
- Search multiple PDF files at once (for example, all PDFs in a specific folder or disk drive).
- Perform advanced searches.

Finding Text in a PDF

Follow this procedure to find text in a PDF file.

STEP 1 Enter your search terms in the Find text box on the toolbar.



NOTE By default, the Find tool is available at the right end of the Acrobat toolbar. If the Find tool does not appear, choose **Edit > Find**.



STEP 2 Optionally, click the arrow next to the Find text box to refine your search by choosing special options such as Whole Words Only.

STEP 3 Press **Enter**.

STEP 4 Acrobat displays the first instance of the search term.

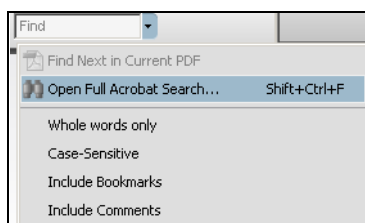
STEP 5 Press **Enter** again to continue to more instances of the term.

Finding Text in Multiple PDF Files

The *Search* window lets you search for terms in multiple PDF files that are stored on your PC or local network. The PDF files do not need to be open.

STEP 1 Start Acrobat Professional or Adobe Reader.

STEP 2 Choose **Edit > Search**, or click the arrow next to the *Find* box and then choose **Open Full Acrobat Search**.

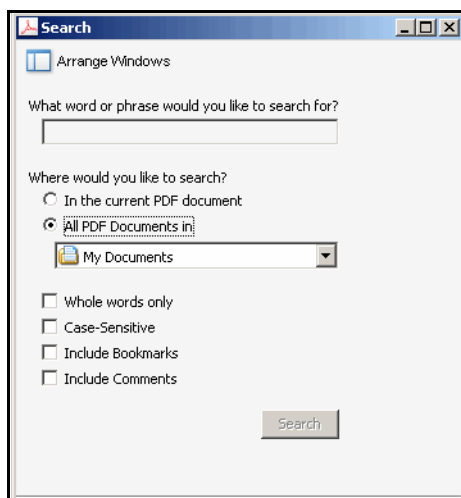


STEP 3 In the *Search* window, complete the following steps:

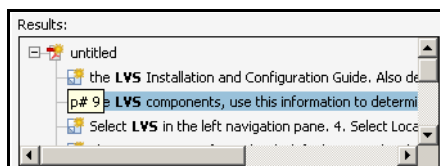
- a. Enter the text that you want to find.
- b. Choose **All PDF Documents** in.

From the drop-down box, choose **Browse for Location**. Then choose the location on your computer or local network, and click **OK**.

- c. If you want to specify additional search criteria, click **Use Advanced Search Options**, and choose the options you want.
- d. Click **Search**.



- STEP 4** When the Results appear, click + to open a folder, and then click any link to open the file where the search terms appear.



For more information about the Find and Search functions, see the Adobe Acrobat online help.

Getting Started

This chapter contains basic information on Cisco SPA and Wireless-G IP Phones. It includes the following sections:

- [Overview, page 15](#)
- [Network Configurations, page 17](#)
- [Prerequisites, page 20](#)
- [Upgrading Firmware, page 20](#)
- [Using the Web Administration User Interface, page 25](#)
- [Viewing Phone Information, page 31](#)

Overview

The Cisco SPA and Wireless-G IP Phone family is a line of full-featured VoIP (Voice over Internet Protocol) phones that provide voice communication over an IP network. They provide all the features of traditional business phones, such as call forwarding, redialing, speed dialing, transferring calls, conference calling and accessing voice mail. Calls can be made or received with a handset, a headset, or a speaker.

The Cisco SPA and Wireless-G IP Phone family includes the following phone models:

- SPA922—Single-line VoIP phone.
- SPA941/942—Four-line VoIP phone with high-resolution monochromatic display.
- SPA962—Six-line VoIP phone with high-resolution color display.
- SPA525G—Five-line VoIP phone with high-resolution color display and configurable protocol (SIP or SCCP).
- WIP310—Single-line Wireless-G IP phone.

Phone model features are summarized in the following table:

Phone Model	Voice Lines	High-Resolution Graphical Display
SPA525G	5	Backlit 320 x 240, 262K color, 3.2" inch LCD with screen saver and customizable logo/background picture. Wireless-G enabled.
SPA922	1	Backlit 128 x 64 pixels with screen saver and customizable logo/background picture (128 x 48 pixels available for background picture).
SPA 942	4	Backlit 128 x 64 pixels with screen saver and customizable logo/background picture (128 x 48 pixels available for background picture).
SPA962	6	Color 320 x 240 pixels with screen saver and customizable logo/background picture (320 x 240 pixels available for background picture). Also provides customizable photo album.
WIP310	1	Backlit color 128 x 160 pixels.

For more information on phone features, see the data sheets for each product.

Cisco SPA932 Attendant Console

The SPA932 Attendant Console has 32 LEDs/buttons for dialing, call transfer, call pick up and call monitoring. Multi-colored LEDs monitor the status of each configured voice line via busy lamp field (BLF). You can attach two attendant consoles to an IP phone, for 64 LEDs/buttons. For more information, see [Chapter 9, “Configuring the SPA932 Attendant Console.”](#)

Network Configurations

The Cisco SPA and Wireless-G IP Phones support Session Initiation Protocol (SIP) or (SPA525G only) Smart Phone Control Protocol (SPCP). You can use the Cisco SPA9X2 and Wireless-G IP Phones as part of a Cisco SPA9000 Voice System phone network, or with any vendor’s IP PBX system that supports SIP. The SPA525G can be used as part of a Cisco SPA9000 Voice System phone network, a SIP network, or as part of the Cisco Unified Communications 500 Series for Small Business.



NOTE

Using the Cisco SPA IP and Wireless-G phones as part of a Cisco SPA9000 Voice System Network provides seamless integration of advanced features, such as paging, call pickup, and shared line appearances.

This document describes some common network configurations; however, your configuration may vary depending on the type of equipment used by your service provider.

Cisco SPA9000 Voice System

The SPA9000 Voice System is an affordable and feature-rich voice over IP (VoIP) telephone system that is designed especially for small businesses. The SPA9000 Voice System uses standard TCP/IP protocols and can provide global connectivity through any Internet Telephony Service Provider (ITSP) that supports Session Initiation Protocol (SIP).

At minimum, the SPA9000 Voice System includes a SPA9000 IP PBX and one or more SPA900 series IP phones. These devices are connected through a switch to a local area network. With an Internet connection, the SPA9000 Voice System can subscribe to ITSP services to take advantage of low calling rates. With the optional SPA400, the SPA9000 Voice System can connect to the Public Switched Telephone Network (PSTN) to support legacy phone lines. The SPA400 also provides local voice mail service.

When you use Cisco SPA IP and Wireless-G IP phones with the Cisco SPA9000 Voice System, the following additional phone features are available:

- Auto attendant for multiple extensions
- Music on hold
- Configurable call routing
- Multiple DID numbers per VoIP line
- Call hunting (sequential, round robin, random)
- Group paging
- Call parking
- Call pick up
- Group call pick up

You can configure and manage the SPA9000 Voice System using an Interactive Voice Response (IVR) system, the Cisco SPA9000 Voice System Setup Wizard, or a built-in web server. For more information, see the *Cisco SPA9000 Voice System Administration Guide*.

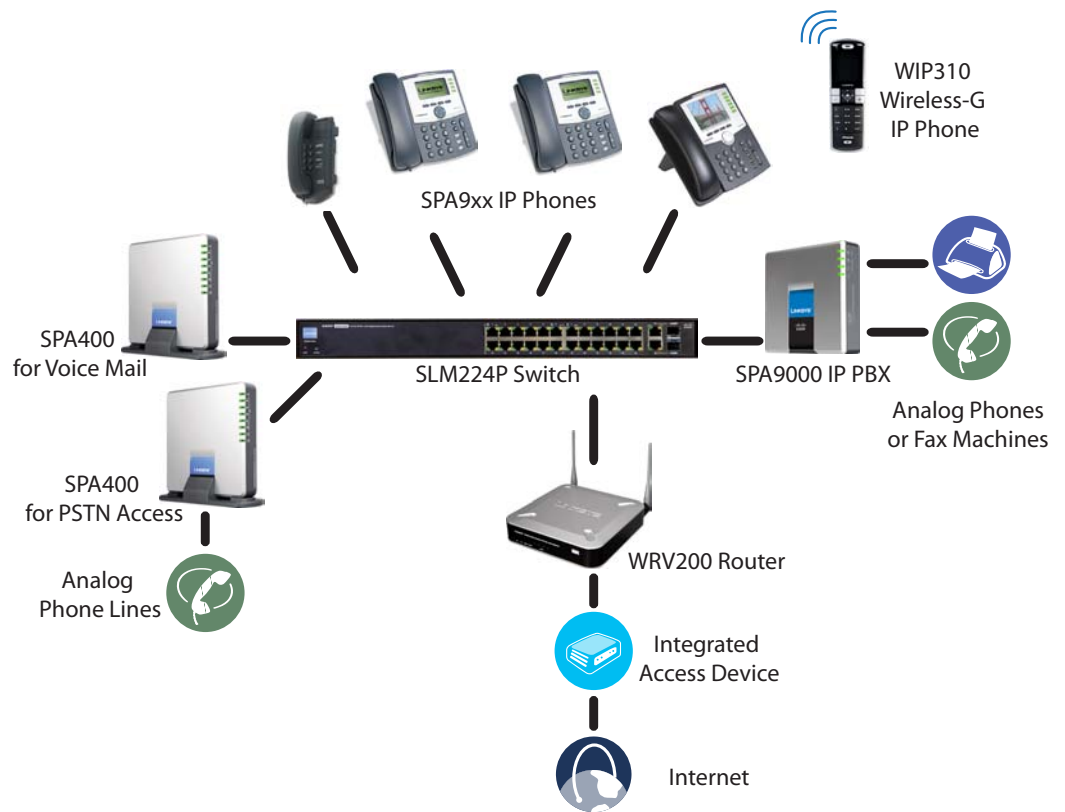


Figure 1 Cisco SPA IP Phones with Cisco SPA9000 IP PBX

Cisco Unified Communications 500 Series for Small Business

The Cisco Unified Communications 500 Series for Small Business is an affordable appliance that provides voice, data, voice mail, Automated Attendant, video, security, and wireless capabilities while integrating with existing desktop applications such as calendar, e-mail, and customer relationship management (CRM) programs. The Cisco SPA525G IP Phone can be configured to work with this system.

Other SIP IP PBX Call Control Systems

The Cisco SPA and Wireless-G IP phones are compatible with other IP PBX call control systems, such as BroadSoft and Asterisk, that use SIP for call processing. Configuration of those systems is not covered in this document. Additional resources for configuring the SPA and Wireless-G IP Phones to work with these systems are available in [Appendix C, “Where to Go From Here.”](#)

Prerequisites

This document assumes that you have performed the following prerequisites before administering your Cisco SPA or Wireless-G IP phone. If you have not completed these prerequisites, see the documentation in [Appendix C, “Where to Go From Here,”](#) for more information.

1. Set up your IP network.
2. Configure the wireless network (required for SPA525G and WIP310).
3. Install and configure the call control system, such as such as a SPA9000, Cisco Unified Communications 500 Series for Small Business, or an Internet-based IP PBX.
4. Update firmware. See [Upgrading Firmware, page 20](#).

Upgrading Firmware

Phones should be upgraded to the latest firmware before using any administration features. There are various ways to upgrade your firmware:

All Phones

- Cisco SPA9000 Voice System Setup Wizard—If you are using the SPA and Wireless IP phones with a SPA9000, you can use the Cisco SPA9000 Voice System Setup Wizard to upgrade your phones. See the *Cisco SPA9000 Voice System Setup Wizard User Guide*.
- Autoprovisioning—A configuration file that includes upgrade information is sent downloaded by a user’s phone when it is powered on. See the [“Upgrading, Resyncing, and Rebooting Phones” section on page 138](#).

SPA9X2 and WIP310

- Firmware Upgrade Executable File (SPA9X2 or WIP310)—Download the firmware upgrade utility from the product page on [Cisco.com](https://www.cisco.com) to your PC desktop and run the upgrade from your PC by double-clicking the executable file. Your computer must be on the same network as the SPA IP phone.

SPA525G

- Web Interface (SPA525G)—You can download the latest firmware onto your PC desktop and use the web interface to upgrade your firmware.

WIP310

- TFTP/HTTP server—The latest firmware image file is loaded onto an HTTP/TFTP server and is accessed by a web browser. See the *Cisco WIP310 User Guide* for more information.

Determining the Current Firmware Version

Before upgrading, determine the current firmware version:

SPA9X2

-
- STEP 1** Press the **Setup** button.
- STEP 2** Scroll to **Product Info** and then press **Select**. The current firmware is displayed under *Software Version*.
-

SPA525G

-
- STEP 1** Press the **Setup** button.
- STEP 2** Scroll to **Status** and press **Select**.
- STEP 3** Select **Product Information**. The current firmware is displayed under *Software Version*.
-

WIP310

-
- STEP 1** In the **Home** screen, press the **Options**, highlight *Phone Info*, and press the **Select** button.
- STEP 2** Scroll to *Software Version*.
-

Determining Your IP Address

Before you upgrade, you'll need the IP address of the phone you are upgrading. To get your IP address:

SPA9X2

-
- STEP 1** Press the **Setup** button.
- STEP 2** Scroll to **Network** and press **Select**. The IP Address is displayed under Current IP.
-

SPA525G

-
- STEP 1** Press the **Setup** button.
- STEP 2** Scroll to **Status** and press **Select**.
- STEP 3** Scroll to **Network Status** and press **Select**. The IP address of your phone is displayed.
-

WIP310

-
- STEP 1** In the Home screen, press the **Select** key and navigate to *Settings*.
- STEP 2** Press the **Select** key and navigate to *Phone Info*.
- STEP 3** Scroll to *Software Version*. The phone lists the software version.
-

Downloading the Firmware

- STEP 1** Direct your browser to the following URL: <http://www.cisco.com/go/smallbiz>.
- STEP 2** Search to locate your product.
- STEP 3** Locate the download page and download the firmware file.
- STEP 4** Click **Save** in the *File Download* dialog box that appears.
- STEP 5** In the *Save As* dialog box, choose a location for the file and click **Save**.
- STEP 6** When the download is complete, if prompted, click **Close**.



NOTE: The name of the file depends on the firmware file of your phone.

- STEP 7** If the firmware file you download is in zip format, double-click the file and extract its contents to a single folder or to the desktop.

Installing the Firmware (SPA9X2)



NOTE: Your computer must be on the same sub-network as the phone you are upgrading.

- STEP 1** Run the executable file for the firmware upgrade (for example, double click **upg-spa942-5-2-8.exe**).
- STEP 2** Click **Continue** after reading the message regarding upgrading and your service provider.

STEP 3 Enter the IP address of your phone.

STEP 4 Follow the on-screen directions.

Installing the Firmware (SPA525G)

STEP 1 Log in to the web interface.

STEP 2 Choose the **Firmware Upgrade** tab.

STEP 3 Click **Firmware Upgrade Window**.

STEP 4 Browse to select the firmware file from your PC. Click **Submit**. The firmware is installed and your phone reboots.

Installing the Firmware (WIP310)



NOTE: Your computer must be on the same sub-network as the phone you are upgrading.

STEP 1 Turn off your WIP310 and connect it to your computer by using the USB cable.

STEP 2 Double-click the executable file for the firmware upgrade (for example, double-click **wip310-5-0-11.exe**).

STEP 3 Follow the on-screen instructions.

STEP 4 When the upgrade is complete, disconnect the phone from your PC and power it on.

Using the Web Administration User Interface

You must be connected to the same network as your phone. For example, if you are connected to a VPN, you must first disconnect.

**NOTE**

If you are using the SPA525G with the Cisco Unified Communications 500 Series for Small Business for Call Control, use Cisco Unified Communication Manager Express or Cisco Configuration Assistant for phone administration. For more information, refer to the *Cisco Unified Communications 500 Office Administrator Guide*.

To access the IP phone administration web user interface (UI):

- STEP 1** Launch a web browser on a computer that can reach the phone on the network.
- STEP 2** Direct the browser to the IP address of the phone. To determine the IP address:
- SPA9X2: Press the **Setup** button, then select **Network**. The *Current IP* field shows the phone's current IP address.
 - WIP310: In the **Home** screen, press **Options** and highlight *Phone Info*. Press the **Select** button.
 - SPA525G: Press the **Setup** button, then select **Status**. Select *Network Status*. The IP address is displayed.
- STEP 3** Enter the IP address in your web browser address bar. For example:

```
http://192.168.1.8
```

If you have trouble accessing the web interface, perform the following steps:

SPA9X2

-
- STEP 1** Press the **Setup** button on the phone.
 - STEP 2** Select **Network**.
 - STEP 3** Scroll to **Enable Web Server** and make sure that it is set to **Yes**. If not, press the **Edit** soft key and press **y/n** soft key to set it to **Yes**.
 - STEP 4** Press **OK**, then press **Save**.



NOTE: If your service provider disabled access to the web UI, you must contact the service provider.

WIP310:

-
- STEP 1** In the **Home** screen, press the **Select** button to choose **Settings**.
 - STEP 2** Press the **Select** button again to reach the **Settings** menu.
 - STEP 3** Scroll to highlight *Misc Settings* and press the **Select** button.
 - STEP 4** Press the left arrow to ensure that **Enable Web Server** is set to **On**.
 - STEP 5** Press the **Select** button to save this setting.

SPA525G:

-
- STEP 1** Press the **Setup** button.
 - STEP 2** Select **Network Configuration**.
 - STEP 3** Scroll to **Web Server** and make sure it is set to **On**.
 - STEP 4** Press **Save**.
-

Understanding Administrator and User Views

Depending on whether you are a VAR or service provider, you might have different privileges. By default, the Administrator account name is **admin**, and the User account name is **user**. These account names cannot be changed.

If the service provider set an Administrator account password, you are prompted for it when you click **Admin Login**.

The Administrator account can modify all web profile parameters, including web parameters available to the user login. The Administrator specifies the parameters that a User account can modify using the **Provisioning** tab of the web UI.



NOTE: No default passwords are assigned to either the Administrator or User accounts. Only the Administrator account can assign and change passwords.

Accessing Administrative Options

To access administrative options, either:

- Log in to the web interface, then click **Admin Login**.
- Enter the following URL when accessing the interface:

`http://phone.ip.address/admin/`



NOTE: To save changes on a web page, click **Submit All Changes** before switching between User and Admin Login or between basic and advanced views. Switching logins or views discards any unsubmitted changes.

Understanding Basic and Advanced Views

These views are similar, but *advanced* view shows more options on each web page. To see all available options for your login, use the *advanced* view.

Using the Web Administration Tabs

The following graphics show all Admin Login *advanced* tabs that appear on your web UI pages. When viewing in User Login or basic view, fewer tabs are visible.

Each tab contains different parameters. Some tasks require you to set parameters in different tabs.

For field reference about each parameter available on the web UI, see [Appendix B, “SPA and Wireless IP Phone Field Reference.”](#)

Roadmap to Web UI Features

The following tables provide a roadmap to features available on the web UI.

To perform these tasks...	On the web UI, click the ...
<p>View phone, extension, and line/call information such, including:</p> <ul style="list-style-type: none">▪ DHCP, current IP address, DNS addresses▪ Software and hardware versions▪ Broadcast, RTP, and SIP information▪ Registration state▪ Packets sent, received, lost, and other information	<p>Info tab</p> <p>See “Viewing Phone Information” section on page 31.</p>
<p>Configure system-level parameters, including network and debug parameters. To:</p> <ul style="list-style-type: none">▪ Enable the web UI and web administrator access▪ Set the Internet connection type to DHCP▪ Configure the syslog and debug servers▪ Enable VLAN and CDP	<p>System tab</p> <p>See Chapter 6, “Configuring Security, Quality, and Network Features.”</p>

To perform these tasks...	On the web UI, click the ...
Configure parameters to adjust SIP stack and protocols. To enable: <ul style="list-style-type: none"> ▪ CTI ▪ SIP-B ▪ STUN 	SIP tab See Chapter 5, “Configuring SIP, SPCP, and NAT.”
Configure provisioning parameters. To: <ul style="list-style-type: none"> ▪ Enable remote provisioning ▪ Enable firmware upgrades ▪ Set general purpose parameters 	Provisioning tab The <i>Provisioning</i> tab is viewable by Admin logins only. See Chapter 7, “Provisioning Basics.” For additional information about provisioning, see the <i>Cisco SPA Provisioning Guide</i> (for Cisco service providers).
Configure parameters that depend on country or region, including: <ul style="list-style-type: none"> ▪ Call progress tones ▪ Ring patterns ▪ Star codes/vertical service activation codes ▪ Vertical service announcement codes ▪ Local date/time and language 	Regional tab See Chapter 8, “Configuring Regional Parameters and Supplementary Services.”
Configure General phone station info, which applies to all extensions configured for the phone, including: <ul style="list-style-type: none"> ▪ Station name, voice mail number, text logos and background pictures ▪ Extension numbers for line keys ▪ Shared call (line) appearance ▪ Enabling call conferencing, call forward, call transfer, and so on. ▪ Select ring tones, audio input, and extension mobility settings 	Phone tab See Chapter 4, “Customizing Cisco SPA and Wireless IP Phones.”

To perform these tasks...	On the web UI, click the ...
<p>Customize individual extension parameters, including:</p> <ul style="list-style-type: none"> ▪ Shared line/call appearance ▪ NAT settings ▪ SIP settings such as SIP debug and SIP port ▪ Mailbox ID, MOH server ▪ Voice mail server ▪ Proxy and registration information ▪ Subscriber information such as user ID and password ▪ Audio settings ▪ Dial plan settings 	<p>Ext tab</p> <p>(1-6, depending on phone model)</p> <p>See Chapter 3, “Configuring Extensions.”</p>
<p>Customize user-level parameters, including:</p> <ul style="list-style-type: none"> ▪ Call forward ▪ Speed dial ▪ Supplementary services ▪ Web information (RSS newsfeeds) ▪ Traffic information settings ▪ Audio volume ▪ Phone GUI settings 	<p>User tab</p> <p>See Chapter 4, “Customizing Cisco SPA and Wireless IP Phones.”</p>
<p>View and change parameters for Unit 1 and Unit 2 (applicable only to SPA962 phones with one or two SPA932 attendant consoles attached)</p>	<p>932 tab</p> <p>See Chapter 9, “Configuring the SPA932 Attendant Console.”</p>

Viewing Phone Information

After you log on to the web UI (see [“Using the Web Administration User Interface” section on page 25](#)), you can check the current status of a the SPA or Wireless IP Phone by clicking the **Info** tab (appears by default). The Info tab shows information about all phone extensions, including phone statistics and the registration status. All fields are read-only.

See [“Info Tab” section on page 191](#) for more information about the fields.

Configuring Lines and Extensions

This chapter contains the following sections:

- [Configuring Lines, page 33](#)
- [Configuring Extensions, page 44](#)

Configuring Lines

The Cisco IP Phones (also called *stations* in this document) have the following number of lines:

- SPA922: One line
- WIP310: One line
- SPA525G: Five lines
- SPA942: Four lines
- SPA962: Six lines
- SPA932 attendant console: 32 lines

Each line corresponds to a phone number (or extension) used for calls.



NOTE: Each line can support two calls. So, for example, the SPA942 can handle eight calls. One call can be active (in conversation) and seven can be on hold.

Shared Line Appearances

Shared Line Appearance (SLA) allows multiple phones to share an extension number and manage a call as a group. Cisco IP phones support SLA in association with a BroadSoft application server.

At any given time, each station sharing a call appearance can monitor the state of the call appearance. A station can select a shared call appearance to make a call only if the call appearance is not being used by another station.

When a call is made to the extension number for the SLA, all sharing stations ring. Any station can answer the call. If the active phone places the call on hold, the call can be resumed from any of the sharing stations by pressing the corresponding line key from another phone (SPA9X2) or the **Select** button when the **Resume** icon is displayed (WIP310).



NOTE

The Cisco SPA525G supports the “private hold” feature for MetaSwitch and BroadSoft. Users who have a shared line can press the “**PrivHold**” softkey, and the call can only be resumed by the user who placed the call on hold.

Each station with an SLA can be configured independently. Although the account information is usually the same for all of the stations, settings such as the dial plan or the preferred codec can vary.

Configuring a Line



NOTE

This section does not apply to the WIP310.

To configure a phone line:

- STEP 1** Log in to the web administration interface.
- STEP 2** Click **Admin Login** and **advanced**.
- STEP 3** Click the **Phone** tab.

STEP 4 Under each line key for the phone, configure the following:

- **Extension**—Assign an extension to the line key:
 - SPA922: 1 or disabled.
 - SPA942: 1, 2, 3, 4, or disabled.
 - SPA962: 1, 2, 3, 4, 5, 6, or disabled.
 - SPA525G: 1, 2, 3, 4, 5, or disabled.

Defaults to 1. Generally you should reserve **EXT 1** on the client station as the primary and private extension of the designated user. Configure shared extensions on **EXTs 2 to 6**.

- **Short Name**—Enter a short name or number to display on the LCD for the line key.
- **Share Call Appearance**—Select **shared** if you want the line key to share incoming call appearances with other phones. See [Configuring Shared Line Appearance, page 36](#). If you select **private**, the call appearance is private and not shared with any other phone. Defaults to private.
- **Extended Function**—See [Assigning Busy Lamp Field, Call Pickup, and Speed Dial Functions to Unused Lines on a SPA525G, SPA962 or SPA942, page 37](#). This feature is currently not available on the SPA525G.



NOTE The number of line keys displayed depends on the type of phone.

STEP 5 Click **Submit All Changes**.

Configuring Shared Line Appearance

After configuring the line and choosing **shared** in the Shared Call Appearance field, perform the following steps:

-
- STEP 1** Click the **Ext <number>** tab of the extension that is shared (do not use Ext 1).
- STEP 2** Under **Share Line Appearance**, in the **Share Ext** field, select **shared**. If you set this extension to private (not shared), the extension does not share calls, regardless of the **Share Call Appearance** setting on the **Phone** tab. If you set this extension to **shared**, calls follow the **Share Call Appearance** setting on the **Phone** tab. A shared line shows two telephone icons next to the line button to indicate that it is shared (SPA9X2).
- STEP 3** In the **Shared User ID** field, enter the user ID (name) of the phone with the extension that is being shared.
- STEP 4** (Optional) In the **Subscription Expires** field, enter the number of seconds before the SIP proxy expires. After proxy expiration, the phone polls the SIP server to determine the status of the shared phone extension. The default is 60 seconds.
- STEP 5** (Optional) In the **Phone** tab, under **Miscellaneous Line Settings**, you can configure line mapping. Each LED (line/extension) can hold two calls. You can assign an extension to two LEDs. The first call always causes the assigned LED to flash. Choose one of the following:
- Vertical first—The next LED on the phone flashes with the second incoming call.
 - Horizontal first—The same LED to flashes with the second incoming call.
- STEP 6** (Optional) Under **SCA Barge-In Enable**, choose **yes** to allow users sharing call appearances to take over the call on a shared line.

For example, Bob and Chris share the extension 401. A caller, Adam, calls extension 401. Bob answers the call. Adam and Bob are connected. If Chris has the SCA Barge-In Enable field on her phone set to **yes**, she can press the line button for extension 401. Then Chris and Adam are connected in a call and Bob is dropped from the call.

**NOTE**

The Cisco SPA525G supports the “private hold” feature for MetaSwitch and Broadsoft. Users who have a shared line can press the “**PrivHold**” softkey, and the call can only be resumed by the user who placed the call on hold. No barge-in can be performed on these calls.

STEP 7 Click **Submit All Changes**.

Assigning Busy Lamp Field, Call Pickup, and Speed Dial Functions to Unused Lines on a SPA525G, SPA962 or SPA942

You can configure unused or idle lines on a SPA525G, SPA962 or SPA942 to interact with another phone line in the system. For example, if you have two idle lines on an assistant's phone, you can configure those lines to show the status of a supervisor's phone (Busy Lamp Field, or BLF). You can also configure the idle lines so that they can be used to speed dial the supervisor's phone, or pick up calls that are ringing on the supervisor's phone.

**NOTE**

A monitored extension must be private, not shared. Additionally, an extension can only be monitored by one other extension.

Configuring Call Pickup and Busy Lamp Field

**NOTE**

You must enable BLF to configure call pickup.

In this example, the assistant Bob (extension 200) has an idle lines (line 4) on his SPA962. He would like to be able to see if his supervisor Stephanie (extension 300) is on the phone, and pick up calls that are ringing at her extension.

To configure this feature for Bob's SPA962:

STEP 1 Log in to the web administration interface.

STEP 2 Click **Admin Login**.

STEP 3 Click **advanced**.

STEP 4 Click the **Phone** tab.

STEP 5 Find the first line to configure (line 4 in this example):

- a. From the *Extension* drop down list, choose **Disabled**.
- b. From the *Share Call Appearance* drop-down list, choose **private**.

- c. Enter the following string in the Extended Function field:

```
fnc=blf+cp;sub=Stephanie@$PROXY;ext=300@$PROXY
```

Using the following syntax:

```
fnc=type;sub=stationname@$PROXY;ext=extension#@$PROXY
```

where:

- fnc: function
- blf: busy lamp field
- cp: call pickup
- sub: station name
- ext: extension

- STEP 6** Click **Submit All Changes**. After the phone reboots, the SPA962 (in this example) should show the following color LEDs for the monitored lines:

- Green: Available
- Red: Busy
- Red Fast Blink: Ringing

If the phone LEDs display orange or slow blinking orange, there is a problem: Orange denotes that the phone failed to subscribe (received 4xx response) and slow blinking orange denotes an undefined problem (there may be no response to subscribe, or BLF).

In this example, after this configuration, Bob will be able to monitor Stephanie's line. He can press line button 4 to pick up a call ringing at Stephanie's line.

Configuring Speed Dial

In this example, the assistant Bob (extension 200) has another idle line (line 5) on his SPA962. He would like to be able to speed dial his supervisor Mark (extension 400) from that line.

To configure this feature for Bob's SPA962:

- STEP 1** Log in to the web administration interface.

- STEP 2** Click **Admin Login**.

STEP 3 Click **advanced**.

STEP 4 Click the **Phone** tab.

STEP 5 Find the first line to configure (line 5 in this example):

- a. From the *Extension* drop down list, choose **Disabled**.
- b. From the *Share Call Appearance* drop-down list, choose **private**.
- c. Enter the following string in the Extended Function field:

```
fnc=sd;sub=Mark@$PROXY;ext=400@$PROXY
```

Using the following syntax:

```
fnc=type;sub=stationname@$PROXY;ext=extension#@$PROXY
```

where:

- fnc: function
- sd: speed dial
- sub: station name
- ext: extension

STEP 6 Click **Submit All Changes**.

In this example, after this configuration, Bob can press line button 5 to dial Mark's line.

Configuring Unused Line Keys for Call Park on the SPA525G (MetaSwitch)

You can configure unused line keys for call park (for the MetaSwitch softswitch) on the SPA525G. Users can then press this line button to park a call or retrieve a parked call. To configure:

STEP 1 Log in to the web administration interface.

STEP 2 Click **Admin Login**.

STEP 3 Click **advanced**.

STEP 4 Click the **Phone** tab.

STEP 5 Find the first line to configure (line 5 in this example):

- a. From the *Extension* drop down list, choose **Disabled**.
- b. From the *Share Call Appearance* drop-down list, choose **private**.
- c. Enter the following string in the Extended Function field:

```
fnc=prk;sub=05@domain.com
```

where:

- fnc: function
- prk: call park
- sub: call park orbit, or location where the call is parked. Value is 05 (specific to the MetaSwitch softswitch)—do not change the “sub=05” value.
- domain.com: phone domain, usually the same as the “proxy” value in the Ext 1 tab. You can also use fnc=prk;sub=05@\$PROXY to use this value.

STEP 6 Click **Submit All Changes**.

Configuring Unused Line Keys to Access Services (SPA525G)

On the SPA525G, unused or idle phone lines can also be configured to access services, such as the following:

- XML services
- MP3 player
- Weather (RSS)
- News (RSS)

To configure line keys to access services:

STEP 1 Log in to the web administration interface.

STEP 2 Click **Admin Login**.

STEP 3 Click **advanced**.

STEP 4 Click the **Phone** tab.

STEP 5 Find the first line to configure (line 4 in this example):

- a. From the *Extension* drop down list, choose **Disabled**.
- b. Enter the following string in the Extended Function field:

```
fnc=type
```

where:

- fnc: function
- type: choose from the following:
 - xml: pressing the line button accesses XML services.



NOTE The XML service configured on the Phone tab under the XML Service field is used (see the **“Configuring XML Services (SPA525G)” section on page 82**). You can specify a different XML service to connect to by using the syntax “fnc=xml;URL=http://xxx.xx.xxx” where xxx.xx.xxx is the URL of the XML service.

- mp3: pressing the line button starts the mp3 player.
- weather: pressing the line button accesses weather information.
- news: pressing the line button accesses news.

For example, to configure line 4 for the mp3 player:

```
fnc=mp3
```

STEP 6 Click **Submit All Changes**. After the phone reboots, configured lines glow orange and display the following icons next to the extension label:

- xml: XML icon
- mp3: mp3 player icon
- news: RSS icon
- weather: thermometer icon

Configuring Line Key LED Patterns (SPA9X2/525G only)

You can customize the LED patterns for the line keys on the phone by entering letters for the color or pattern in the LED pattern fields.

To configure Line Key LED patterns:

-
- STEP 1** Log in to the web administration interface.
- STEP 2** Click **Admin Login** and **advanced**.
- STEP 3** Click the **Phone** tab.
- STEP 4** Under **Line Key LED Pattern**, use the following letters to customize the fields shown in the following table:
- “p” indicates “pattern”: the blinking pattern of the LED
 - “c” indicates “color”: the color of the LED
 - “r” indicates “red”: a red-colored LED
 - “g” stands for “green”: a green-colored LED

Parameters	Description
Idle LED	Appears when the line is idle. Defaults to blank (c=r).
Remote Undefined LED	LED pattern during the Remote Undefined state, where the shared call state is undefined (the station is still waiting for the state information from the application server). Not applicable if the call appearance is not shared. Leaving this entry blank indicates the default value of c=r;p=d.
Local Seized LED	Appears when this station seizes the call appearance to prepare for a new outbound call. Defaults to blank (c=r).
Remote Seized LED (applicable only to shared call appearance)	Appears when the shared call appearance is seized by another station. Defaults to blank (c=r; p=d).

Parameters	Description
Local Progressing LED	Appears when this station attempts an outgoing call on this call appearance (the called number is ringing). Defaults to blank (c=r).
Remote Progressing LED (applicable only to shared call appearance)	Appears when another station attempts an outbound call on this shared call appearance. Defaults to blank (c=r; p=d).
Local Ringing LED	Appears when the call appearance is ringing. Defaults to blank (c=r;p=f).
Remote Ringing LED (applicable only to shared call appearance)	Appears when the shared call appearance is in ringing on another station. Defaults to blank (c=r;p=d).
Local Active LED	Appears when the call appearance is engaged in an active call. Defaults to blank (c=r).
Remote Active LED (applicable only to shared call appearance)	Appears when another station is engaged in an active call on this shared call appearance. Defaults to blank (c=r;p=d).
Local Held LED	Appears when the call appearance is held by this station. Defaults to blank (c=r;p=s).
Remote Held LED (applicable only to shared call appearance)	Appears when another station places this call appearance on hold. Defaults to blank (c=4,p=s).
Register Failed LED	LED pattern when the corresponding extension has failed to register with the proxy server. Leaving this entry blank indicates the default value of c=a.
Disabled LED	LED pattern when the Call Appearance is disabled (not available for any incoming or outgoing call). Leaving this entry blank indicates the default value of c=o.

Parameters	Description
Registering LED	Appears when the corresponding extension tries to register with the proxy server. Defaults to blanks (c=r;p=s).
Call Back Active LED	Indicates Call Back operation is currently active on this call. Defaults to blank (c=r;p=s).

STEP 5 Click **Submit All Changes**.

For more information on LEDs, see the [“Creating an LED Script”](#) section on page 187.

Configuring Extensions

STEP 1 Log in to the web administration interface.

STEP 2 Click **Admin Login** and **advanced**.

STEP 3 Click the **Ext <number>** tab for the extension you want to configure.

STEP 4 In the **General** section, make sure that **Line Enable** is set to **yes**.

You can configure many parameters differently for different extensions. These parameters are grouped on the **Ext <number>** tab. These parameters are explained in other sections of this document:

- NAT Settings—[Chapter 5, “Configuring SIP, SPCP, and NAT.”](#)
- Network Settings—[Chapter 5, “Configuring SIP, SPCP, and NAT.”](#)
- SIP Settings—[Chapter 5, “Configuring SIP, SPCP, and NAT.”](#)
- Call Feature Settings—[Chapter 4, “Customizing Cisco SPA and Wireless IP Phones.”](#)
- Proxy and Registration—[Chapter 5, “Configuring SIP, SPCP, and NAT.”](#)
- Subscriber Information—[Chapter 5, “Configuring SIP, SPCP, and NAT.”](#)

- Audio (Codec) Configuration—**Chapter 6, “Configuring Security, Quality, and Network Features.”**
- Dial Plan—**Chapter 4, “Customizing Cisco SPA and Wireless IP Phones.”**

Customizing Cisco SPA and Wireless IP Phones

This chapter describes customizing the SPA and Wireless IP phones and contains the following sections:

- [Configuring Phone Information and Display Settings, page 47](#)
- [Enabling Call Features, page 56](#)
- [Configuring Phone Features, page 65](#)
- [Enabling SMS Messaging, page 74](#)
- [Enabling the Web Server, page 76](#)
- [Configuring Lightweight Directory Access Protocol \(LDAP\), page 77](#)
- [Configuring BroadSoft Settings \(SPA525G\), page 81](#)
- [Configuring Music On Hold, page 83](#)
- [Configuring Extension Mobility with a BroadSoft Server, page 84](#)

Configuring Phone Information and Display Settings

The web administration interface allows you to customize the phone and configure settings such as the phone name, background photo, logo, and screen saver.

Configuring the Phone Name

-
- STEP 1** Log in to the web administration interface.
 - STEP 2** Click **Admin Login** and **advanced**.
 - STEP 3** Click the **Phone** tab.
 - STEP 4** Under **General**, enter the Station Name, or name for the phone. This name shows up in the corporate directory.
 - STEP 5** Click **Submit All Changes**. The phone reboots.
-

Configuring Voice Mail

To configure the phone to connect to voice mail:

-
- STEP 1** Log in to the web administration interface.
 - STEP 2** Click **Admin Login** and **advanced**.
 - STEP 3** Click the **Phone** tab.
 - STEP 4** Under **General**, enter the Voice Mail Number. This is the internal or external phone number or URL to access the voice mail system. If using an external voice-mail service, the number must include any digits required to dial out and any required area code.
 - STEP 5** Click **Submit All Changes**. The phone reboots.
-

Configuring Internal Voice Mail for Each Extension (When Using SPA400 for Voice Mail)

-
- STEP 1** Log in to the web administration interface.
 - STEP 2** Click **Admin Login** and **advanced**.
 - STEP 3** Click the **Ext <number>** tab.
 - STEP 4** Under **Call Feature Settings**, enter the voice mail line number and phone extension in the **Mailbox ID** field. For example, 2101 indicates that the SPA400 voice mail server is configured on the SPA9000 Line 2, phone extension 101.

STEP 5 Enter the IP address of the voice mail server.

STEP 6 Click **Submit All Changes**.

Customizing the Startup Screen

You can create a text logo to display when the IP phone boots up. (Not applicable to the WIP310.)

SPA9X2:

STEP 1 Log in to the web administration interface.

STEP 2 Click **Admin Login** and **advanced**.

STEP 3 Click the **Phone** tab. In the Text Logo field, enter logo text as follows:

- Up to two lines of text
- Each line must be fewer than 32 characters
- Insert a new line character (\n) and escape code (%0a) between lines

For example, “Super\n%0aTelecom” will display:

```
Super
Telecom
```

STEP 4 Click **Submit All Changes**. The phone reboots.

SPA525G:

STEP 1 Log in to the web administration interface.

STEP 2 **Admin Login** and **advanced**.

STEP 3 Click the **User** tab. In the Screen section, Text Logo field, enter logo text as follows:

- Up to two lines of text
- Each line must be fewer than 32 characters
- Insert a new line character (\n) and escape code (%0a) between lines

For example, "Super\n%0aTelecom" will display:

Super

Telecom

STEP 4 In the Logo Type field, select **Text Logo**.

STEP 5 Click **Submit All Changes**. The phone reboots.

Changing the Display Background (SPA942/962/525G)

You can use a logo and picture to customize the background on your IP phone LCD displays:

- SPA942: Use 128 x 48 black and white image (1-bit color).
- SPA962/525G: Use 320 x 240 pixel bitmap (.bmp) image (1, 4, 8, 24, or 32-bit color).



NOTE: The phone does not reboot after you change the background image URL.

SPA942/962:

STEP 1 Copy the .BMP image to a TFTP or HTTP server that is accessible from the phone.

STEP 2 Log in to the web administration interface.

STEP 3 Click **Admin Login** and **advanced**.

STEP 4 Click the **Phone** tab.

STEP 5 In the Select Background Picture field, select **BMP Picture**.

- STEP 6** Enter the URL of the .bmp file you want in the *BMP Picture Download URL* field. The URL must include the TFTP/HTTP server name (or IP address), directory, and filename, for example:

```
tftp://myserver.mydomain.com/images/  
downloadablebmppicture.bmp
```

or

```
http://myserver.mydomain.com/images/  
downloadablebmppicture.bmp
```

If the HTTP Refresh Timer is set in the server's response to **BMP Picture Download URL**, the phone downloads the picture from the link and displays it on the screen. The phone automatically retrieves the picture after the specified number of seconds.

- STEP 7** Click **Submit All Changes**.

When the BMP Picture Download URL is changed, the phone compares the URL to the previous image's URL. (If the URLs are the same, the phone does not perform the download.) If the URLs are different, the SPA9X2 downloads the new image and displays it (providing the *Select Background Picture* field is set to **BMP Picture**).

SPA525G:

-
- STEP 1** Copy the .BMP image to a TFTP or HTTP server that is accessible from the phone.
- STEP 2** Log in to the web administration interface.
- STEP 3** Click **Admin Login** and **advanced**.
- STEP 4** Click the **User** tab.
- STEP 5** In the Screen section, Background Picture Type field, select **Download BMP Picture**.

- STEP 6** Enter the URL of the .bmp file you want in the *BMP Picture Download URL* field. The URL must include the TFTP/HTTP server name (or IP address), directory, and filename, for example:

```
tftp://myserver.mydomain.com/images/  
downloadablebmppicture.bmp
```

or

```
http://myserver.mydomain.com/images/  
downloadablebmppicture.bmp
```

If the HTTP Refresh Timer is set in the server's response to **BMP Picture Download URL**, the phone downloads the picture from the link and displays it on the screen. The phone automatically retrieves the picture after the specified number of seconds.

- STEP 7** Click **Submit All Changes**.

When the BMP Picture Download URL is changed, the phone compares the URL to the previous image's URL. (If the URLs are the same, the phone does not perform the download.) If the URLs are different, the SPA525G downloads the new image and displays it (providing the *Select Background Picture* field is set to **Download BMP Picture**).

Configuring the Screen Saver

You can configure a screen saver for the SPA9X2 or SPA525G phone. (Not applicable to WIP310.)

This option enables a screen saver on the phone's LCD. When the phone is idle for a specified time, it enters screen saver mode. (Users can set up screen savers directly using phone's **Setup** button.)

Any button press or on/off hook event triggers the phone to return to its normal mode. If a user password is set, the user must enter it to exit screen saver mode.

To configure the screen saver:

SPA9X2

-
- STEP 1** Log in to the web administration interface.
- STEP 2** Click **Admin Login** and **advanced**.

STEP 3 Click the **Phone** tab.

STEP 4 In the **General** section, in the **Screen Saver Enable** field, choose **yes**.

STEP 5 In the **Screen Saver Wait** field, enter the number of seconds of idle time to elapse before the screen saver starts.

STEP 6 In the **Screen Saver Icon** field, choose the display type:

- A background picture.
- The station time in the middle of the screen.
- A moving padlock icon. When the phone is locked, the status line displays a scrolling message “Press any key to unlock your phone.”
- A moving phone icon.
- The station date and time in the middle of the screen.
- A blank “power save” screen.”

STEP 7 Click **Submit All Changes**.

SPA525G

STEP 1 Log in to the web administration interface.

STEP 2 Click **Admin Login** and **advanced**.

STEP 3 Click the **User** tab.

STEP 4 Under **Screen**, in the **Screen Saver Enable** field, choose **yes**.

STEP 5 In the **Screen Saver Type** field, choose the display type:

- **Black Background**—Displays a black screen.
- **Gray Background**—Displays a gray screen.
- **Black/Gray Rotation**—The screen incrementally cycles from black to gray.
- **Picture Rotation**—The screen rotates through available pictures on the phone.
- **Digital Frame**—Shows the background picture.

STEP 6 In the **Screen Saver Trigger Time** field, enter the number of seconds that the phone remains idle before the screen saver turns on.

- STEP 7** In the **Screen Saver Refresh Time** field, enter the number of seconds before the screen saver should refresh (if, for example, you chose a rotation of pictures).
- STEP 8** Click **Submit All Changes**.
-

Configuring the Phone Display Menu Color Settings (SPA962 Only)

To configure the phone display menu color settings:

- STEP 1** Log in to the web administration interface.
- STEP 2** Click **Admin Login** and **advanced**.
- STEP 3** Click the **User** tab.
- STEP 4** Under **Phone GUI Menu Color Settings**, select the color scheme:
- Light Blue (default)
 - Light Green
 - Pink
 - Silver
- STEP 5** Click **Submit All Changes**.
-

Configuring the LCD Contrast

You can configure the LCD contrast on the SPA9X2 or SPA525G phone. (Not applicable to the WIP310.) To configure the contrast for the LCD screen on the phone:

SPA9X2

- STEP 1** Log in to the web administration interface.
- STEP 2** Click **Admin Login** and **advanced**.
- STEP 3** Click the **User** tab.

STEP 4 Under **Audio Volume**, in the **LCD Contrast** field, enter a number value from 1 to 30. The higher the number, the greater the contrast on the screen.

STEP 5 Click **Submit All Changes**.

SPA525G

STEP 1 Log in to the web administration interface.

STEP 2 Click **Admin Login** and **advanced**.

STEP 3 Click the **User** tab.

STEP 4 Under **Screen**, in the **LCD Contrast** field, enter a number value from 1 to 30. The higher the number, the greater the contrast on the screen.

STEP 5 Click **Submit All Changes**.

Configuring Back Light Settings (SPA525G)

To configure the back light settings for the LCD screen on the phone:

STEP 1 Log in to the web administration interface.

STEP 2 Click **Admin Login** and **advanced**.

STEP 3 Click the **User** tab.

STEP 4 Under **Screen**, in the **Back Light Enable** field, choose **yes** to enable the screen back light.

STEP 5 In the **Back Light Timer** field, enter the number of seconds of idle time that can elapse before the back light turns off.

STEP 6 Click **Submit All Changes**.

Configuring Linksys Key System Parameters

To configure the phone as part of a Linksys Key System (for use with the SPA9000):

-
- STEP 1** Log in to the web administration interface.
- STEP 2** Click **Admin Login** and **advanced**.
- STEP 3** Click the **SIP** tab.
- STEP 4** Configure the following fields:
- **Linksys Key System**—Enables or disables the Linksys Key System for use with the LVS SPA9000. Defaults to yes. See the *Cisco SPA9000 System Administration Guide* for more details.
 - **Multicast Address**—Used by the SPA9000 to communicate with Cisco IP phones. Defaults to 224.168.168.6061.
 - **Force LAN Codec**—Used with the SPA9000. Choices are none, G.711u, or G.711a. Defaults to none.
- STEP 5** Click **Submit All Changes**.
-

Enabling Call Features

This section describes how to enable and disable call features on the phone.

Enabling Anonymous Call and Caller ID Blocking

-
- STEP 1** Log in to the web administration interface.
- STEP 2** Click **Admin Login** and **advanced**.
- STEP 3** Click the **Phone** tab.

STEP 4 Under **Supplementary Services**, under the type of call blocking to enable, choose **yes**.

- **Block ANC**—Blocks anonymous calls.
- **Block CID**—Blocks outbound caller ID.

**NOTE**

These features can also be configured from the **User** tab, under **Supplementary Services**.

STEP 5 Click **Submit All Changes**.

Enabling Automatic Call Distribution (ACD)

Typically used for call centers, Automatic Call Distribution (ACD) handles incoming calls and manages them based on a database of instructions. You can enable this with the SIP B parameter ([“Configuring SIP” section on page 89](#)).

Defaults to no (disabled).

STEP 1 Log in to the web administration interface.

STEP 2 Click **Admin Login** and **advanced**.

STEP 3 Click the **Phone** tab.

STEP 4 Under **Supplementary Services**, in the **ACD Login Serv** field, choose **yes**.

STEP 5 In the **ACD Ext** field, choose the extension used for handling ACD calls. Select 1-6, depending on your phone model. Defaults to 1.

STEP 6 Click **Submit All Changes**.

Enabling Call Back

Call back is a feature that forces the phone to repeatedly try a number that has been dialed and received a busy response. The busy number is tried until the call goes through and the phone rings on the user's end.

To enable call back:

-
- STEP 1** Log in to the web administration interface.
 - STEP 2** Click **Admin Login** and **advanced**.
 - STEP 3** Click the **Phone** tab.
 - STEP 4** In the **Phone** tab, under **Supplementary Services**, in the **Call Back Serv** field, choose **yes**.
 - STEP 5** Click **Submit All Changes**.
-

Enabling Call Park and Call Pickup

Call park and call pickup are features available on IP phones in a SPA9000 system. Call park allows users to put a call on a line and make it available for another user to pick up. Call pickup allows a user to pick up a phone that is ringing at another user's phone.

-
- STEP 1** Log in to the web administration interface.
 - STEP 2** Click **Admin Login** and **advanced**.
 - STEP 3** Click the **Phone** tab.
 - STEP 4** Under **Supplementary Services**, under the type of call feature to enable, choose **yes**.
 - **Call Park Serv**—Enables call parking.
 - **Call Pickup Serv**—Enables call pickup.
 - STEP 5** Click **Submit All Changes**.
-

Enabling Call Transfer and Call Forwarding

-
- STEP 1** Log in to the web administration interface.
- STEP 2** Click **Admin Login** and **advanced**.
- STEP 3** Click the **Phone** tab.
- STEP 4** Under **Supplementary Services**, under the transfer type you want to enable, choose **yes**:
- **Attn Transfer**—Attended call transfer. The user answers the call before transferring it.
 - **Blind Transfer**—Blind call transfer. The user transfers the call without speaking to the caller.
- You can also enable or disable call forwarding:
- **Cfwd All**—Forwards all calls.
 - **Cfwd Busy**—Forwards calls only if the line is busy.
 - **Cfwd No Ans**—Forwards calls only if the line is not answered.
- STEP 5** Click **Submit All Changes**.
-

Enabling Call Waiting

To enable call waiting:

-
- STEP 1** Log in to the web administration interface.
- STEP 2** Click **Admin Login** and **advanced**.
- STEP 3** Click the **Phone** tab.
- STEP 4** Under **Supplementary Services**, in the CW Setting field, choose **yes**.
- STEP 5** Click **Submit All Changes**.
-

Enabling Conferencing

To allow the user to perform call conferencing on the phone:

-
- STEP 1** Log in to the web administration interface.
 - STEP 2** Click **Admin Login** and **advanced**.
 - STEP 3** Click the **Phone** tab.
 - STEP 4** Under **Supplementary Services**, in the **Conference Serv** field, choose **yes**.
 - STEP 5** Click **Submit All Changes**.
-

Enabling Dial Assistance

Dial assistance can help users to place calls more quickly. When a user begins dialing, the phone displays a list of closely-matched phone numbers on the screen.

To enable dial assistance:

-
- STEP 1** Click **Admin Login** and **advanced**.
 - STEP 2** Click the **User** tab.
 - STEP 3** Under **Supplementary Services**, in the **Dial Assistance** field, choose **yes**.
 - STEP 4** Click **Submit All Changes**.
-

Enabling Do Not Disturb

You can allow users to turn the Do Not Disturb feature on and off. This feature directs all incoming calls to voice mail or, if voice mail is not configured, plays a message to the caller saying the user is unavailable.



NOTE

On the SPA525G, users can press the **Ignore** softkey to divert a ringing call to the forwarded destination.

To allow users to use Do Not Disturb:

-
- STEP 1** Log in to the web administration interface.
 - STEP 2** Click **Admin Login** and **advanced**.
 - STEP 3** Click the **Phone** tab.
 - STEP 4** Under **Supplementary Services**, under **DND Serv**, choose **yes**.



NOTE

This feature can also be configured from the **User** tab, under **Supplementary Services**.

- STEP 5** Click **Submit All Changes**.
-

Enabling the Missed Call Shortcut

The IP phones can display a notification that a call has been missed. (Not applicable to WIP310.) To enable this notification:

-
- STEP 1** Log in to the web administration interface.
 - STEP 2** Click **Admin Login** and **advanced**.
 - STEP 3** Click the **User** tab.
 - STEP 4** Under **Supplementary Services**, in the **Miss Call Shortcut** field, choose **yes**.
 - STEP 5** Click **Submit All Changes**.
-

Logging Missed Calls (SPA9X2/SPA525G Only)

You may want to disable or enable missed call logging per extension. For example, if you have set up a line to monitor another user's line, you may want to disable missed call logging for the monitored line.

-
- STEP 1** Log in to the web administration interface.
 - STEP 2** Click **Admin Login** and **advanced**.
 - STEP 3** Click the **User** tab.
 - STEP 4** Under **Supplementary Services**, in the **Log Missed Calls for EXT** <number> field, choose **yes**.
 - STEP 5** Click **Submit All Changes**.
-

Enabling Paging (Intercom)

The paging, or intercom feature, allows two types of paging:

Single Page

A user can directly contact another user by phone. If the person being paged has configured their phone to automatically accept pages (see [Configuring a Phone to Automatically Accept Pages, page 63](#)), the phone does not ring and a direct connection between the two phones is automatically established when paging is initiated.

Group Paging (Limited to specific models; see below)

Group Paging lets the user page all the client stations at once (SPA9X2/SPA525G), or page groups of phones (SPA525G). If the client station is on an active call while a group page starts, the incoming page is ignored.

When paging occurs, the speaker on the paged stations is automatically powered on unless the handset or headset is being used.

Group page is one-way only. The paged client stations can only listen to the call from the originator.

To enable paging:

-
- STEP 1** Log in to the web administration interface.
 - STEP 2** Click **Admin Login** and **advanced**.
 - STEP 3** Click the **Phone** tab.
 - STEP 4** Under **Supplementary Services**, under **Paging Serv**, choose **yes**.

STEP 5 Click **Submit All Changes**.

Configuring a Phone to Automatically Accept Pages

To configure a phone to automatically accept pages:

STEP 1 Log in to the web administration interface.

STEP 2 Click **Admin Login** and **advanced**.

STEP 3 Click the **User** tab.

STEP 4 Under **Supplementary Services**, in the **Auto Answer Page** field, choose **yes**.

STEP 5 Click **Submit All Changes**. The phone reboots.

Configuring Paging Groups

You can configure a phone as part of a paging group. Users can then direct pages to specific groups of phones.

Limitations:

- A phone can be part of no more than two paging groups
- A user can page a maximum of five paging groups

To configure a phone as part of a paging group:

STEP 1 Log in to the web administration interface for the phone.

STEP 2 Click **Admin Login** and **advanced**.

STEP 3 Click the **Phone** tab.

STEP 4 Under Multiple Paging Group Parameters, enter the paging commands into the Group Paging Script box. The syntax is as follows:

```
pggrp=multicast-address:port;[name=xxx;]num=xxx;[listen={yes|no}]];
```

Where:

- IP address: Multicast IP address of the phone that will listen for and receive pages.
- port: Port on which to page; you must use different ports for each paging group.
- name (optional): The name of the paging group.
- num: The number users will dial to access the paging group; must be unique to the group.
- listen: If the phone should listen on the page group. Only the first two groups with listen set to yes will listen to group pages. If the field is not defined, the default value is no, so you must set this field to listen to the group pages.

For example:

```
pggrp=224.123.123.121:43210;name=All; num=801;listen=yes;  
pggrp=224.123.123.121:43211;name=Sales;num=802; listen=yes;  
pggrp=224.123.123.121:43212;name=Support;num=803;  
pggrp=224.123.123.121:43213;name=Engineering;num=804;
```

The previous example creates four paging groups: *All*, *Sales*, *Support*, and *Engineering*. Users will press 801 to send pages to all phones, 802 to send pages to phones configured as part of the sales group, 803 to send pages to phones configured as part of the Support group, and 804 to send pages to phones configured as part of the Engineering group. In this example, the configured phone will receive pages directed to the “All” and “Sales” groups.

STEP 5 Click **Submit All Changes**.

Enabling Secure Call

See [Setting Security Features, page 115](#).

Enabling Service Announcements

The Service Announcements features allows a user to send announcement requests to a customer-supplied announcement server. (Not applicable to the WIP310.)

-
- STEP 1** Log in to the web administration interface.
 - STEP 2** Click **Admin Login** and **advanced**.
 - STEP 3** Click the **Phone** tab.
 - STEP 4** Under **Service Annc Serv**, choose **yes**.
 - STEP 5** Click **Submit All Changes**.
-

Configuring Phone Features

The following sections describe how to configure features on the phone such as the message waiting indicator, ring tones, and audio features.

Configuring the Message Waiting Indicator

You can configure the message waiting indicator for separate extensions on the phone. If the indicator at the top of your SPA9X2 is not lighting when voice mail is left, or you are not seeing message waiting notifications on your WIP310:

-
- STEP 1** Log in to the web administration interface.
 - STEP 2** Click **Admin Login** and **advanced**.
 - STEP 3** Click the **Ext <number>** tab.
 - STEP 4** Under **Call Feature Settings**, in the **Message Waiting** field, choose **yes**.
 - STEP 5** Click **Submit All Changes**.
-

Configuring Ring Tones

You can define up to ten ring tones for a SPA9X2 or SPA525G phone.

**NOTE**

WIP310 ring tones are not configurable from the web administration interface.

You can define:

- The default ring tone for the extension
- Specific ring tones assigned to individual callers in the personal directory. These override the default ring tone.

To configure ring tones:

-
- STEP 1** Log in to the web administration interface.
- STEP 2** Click **Admin Login** and **advanced**.
- STEP 3** Click the **Phone** tab and proceed to the **Ring Tone** section.

Ring Tone	Value
Ring1:	n=Classic-1;w=3;c=1
Ring2:	n=Classic-2;w=3;c=2
Ring3:	n=Classic-3;w=3;c=3
Ring4:	n=Classic-4;w=3;c=4
Ring5:	n=Simple-1;w=2;c=1
Ring6:	n=Simple-2;w=2;c=2
Ring7:	n=Simple-3;w=2;c=3
Ring8:	n=Simple-4;w=2;c=4
Ring9:	n=Simple-5;w=2;c=5
Ring10:	n=Office;w=4;c=1

Audio Input Gain (dB)
Handset Input Gain: 0

- STEP 4** After configuration, click **Submit All Changes**.

Managing Ring Tones

Ten ring tones appear by default.

You can configure the characteristics of each ring tone using a Ring Tone script. In a Ring Tone script, you can assign a name for the ring tone, and specify:

- Name (n)—Ring tone name, such as Classic, Simple, and Office
- Waveform (w)—1, 2, 3, or 4
- Cadence (c)—1, 2, 3, 4, or 5

You can also download one of two available ring tones (user ring tone 1 or 2) using TFTP:

```
http://phone_ip_addr/ringtone1?[url]
```

- Where the URL syntax is `tftp://host[:port]/path`.
- The default host is the TFTP host.
- Port is optional. The default port is 69.
- The link is case sensitive.

On the SPA phones, user-downloaded ring tones are labeled User 1 and User 2 in the choices for the Default Ring. On the phone ring tone menu, the User 1 and 2 choices are replaced by the corresponding name of the ring tone. “Not Installed” appears if the user ring tone slots are not used.

For ring tone User 1 and User2, the cadence is fixed with the on-time equals to the duration of the ring tone file and off-time equals to four seconds. The total ring duration is fixed at 60 seconds. The user ring tone names displayed on the phone LCD are derived from the ring tone file header file.

The phone does not require rebooting after downloading a ring tone.

To remove the User 1 ring tone from the phone, set the *path* to delete, as follows:

```
http://phone_ip_addr/ringtone1/delete
```

Configuring On-Demand Ring Tones (SPA525G)

The SPA525G supports on-demand ring tones, which means that ring tones are downloaded and played from a TFTP server when a call comes in. To configure:

STEP 1 Log in to the web administration interface.

STEP 2 Click **Admin Login** and **advanced**.

STEP 3 Click the **Phone** tab.

STEP 4 Under **Ring Tone**, in one or more of the ten ring tone fields, enter the following:

```
n=office;w=[tftp://]host[:port]/path;c=0
```

and specify the URL to download in the host/port/path field. If the connection cannot be established, a default ring tone is played.

STEP 5 Click **Submit All Changes**.

User-Created MP3 Ring Tones (SPA525G)

SPA525G users can create up to two ring tones from an MP3 audio file stored on a USB memory device. For instructions, see the *Cisco Small Business Pro SPA525G User Guide (SIP)*, located on Cisco.com. (See **Appendix C, “Where to Go From Here,”** for the location of this document.)

Creating and Uploading Ring Tones Using the Ring Tone Utility (SPA9X2 only)

To convert a file for use as a ring tone, use the SPA Ring Tone Utility, available at:

<https://www.myciscocommunity.com/docs/DOC-6672>

You must have a wav file less than 8 seconds in length saved to your computer. You can also use a sound editor to create the file with the following restrictions:

- 16-bit PCM mono
- 8000 samples per second
- less than 6000 ms in length

To create a ring tone and upload it to a phone:

STEP 1 Open the Ring Tone Utility.

STEP 2 Enter the IP address of the phone.

STEP 3 Click **Browse** and navigate to the directory on your computer where the source .wav file is stored. Select the wav file and click **Open**.

STEP 4 Click **Load Source File**.

STEP 5 Enter a name for the ring tone. This name will appear in the display on the phone. You choose the file name later.

-
- STEP 6** Enter the target. You can have up to two customized ring tones uploaded to the phone.
 - STEP 7** (Optional) Click **Preview** to preview the ring tone. Click **Options** to change the start or end positions, or to squeeze or stretch the audio.
 - STEP 8** Click **Upload to Phone** to upload the ring tone to the phone. Click **OK** when the success status message appears.
 - STEP 9** Close the open Ring Tone Utility windows.
-

To create a ring tone and save it to a file:

-
- STEP 1** Open the Ring Tone Utility.
 - STEP 2** Enter the IP address of the user's phone or press **Skip** to create the ring tone and save it as a file.
 - STEP 3** Click **Browse** and navigate to the directory on your computer where the source wav file is stored. Select the wav file and click **Open**.
 - STEP 4** Click **Load**.
 - STEP 5** Enter a name for the ring tone. This name will appear in the phone display. You choose the file name later.
 - STEP 6** (Optional) Click **Preview** to preview the ring tone. Click **Options** to change the start or end positions, or to squeeze or stretch the audio.
 - STEP 7** Click **Save As** to save the file to your computer. Enter the file name and press **Save**.
 - STEP 8** Close the open Ring Tone Utility windows.
-

To delete a ring tone from a phone:

-
- STEP 1** Open the Ring Tone Utility.
 - STEP 2** Enter the IP address of the phone.
 - STEP 3** Click the **Delete** button next to the ring tone you want to delete.
 - STEP 4** Click **OK**.

STEP 5 Close the open Ring Tone Utility windows.

Assigning a Ring Tone to an Extension

STEP 1 Log in to the web administration interface.

STEP 2 Click **Admin Login** and **advanced**.

STEP 3 Click the **Ext** <number> tab.

STEP 4 Under **Call Feature Settings**, in the **Default Ring** field, choose from the following:

- No Ring
- 1 through 10
- User 1
- User 2

STEP 5 Click **Submit All Changes**.

Configuring RSS Newsfeeds on the SPA962/SPA525G Phone

The SPA962 and SPA525G provide the option to view RSS newsfeeds for news in the categories of local, world, finance, sports, and politics. Newsfeeds that are provided by Yahoo are supported for U.S. customers only.

To configure newsfeeds:

STEP 1 Log in to the web administration interface.

STEP 2 Click **Admin Login** and **advanced**.

STEP 3 Click the **User** tab.

STEP 4 Under **Web Information Service Settings**, you can edit the following fields:

Parameter	Description
RSS Feed URLs 1-5	<p>URLs for Local and World news, Finance, Sports, and Politics. Default values are:</p> <ul style="list-style-type: none"> 1—Local News (defaults to URL http://rss.cnn.com/rss/cnn_us.rss) 2—World News (defaults to URL http://newsrss.bbc.co.uk/rss/newsonline_uk_edition/world/rss.xml) 3—Finance News (defaults to URL http://finance.yahoo.com/rss/topstories) 4—Sports News (defaults to URL http://rss.news.yahoo.com/rss/sports) 5—Politics News (defaults to URL http://rss.news.yahoo.com/rss/politics)
Weather Temperature Unit	<p>Choose which unit to display for weather information (Fahrenheit or Celsius).</p> <p>NOTE For the SPA525G only.</p>
Feed Data Update Interval	<p>Interval at which the news feed is updated</p> <p>Defaults to 15 minutes.</p> <p>NOTE Not applicable to the SPA525G.</p>

STEP 5 Click **Submit All Changes**. The phone reboots.

Configuring Traffic Information Settings on the SPA9X2 Phone

You can change the defaults to your local state, city, street, and zip code for local traffic information. (This feature is not available on the SPA525G or WIP310.)



NOTE: Yahoo! Traffic Alerts provides this information to U.S. customers only.

To configure traffic information settings:

-
- STEP 1** Log in to the web administration interface.
- STEP 2** Click **Admin Login** and **advanced**.
- STEP 3** Click the **User** tab.
- STEP 4** Under **Traffic Service Information Settings**, you can change the following fields:
- State
 - City
 - Street
 - Zip Code
- STEP 5** Click **Submit All Changes**. The phone reboots.
-

Configuring Audio Settings

You can configure default audio volume settings for the phone. These settings can be modified by the user by pressing the volume control button on the phone, then pressing the **Save** soft button. (Not applicable to the WIP310.)

To configure the audio volume settings:

-
- STEP 1** Log in to the web administration interface.
- STEP 2** Click **Admin Login** and **advanced**.
- STEP 3** Click the **User** tab. You can configure the following settings:

Parameter	Description
Ringer Volume	Sets the volume for the ringer.
Speaker Volume	Sets the volume for the full-duplex speakerphone.
Handset Volume	Sets the volume for the handset.
Headset Volume	Sets the volume for the headset.

Parameter	Description
Bluetooth Volume	Sets the volume for the Bluetooth device. NOTE Applies to SPA525G only.

STEP 4 Click **Submit All Changes**.

Configuring Audio Input Gain (SPA9X2/SPA525G only)

You can amplify or deamplify the sound on your phone's handset, headset, and speakerphone.

STEP 1 Log in to the web administration interface.

STEP 2 Click **Admin Login** and **advanced**.

STEP 3 Click the **Phone** tab.

STEP 4 Under Audio Input Gain (dB), choose the item to configure.

- If you enter a positive value, amplification increases (sound is louder).
- If you enter a negative value, amplification decreases (sound is softer).
- You can enter a value from -6 decibels to +6 decibels. All fields default to zero.
- Try a value that is loud enough without producing echo (an issue if the input gain is too high).

STEP 5 Click **Submit All Changes**.

Enabling Wireless (SPA525G only)

The SPA525G supports Wireless-G. To enable wireless:

STEP 1 Log in to the web administration interface.

STEP 2 Click **Admin Login** and **advanced**.

STEP 3 Click the **System** tab.

STEP 4 Under **Wi-Fi Settings**, in the **SPA525-wifi-on** field, choose **yes**.

STEP 5 Click **Submit All Changes**.

Enabling Bluetooth (SPA525G only)

The SPA525G supports Bluetooth to allow use of the phone with a wireless Bluetooth-enabled headset. To enable Bluetooth:

STEP 1 Log in to the web administration interface.

STEP 2 Click **Admin Login** and **advanced**.

STEP 3 Click the **System** tab.

STEP 4 Under **Bluetooth Settings**, in the **Enable BT** field, choose **yes**.

STEP 5 Click **Submit All Changes**.

Enabling SMS Messaging

These Cisco IP phones can receive and display text messages via SIP according to RFC3428. Users can receive text messages. WIP310 users can send *and* receive text messages.

When this feature is enabled, the phone displays messages up to 255 characters in length. The message appears on the phone display along with the date and time.

Service providers could use text messages to:

- Send billing information, calling minutes consumed, minutes available
- Include additional text with a call to facilitate call processing

SPA9X2

To enable text message receipt on the SPA9X2 phones:

-
- STEP 1** Log in to the web administration interface.
 - STEP 2** Click **Admin Login** and **advanced**.
 - STEP 3** Click the **User** tab.
 - STEP 4** Under **Supplementary Services**, in the **Text Message** field, choose **yes**.
 - STEP 5** (Optional) To enable receipt of text messages from a third party directly without proxy involvement, in the **Text Message from 3rd Party** field, choose **yes**.
 - STEP 6** Click **Submit All Changes**.
-

SPA525G

To enable text messaging on the SPA525G phones:

-
- STEP 1** Log in to the web administration interface.
 - STEP 2** Click **Admin Login** and **advanced**.
 - STEP 3** Click the **User** tab.
 - STEP 4** Under **Supplementary Services**, in the **Display Text Message on Recv** field, choose **yes**.
 - STEP 5** (Optional) To enable receipt of text messages from a third party directly without proxy involvement, in the **Text Message from 3rd Party** field, choose **yes**.
 - STEP 6** Click **Submit All Changes**.
-

WIP310

-
- STEP 1** Log in to the web administration interface.
 - STEP 2** Click **Admin Login** and **advanced**.
 - STEP 3** Click the **Phone** tab.

STEP 4 Under **SMS Enable**, choose **yes**.

STEP 5 Click **Submit All Changes**.

Enabling the Web Server

The web server allows administrators and users to log in to the phone using a web interface. Administrators and users have different privileges and see different options for the phone based on their role.

To enable the web server:

STEP 1 Log in to the web administration interface.

STEP 2 Click **Admin Login** and **advanced**.

STEP 3 Click the **System** tab.

STEP 4 In the **Enable Web Server** field, choose **yes** to enable the web administration server for the phone.

STEP 5 In the **Web Server Port** field, enter the port to access the server. The default is port 80 for the web UI.

STEP 6 In the **Enable Web Admin Access** field, you can enable or disable local access to the **Admin Login** of the web UI. Defaults to **yes** (enabled).

STEP 7 In the **Admin Passwd** field, enter a password if you want the system administrator to log on to the web UI with a password. The password prompt will appear when an administrator clicks **Admin Login**. The maximum password length is 32 characters.

STEP 8 In the **User Password** field, enter a password if you want users to log on to the web UI with a password. The password prompt will appear users click **User Login**. The maximum password length is 32 characters

STEP 9 Click **Submit All Changes**.

You can also enable the web administration interface from the **Phone** tab (does not apply to the WIP310):

STEP 1 Log in to the web administration interface.

STEP 2 Click **Admin Login** and **advanced**.

STEP 3 Click the **Phone** tab.

STEP 4 Under **Web Serv**, choose **yes**.

STEP 5 Click **Submit All Changes**.

Configuring Lightweight Directory Access Protocol (LDAP)

The SPA IP phones support Lightweight Directory Access Protocol v3 to enable the retrieval of directory information. The LDAP Corporate Directory Search feature, when configured and enabled on a SPA IP phone, allows a user to search a specified LDAP directory for a name, phone number, or both. (LDAP is not supported on the WIP310.)

LDAP-based directories, such as Microsoft Active Directory 2003 and OpenLDAP-based databases, are supported.

These instructions assume you have the following equipment and services:

- A functional LDAP server such as OpenLDAP or Microsoft's Active Directory Server 2003
- A SPA9x2 or SPA525G phone running at least 6.1.3a software on a functional network

Users access LDAP from the **Directory** menu on their IP phone. There is a limit of 20 records returned from an LDAP search.

Before you use the LDAP Corporate Directory Search feature of your phone, you need to configure some basic information.

STEP 1 Log in to the web administration interface.

STEP 2 Click **Admin Login** and **advanced**.

STEP 3 Click the **System** tab.

STEP 4 In the **Optional Network Configuration** section, under **Primary DNS**, enter the IP address of the DNS server. (Only required if using Active Directory with authentication set to MD5.)

STEP 5 In the **Optional Network Configuration** section, under **Domain**, enter the LDAP domain. (Only required if using Active Directory with authentication set to MD5.)

**NOTE**

Some sites may not deploy DNS internally and instead use Active Directory 2003. In this case, it is not necessary to enter a Primary DNS address and an LDAP Domain. However, with Active Directory 2003, the authentication method is restricted to Simple.

STEP 6 Click the **Phone** tab.

STEP 7 Under **LDAP Corporate Directory Search**, in the **LDAP Dir Enable** field, choose **yes** to enable LDAP and cause the name defined in **LDAP Corp Dir Name** to appear in the phone's Directory menu.

STEP 8 Configure values for the fields in the following table and click **Submit All Changes**.

Parameter	Description
LDAP Corp Dir Name	Enter a free-form text name, such as "Corporate Directory."
LDAP Server	Enter a fully qualified domain name or IP address of LDAP server, in the following format: nnn . nnn . nnn . nnn
LDAP Auth Method	Select the authentication method that the LDAP server requires. Choices are: <ul style="list-style-type: none"> None—No authentication is used between the client and the server. Simple—The client sends its fully-qualified domain name and password to the LDAP server. May present security issues. Digest-MD5—The LDAP server sends authentication options and a token to the client. The client returns an encrypted response that is decrypted and verified by the server.

Parameter	Description
LDAP Client DN	<p>Enter the distinguished name domain components [dc] ; for example:</p> <p><code>dc=cv2bu,dc=com</code></p> <p>If using the default Active Directory schema (Name(cn)->Users->Domain), an example of the client DN follows:</p> <p><code>cn="David Lee",dc=users,dc=cv2bu,dc=com</code></p>
LDAP Username	Enter the username for a credentialed user on the LDAP server.
LDAP Password	Enter the password for the LDAP username.
LDAP Search Base	<p>Specify a starting point in the directory tree from which to search.</p> <p>Separate domain components [dc] with a comma. For example:</p> <p><code>dc=cv2bu,dc=com</code></p>
LDAP Last Name Filter	This defines the search for surnames [sn], known as last name in some parts of the world. For example, <code>sn:(sn=*\$VALUE*)</code> . This search allows the provided text to appear anywhere in a name, beginning, middle, or end.
LDAP First Name Filter	This defines the search for the common name [cn]. For example, <code>cn:(cn=*\$VALUE*)</code> . This search allows the provided text to appear anywhere in a name, beginning, middle, or end.
LDAP Search Item 3	Additional customized search item. Can be blank if not needed.
LDAP Item 3 Filter	Customized filter for the searched item. Can be blank if not needed.
LDAP Search Item 4	Additional customized search item. Can be blank if not needed.
LDAP Item 4 Filter	Customized filter for the searched item. Can be blank if not needed.

Parameter	Description
LDAP Display Attrs	<p>Format of LDAP results display on phone where:</p> <ul style="list-style-type: none"> a—Attribute name cn—Common name sn—Surname (last name) telephoneNumber—phone number n—Display name t—type p—phone number
LDAP Number Mapping	<p>Can be blank if not needed.</p> <p>NOTE With the LDAP number mapping you can manipulate the number that was retrieved from the LDAP server. For example, you can append 9 to the number if your dial plan requires a user to enter 9 before dialing. If you do not manipulate the number in this fashion, a user can use the Edit Dial feature to edit the number before dialing out.</p>

The following graphic is an example of an LDAP configuration:

LDAP Corporate Directory Search

LDAP Dir Enable: LDAP Corp Dir Name:

LDAP Server: LDAP Auth Method:

LDAP Client DN:

LDAP Username: LDAP Password:

LDAP Search Base:

LDAP Last Name Filter: LDAP First Name Filter:

LDAP Search Item 3:

LDAP Search Item 4: LDAP Item 3 Filter:

LDAP Display Attrs:

LDAP Number Mapping:

For more information on LDAP, including troubleshooting information, see the *Configuring LDAP Directory Search on SPA SIP IP Phones* Application Note, available from http://www.cisco.com/web/partners/sell/smb/products/voice_and_conferencing.html#~vc_technical_resources (partner log on required).

Configuring BroadSoft Settings (SPA525G)

Configuring BroadSoft Directory

The BroadSoft directory service enables users to search and view their personal, group, or enterprise contacts. This application feature uses BroadSoft's Extended Services Interface (XSI).

To configure the BroadSoft Directory service:

-
- STEP 1** Log in to the web administration interface.
- STEP 2** Click **Admin Login** and **advanced**.
- STEP 3** Click the **Phone** tab.
- STEP 4** Under **Broadsoft Settings**, configure the following:
- Directory Enable: Set to **yes**.
 - XSI Host Server: Enter the name of the server; for example, xsp.xdp.broadsoft.com.
 - Directory Name: Name of the directory. Displays on the user's phone as a directory choice.
 - Directory Type: Select the type of BroadSoft directory:
 - Enterprise (default): Allows users to search on last name, first name, user or group ID, phone number, extension, department, or email address.
 - Group: Allows users to search on last name, first name, user ID, phone number, extension, department, or email address.
 - Personal: Allows users to search on last name, first name, or telephone number.
 - Directory UserID: BroadSoft User ID of the phone user; for example, johndoe@xdp.broadsoft.com.
 - Directory Password: Alphanumeric password associated with the User ID.
- STEP 5** Click **Submit All Changes**.
-

Configuring Synchronization of Do Not Disturb and Call Forward

Enabling synchronization of Do Not Disturb and Call Forward allows the phone to synchronize with the call server so that if Do Not Disturb or Call Forwarding settings are changed on the phone, changes are also made on the server; if changes are made on the server, they are propagated to the phone.

This feature is disabled by default.

To enable synchronization:

-
- STEP 1** Log in to the web administration interface.
 - STEP 2** Click **Admin Login** and **advanced**.
 - STEP 3** Click the **Phone** tab.
 - STEP 4** Under **Broadsoft Settings**, in the **Call Feature Sync Ext** field, choose the extension (1 through 5) that is registered to the BroadSoft server.
 - STEP 5** Click **Submit All Changes**.
-

Configuring XML Services (SPA525G)

The SPA525G provides support for XML services, such as an XML Directory Service or other XML applications.

To configure the phone to connect to an XML Directory service:

-
- STEP 1** Log in to the web administration interface.
 - STEP 2** Click **Admin Login** and **advanced**.
 - STEP 3** Click the **Phone** tab.
 - STEP 4** Enter the following information:
 - XML Directory Service Name: Name of the XML Directory. Displays on the user's phone as a directory choice.
 - XML Directory Service URL: URL where the XML Directory is located.

STEP 5 Click **Submit All Changes**.

To configure the phone to connect to an XML application:

STEP 1 Log in to the web administration interface.

STEP 2 Click **Admin Login** and **advanced**.

STEP 3 Click the **Phone** tab.

STEP 4 Enter the following information:

- XML Application Service Name: Name of the XML application. Displays on the user's phone as a menu item.
- XML Application Service URL: URL where the XML application is located.



NOTE If you have configured an unused line button to connect to an XML application, the button connects to the URL configured here, unless you enter a different URL when configuring the line button. See the [“Configuring Unused Line Keys to Access Services \(SPA525G\)”](#) section on page 40.

STEP 5 Click **Submit All Changes**.

Configuring Music On Hold

Your phone can play music on hold if it is part of a system that has a music-on-hold (MOH) server. To configure music on hold:

STEP 1 Log in to the web administration interface.

STEP 2 Click **Admin Login** and **advanced**.

STEP 3 Click the **Ext <number>** tab.

STEP 4 Under **Call Feature Settings**, in the **MOH Server** field, enter the user ID or the URL of the MOH streaming audio server. If you enter a user ID (no server), the current or outbound proxy is contacted. Defaults to blank (no MOH). If used with a Cisco

SPA9000 Voice System, defaults to *imusic*. For more information, see the *Cisco SPA9000 Voice System Administration Guide*.

STEP 5 Click **Submit All Changes**.

Configuring Extension Mobility with a BroadSoft Server



NOTE

This feature is not available on the WIP310.

If your phones use a BroadSoft server, users can dynamically log in on their phones using extension mobility (EM).

EM lets people who work different shifts or who work at different desks during the week share an extension. EM dynamically configures a phone according to the current user. A Login prompt appears on the phone display when EM is enabled. The user must enter their User ID and Password.

For example, when User A logs in to the phone, all of her personal phone settings are available. Later in the day, User B can log in to the same phone and use his personal settings. After logging on, users have access to personal directory numbers, services, speed dials, and other properties on their phone.

When a user logs out, the phone reverts to a basic profile with limited features enabled.

To configure extension mobility:

-
- STEP 1** Log in to the web administration interface.
 - STEP 2** Click **Admin Login** and **advanced**.
 - STEP 3** Click the **Phone** tab.
 - STEP 4** Under **Extension Mobility**, in the **EM Enable** field, choose **yes**.
 - STEP 5** In the **EM User Domain** field, enter the BroadSoft domain for the phone.
 - STEP 6** Click **Submit All Changes**. The phone reboots.
-

You must also configure the Extension Mobility parameters in the profile rule field in the Provisioning tab. See the *Cisco Small Business IP Telephony Devices Provisioning Guide* for more information.

For more information on extension mobility and BroadSoft, see <http://www.broadsoft.com>.

Configuring SIP, SPCP, and NAT

The SPA and Wireless IP Phones use the following protocols:

- Session Initiation Protocol (SIP)—SPA9X2, WIP310, SPA525G
- Cisco Smart Phone Control Protocol (SPCP)—SPA525G only

This chapter describes how to configure the phone protocols and other parameters. It contains the following sections:

- [Session Initiation Protocol and Cisco IP Phones, page 87](#)
- [Configuring SIP, page 89](#)
- [Configuring SPCP on the SPA525G, page 109](#)
- [Network Address Translation \(NAT\) and Cisco IP Phones, page 109](#)

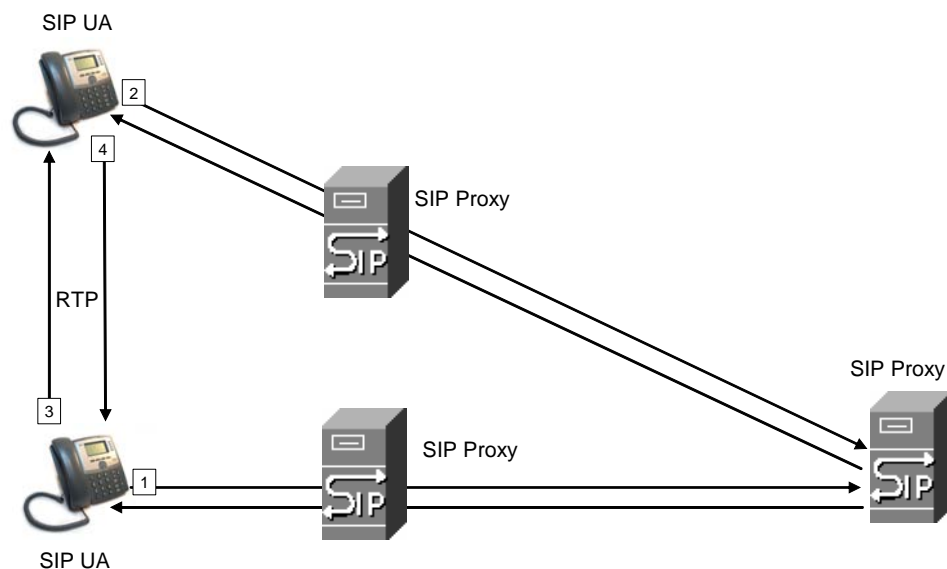
Session Initiation Protocol and Cisco IP Phones

Cisco IP phones use Session Initiation Protocol (SIP), allowing interoperation with all ITSPs supporting SIP.

SIP handles signaling and session management within a packet telephony network. *Signaling* allows call information to be carried across network boundaries. *Session management* controls the attributes of an end-to-end call.

The following diagram shows a SIP request for connection to another subscriber in the network.

In typical commercial IP telephony deployments, all calls go through a SIP proxy server. The requesting phone is called the SIP user agent server (UAS), while the receiving phone is called the user agent client (UAC).



SIP message routing is dynamic. If a SIP proxy receives a request from a UAS for a connection but cannot locate the UAC, the proxy forwards the message to another SIP proxy in the network. When the UAC is located, the response is routed back to the UAS, and a direct peer-to-peer session is established between the two UAs. Voice traffic is transmitted between UAs over dynamically-assigned ports using Real-time Protocol (RTP).

The Internet protocol RTP transmits real-time data such as audio and video; it does not guarantee real-time delivery of data. RTP provides mechanisms for the sending and receiving applications to support streaming data. Typically, RTP runs on top of the UDP protocol. See [“Configuring NAT Mapping with STUN” section on page 111](#).

SIP Over TCP

To guarantee state-oriented communications, Cisco IP phones can use TCP as the transport protocol for SIP. This protocol is “guaranteed delivery”, which assures that lost packets are retransmitted. TCP also guarantees that the SIP packages are received in the same order that they were sent.

TCP overcomes the problem with UDP ports being blocked by corporate firewalls. With TCP, new ports do not need to be opened or packets dropped, because TCP is already in use for basic activities such as Internet browsing or e-commerce.

SIP Proxy Redundancy

An average SIP proxy server may handle tens of thousands of subscribers. A backup server allows an active server to be temporarily switched out for maintenance. Cisco phones support the use of backup SIP proxy servers to minimize or eliminate service disruption.

A static list of proxy servers is not always adequate. If your user agents are served by different domains, for example, you would not want to configure a static list of proxy servers for each domain into every Cisco IP phone.

A simple way to support proxy redundancy is to configure a SIP proxy server in the Cisco IP phone configuration profile. The DNS SRV records instruct the phones to contact a SIP proxy server in a domain named in SIP messages. The phone consults the DNS server. If configured, the DNS server returns an SRV record that contains a list of SIP proxy servers for the domain, with their host names, priority, listening ports, and so on. The Cisco IP phone tries to contact the hosts in the order of their priority.

If the Cisco IP phone currently uses a lower-priority proxy server, the phone periodically probes the higher-priority proxy and switches to the higher-priority proxy when available.

RFC3311 Support

The Cisco SPA525G supports RFC3311, the SIP UPDATE Method.

Configuring SIP

SIP settings for the SPA and Wireless IP phones are configured for the phone in general and for individual extensions. The following sections describe SIP configuration.

Configuring SIP Parameters

To configure general SIP parameters, including enabling CTI:

-
- STEP 1** Log in to the web administration interface.
- STEP 2** Click **Admin Login** and **advanced**.
- STEP 3** Click the **SIP** tab.
- STEP 4** Under **SIP Parameters**, make the necessary configuration changes to the fields shown in the following table and click **Submit All Changes**.
-

Parameter	Description
SIP Reg User Agent Name	User-Agent name used in a REGISTER request. If not specified, the SIP User Agent Name is also used for the REGISTER request. Defaults to blank.
SIP Accept Language	Accept-Language header used. If empty, the header is not included. Defaults to blank.
DTMF Relay MIME Type	MIME Type used in a SIP INFO message to signal a DTMF event. This parameter must match that of the service provider. Defaults to application/dtmf-relay.
Remove Last Reg	If set to yes, removes the last registration before re-registering (if the value is different). Defaults to no.
Use Compact Header	If set to yes, the Cisco IP phone uses compact SIP headers in outbound SIP messages. If inbound SIP requests contain normal (non-compact) headers, the phone substitutes incoming headers with compact headers. If set to no, the Cisco IP phone uses normal SIP headers. If inbound SIP requests contain compact headers, the phone reuses the same compact headers when generating the response, regardless of this setting. Defaults to no.
Escape Display Name	Setting this parameter to yes encloses the configured Display Name string in a pair of double quotes for outbound SIP messages. Any occurrences of ' or \ in the string is escaped with \ and \\ inside the pair of double quotes. Defaults to yes.

Parameter	Description
SIP-B Enable	If set to yes, enables Sylanro call features. See www.sylanro.com for more information. Defaults to no.
Talk Package	If set to yes enables support for the BroadSoft Talk Package, which lets users answer or resume a call by clicking a button in an external application. Defaults to no.
Hold Package	If set to yes, enables support for the BroadSoft Hold Package, which lets users place a call on hold by clicking a button in an external application. Defaults to no.
Conference Package	If set to yes, enables support for the BroadSoft Conference Package, which enables users to start a conference call by clicking a button in an external application. Defaults to no.
Notify Conference	If set to yes, the Cisco IP phone sends out a NOTIFY with event=conference when starting a conference call (with the BroadSoft Conference Package). Defaults to no.
RFC 2543 Call Hold	If set to yes, the Cisco IP phone includes SDP syntax c=0.0.0.0 when sending a SIP re-INVITE to a peer to hold the call. If set to no, the Cisco IP phone does not include the c=0.0.0.0 syntax in the SDP. With either setting, the phone includes a=sendonly syntax in the SDP. Defaults to yes.
Random REG CID On Reboot	If set to yes, the SPA9X2 uses a different random call-ID for registration after the next software reboot. If set to no, the SPA9X2 tries to use the same call-ID for registration after the next software reboot. With either setting the phone uses a new random call-ID for registration after a power-cycle. Defaults to no. NOTE: Not applicable to the WIP310.
Mark All AVT packets	If set to yes, all audio video transport (AVT) tone packets (encoded for redundancy) have the marker bit set. If set to no, only the first packet has the marker bit set for each DTMF event. Defaults to yes.
SIP TCP Port Min	Specifies the lowest TCP port number that can be used for SIP sessions. Defaults to 5060.
SIP TCP Port Max	Specifies the highest TCP port number that can be used for SIP sessions. Defaults to 5080.

Parameter	Description
CTI Enable	<p>If set to yes, enables the computer telephony integration (CTI), where a computer can act as a call center handling all sorts of incoming and outgoing communications, including phone calls, faxes, and text messages. The CTI interface allows a third-party application to control and monitor the state of a Cisco IP phone and, for example, initiate or answer a call by clicking a mouse on a PC,</p> <p>NOTE: CTI must be enabled on the SPA962 for an attached SPA932 to properly monitor the SPA962's line status. If setting up a SPA932, see Chapter 9, “Configuring the SPA932 Attendant Console.”</p> <p>Defaults to no.</p>
Caller ID Header	<p>Select where the SPA9X2/525G gets its caller ID from:</p> <p>PAID-RPID-FROM</p> <p>P-ASSERTED-IDENTITY</p> <p>REMOTE-PARTY-ID</p> <p>FROM header</p> <p>Defaults to PAID-RPID-FROM.</p> <p>NOTE: Not applicable to the WIP310.</p>
SRTP Method	<p>Selects the method to use for SRTP. Two choices are available:</p> <ul style="list-style-type: none"> ▪ x-sipura—legacy SRPT method ▪ s-descriptor—new method compliant with RFC-3711 and RFC-4568 <p>The default value is "x-sipura."</p> <p>NOTE Not applicable to WIP310.</p>
Hold Target Before REFER	<p>Controls whether to hold call leg with transfer target before sending REFER to the transferee when initiating a fully-attended call transfer (where the transfer target has answered). Default value is "no," where the call leg is not held.</p> <p>NOTE Not applicable to WIP310.</p>

Configuring SIP Timer Values

To configure SIP timer values:

-
- STEP 1** Log in to the web administration interface.
- STEP 2** Click **Admin Login** and **advanced**.
- STEP 3** Click the **SIP** tab.
- STEP 4** Under **SIP Timer Values**, make the necessary configuration changes to the fields shown in the following table and click **Submit All Changes**.
-

Parameter	Description
SIP T1	RFC 3261 T1 value (RTT estimate). Ranges from 0 to 64 seconds. Defaults to .5 seconds.
SIP T2	RFC 3261 T2 value, which is the maximum retransmit interval for non-INVITE requests and INVITE responses. Ranges from 0 to 64 seconds. Defaults to 4 seconds.
SIP T4	RFC 3261 T4 value, which is the maximum duration a message remains in the network. Ranges from 0 to 64 seconds. Defaults to 5 seconds.
SIP Timer B	RFC 3261 INVITE transaction time-out value. Ranges from 0 to 64 seconds. Defaults to 16 seconds.
SIP Timer F	RFC 3261 Non-INVITE transaction time-out value. Ranges from 0 to 64 seconds. Defaults to 16 seconds.
SIP Timer H	RFC 3261 INVITE final response time-out value for ACK receipt. Ranges from 0 to 64 seconds. Defaults to 16 seconds.

Parameter	Description
SIP Timer D	FRC 3261 wait time for response retransmits. Ranges from 0 to 64 seconds. Defaults to 16 seconds.
SIP Timer J	RFC 3261 Wait time for Non-INVITE request retransmits. Ranges from 0 to 64 seconds. Defaults to 16 seconds.
INVITE Expires	INVITE request Expires header value. If you enter 0, the Expires header is not included in the request. Ranges from 0 to 19999999999999999999999999999999. Defaults to 240 seconds.
ReINVITE Expires	ReINVITE request Expires header value. If you enter 0, the Expires header is not included in the request. Ranges from 0 to 19999999999999999999999999999999. Defaults to 30
Reg Min Expires	Minimum registration expiration time allowed from the proxy in the Expires header or as a Contact header parameter. If the proxy returns a value less than this setting, the minimum value is used. Defaults to 1 second.
Reg Max Expires	Maximum registration expiration time allowed from the proxy in the Min-Expires header. If the value is greater than this setting, the maximum value is used. Defaults to 7200 seconds.
Reg Retry Intvl (see Note below)	Interval to wait before the Cisco IP phone retries registration after failing during the previous registration. Do not enter 0. Defaults to 30 seconds.
Reg Retry Long Intvl (see Note below)	When registration fails with a SIP response code that does not match the Retry Reg response status code (RSC) value (see next table), the SPA9000 waits for this length of time before retrying. If this interval is 0, the Cisco IP phone stops trying. This value should be much larger than the Reg Retry Intvl value. Defaults to 1200 seconds.

Parameter	Description
Reg Retry Random Delay	<p>Random delay added to the Register Retry Intvl value when retrying REGISTER after a failure. Minimum and maximum random delay to be added to the short timer.</p> <p>Defaults to 0, which disables this feature.</p>
Reg Retry Long Random Delay	<p>Random delay add ed to Register Retry Long Intvl value when retrying REGISTER after a failure.</p> <p>Minimum and maximum random delay to be added to the long timer. Random delay range (in seconds) to add to the Register Retry Long Intvl when retrying REGISTER after a failure.</p> <p>Defaults to 0, which disables this feature.</p> <p>NOTE Not applicable to WIP310.</p>
Reg Retry Intvl Cap	<p>Reg_Retry_Intvl_Cap—Maximum value of the exponential delay. The maximum value to cap the exponential backoff retry delay (which starts at the Register Retry Intvl and doubles every retry).</p> <p>Defaults to 0, which disables the exponential backoff feature (that is, the error retry interval is always at the Register Retry Intvl). If this feature is enabled, the Reg Retry Random Delay is added on top of the exponential backoff delay value.</p> <p>NOTE Not applicable to WIP310.</p>
Sub Min Expires	<p>The lower limit of the REGISTER (subscribe) expires value returned from the proxy server.</p> <p>Defaults to 10 seconds.</p>
Sub Max Expires	<p>The upper limit of the REGISTER (subscribe) min-expires value returned from the proxy server in the Min-Expires header.</p> <p>Defaults to 7200 seconds.</p>
Sub Retry Intvl	<p>The retry interval when the last Subscribe request fails.</p> <p>Defaults to 10 seconds.</p>



NOTE: Cisco IP phones can use a RETRY-AFTER value when received from a SIP proxy server that is too busy to process a request (503 Service Unavailable message). If the response message includes a RETRY-AFTER header, the phone waits for the specified length of time before retrying to REGISTER again. If a RETRY-AFTER header is not present, the phone waits for the value specified in the *Reg Retry Interval* or the *Reg Retry Long Interval* parameter.

Configuring Response Status Code Handling

To configure response status code handling:

-
- STEP 1** Log in to the web administration interface.
- STEP 2** Click **Admin Login** and **advanced**.
- STEP 3** Click the **SIP** tab.
- STEP 4** Under **Response Status Code Handling**, configure the following:
- **SIT1 through SIT4 RSC**—SIP response status code for the appropriate Special Information Tone (SIT). For example, if you set the SIT1 RSC to 404, when the user makes a call and a failure code of 404 is returned, the SIT1 tone is played. Reorder or Busy Tone is played by default for all unsuccessful response status code for SIT 1 RSC through SIT 4 RSC. Defaults to blank.
 - **Try Backup RSC**—SIP response code that retries a backup server for the current request. Defaults to blank.
 - **Retry Reg RSC**—Interval the SPA9000 waits before re-trying registration after a failed registration. Defaults to blank.
- STEP 5** Click **Submit All Changes**.
-

Configuring RTP Parameters

To configure individual RTP parameters:

-
- STEP 1** Log in to the web administration interface.
- STEP 2** Click **Admin Login** and **advanced**.
- STEP 3** Click the **SIP** tab.
- STEP 4** Under **RTP Parameters**, configure the following fields:
- **RTP Port Min**—Minimum port number for RTP transmission and reception. <RTP Port Min> and <RTP Port Max> should define a range that contains at least 10 even number ports (twice the number of lines); for example, 100 – 106. Defaults to 16384.
 - **RTP Port Max**—Maximum port number for RTP transmission and reception. <RTP Port Min> and <RTP Port Max> should define a range that contains at least 10 even number ports (twice the number of lines); for example, 100 – 106. Defaults to 16482.
 - **RTP Packet Size**—Packet size in seconds, which can range from 0.01 to 0.16. Valid values must be a multiple of 0.01 seconds. Defaults to 0.030.
 - **Max RTP ICMP Err**—Number of successive ICMP errors allowed when transmitting RTP packets to the peer before the Cisco IP phone terminates the call. If the value is set to 0 (the default), the Cisco IP phone ignores the limit on ICMP errors, disabling the feature.
 - **RTCP Tx Interval**—Interval for sending out RTCP sender reports on an active connection. During an active connection, the Cisco IP phone can be programmed to send out compound RTCP packet on the connection. Each compound RTP packet except the last one contains a sender report (SR) and a source description (SDES). The last RTCP packet contains an additional BYE packet. Each SR except the last one contains exactly 1 receiver report (RR); the last SR carries no RR.

The SDDES contains CNAME, NAME, and TOOL identifiers.:

- CNAME is set to *User ID@Proxy*
- NAME is set to *Display Name* (or *Anonymous* if user blocks caller ID)
- TOOL is set to the Vendor/Hardware-platform-software-version (such as Cisco/SPA9000-5.2.2(SCb)).

- The NTP timestamp used in the SR is a snapshot of the Cisco IP phone's local time, not the time reported by an NTP server.
- If the Cisco IP phone receives a RR from the peer, it tries to compute the round trip delay and show it as the *Call Round Trip Delay* value in the Info section of the web GUI administration page. It can range from 0 to 255 seconds. Defaults to 0 (recommended).
- **No UDP Checksum**—Select yes if you want the Cisco IP phone to calculate the UDP header checksum for SIP messages. Since this involves computation load, you should keep the default value (no) to disable it.
- **Symmetric RTP**—Enable symmetric RTP operation. If enabled, sends RTP packets to the source address and port of the last received valid inbound RTP packet. If disabled (or before the first RTP packet arrives) sends RTP to the destination as indicated in the inbound SDP. Defaults to no.
- **Stats in BYE**—Determines whether the IP phone includes the P-RTP-Stat header or response to a BYE message. The header contains the RTP statistics of the current call. Select yes or no from the drop-down menu. The format of the P-RTP-Stat header is:

```
P-RTP-State: PS=<packets sent>,OS=<octets
sent>,PR=<packets received>,OR=<octets
received>,PL=<packets lost>,JI=<jitter in ms>,LA=<delay
in ms>,DU=<call duration in s>,EN=<encoder>,DE=<decoder>
```

Defaults to no.

STEP 5 Click **Submit All Changes**.

Configuring SDP Payload Types

Configured dynamic payloads are used for outbound calls only when the Cisco IP phone presents an SDP offer. For inbound calls with an SDP offer, the phone follows the caller's assigned dynamic payload type.

Cisco IP phones use the configured codec names in outbound SDP. For incoming SDP with standard payload types of 0-95, the Cisco IP phone ignores the codec names. For dynamic payload types, the Cisco IP phone identifies the codec by the configured codec names (comparison is case-sensitive).

To configure SDP payload types:

-
- STEP 1** Log in to the web administration interface.
- STEP 2** Click **Admin Login** and **advanced**.
- STEP 3** Click the **SIP** tab.
- STEP 4** Under **SDP Payload Types**, configure the fields shown in the following table and click **Submit All Changes**.
-

Parameter	Description
AVT Dynamic Payload	AVT dynamic payload type. Ranges from 96-127. Defaults to 101.
INFOREQ Dynamic Payload	This parameter defines the Codec Number used in the SIP messaging for the Dynamic Payload size mechanism. This number should match the number configured in the network/other party to enable the use of Dynamic Payload. The best range is 96-127 for any dynamic payload type. Defaults to blank.
G726r16 Dynamic Payload	G.726-16 dynamic payload type. Ranges from 96-127. Defaults to 98. NOTE Not applicable to SPA525G.
G726r24 Dynamic Payload	G.726-24 dynamic payload type. Ranges from 96-127. Defaults to 97. NOTE Not applicable to SPA525G.
G726r32 Dynamic Payload	G726r32 dynamic payload type. The default is 2.
G726r40 Dynamic Payload	G.726-40 dynamic payload type. Ranges from 96-127. Defaults to 96. NOTE Not applicable to SPA525G.

Parameter	Description
G729b Dynamic Payload	G729b Dynamic Payload type. Defaults to 99.
EncapRTP Dynamic Payload	EncapRTP Dynamic Payload type. Defaults to 112.
RTP-Start-LoopbackDynamic	RTP-Start-Loopback Dynamic Payload. Defaults to 113.
RTP-Start-Loopback Codec	RTP-Start-Loopback Codec. Select one of following: G711u, G711a, G726-16, G726-24, G726-32, G726-40, G729a, or G723. Defaults to G711u.
AVT Codec Name	AVT codec name used in SDP. Defaults to telephone-event.
G711u Codec Name	G.711u codec name used in SDP. Defaults to PCMU.
G711a Codec Name	G.711a codec name used in SDP. Defaults to PCMA.
G726r16 Codec Name	G.726-16 codec name used in SDP. Defaults to G726-16. NOTE Not applicable to SPA525G.
G726r24 Codec Name	G.726-24 codec name used in SDP. Defaults to G726-24. NOTE Not applicable to SPA525G.
G726r32 Codec Name	G.726-32 codec name used in SDP. Defaults to G726-32.
G726r40 Codec Name	G.726-40 codec name used in SDP. Defaults to G726-40. NOTE Not applicable to SPA525G.

Parameter	Description
G729a Codec Name	G.729a codec name used in SDP. Defaults to G729a.
G729b Codec Name	G.729b codec name used in SDP. Defaults to G729ab.
G723 Codec Name	G.723 codec name used in SDP. Defaults to G723. NOTE Not applicable to the WIP310 or SPA525G.
EncapRTP Codec Name	EncapRTP codec name used in SDP. Defaults to encaprtp.

Configuring SIP Settings for Extensions

- STEP 1** Log in to the web administration interface.
- STEP 2** Click **Admin Login** and **advanced**.
- STEP 3** Click the **Extension** <number> tab.
- STEP 4** Under **Network Settings**, configure the following fields:

Parameter	Description
SIP ToS/DiffServ Value	Time of service (ToS)/differentiated services (DiffServ) field value in UDP IP packets carrying a SIP message. Defaults to 0x68.
SIP CoS Value [0-7]	Class of service (CoS) value for SIP messages. Defaults to 3.
RTP ToS/DiffServ Value	ToS/DiffServ field value in UDP IP packets carrying RTP data. Defaults to 0xb8.

Parameter	Description
RTP CoS Value [0-7]	CoS value for RTP data. Defaults to 6.
Network Jitter Level	Determines how jitter buffer size is adjusted by the SPA9000. Jitter buffer size is adjusted dynamically. The minimum jitter buffer size is 30 milliseconds or (10 milliseconds + current RTP frame size), whichever is larger, for all jitter level settings. However, the starting jitter buffer size value is larger for higher jitter levels. This setting controls the rate at which the jitter buffer size is adjusted to reach the minimum. Select the appropriate setting: low, medium, high, very high, or extremely high. Defaults to high.
Jitter Buffer Adjustment	Controls how the jitter buffer should be adjusted. Select the appropriate setting: up and down, up only, down only, or disable. Defaults to up and down.

STEP 5 Under **SIP Settings**, configure the following fields:

Parameter	Description
SIP Transport	Select from UDP, TCP, or TLS. Defaults to UDP.
SIP Port	Port number of the SIP message listening and transmission port. Defaults to 5060.
SIP 100REL Enable	To enable the support of 100REL SIP extension for reliable transmission of provisional responses (18x) and use of PRACK requests, select yes. Otherwise, select no. Defaults to no.
EXT SIP Port	The external SIP port number.

Parameter	Description
Auth Resync-Reboot	If this feature is enabled, the Cisco IP phone authenticates the sender when it receives the NOTIFY resync reboot (RFC 2617) message. To use this feature, select yes. Otherwise, select no. Defaults to yes.
SIP Proxy-Require	The SIP proxy can support a specific extension or behavior when it sees this header from the user agent. If this field is configured and the proxy does not support it, it responds with the message, unsupported. Enter the appropriate header in the field provided.
SIP Remote-Party-ID	To use the Remote-Party-ID header instead of the From header, select yes. Otherwise, select no. Defaults to yes.
Referror Bye Delay	Controls when the Cisco IP phone sends BYE to terminate stale call legs upon completion of call transfers. Multiple delay settings (Referror, Refer Target, Referee, and Refer-To Target) are configured on this screen. For the Referror Bye Delay, enter the appropriate period of time in seconds. Defaults to 4.
Refer-To Target Contact	To contact the refer-to target, select yes. Otherwise, select no. Default: no.
Referee Bye Delay	For the Referee Bye Delay, enter the appropriate period of time in seconds. Defaults to 0.

Parameter	Description
SIP Debug Option	<p>SIP messages are received at or sent from the proxy listen port. This feature controls which SIP messages to log. Choices are as follows:</p> <ul style="list-style-type: none"> ▪ none—No logging. ▪ 1-line—Logs the start-line only for all messages. ▪ 1-line excl. OPT—Logs the start-line only for all messages except OPTIONS requests/responses. ▪ 1-line excl. NTFY—Logs the start-line only for all messages except NOTIFY requests/responses. ▪ 1-line excl. REG—Logs the start-line only for all messages except REGISTER requests/responses. ▪ 1-line excl. OPTINTFYIREG—Logs the start-line only for all messages except OPTIONS, NOTIFY, and REGISTER requests/responses. ▪ full—Logs all SIP messages in full text. ▪ full excl. OPT—Logs all SIP messages in full text except OPTIONS requests/responses. ▪ full excl. NTFY—Logs all SIP messages in full text except NOTIFY requests/responses. ▪ full excl. REG—Logs all SIP messages in full text except REGISTER requests/responses. ▪ full excl. OPTINTFYIREG—Logs all SIP messages in full text except for OPTIONS, NOTIFY, and REGISTER requests/responses. <p>Defaults to none.</p>
Refer Target Bye Delay	<p>For the Refer Target Bye Delay, enter the appropriate period of time in seconds.</p> <p>Defaults to 0.</p>
Sticky 183	<p>If this feature is enabled, the IP telephony ignores further 180 SIP responses after receiving the first 183 SIP response for an outbound INVITE. To enable this feature, select yes. Otherwise, select no.</p> <p>Defaults to no.</p>

Parameter	Description
Auth INVITE	When enabled, authorization is required for initial incoming INVITE requests from the SIP proxy. NOTE: Not applicable to the WIP310.
Ntfy Refer On 1xx-To-Inv	If set to yes , as a transferee, the phone will send a NOTIFY with Event:Refer to the transferor for any 1xx response returned by the transfer target, on the transfer call leg. If set to no , the phone will only send a NOTIFY for final responses (200 and higher). NOTE: Not applicable to the WIP310.
Use Anonymous with RPID	This parameter applies only if <SIP Remote-Party-ID> is set to yes ; otherwise, it is ignored. If the parameter is set to yes , the FROM header's display-name and user-id fields are set to anonymous when the caller blocks his caller-id. If the parameter is set to no , the FROM header's display-name and user-id are not masked. The Remote-Party-ID header indicates privacy=full when the caller wishes to block his caller-id. Default: yes. NOTE: Not applicable to the WIP310.
Set G729 annexb	Configure G.729 Annex B settings. NOTE: Not applicable to the SPA525G.

STEP 6 Click **Submit All Changes**.

Configuring a SIP Proxy Server

To configure SIP proxy and registration parameters:

- STEP 1** Log in to the web administration interface.
- STEP 2** Click **Admin Login** and **advanced**.
- STEP 3** Click the **Extension** <number> tab.
- STEP 4** Configure the proxy and registration parameters for each extension.

Parameter	Description
Proxy	SIP proxy server and port number set by the service provider for all outbound requests. For example: 192.168.2.100:6060.
Use Outbound Proxy	Enables an outbound proxy. If set to no, the Outbound Proxy and Use OB Proxy in Dialog fields are ignored. Defaults to no.
Outbound Proxy	SIP outbound proxy server where all outbound requests are sent as the first hop.
Use OB Proxy In Dialog	Select yes for SIP requests to be sent to the outbound proxy within a dialog. This field is ignored if: <ul style="list-style-type: none"> Use Outbound Proxy is set to no or <ul style="list-style-type: none"> Outbound Proxy is blank Defaults to yes.
Register	Enables periodic registration with the proxy. This parameter is ignored if a proxy is not specified. Defaults to yes.
Make Call Without Reg	Enables making outbound calls without successful (dynamic) registration by the phone. If set to no, the dial tone plays only when registration is successful. Defaults to no.
Register Expires	Enables answering inbound calls without successful (dynamic) registration by the phone. If the proxy responds to a REGISTER with a lower expires value, the phone renews registration based on that lower value instead of the configured value. If registration fails with an “Expires too brief” error response, the phone retries with the value specified in the Min-Expires header of the error. Defaults to 60 seconds.

Parameter	Description
Ans Call Without Reg	<p>If enabled, the user does not have to be registered with the proxy to answer calls.</p> <p>Defaults to no.</p>
Use DNS SRV	<p>Enables DNS SRV lookup for the proxy and outbound proxy.</p> <p>Defaults to no.</p>
DNS SRV Auto Prefix	<p>Enables the phone to automatically prepend the proxy or outbound proxy name with _sip._udp when performing a DNS SRV lookup on that name.</p> <p>Defaults to no.</p>
Proxy Fallback Intvl	<p>Sets the delay after which the phone retries from the highest priority proxy (or outbound proxy) after it has failed over to a lower priority server.</p> <p>The phone should have the primary and backup proxy server list via DNS SRV record lookup on the server name. It needs to know proxy priority; otherwise, it does not retry.</p> <p>Defaults to 3600 seconds.</p>
Proxy Redundancy Method	<p>Select Normal or Based on SRV port. The phone creates an internal list of proxies returned in the DNS SRV records.</p> <p>If you select Normal, the list contains proxies ranked by weight and priority.</p> <p>If you select Based on SRV, the phone uses normal, then inspects the port number based on the first-listed proxy port.</p> <p>Defaults to Normal.</p>

STEP 5 Click **Submit All Changes**.

Configuring Subscriber Information Parameters

To configure subscriber information parameters for each extension:

STEP 1 Log in to the web administration interface.

STEP 2 Click **Admin Login** and **advanced**.

STEP 3 Click the **Ext <number>** tab.

STEP 4 Enter the subscriber information:

Parameter	Description
Display Name	Display name for caller ID.
User ID	Extension number for this line.
Password	Password for this line. Defaults to blank.
Use Auth ID	To use the authentication ID and password for SIP authentication, select yes. Otherwise, select no to use the user ID and password. Defaults to no.
Auth ID	Authentication ID for SIP authentication. Defaults to blank.
Mini Certificate	Base64 encoded of Mini-Certificate concatenated with the 1024-bit public key of the CA signing the MC of all subscribers in the group. Defaults to blank.
SRTP Private Key	Base64 encoded of the 512-bit private key per subscriber for establishment of a secure call. Defaults to blank.

STEP 5 Click **Submit All Changes**.

Configuring SPCP on the SPA525G

The SPA525G can be used as part of a Cisco Unified Communications System. This system uses SPCP (also called SCCP) for call control features.

To configure SPCP on the SPA525G:

-
- STEP 1** Log in to the web administration interface.
 - STEP 2** Click **Admin Login** and **advanced**.
 - STEP 3** Click the **System** tab.
 - STEP 4** Under **System Configuration**, in the **SPA525-protocol** field, choose **SCCP**.
 - STEP 5** (Optional) To configure the phone to automatically detect the protocol being used on the network to which it is connected, in the **SPA525-auto-detect-sccp** field, choose **yes**.
 - STEP 6** Click **Submit All Changes**.
-

Network Address Translation (NAT) and Cisco IP Phones

NAT is a function that allows multiple devices to share the same public, routable, IP address to establish connections over the Internet. NAT is present in many broadband access devices to translate public and private IP addresses. To enable VoIP to co-exist with NAT, some form of NAT traversal is required.

Some ITSPs provide NAT traversal, but some do not. If your ITSP does not provide NAT traversal, you have several options.

- **NAT Mapping with Session Border Controller, page 110**
- **NAT Mapping with SIP-ALG Router, page 110**
- **Configuring NAT Mapping with a Static IP Address, page 110**
- **Configuring NAT Mapping with STUN, page 111**

NAT Mapping with Session Border Controller

It is strongly recommended that you choose an ITSP that supports NAT mapping through a Session Border Controller. With NAT mapping provided by the ITSP, you have more choices in selecting a router.

NAT Mapping with SIP-ALG Router

If the ITSP network does not provide a Session Border Controller functionality, you can achieve NAT mapping by using a router that has a SIP ALG (Application Layer Gateway). The WRV200 router is recommended for this purpose, although any router with a SIP-ALG can be used. By using a SIP-ALG router, you have more choices in selecting an ITSP.

Configuring NAT Mapping with a Static IP Address

If the ITSP network does not provide a Session Border Controller functionality, and if other requirements are met, you can configure NAT mapping to ensure interoperability with the ITSP.

Requirements

- You must have an external (public) IP address that is static.
- The NAT mechanism used in the router must be symmetric. See **“Determining Whether the Router Uses Symmetric or Asymmetric NAT,” on page 113.**
- The LAN switch must be configured to enable Spanning Tree Protocol and Port Fast on the ports to which the SPA devices are connected.

**NOTE**

Use NAT mapping only if the ITSP network does not provide a Session Border Controller functionality.

STEP 1 Click **Admin Login** and **advanced**.

STEP 2 Click the **SIP** tab.

STEP 3 Under **NAT Support Parameters**, configure the following:

- **Handle VIA received, Insert VIA received, Substitute VIA Addr:** yes
- **Handle VIA rport, Insert VIA rport, Send Resp To Src Port:** yes
- **EXT IP:** Enter the public IP address for your router.

STEP 4 Click the **Ext <number>** tab. Configure the following:

- **NAT Mapping Enable:** Choose **yes**.
- **NAT Keep Alive Enable:** Choose **yes** (optional).

STEP 5 Click **Submit All Changes**.



NOTE You also need to configure the firewall settings on your router to allow SIP traffic. See **“Configuring SIP,” on page 89**.

Configuring NAT Mapping with STUN

If the ITSP network does not provide a Session Border Controller functionality, and if other requirements are met, it is possible to use STUN as a mechanism to discover the NAT mapping. This option is considered a practice of last resort and should be used only if the other methods are unavailable.

Requirements

- STUN is a viable option only if your router uses asymmetric NAT. See **“Determining Whether the Router Uses Symmetric or Asymmetric NAT,” on page 113**.
- You must have a computer running STUN server software.
- The LAN switch must be configured to enable Spanning Tree Protocol and Port Fast on the ports to which the SPA devices are connected.



NOTE Use NAT mapping only if the ITSP network does not provide a Session Border Controller functionality.

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- STEP 1** Log in to the web administration interface.
- STEP 2** Click **Admin Login** and **advanced**.
- STEP 3** Click the **SIP** tab.
- STEP 4** Under **NAT Support Parameters**, configure the following:
- **Handle VIA received:** yes
 - **Handle VIA rport:** yes
 - **Insert VIA received:** yes
 - **Insert VIA rport:** yes
 - **Substitute VIA Addr:** yes
 - **Send Resp To Src Port:** yes
 - **STUN Enable:** Choose **yes**.
 - **STUN Server:** Enter the IP address for your STUN server.
- STEP 5** Click the **Ext <number>** tab. Configure the following:
- **NAT Mapping Enable:** Choose **yes**.
 - **NAT Keep Alive Enable:** Choose **yes** (optional).



NOTE Your ITSP may require the phone to send NAT keep alive messages to keep the NAT ports open permanently. Check with your ITSP to determine the requirements.

- STEP 6** Click **Submit All Changes**.



NOTE You also need to configure the firewall settings on your router to allow SIP traffic. See **“Configuring SIP,” on page 89**.

Determining Whether the Router Uses Symmetric or Asymmetric NAT

STUN does not work on routers with symmetric NAT. With symmetric NAT, IP addresses are mapped from one internal IP address and port to one external, routable destination IP address and port. If another packet is sent from the same source IP address and port to a different destination, then a different IP address and port number combination is used. This method is restrictive because an external host can send a packet to a particular port on the internal host *only if* the internal host first sent a packet from that port to the external host.

**NOTE**

This procedure assumes that a syslog server is configured and is ready to receive syslog messages.

-
- STEP 1** Make sure you do not have firewall running on your PC that could block the syslog port (by default this is 514).
 - STEP 2** Log on to the phone's web UI. For information about this, see the [“Using the Web Administration User Interface” section on page 25](#).
 - STEP 3** Click **Admin Login > Advanced**. (For WIP310, click **Admin Login**.)
 - STEP 4** Click the **System** tab, then set *Debug Server* to the IP address and port number of your syslog server. Note that this address and port number has to be reachable from the Cisco IP phone. This port number appears on the output file name. The default port number is 514.
 - STEP 5** Set *Debug Level* to **3**. Do not change the value of the *Syslog Server* parameter.
 - STEP 6** To capture SIP signaling messages, click the **Ext** tab.
 - STEP 7** Set *SIP Debug Option* to **Full**. The default output is named syslog.514.log (if not port number was specified).
 - STEP 8** To collect information about what type of NAT your router uses click the **SIP** tab and scroll to NAT Support Parameters.

STEP 9 Back in the *SIP* tab, select **yes** in the *STUN Test Enable* drop-down box.

View the debug messages to determine if your network uses symmetric NAT. Look for the Warning header in REGISTER messages, for example, Warning: 399 Spa "Full Cone NAT detected."

STEP 10 Click **Submit All Changes**.

Configuring Security, Quality, and Network Features

This chapter describes how to configure security, quality, and network features for the phone. It contains the following sections:

- **“Setting Security Features” section on page 115**
- **“Ensuring Voice Quality” section on page 118**
- **“Configuring Voice Codecs” section on page 123**
- **“Configuring Domain and Internet Settings” section on page 126**
- **“Setting Optional Network Parameters” section on page 129**
- **“Configuring VLAN Settings” section on page 131**

Setting Security Features

The following features help ensure that calls are secure and authenticated.

- **“SIP Initial INVITE and MWI Challenge” section on page 115**
- **“SIP Over TLS” section on page 116**
- **“SRTP and Securing Calls” section on page 117**

SIP Initial INVITE and MWI Challenge

SIP INVITE (initial) and MWI message in a session can be challenged by the endpoint. The purpose of this challenge is to restrict the SIP servers that are permitted to interact with the devices on the service provider network, which significantly increases the security of the VoIP network by preventing malicious attacks against the device.

In addition, the Auth INVITE option for Lines 1 and 2 enables the challenging of incoming initial SIP INVITE requests.

SIP Over TLS

Transport layer security (TLS) is a standard protocol for securing and authenticating communications over the Internet.

SIP Over TLS eliminates the possibility of malicious activity by encrypting the SIP messages by the SIP proxy of the service provider and the end user. SIP Over TLS relies on the widely-deployed and standardized Transport Layer Security (TLS) protocol. Note that SIP Over TLS encrypts only the signaling messages and not the media. A separate secure protocol such as Secure Real-Time Transport Protocol (SRTP) (see below) can be used to encrypt voice packets.

The TLS protocol has two layers:

- TLS Record Protocol -- layered on top of a reliable transport protocol, such as SIP or TCH, it ensures that the connection is private by using symmetric data encryption and it ensures that the connection is reliable.
- TLS Handshake Protocol -- allows authentication between the server and client and the negotiation of an encryption algorithm and cryptographic keys before the application protocol transmits or receives any data.

TLS is application protocol-independent. Higher-level protocols such as SIP can layer on top of the TLS protocol transparently.

The IP phones use UDP as a standard for SIP transport, but they also support SIP over TLS for added security.

To enable TLS for the phone:

-
- STEP 1** Log in to the web administration interface.
- STEP 2** Click **Admin Login** and **advanced**.
- STEP 3** Click **Ext <number>**, then scroll to the *SIP Settings* section.
- STEP 4** Select **TLS** from the *SIP Transport* drop-down box.
- STEP 5** Click **Submit All Changes**.
-

SRTP and Securing Calls

Secure Real-Time Transport Protocol (SRTP) is a secure protocol for transporting real-time data over networks. SPA9X2 phones use SRTP to securely send and receive real-time voice traffic from other phones and gateways. Security Description (RFC 4568) is supported.

SRTP provides media encryption to ensure that media streams between devices are secure and that only the intended devices receive and read the data.

When a call is secure, the voice conversation is encrypted so that others cannot eavesdrop on the conversation. To enable this feature the SPA9X2 phone must have a mini-certificate installed.

The supplementary service Secure All Calls (*16)—Defaults to prefer to use encrypted media (voice codecs). Audio packets in both directions of outbound calls are encrypted using SRTP.

To use Secure Call on an extension, you must configure *Mini Certificate* and *SRTP Private Key* for that extension. These parameters appear on the *Ext* tabs.

Secure Call Service activates secure encryption of RTP streams between the two endpoints. You can disable this if the other endpoint (or gateway) does not support this proprietary method.

To enable the secure call service:

-
- STEP 1** Log in to the web administration interface.
 - STEP 2** Click **Admin Login** and **advanced**.
 - STEP 3** Click the **Phone** tab.
 - STEP 4** In the **Supplementary Services** section, under **Secure Call Serv**, choose **yes**.
 - STEP 5** Click **Submit All Changes**.
-

**NOTE**

This feature can also be configured from the **User** tab, under **Supplementary Services**.

Users can enter *18 to Secure Next Call—Uses encrypted media for the next outbound call (on this call appearance only). This star code is redundant if all outbound calls are secure by default.

The phone can be configured for secure provisioning using the factory-installed security certificate. To determine if the **Client Certificate** is installed on the phone:

- SPA9X2: Press the **Setup** button and select **Product Info**. Scroll to **Client Cert**.
- SPA525G: Press the **Setup** button and select **Status**. Select **Product Information**. Scroll to **Certificate**.
- WIP310: Log in to the web administration interface. In the **Info** tab, under **Product Information**, certificate information is listed in the **Client Certificate** field.

Secure Call Indication Tone

This tone is played when a call has been successfully switched to secure mode. It should be played only for a short while (less than 30 seconds) and at a reduced level (less than -19 dBm), so it does not interfere with the conversation. You can configure it on the *Regional* web page under *Call Progress Tones*.

Defaults to 397@-19,507@-19;15(0/2/0,,2/.1/1,,1/2.1/2)

Ensuring Voice Quality

Voice quality perceived by the subscribers of the IP Telephony service should be indistinguishable from that of the PSTN. Cisco IP phones support several codecs. See:

- **“Supported Codecs” section on page 118**
- **“Bandwidth Requirements” section on page 120**
- **“Factors Affecting Voice Quality” section on page 121**

Supported Codecs

Negotiation of the optimal voice codec sometimes depends on the ability of the Cisco IP Phone to “match” a codec name with the far-end device/gateway codec name. Cisco IP phones allow the network administrator to individually name the various codecs that are supported such that the correct codec successfully negotiates with the far-end equipment.

Note that Cisco IP phones support voice codec priority. You can select up to three preferred codecs.

The administrator can select the low-bit-rate codec used for each line. G.711a and G.711u are always enabled. The following table shows the codecs supported by Cisco IP phones. The third column shows the voice quality Mean Opinion Score (MOS), with a scale of 1–5, in which higher is better.

Codec (Voice Compression Algorithm)	Complexity and Description	MOS Score
G.711 (A-law and u-law)	Very low complexity. Supports uncompressed 64 kbps digitized voice transmission at one through ten 5 ms voice frames per packet. This codec provides the highest voice quality and uses the most bandwidth of any of the available codecs.	4.5 Highest voice quality
G.726	Low complexity. Supports compressed 16, 24, 32, and 40 kbps digitized voice transmission at one through ten 10 ms voice frames per packet. When no static payload value is assigned per RFC 1890, Cisco IP phones can support dynamic payloads for G.726. NOTE G.726 is supported only for 32kbps on the SPA525G.	4.1 (32 kbps)
G.729 and G.729A	G.729A low-medium complexity. G.729 medium complexity. G.729A requires about half the processing power of G.729. The G.729 and G.729A bit streams are compatible and interoperable, but not identical.	4
G.723.1	High complexity. Cisco IP phones support the use of ITU G.723.1 audio codec at 6.4 kbps. Up to two channels of G.723.1 can be used simultaneously. For example, Line 1 and Line 2 can be using G.723.1 simultaneously, or Line 1 or Line 2 can initiate a three-way conference with both call legs using G.723.1. NOTE G.723.1 is not supported on the 525G or WIP310.	3.8
G.722	Supported on the SPA525G only. Only one G.722 call at a time is allowed. If a conference call is placed, a SIP re-invite message is sent to switch the calls to narrowband audio.	4.3 (approx)

Bandwidth Requirements

Depending on how you have your IP phones configured, each call requires 55 to 110 kbps in each direction. Therefore, using G.729 as the voice codec setting, and with an average business-grade broadband Internet connection supporting 1.5 Mbps downstream and 384 kbps upstream, a total of seven (7) simultaneous conversations can be reliably supported with adequate bandwidth available for file downloads.

Cisco recommends using the Cisco IP phones with QoS-capable networking equipment that can prioritize the VoIP application traffic. QoS features are available on many data networking switches and routers. A QoS-enabled router prioritizes the packets going upstream to the ISP.

The following table approximates the bandwidth budget for each side of the conversation (in each direction) using different codecs and number of calls. This table is based on the following assumptions:

- Bandwidth calculated with no silence suppression
- 20 millisecond of payload per RTP packet

Codec	Est. Bandwidth Budget	2 Calls	4 Calls	6 Calls	8 Calls
G.711	110 kbps	220 kbps	440 kbps	660 kbps	880 kbps
G.726-40	87 kbps	174 kbps	348 kbps	522 kbps	696 kbps
G.726-32	79 kbps	158 kbps	316 kbps	474 kbps	632 kbps
G.726-24	71 kbps	142 kbps	284 kbps	426 kbps	568 kbps
G.726-16	63 kbps	126 kbps	252 kbps	378 kbps	504 kbps
G.729	55 kbps	110 kbps	220 kbps	330 kbps	440 kbps



NOTE: The use of silence suppression can reduce the average bandwidth budget by 30% or more.

For more information about bandwidth calculation, refer to the following websites:

<http://www.erlang.com/calculator/lipb/>

<http://www.packetizer.com/voip/diagnostics/bandcalc.html>

Factors Affecting Voice Quality

The following factors contribute to voice quality:

- **Audio compression algorithm**—Speech signals are sampled, quantized, and compressed before they are packetized and transmitted to the other end. For IP Telephony, speech signals are usually sampled at 8000 samples per second with 12–16 bits per sample. The compression algorithm plays a large role in determining the voice quality of the reconstructed speech signal at the other end. Cisco IP phones support the most popular audio compression algorithms for IP Telephony: G.711 a-law and u-law, G.726, G.729a, and G.723.1. (G.723.1 is not supported on the 525G or WIP310.)
- **The encoder and decoder pair in a compression algorithm is known as a codec.** The compression ratio of a codec is expressed in terms of the bit rate of the compressed speech. The lower the bit rate, the smaller the bandwidth required to transmit the audio packets. Although voice quality is usually lower with a lower bit rate, it is usually higher as the complexity of the codec gets higher at the same bit rate.
- **Silence suppression**—Cisco IP phones apply silence suppression so that silence packets are not sent to the other end to conserve more transmission bandwidth. IP bandwidth is used only when someone is speaking. Voice activity detection (VAD) with silence suppression is a means of increasing the number of calls supported by the network by reducing the required bidirectional bandwidth for a single call. A noise level measurement is sent periodically during silence suppressed intervals so that the other end can generate artificial comfort noise (comfort noise generator, or CNG).
- **Packet loss**—Audio packets are transported by UDP, which does not guarantee the delivery of the packets. Packets may be lost or contain errors that can lead to audio sample drop-outs and distortions and lower the perceived voice quality. SPA9X2 phones apply an error concealment algorithm to alleviate the effect of packet loss.
- **Network jitter**—The IP network can induce varying delay of received packets. The RTP receiver in Cisco IP phones keeps a reserve of samples to absorb the network jitter, instead of playing out all the samples as soon

as they arrive. This reserve is known as a jitter buffer. The bigger the jitter buffer, the more jitter it can absorb, but this also introduces bigger delay.

Jitter buffer size should be kept to a relatively small size whenever possible. If jitter buffer size is too small, many late packets may be considered as lost and thus lowers the voice quality. Cisco IP phones dynamically adjust the size of the jitter buffer according to the network conditions that exist during a call.

The minimum jitter buffer size is 30 milliseconds or (10 milliseconds + current RTP frame size), whichever is larger, for all jitter level settings. However, the starting jitter buffer size value is larger for higher jitter levels. This setting controls the rate at which the jitter buffer size is adjusted to reach the minimum. Select the appropriate setting: low, medium, high, very high, or extremely high. Defaults to high.

Jitter Buffer Adjustment—Controls how the jitter buffer should be adjusted. Select the appropriate setting: up and down, up only, down only, or disable. Defaults to up and down.

- Echo—Impedance mismatch between the telephone and the IP Telephony gateway phone port can lead to near-end echo. Cisco IP phones have a near-end echo canceller with at least 8 ms tail length to compensate for impedance match. Cisco IP phones implement an echo suppressor with comfort noise generator (CNG) so that any residual echo is not noticeable.
- Hardware noise—Certain levels of noise can be coupled into the conversational audio signals because of the hardware design. The source can be ambient noise or 60 Hz noise from the power adaptor. The Cisco hardware design minimizes noise coupling.
- End-to-end delay—End-to-end delay does not affect voice quality directly but is an important factor in determining whether IP phone subscribers can interact normally in a conversation. A reasonable delay should be about 50–100 ms. End-to-end delay larger than 300 ms is unacceptable to most callers. Cisco IP phones support end-to-end delays well within acceptable thresholds.
- Adjustable Audio Frames Per Packet—Allows you to set the number of audio frames contained in one RTP packet. Packets can be adjusted to contain from 1–10 audio frames. Increasing the number of packets decreases the bandwidth utilized, but it also increases delay and may affect voice quality.

Configuring Voice Codecs

A codec resource is considered allocated if it has been included in the SDP codec list of an active call, even though it eventually might not be chosen for the connection. If the G.729a codec is enabled and included in the codec list, that resource is tied up until the end of the call whether or not the call actually uses G.729a. If the G.729a resource is already allocated (and since only one G.729a resource is allowed per SPA), no other low-bit-rate codec can be allocated for subsequent calls. The only choices are G.711a and G.711u.

Since two G.723.1/G.726 resources are available per SPA9X2, you should disable the use of G.729a to guarantee support for two simultaneous G.723/G.726 codecs.

To configure the voice codecs on each extension:

-
- STEP 1** Log in to the web administration interface.
 - STEP 2** Click **Admin Login** and **advanced**.
 - STEP 3** Click the **Ext <number>** tab for the extension you want to configure.
 - STEP 4** Under **Audio Configuration**, configure the following parameters:

Parameter	Description
Preferred Codec	<p>Preferred codec for all calls. (The actual codec used in a call still depends on the outcome of the codec negotiation protocol.) Select one of the following:</p> <ul style="list-style-type: none"> ▪ G711u (all models) ▪ G711a (all models) ▪ G726-16 (not supported on WIP310, SPA525G) ▪ G726-24 (not supported on WIP310, SPA525G) ▪ G726-32 ▪ G726-40 (not supported on WIP310, SPA525G) ▪ G729a ▪ G723 (not supported on WIP310, SPA525G) ▪ G722 (SPA525G) <p>See “Supported Codecs” section on page 118.</p> <p>Defaults to G711u.</p>
Use Pref Codec Only	<p>To use only the preferred codec for all calls, select yes. (The call fails if the far end does not support this codec.) Otherwise, select no.</p> <p>Defaults to no.</p>
Second Preferred Codec	<p>If the first codec fails, this codec is tried.</p> <p>Defaults to unspecified.</p> <p>NOTE: Not applicable to the WIP310.</p>
Third Preferred Codec	<p>If the second codec fails, this codec is tried.</p> <p>Defaults to unspecified.</p> <p>NOTE: Not applicable to the WIP310.</p>
G729a Enable	<p>To enable the use of the G.729a codec at 8 kbps, select yes. Otherwise, select no.</p> <p>Defaults to yes.</p>

Parameter	Description
G723 Enable	<p>To enable the use of the G.723a codec at 6.3 kbps, select yes. Otherwise, select no.</p> <p>Defaults to yes.</p> <p>NOTE: Not applicable to the WIP310 or SPA525G.</p>
G726-16 Enable	<p>To enable the use of the G.726 codec at 16 kbps, select yes. Otherwise, select no.</p> <p>Defaults to yes.</p> <p>NOTE: Not applicable to the WIP310 or SPA525G.</p>
G726-24 Enable	<p>To enable the use of the G.726 codec at 24 kbps, select yes. Otherwise, select no.</p> <p>Defaults to yes.</p> <p>NOTE: Not applicable to the WIP310 or SPA525G.</p>
G726-32 Enable	<p>To enable the use of the G.726 codec at 32 kbps, select yes. Otherwise, select no.</p> <p>Defaults to yes.</p>
G726-40 Enable	<p>To enable the use of the G.726 codec at 40 kbps, select yes. Otherwise, select no.</p> <p>Defaults to yes.</p> <p>NOTE: Not applicable to the WIP310 or SPA525G.</p>
Release Unused Codec	<p>Allows the release of codecs not used after codec negotiation on the first call so that other codecs can be used for the second line. To use this feature, select yes.</p> <p>Defaults to yes.</p>
DTMF Process AVT	<p>Select yes to process RTP DTMF events. Otherwise, select no. If this parameter is set to no, the AVT payload type is not included in outbound SDP.</p> <p>Defaults to yes.</p>
Silence Supp Enable	<p>To enable silence suppression so that silent audio frames are not transmitted, select yes. Otherwise, select no. See “Ensuring Voice Quality” section on page 118.</p> <p>Defaults to no.</p>

Parameter	Description
DTMF Tx Method	Select the method to transmit DTMF signals to the far end: InBand, audio video transport (AVT), INFO, Auto, InBand+INFO, or AVT+INFO. InBand sends DTMF using the audio path. AVT sends DTMF as AVT events. INFO uses the SIP INFO method. Auto uses InBand or AVT based on the outcome of codec negotiation. Defaults to Auto.

STEP 5 Click **Submit All Changes**.

Configuring Domain and Internet Settings

Configuring Restricted Access Domains

You can configure restricted access domains. If you enter domains, the Cisco IP phones will respond to SIP messages only from the entered servers.

-
- STEP 1** Log in to the web administration interface.
- STEP 2** Click **Admin Login** and **advanced**.
- STEP 3** Click the **System** tab.
- STEP 4** Enter fully-qualified domain names (FQDNs) for each SIP server you want the phone to respond to. Separate with semicolons.
- STEP 5** Click **Submit All Changes**.
-

Configuring DHCP, Static IP, and PPPoE Information

-
- STEP 1** Log in to the web administration interface.
- STEP 2** Click **Admin Login** and **advanced**.
- STEP 3** Click the **System** tab.

STEP 4 Configure the Internet Connection Type. Choose one of the following:

- **Dynamic Host Configuration Protocol (DHCP):** Configure the phone to receive an IP address from the network DHCP. Cisco IP phones typically operate in a network where a DHCP server assigns the device its IP address. Because IP addresses are a limited resource, the DHCP server periodically renews the device lease on the IP address. If a phone loses its IP address for any reason, or if some other device on the network is assigned its IP address, the communication between the SIP proxy and the phone is either severed or degraded. Whenever an expected SIP response is not received within a programmable amount of time after the corresponding SIP command is sent, the *DHCP Timeout on Renewal* feature causes the device to request a renewal of its IP address. If the DHCP server returns the IP address that it originally assigned to the phone, the DHCP assignment is presumed to be operating correctly. Otherwise, the phone resets to try to fix the issue.
- **Static IP—**Configure a static IP address for the phone.
- **PPPoE—**Point-to-Point Protocol over Ethernet (PPPoE) relies on two widely accepted standards: PPP and Ethernet. PPPoE is a specification for connecting users on an Ethernet to the Internet through a common broadband medium, such as a single DSL line, wireless device, or cable modem. All users on an Ethernet share a common connection, so the Ethernet principles supporting multiple users in a LAN combine with the principles of PPP, which apply to serial connections.

**NOTE:** PPPoE is not applicable to WIP310.**STEP 5** Click **Submit All Changes**.**Setting a Static IP Address**

If you configured Static IP as the internet connection type:

STEP 1 Log in to the web administration interface.**STEP 2** Click **Admin Login** and **advanced**.

STEP 3 Click the **System** tab.

STEP 4 Configure the following fields:

- Static IP Address—Enter the static IP address of the phone.
- Netmask—Enter the subnet mask of the phone.
- Gateway—Enter the IP address of the gateway.

For the SPA525G, you also have the following fields available:

- LAN MTU—LAN Maximum Transmission Unit size. Default value: 1500.
- Duplex Mode—Choose one of the following to configure the speed/duplex for the phone's Ethernet ports:
 - Auto
 - 10MBps/Duplex
 - 10MBps/Half
 - 100Mbps/Duplex
 - 100Mbps/Half

STEP 5 Click **Submit All Changes**.

Configuring PPPoE Settings

If you configured PPPoE as the internet connection type:

STEP 1 Log in to the web administration interface.

STEP 2 Click **Admin Login** and **advanced**.

STEP 3 Click the **System** tab.

STEP 4 Configure the following fields:

Parameter	Description
PPPoE Login Name	Specifies the account name assigned by the ISP for connecting on a Point-to-Point Protocol over Ethernet (PPPoE) link.

Parameter	Description
PPPoE Login Password	Specifies the password assigned by the ISP for connecting on a Point-to-Point Protocol over Ethernet (PPPoE) link.
PPPoE Service Name	Specifies the service name assigned by the ISP for connecting on a Point-to-Point Protocol over Ethernet (PPPoE) link.

STEP 5 Click **Submit All Changes**.

Setting Optional Network Parameters

STEP 1 Log in to the web administration interface.

STEP 2 Click **Admin Login** and **advanced**.

STEP 3 Click the **System** tab.

STEP 4 Configure the following fields:

Parameter	Description
Host Name	The host name of the phone.
Domain	The network domain of the phone. However, if using LDAP, see Configuring Lightweight Directory Access Protocol (LDAP) , page 77.
Primary DNS	DNS server used by the phone in addition to DHCP supplied DNS servers if DHCP is enabled; when DHCP is disabled, this is the primary DNS server. Defaults to 0.0.0.0. However, if using LDAP, see Configuring Lightweight Directory Access Protocol (LDAP) , page 77.
Secondary DNS	DNS server used by the phone in addition to DHCP supplied DNS servers if DHCP is enabled; when DHCP is disabled, this is the secondary DNS server. Defaults to 0.0.0.0.

Parameter	Description
DNS Server Order	Specifies the method for selecting the DNS server. The options are Manual, Manual/DHCP, and DHCP/Manual.
DNS Query Mode	Do parallel or sequential DNS Query. With parallel DNS query mode, the phone sends the same request to all the DNS servers at the same time when doing a DNS lookup, the first incoming reply is accepted by the phone. Defaults to parallel. Not available on WIP310.
Syslog Server	Specify the syslog server name and port. This feature specifies the server for logging system information and critical events. If both Debug Server and Syslog Server are specified, Syslog messages are also logged to the Debug Server.
Debug Server	The debug server name and port. This feature specifies the server for logging debug information. The level of detailed output depends on the debug level parameter setting.
Debug Level	The debug level from 0-3. The higher the level, the more debug information is generated. Zero (0) means no debug information is generated. To log SIP messages, you must set the Debug Level to at least 2. Defaults to 0.
NTP Enable	Enables Network Time Protocol (NTP). Applies to the SPA525G only.
Primary NTP Server	IP address or name of primary NTP server. The phones use these servers to synchronize its time. Defaults to blank.
Secondary NTP Server	IP address or name of secondary NTP server. The phones use these servers to synchronize its time. Defaults to blank.

STEP 5 Click **Submit All Changes**.

Configuring VLAN Settings

Using the IP Phones in a VLAN

If you use a VLAN your IP phone voice packets are tagged with the VLAN ID. (This section is not applicable to the WIP310.)

If you are using a Cisco switch, Cisco discovery protocol (CDP) is enabled (this is the default). CDP is negotiation-based and determines which VLAN the IP phone resides in. CDP:

- Obtains the protocol addresses of neighboring devices and also discovers the platform of those devices.
- Shows information about the interfaces your router uses.
- Is media and protocol-independent.

If you are using a VLAN without CDP, you must enter a VLAN ID for the SPA9X2.

STEP 1 Log in to the web administration interface.

STEP 2 Click **Admin Login** and **advanced**.

STEP 3 Click the **System** tab.

STEP 4 Enter the following parameters:

Parameter	Description
Enable VLAN	Choose Yes to enable VLAN. Choose no to disable.
Enable CDP	<i>Enable CDP</i> only if you are using a Linksys switch. CDP is negotiation based and determines which VLAN the SPA9X2 resides in.
VLAN ID	If you use a VLAN without CDP (VLAN enabled and CDP disabled), enter a <i>VLAN ID</i> for the SPA9X2. Note that only voice packets are tagged with the VLAN ID.

Parameter	Description
Enable PC Port VLAN Tagging	Enables VLAN and priority tagging on the phone data port (802.1p/q). This feature facilitates tagging of the VLAN ID (802.1Q) and priority bits (802.1p) of the traffic coming from the PC port of the SPA9x2 phone. Default No. Choose Yes to enable the tagging algorithm.
PC Port VLAN Highest Priority	0-7 (default 0). The priority applied to all frames, tagged and untagged. The phone modifies the frame priority only if the incoming frame priority is higher than this value.
PC Port VLAN ID	0-4095 (default 0). Value of the VLAN ID. The phone tags all the untagged frames coming from the PC (it will not tag frames with an existing tag).

STEP 5 Click **Submit All Changes**.

Provisioning Basics

The Provisioning Tab and its fields are for service provider use only and are not needed in non-SP deployments. This chapter discusses:

- **Provisioning Capabilities, page 134**
- **IP Phone Configuration Profiles, page 136**
- **Sample Configuration File, page 138**
- **Upgrading, Resyncing, and Rebooting Phones, page 138**
- **Redundant Provisioning Servers, page 142**
- **Retail Provisioning, page 143**
- **Automatic In-House Preprovisioning, page 143**
- **Configuration Access Control, page 144**
- **Using HTTPS, page 145**

VARs and service providers should refer to other documentation, depending on your configuration:

- *Cisco SPA Provisioning Guide* (service provider login required)
- Cisco SPA9000 Voice System Administration Guide
- Service provider documentation

Provisioning Capabilities

The Cisco IP phones provide for secure provisioning and remote upgrade. Provisioning is achieved through configuration profiles transferred to the device via TFTP, HTTP, or HTTPS.

The Cisco IP phones can be configured to automatically resync their internal configuration state to a remote profile periodically and on power up. The automatic resyncs are controlled by configuring the desired profile URL into the device.

The SPA9000 accepts profiles in XML format, or alternatively in a proprietary binary format, which is generated by a profile compiler tool, SIP Profile Compiler (SPC), available from Cisco. The SPA9000 supports up to 256-bit symmetric key encryption of profiles. For the initial transfer of the profile encryption key (initial provisioning stage), the SPA9000 can receive a profile from an encrypted channel (HTTPS with client authentication), or it can resync to a binary profile generated by the Cisco SIP profile compiler. In the latter case, the SIP profile compiler can encrypt the profile specifically for the target SPA9000, without requiring an explicit key exchange.

Remote firmware upgrade is achieved via TFTP or HTTP or HTTPS (TFTP or HTTP for WIP310). Remote upgrades are controlled by configuring the desired firmware image URL into the Cisco IP phone via a remote profile resync.

Provisioning Configuration from Phone Keypad

Remote provisioning can be performed from a phone keypad. After the user enters the IP address of the provisioning server, the unit resyncs to a known path name. This feature enables service providers to have VARs install and provision Cisco phones.

To provision from the phone:

SPA9X2

STEP 1 Press **Setup**, then scroll to **Profile Rule**.

STEP 2 Enter the profile rule using the following format, then press the **Resync** soft button.

protocol://server[:port]/profile_pathname

For example:

tftp://192.168.1.5/spa942.cfg

If no protocol is specified, TFTP is assumed. If no server-name is specified, the host that requests the URL is used as *server-name*.

If no port is specified, the default port is used (69 for TFTP, 80 for HTTP, and 443 for HTTPS). then the address can be entered in and press **Resync**.

The status of the remote customization process is shown by the phone's mute button blinking in the following patterns:

- Red/orange slow blink (1.0 seconds on, 1.0 seconds off): Contacting server, server not resolvable, not reachable, or down
- Red/orange slow blink (0.2 seconds on, 0.2 seconds off, 0.2 seconds on, 1.4 seconds off): Server responded with file not found or corrupt file

WIP310

STEP 1 Press the **Select** button to choose *Settings* and press the **Select** button again.

STEP 2 Navigate to *Misc Settings*.

STEP 3 Navigate to profile rule. Enter the profile rule in the following format:

protocol://server[:port]/profile_pathname

For example, to have the WIP310 provisioning done by the Cisco SPA9000 Voice System, enter:

192.168.2.64/cfg/generic.xml

SPA525G

- STEP 1** Press the **Setup** button.
- STEP 2** Scroll to **Device Administration** and press **Select**.
- STEP 3** Scroll to **Profile Rule** and press **Select**.
- STEP 4** Enter the profile rule using the following format, then press the **Resync** soft button.

protocol://server[:port]/profile_pathname

For example:

tftp://192.168.1.5/spa525.cfg

IP Phone Configuration Profiles

The IP phone configuration profile defines the parameter values for a specific device. The configuration profile can be used in two formats:

- Open (XML-style) format
- Proprietary, plain-text format

The XML-style format lets you use standard tools to compile the parameters and values. To protect confidential information contained in the configuration profile, this type of file is generally delivered from the provisioning server to the IP phone over a secure channel provided by HTTPS.

The XML file consists of a series of elements (one per configuration parameter), encapsulated within the element tags `<flat-profile> ... </flat-profile>`. The encapsulated elements specify values for individual parameters. The following is an example of a valid XML profile:

```
<flat-profile>
<Admin_Passwd>some secret</Admin_Passwd>
<Upgrade_Enable>Yes</Upgrade_Enable>
</flat-profile>
```

The names of parameters in XML profiles can generally be inferred from the SPA9000 configuration web pages, by substituting underscores (_) for spaces and other control characters. Further, to distinguish between Lines 1, 2, 3, and 4, corresponding parameter names are augmented by the strings _1_, _2_, _3_, and _4_. For example, Line 1 Proxy is named Proxy_1_ in XML profiles.

The plain-text configuration file uses a proprietary format, which can be encrypted to prevent unauthorized use of confidential information. By convention, the profile is named with the extension .cfg (for example, spa942.cfg). The Cisco SIP Profile Compiler (SPC) tool is provided for compiling the plain-text file containing parameter-value pairs into an encrypted CFG file. The SPC tool is available from Cisco for the Win32 environment (spc.exe) and Linux-i386-elf environment (spc-linux-i386-static). Availability of the SPC tool for the OpenBSD environment is available on a case-by-case basis.

The syntax of the plain-text file accepted by the profile compiler is a series of parameter-value pairs, with the value in double quotes. Each parameter-value pair is followed by a semicolon. The following is an example of a valid text source profile for input to the SPC tool:

```
Admin_Passwd "some secret";  
Upgrade_Enable "Yes";
```

Parameters in the case of source text files for the SPC tool are similarly named, except that to differentiate Line 1, 2, 3, and 4, the appended strings ([1], [2], [3], or [4]) are used. For example, the Line 1 Proxy is named Proxy[1] in source text profiles for input to the SPC.

General Purpose Parameters

These are configured in the **General Purpose Parameters** section of the **Provisioning** tab. These parameters can be used as variables in provisioning and upgrade rules. They are referenced by prepending the variable name with a '\$' character, such as \$GPP_A.

You can optionally Require Admin Password to Reset Unit to Factory Defaults (see last line of sample config file).

Sample Configuration File

Following is a sample configuration file:

```
Set_Local_Date_(mm/dd) " " ;
Set_Local_Time_(HH/mm) " " ;
Time_Zone "GMT-07:00" ; # options: GMT-12:00/GMT-11:00/GMT-10:00/GMT-09:00/
GMT-08:00/GMT-07:00/GMT-06:00/GMT-05:00/GMT-04:00/GMT-03:30/GMT-03:00/GMT-
02:00/GMT-01:00/GMT/GMT+01:00/GMT+02:00/GMT+03:00/GMT+03:30/GMT+04:00/
GMT+05:00/GMT+05:30/GMT+05:45/GMT+06:00/GMT+06:30/GMT+07:00/GMT+08:00/
GMT+09:00/GMT+09:30/GMT+10:00/GMT+11:00/GMT+12:00/GMT+13:00
FXS_Port_Impedance "600" ; # options: 600/900/600+2.16uF/900+2.16uF/
270+750||150nF/220+820||120nF/220+820||115nF/370+620||310nF
FXS_Port_Input_Gain "-3" ;
FXS_Port_Output_Gain "-3" ;
DTMF_Playback_Level "-16" ;
DTMF_Playback_Length ".1" ;
Detect_ABCD "Yes" ;
Playback_ABCD "Yes" ;
Caller_ID_Method "Bellcore(N.Amer,China)" ; # options:
Bellcore(N.Amer,China)/DTMF(Finland,Sweden)/DTMF(Denmark)/ETSI DTMF/ETSI
DTMF With PR/ETSI DTMF After Ring/ETSI FSK/ETSI FSK With PR(UK)
FXS_Port_Power_Limit "3" ; # options: 1/2/3/4/5/6/7/8
Protect_IVR_FactoryReset "No" ;
```



NOTE: You can optionally require an admin password to reset the phone to factory defaults by setting the last line parameter to "yes."

If you are a service provider with a password, see the *Cisco SPA Provisioning Guide*.

Upgrading, Resyncing, and Rebooting Phones

Cisco IP phones support secure remote provisioning and firmware upgrades. You can generate configuration profiles using common, open-source tools that integrate with service provider provisioning systems. Supported transport protocols include TFTP, HTTP, and HTTPS with client certificates.

256-bit symmetric key encryption of profiles is supported. In addition, an unprovisioned Cisco IP phone can receive an encrypted profile specifically targeted for that device without requiring an explicit key, a secure first-time provisioning mechanism using SSL functionality.

User intervention is not required to initiate or complete a profile update or firmware upgrade. The Cisco IP phone upgrade logic is capable of automating multi-stage upgrades, if intermediate upgrades are required to reach a future upgrade state from an older release. A profile resync is only attempted when the Cisco IP phone is idle, because this may trigger a software reboot.

General purpose parameters are provided to help service providers manage the provisioning process. Each Cisco IP phone can be configured to periodically contact a normal provisioning server (NPS). Communication with the NPS does not require the use of a secure protocol because the updated profile is encrypted by a shared secret key. The NPS can be a standard TFTP, HTTP or HTTPS server.

The administrator can upgrade, reboot, or resync Cisco IP phones using the web interface.

Upgrading Firmware on a Phone

Use the upgrade URL to upgrade firmware on the Cisco IP phone. You can upgrade from either a TFTP or HTTP server.

The Upgrade Enable parameter on the Provisioning web page must be set to Yes:

Cisco IP phone web UI: **Provisioning > Firmware Upgrade > Upgrade Enable: yes**

Use the following syntax to upgrade firmware on a phone:

```
http://phone-ip-address/admin/upgrade?protocol://server-  
name[:port]]/firmware-path
```

- Protocol defaults to TFTP.
- Server name is the host requesting the URL.
- Port is the port of the protocol being used (for example, 69 for TFTP or 80 for HTTP).
- *Firmware-path* defaults to /spa.bin (for example, http://192.168.2.217/admin/upgrade?tftp://192.168.2.251/spa.bin) for SPA phones and /wip310.img for the WIP310. The firmware-pathname is typically the file name of the binary located in a directory on the TFTP or HTTP server.

Firmware Upgrade Parameters

The following table defines the function and usage of each parameter in the Firmware Upgrade section of the *Provisioning* tab.

Parameter	Description
Upgrade_Enable	Enables firmware upgrade operations independently of resync actions. Defaults to Yes.
Upgrade_Error_Retry_Delay	The upgrade retry interval (in seconds) applied in case of upgrade failure. The device has a firmware upgrade error timer that activates after a failed firmware upgrade attempt. The timer is initialized with the value in this parameter. The next firmware upgrade attempt occurs when this timer counts down to zero. The default is 3600 seconds.
Downgrade_Rev_Limit	Enforces a lower limit on the acceptable version number during a firmware upgrade or downgrade. The device does not complete a firmware upgrade operation unless the firmware version is greater than or equal to this parameter. The default is (empty).
Upgrade_Rule	This parameter is a firmware upgrade script with the same syntax as Profile_Rule. Defines upgrade conditions and associated firmware URLs. The default is (empty).

Parameter	Description
Log_Upgrade_Request_Msg	Syslog message issued at the start of a firmware upgrade attempt. The default is \$PN \$MAC -- Requesting upgrade \$SCHEME://\$SERVIP:\$PORT\$PATH
Log_Upgrade_Success_Msg	Syslog message issued after a firmware upgrade attempt completes successfully. The default is \$PN \$MAC -- Successful upgrade \$SCHEME://\$SERVIP:\$PORT\$PATH -- \$ERR
Log_Upgrade_Failure_Msg	Syslog message issued after a failed firmware upgrade attempt. The default is \$PN \$MAC -- Upgrade failed: \$ERR.
License Keys	This field is empty.

Resyncing a Phone

You can resync an IP phone to a specific remote profile. The configuration of the phone you resync will match the configuration of the remote phone. The phone can be configured to resync its internal configuration state to a remote profile periodically and on power up.



NOTE: The phone resyncs only when it is idle.

Use the following syntax to resync a phone's profile to a profile on a TFTP, HTTP, or HTTPS server:

```
http://phone-ip-addr/admin/resync?protocol://server-name[:port]/profile-pathname
```

- Parameter following `resync?` defaults to the Profile Rule setting on the web server Provisioning page.
- Protocol defaults to TFTP.
- Server-name defaults to the host requesting the URL.
- Port defaults to:
 - 69 for TFTP
 - 80 for HTTP
 - 443 for HTTPS
- Profile-*path* defaults to the path to the new resync profile (for example, `http://192.168.2.217admin/resync?tftp://192.168.2.251/spaconf.cfg`).

Rebooting a Phone

You can remotely reboot a Cisco IP phone if needed.

Use the following syntax to reboot a phone:

```
http://phone-ip-address/admin/reboot
```

Redundant Provisioning Servers

The provisioning server may be specified as an IP address or as a fully qualified domain name (FQDN). The use of a FQDN facilitates the deployment of redundant provisioning servers. When the provisioning server is identified through a FQDN, the Cisco IP phone attempts to resolve the FQDN to an IP address through DNS. Only DNS A-records are supported for provisioning; DNS SRV address resolution is not available for provisioning. The Cisco IP phone continues to process A-records until the first server responds. If no server associated with the A-records responds, the Cisco IP phone logs an error to the syslog server.

Retail Provisioning

The Cisco IP phone includes a web UI that displays internal configuration and accepts new configuration parameter values. The server also accepts a special URL command syntax for performing remote profile resync and firmware upgrade operations.

In a retail distribution model, a customer purchases a Cisco voice endpoint device, and subsequently subscribes to a particular service. The customer first signs on to the service and establishes a VoIP account, possibly through an online portal. Subsequently, the customer binds the particular device to the assigned service account.

To do so, the unprovisioned Cisco IP phone is instructed to resync with a specific provisioning server through a resync URL command. The URL command typically includes an account PIN number or alphanumeric code to associate the device with the new account.

In the following example, a device at the DHCP-assigned IP address 192.168.1.102 is instructed to provision itself to the SuperVoIP service:

```
http://192.168.1.102/admin/resync?https://prov.supervoip.com/cisco-init/1234abcd
```

In this example, 1234abcd is the PIN number of the new account. The remote provisioning server is configured to associate the Cisco IP phone that is performing the resync request with the new account, based on the URL and the supplied PIN. Through this initial resync operation, the Cisco IP phone is configured in a single step, and is automatically directed to resync thereafter to a permanent URL on the server. For example:

```
https://prov.supervoip.com/cisco-init
```

For both initial and permanent access, the provisioning server relies on the Cisco IP phone client certificate for authentication and supplies correct configuration parameter values based on the associated service account.

Automatic In-House Preprovisioning

Using the web UI and issuing a resync URL is convenient for a customer in the retail deployment model, but it is not as convenient for preprovisioning a large number of units.

The Cisco IP phone supports a more convenient mechanism for in-house preprovisioning. With the factory default configuration, a Cisco IP phone automatically tries to resync to a specific file on a TFTP server, whose IP address is offered as one of the DHCP-provided parameters. This lets a service provider connect each new Cisco IP phone to a LAN environment configured to preprovision phones. Any new Cisco IP phone connected to this LAN automatically resyncs to the local TFTP server, initializing its internal state in preparation for deployment. Among other parameters, this preprovisioning step configures the URL of the Cisco IP phone provisioning server.

Subsequently, when a new customer signs up for service, the preprovisioned Cisco IP phone can be simply bar-code scanned, to record its MAC address or serial number, before being shipped to the customer. Upon receiving the unit, the customer connects the unit to the broadband link. On power-up the Cisco IP phone already knows the server to contact for its periodic resync update.

Configuration Access Control

Besides configuration parameters that control resync and upgrade behavior, the Cisco IP phone provides mechanisms for restricting end-user access to various parameters.

The Cisco IP phone firmware provides specific privileges for login to a User account and an Admin account. The Admin account is designed to give the service provider or VAR configuration access to the Cisco IP phone, while the User account is designed to give limited and configurable control to the end user of the device.

The User and Admin accounts can be independently password protected. The configuration parameters available to the User account are completely configurable in the Cisco IP phone, on a parameter-by-parameter basis. Optionally, user access to the Cisco IP phone web UI can be totally disabled.

The Internet domains accessed by the Cisco IP phone for resync, upgrades, and SIP registration for Line 1 can be restricted.

Using HTTPS

The Cisco IP phone provides a reliable and secure provisioning strategy based on HTTPS requests from the Cisco IP phone to the provisioning server, using both server and client certificates for authenticating the client to the server and the server to the client.

To use HTTPS with Cisco IP phones, you must generate a Certificate Signing Request (CSR) and submit it to Cisco. The Cisco IP phone generates a certificate for installation on the provisioning server that is accepted by Cisco IP phones when they seek to establish an HTTPS connection with the provisioning server.

The Cisco IP phone implements up to 256-bit symmetric encryption, using the American Encryption Standard (AES), in addition to 128-bit RC4. The Cisco IP phone supports the Rivest, Shamir, and Adelman (RSA) algorithm for public/private key cryptography.

Server Certificates

Each secure provisioning server is issued an secure sockets layer (SSL) server certificate, directly signed by Cisco. The firmware running on the Cisco IP phone clients recognizes only these certificates as valid. The clients try to authenticate the server certificate when connecting via HTTPS, and reject any server certificate not signed by Cisco.

This mechanism protects the service provider from unauthorized access to the Cisco IP phone endpoint, or any attempt to spoof the provisioning server. This might allow the attacker to reprovision the Cisco IP phone to gain configuration information, or to use a different VoIP service. Without the private key corresponding to a valid server certificate, the attacker is unable to establish communication with a Cisco IP phone.

Client Certificates

In addition to a direct attack on the Cisco IP phone, an attacker might attempt to contact a provisioning server using a standard web browser, or other HTTPS client, to obtain the Cisco IP phone configuration profile from the provisioning server. To prevent this kind of attack, each Cisco IP phone also carries a unique client certificate, also signed by Cisco, including identifying information about

each individual endpoint. A certificate authority root certificate capable of authenticating the device client certificate is given to each service provider. This authentication path allows the provisioning server to reject unauthorized requests for configuration profiles.

Obtaining a Server Certificate

To obtain a server certificate:

STEP 1 Contact a Cisco/Linksys support person who will work with you on the certificate process.

STEP 2 Generate a private key that will be used in a CSR (Certificate Signing Request). This key is private and you do not need to provide this key to Cisco support. Use open source "openssl" to generate the key. For example:

```
openssl genrsa -out <file.key> 1024
```

STEP 3 Generate CSR a that contains fields that identify your organization, and location. For example:

```
openssl req -new -key <file.key> -out <file.csr>
```

You must have the following information:

- Subject field—Enter the Common Name (CN) that must be a FQDN (Fully Qualified Domain Name) syntax. During SSL authentication handshake, the SPA9000 verifies that the certificate it receives is from the machine that presented it.
- Server's hostname—For example, provserv.domain.com.
- Email address—Enter an email address so that customer support can contact you if needed. This email address is visible in the CSR.

STEP 4 Email the CSR (in zip file format) to the Cisco/Linksys support person. The certificate is signed by Cisco and given to you.

Configuring Regional Parameters and Supplementary Services

Use the *Regional* tab to configure regional and local settings, such as Vertical Service Activation codes (star codes), Vertical Service Announcement Codes, and local language and dictionary. See the following sections:

- [Advanced Scripting for Cadences, Call Progress Tones, and Ring Tones, page 148](#)
- [Call Progress Tones, page 151](#)
- [Distinctive Ring Patterns, page 151](#)
- [Control Timer Values \(sec\), page 152](#)
- [Vertical Service Announcement Codes \(SPA 9X2 only\), page 158](#)
- [Using a Dial Plan, page 161](#)
- [Miscellaneous Parameters, page 164](#)
- [Localizing Your IP Phone, page 165](#)
- [Selecting a Display Language, page 169](#)

Cisco IP phones have configurable call progress tones. Parameters for each type of tone can include number of frequency components, frequency and amplitude of each component, and cadence information.

The call progress tone pass-through feature lets you hear call progress tones (such as ringing) that are generated from the far-end network.

Advanced Scripting for Cadences, Call Progress Tones, and Ring Tones

Advanced information on defining tones and cadences follows.

A CadScript is a mini-script that specifies the cadence parameters of a signal. It can be up to 127 characters. The syntax follows:

$1[S_2]$ —where $S_i = D_i(\text{oni}, 1/\text{offi}, 1[\text{oni}, 2/\text{offi}, 2[\text{oni}, 3/\text{offi}, 3[\text{oni}, 4/\text{offi}, 4[\text{oni}, 5/\text{offi}, 5[\text{oni}, 6/\text{offi}, 6]]]])$ and is known as a *section*, oni, j and offi, j are the on/off duration in seconds of a *segment* and $i = 1$ or 2 , and $j = 1$ to 6 .

D_i is the total duration of the section in seconds. All durations can have up to 3 decimal places to provide 1 ms resolution. The wildcard character “*” stands for infinite duration. The segments within a section are played in order and repeated until the total duration is played.

Example 1: Normal Ring

```
60(2/4)
Number of Cadence Sections = 1
Cadence Section 1: Section Length = 60 s
Number of Segments = 1
Segment 1: On=2s, Off=4s
Total Ring Length = 60s
```

Example 2: Distinctive Ring (short,short,short,long)

```
60(.2/.2,.2/.2,.2/.2,1/4)
Number of Cadence Sections = 1
Cadence Section 1: Section Length = 60s
Number of Segments = 4
Segment 1: On=0.2s, Off=0.2s
Segment 2: On=0.2s, Off=0.2s
Segment 3: On=0.2s, Off=0.2s
Segment 4: On=1.0s, Off=4.0s
Total Ring Length=60s
```

A ToneScript is a mini-script that specifies the frequency, level and cadence parameters of a call progress tone. It can contain up to 127 characters. The syntax follows:

$\text{FreqScript}; Z_1[Z_2]$. The section Z_i is similar to the S_i section in a CadScript except that each on/off segment is followed by a frequency components parameter: $Z_i = D_i(\text{oni}, 1/\text{offi}, 1/\text{fi}, 1[\text{oni}, 2/\text{offi}, 2/\text{fi}, 2[\text{oni}, 3/\text{offi}, 3/\text{fi}, 3[\text{oni}, 4/\text{offi}, 4/\text{fi}, 4[\text{oni}, 5/\text{offi}, 5/\text{fi}, 5[\text{oni}, 6/\text{offi}, 6/\text{fi}, 6]]]])$

where $f_{i,j} = n1[+n2]+n3[+n4[+n5[+n6]]]$ and $1 < n_k < 6$ indicates which of the frequency components given in the FreqScript are used in that segment; if more than one frequency component is used in a segment, the components are summed together.

Example 1: Dial Tone

```
350@-19,440@-19;10(*0/1+2)
Number of Frequencies = 2
Frequency 1 = 350 Hz at -19 dBm
Frequency 2 = 440 Hz at -19 dBm
Number of Cadence Sections = 1
Cadence Section 1: Section Length = 10 s
Number of Segments = 1
Segment 1: On=forever, with Frequencies 1 and 2
Total Tone Length = 10s
Example 2: Stutter Tone
350@-19,440@-19;2(.1/.1/1+2);10(*0/1+2)
Number of Frequencies = 2
Frequency 1 = 350 Hz at -19 dBm
Frequency 2 = 440 Hz at -19 dBm
Number of Cadence Sections = 2
Cadence Section 1: Section Length = 2s
Number of Segments = 1
Segment 1: On=0.1s, Off=0.1s with Frequencies 1 and 2
Cadence Section 2: Section Length = 10s
Number of Segments = 1
Segment 1: On=forever, with Frequencies 1 and 2
Total Tone Length = 12s
```

Example 3: SIT Tone

```
985@-16,1428@-16,1777@-16;20(.380/0/1,.380/0/2,.380/0/3,0/4/0)
Number of Frequencies = 3
Frequency 1 = 985 Hz at -16 dBm
Frequency 2 = 1428 Hz at -16 dBm
Frequency 3 = 1777 Hz at -16 dBm
Number of Cadence Sections = 1
Cadence Section 1: Section Length = 20s
Number of Segments = 4
Segment 1: On=0.38s, Off=0s, with Frequency 1
Segment 2: On=0.38s, Off=0s, with Frequency 2
Segment 3: On=0.38s, Off=0s, with Frequency 3
Segment 4: On=0s, Off=4s, with no frequency components
Total Tone Length = 20s
```

A RingScript is a mini-script that describes a ring tone. The syntax follows:

n=ring-tone-name;w=waveform-id-or-path;c=cadence-id;b=break-time;t=total-time

ring-tone-name is a name to identify this ring tone specification. This name will appear on the Ring Tone menu of the phone. The same name can be used in a SIP Alert-Info header in an inbound INVITE request to tell the phone to play the corresponding ring tone specification. Because of this, the name should contain characters allowed in a URL only.

Waveform-id is the index of the desired waveform to use in this ring tone specification. There are 4 built-in waveforms:

- 1 = A classic phone with mechanical bell
- 2 = Typical phone ring
- 3 = A classic ring tone
- 4 = A wide-band frequency sweep signal

This field can also be a network path (url) to download a ring tone data file from a server on-the-fly. In this case, the syntax of the field is

w=[tftp://]hostname[:port]/path.

cadence-id is the index of the desired cadence to play the given waveform. 8 cadences (1–8) as defined in <Cadence 1> through <Cadence 8>. *Cadence-id* can be 0 if *w*=3,4, or an url. Setting *c*=0 implies the on-time is the natural length of the ring tone file.

break-time specifies the number of seconds to break between two bursts of ring tone, such as *b*=2.5

total-time specifies the total number of seconds to play the ring tone before it times out

Example 1: SIT Tone

```
n=Classic-1,w=3;c=1
n=Simple-1,w=2;c=1
```


Call Progress Tones

For definitions of all call progress tones, see “**Call Progress Tones**” section on page 217.

Distinctive Ring Patterns

Ring cadence defines the ringing pattern that announces a telephone call.



NOTE: The WIP310 has only eight distinctive ring pattern fields.

Alert Tone:		600@-19;.2(.05/0.05/1)	
Distinctive Ring Patterns			
Cadence 1:	60(2/4)	Cadence 2:	60(.3/.2,1/.2,.3/4)
Cadence 3:	60(.8/.4,.8/4)	Cadence 4:	60(.4/.2,.3/.2,.8/4)
Cadence 5:	60(.2/.2,.2/.2,.2/.2,1/4)	Cadence 6:	60(.2/.4,.2/.4,.2/4)
Cadence 7:	60(4.5/4)	Cadence 8:	60(0.25/9.75)
Cadence 9:	60(.4/.2,.4/2)		
Control Timer Values (sec)			
Reorder Delay:	5	Call Back Expires:	1800

Ring Pattern Notes

The default cadence script for eight ring patterns is shown in the figure above.

Control Timer Values (sec)

Cadence 9:	60(.4/.2, .4/.2)
Control Timer Values (sec)	
Reorder Delay:	5
Call Back Expires:	1800
Call Back Retry Intvl:	30
Call Back Delay:	.5
Interdigit Long Timer:	10
Interdigit Short Timer:	3
Vertical Service Activation Codes	
Call Return Code:	*69
Blind Transfer Code:	*98

Control Timer Value Notes

The following table describes all control timer parameters. Each value is displayed in seconds.

Field	Description
Reorder Delay	Delay after far end hangs up before reorder tone is played. 0 = plays immediately, inf = never plays. Range: 0–255 seconds. Defaults to 5.
Call Back Expires	Expiration time in seconds of a call back activation. Range: 0–65535 seconds. Defaults to 1800.
Call Back Retry Intvl	Call back retry interval in seconds. Range: 0–255 seconds. Defaults to 30.
Call Back Delay	Delay after receiving the first SIP 18x response before declaring the remote end is ringing. If a busy response is received during this time, the Cisco IP phone still considers the call as failed and keeps on retrying. Defaults to 0.5.

Field	Description
Interdigit Long Timer	<p>Long timeout between entering digits when dialing. The interdigit timer values are used as defaults when dialing. The <i>Interdigit Long Timer</i> is used after any one digit, if all valid matching sequences in the dial plan are incomplete as dialed. Range: 0–64 seconds.</p> <p>Setting this value high can result in a longer post dialing delay (PDD), which is the time between the start of a call and the time the phone starts ringing. A value that is too low can result in dialed digits not being correctly recognized.</p> <p>Defaults to 10.</p>
Interdigit Short Timer	<p>Short timeout between entering digits when dialing. The <i>Interdigit Short Timer</i> is used after any one digit, if at least one matching sequence is complete as dialed, but more dialed digits would match other as yet incomplete sequences. Range: 0–64 seconds.</p> <p>Defaults to 3.</p>

Configuring Supplementary Services (Star Codes)

The Cisco IP phones provides native support of a large set of enhanced or supplementary services (also known as star codes). A user can enter star codes (such as *21 for call forward, followed by the target number) to perform call features such as call return, blind call transfers, call pickup, and so on. These codes can be handled locally by the phone or to be sent to the network as an INVITE to the service provider.



NOTE: Some service providers choose to disable star codes. See “[Configuring Supplementary Services \(Star Codes\)](#)” section on page 153.

Entering Star Code Values

The phone provides default values for star codes. You can change the default star code values for your area or region.

To configure star code values:

-
- STEP 1** Log in to the web administration interface.
- STEP 2** Click **Admin Login** and **advanced**.
- STEP 3** Click the **Regional** tab.
- STEP 4** Under **Vertical Service Activation Codes**, enter the values you want to change for the codes.
- STEP 5** Click **Submit All Changes**.
-

The codes are as follows:

- Call Return (*69)—Calls the last caller, regardless which extension.
- Blind Transfer (*98)—Allows the user to transfer a call to another number without waiting for the other party to pick up.
- Call Back Act (*66)—Periodically redials the last busy number (every 30 seconds by default) until it rings or until the attempt expires (30 min by default), regardless which extension. Only one call back operation can be ordered at a time. A new order automatically cancels the previous order.
- Call Back Deact (*86)—Cancels the last call back operation.
- Call Forward All Act (*72)—Call forwards all inbound calls. Applies to primary extension only.
- Call Forward All Deact (*73)—Cancels call forward all. Applies to primary extension only.
- Call Forward Busy Act (*90)—Call forwards on busy. Applies to primary extension only.
- Call Forward Busy Deact (*91)—Cancels call forward on busy. Applies to primary extension only.
- Call Forward No Answer Act (*92)—Call forwards if no answer. Applies to primary extension only.
- Call Forward No Answer Deact (*93)—Cancels call forward no answer. Applies to primary extension only.

- **CW Act (*56)**—Enables call waiting. For example, if call waiting is turned off globally, this star code will turn on call waiting until the CW Deact code is entered.
- **CW Deact (*57)**—Deactivates call waiting. For example, if call waiting is turned on globally, this star code deactivates call waiting until the CW Act code is entered.
- **CW Per Call Act (*71)**—Enables call waiting for a single call. For example, if call waiting is turned off globally, this star code will turn on call waiting for that call.
- **CW Per Call Deact (*70)**—Deactivates call waiting for a single call. For example, if call waiting is turned on globally, this star code deactivates call waiting for that call.
- **Block CID Act (*67)**—Blocks caller ID on all outbound calls. Applies to all extensions.
- **Block CID Deact (*68)**—Deactivates caller ID blocking on outbound calls. Applies to all extensions.
- **Block CID Per Call (*81)**—Blocks caller ID on the next outbound call (on the current call appearance only).
- **Block CID Per Call Deact (*82)**—Deactivates caller ID blocking on the next outbound call (on the current call appearance only).
- **Block ANC Act**—Blocks anonymous calls. Applies to all extensions.
- **Block ANC Deact**—Deactivates anonymous call blocking. Applies to all extensions.
- **DND Act (*78)**—Activates Do Not Disturb. Applies to all extensions.
- **DND Deact (*79)**—Deactivates Do Not Disturb. Applies to all extensions.
- **Secure All Call Act (*16)**—Defaults to prefer to use encrypted media (voice codecs).
- **Secure No Call Act (*17)**—Defaults to prefer to use unencrypted media for all outbound calls. Applies to all extensions.
- **Secure One Call Act (*18)**—Prefers to use encrypted media for the outbound call (on this call appearance only).
- **Secure One Call Deact (*19)**—Prefers to use unencrypted media for the outbound call (on this call appearance only).

- Paging (*96)—Pages the number called.
- Call Park (*38)—Parks a call on an entered line number.
- Call UnPark Code (*39)—Retrieves a call from an entered line number.
- Call Pickup (*36)—Picks up a call at an entered extension.
- Group Call Pickup (*37)—Picks up a ringing call at a group of extensions.
- Media Loopback Code (*03)—A service provider can set up a test call from an IP media loopback server (the source) to a subscriber's VoIP device (the mirror). The test call provides statistical reporting on network performance and audio quality.

Depending on the source's capabilities, the SP can see packet jitter, loss, and delay (although Media Loopback cannot identify an offending hop). This helps the SP identify an offending hop that could be causing issues in VoIP calls to a subscriber. The test results can also provide audio quality scoring, which lets a SP better understand the subscriber's experience.

- Referral Services Codes—One or more * codes can be configured into this parameter, such as *98, or *97!*98!*123, and so on. The maximum total length is 79 characters.

This parameter applies when the user places the current call on hold (by Hook Flash) and is listening to second dial tone. Each * code (and the following valid target number according to current dial plan) entered on the second dial-tone triggers the Cisco IP phone to perform a blind transfer to a target number that is prepended by the service * code. For example:

- a. After the user dials *98, the Cisco IP phone plays a special prompt tone while waiting for the user to enter a target number (which is validated according to the dial plan as in normal dialing).
- b. When a complete number is entered, the Cisco IP phone sends a blind REFER to the holding party with the Refer-To target equals to *98 *target_number*. This feature allows the Cisco IP phone to hand off a call to an application server to perform further processing, such as call park.

The * codes should not conflict with any of the other vertical service codes internally processed by the Cisco IP phone. You can delete any * code that you do not want to SPA9000 to process.

Feature Dial Services Codes: Tells the Cisco IP phone what to do when the user is listening to the first or second dial tone.

You can configure one or more * codes into this parameter, such as *72, or *72*74*67*82, and so on. The maximum total length is 79 characters. When the user has a dial tone (first or second dial tone), they can enter a * code (and the following target number according to current dial plan) to trigger the Cisco IP phone to call the target number prepended by the * code. For example:

- a. After the user dials *72, the Cisco IP phone plays a special prompt tone while waiting for the user to enter a target number (which is validated according to the dial plan as in normal dialing).
- b. When a complete number is entered, the Cisco IP phone sends an INVITE to *72 *target_number* as in a normal call. This feature allows the proxy to process features such as call forward (*72) or BLock Caller ID (*67).

The * codes should not conflict with any of the other vertical service codes internally processed by the Cisco IP phone. You can delete any * code you do not want to SPA9000 to process.

You can add a parameter to each * code in *Features Dial Services Codes* to indicate what tone to play after the * code is entered, such as *72'c'*67'p'. Following is a list of allowed dial tone parameters (note the use of back quotes surrounding the parameter without spaces).

- 'c' = Cfwd dial tone
- 'd' = Dial tone
- 'm' = MWI dial tone
- 'o' = Outside dial tone
- 'p' = Prompt dial tone
- 's' = Second dial tone
- 'x' = No tones are placed, x is any digit not used above

If no tone parameter is specified, the Cisco IP phone plays the prompt tone by default.

If the * code is not to be followed by a phone number, such as *73 to cancel call forwarding, do not include it in this parameter. In that case, add that * code in the dial plan.

Activating or Deactivating Supplementary Services

You can disable services handled locally by the phone in one of two ways:

- Delete the star code in the *Vertical Service Activation* section in the **Regional** tab.
- Disable the service in the **Phone** tab. See [Configuring Supplementary Services \(Star Codes\)](#), page 153.



NOTE: If a service is enabled in the *Phone* tab but cleared in the *Regional* tab, the service can still be enabled/disabled by the end- user from the phone LCD or the web UI. If a service is disabled, the soft button associated with that service is hidden on the LCD. Also, any menu item associated with a disabled service is preceded with an exclamation mark (!).

A supplementary service should be disabled if

- the user has not subscribed for it
- or
- the service provider intends to support similar service using other means than relying on the Cisco IP phone.

Vertical Service Announcement Codes (SPA 9X2 only)

SPA9x2 phones support all services that can be activated on a phone (call forward, do not disturb, and so on). Vertical service announcement codes apply only when the user dials the corresponding star code.

Feature Dial Services Codes:		
Vertical Service Announcement Codes		
Service Annc Base Number:		
Service Annc Extension Codes:		
Outbound Call Codec Selection Codes		
Prefer G711u Code:	*017110	Force G711u Code:
Prefer G711a Code:	*017111	Force G711a Code:

Vertical Service Announcement Notes

Following is an example of how you can use these fields:

```
<Service Annc Base Number> = 1234
<Service Annc Extension Codes>=
"CWT:00;CWF:01;FAT:02;FAF:05;FBT:03;FBF:05;FNT:04;FNF:05;"
Here CWT: Call waiting service enabled;
CWF: Call waiting service disabled;
FAT: Call forward all service enabled;
FAF: Call forward all service disabled;
FBT: Call forward busy service enabled;
FBF: Call forward busy service disabled;
FNT: Call forward no answer enabled;
FNF: Call forward no answer disabled;
```

When the user enables call waiting service, the SPA9x2 automatically calls "123400@\$proxy".

When the user *disables* the call waiting service, SPA9x2 connects to "123401@\$proxy".

If the <Service Annc Extension Codes> do not define CWT/CWF extension codes, the SPA9x2 defaults to normal.

Bonus Services Announcement description

When the user enables the callback service using the *code, the SPA9x2 automatically calls "123400@\$proxy."

When the user disables the callback service using the *code, the SPA9x2 automatically connects to the "123401@\$proxy."

If the *Service Annc Extension Codes* do not define CBT/CBF extension codes, the SPA9x2 does not use this feature.

```
[Line1/2]<Service Announcement Serv> = Yes
[Regional]<Service Annc Base Number> = {announcement server base number}
[Regional]<Service Annc Extension Codes> = {SAEC Script}
```

SAEC Script format:{SA_map;}* Here * means 0 or multiple
 SA_map syntax:
 SA_serv=SA_extcode
 SA_serv is the name of service plus the current condition;
 SA_extcode is the extension code which the ANNC server will route to.

Appendix: SA_serv list

- 1) Call Back
 - CBT: Call back enabled
 - CBF: Call back disabled
 - CBB: Call back busy enabled
- 2) Call Forward
 - FAT: Call forward all enabled
 - FAF: Call forward all disabled
 - FBT: Call forward busy enabled
 - FBF: Call forward busy disabled
 - FNT: Call forward no answer enabled
 - FNF: Call forward no answer disabled
 - FLT: Call forward last enabled
 - FLF: Call forward last disabled
- 3) Call Waiting
 - CWT: Call waiting enabled
 - CWF: Call waiting disabled
- 4) Block Last Call
 - BLT: Block last call enabled
 - BLF: Block last call disabled
- 5) Accept Last Call
 - ALT: Accept last call enabled
 - ALF: Accept last call disabled
- 6) Block Caller ID
 - BCT: Block caller id enabled
 - BCF: Block caller id disabled
- 7) Distinctive Ringing
 - DRT: Distinctive ringing enabled
 - DRF: Distinctive ringing disabled
- 8) Speed Dial
 - SDT: Speed dial enabled
 - SDF: Speed dial disabled
- 9) Secure Call
 - SCT: Secure call enabled
 - SCF: Secure call disabled
- 10) Do Not Disturb
 - DDT: DND enabled
 - DDF: DND disabled
- 11) Caller ID
 - CDT: Caller ID enabled
 - CDF: Caller ID disabled
- 12) CW CID
 - WDT: CWCID enabled
 - WDF: CWCID disabled
- 13) Block Anonymous call
 - BAT: Block anonymous call enabled
 - BAF: Block anonymous call disabled

Outbound Call Codec Selection Codes

Codec call selection codes affect voice quality. For more information about voice codecs, see the [“Supported Codecs” section on page 118](#).

Service Annnc Extension Codes:			
Outbound Call Codec Selection Codes			
Prefer G711u Code:	*017110	Force G711u Code:	*027110
Prefer G711a Code:	*017111	Force G711a Code:	*027111
Prefer G723 Code:	*01723	Force G723 Code:	*02723
Prefer G726r16 Code:	*0172616	Force G726r16 Code:	*0272616
Prefer G726r24 Code:	*0172624	Force G726r24 Code:	*0272624
Prefer G726r32 Code:	*0172632	Force G726r32 Code:	*0272632
Prefer G726r40 Code:	*0172640	Force G726r40 Code:	*0272640
Prefer G729a Code:	*01729	Force G729a Code:	*02729
Miscellaneous			

Outbound Call Codec Selection Notes

- You can choose a *preferred* codec for a call or *force* a call to use a specific codec.
- *Prefer G.711u (*017110) through G.729a (*01729)*—Sets the preferred codec for next outbound call. If the preferred codec is unavailable, the second, then the third preferred codec is used, if specified (see the [“Configuring Voice Codecs” section on page 123](#)).
- *Force G.711u (*027110) through G.729a (*02729)*—Forces the specified codec for next outbound call. If the specified codec is unavailable, the preferred codecs are used in order, if specified (see the [“Configuring Voice Codecs” section on page 123](#)).

Using a Dial Plan

Dial plans determine how the digits are interpreted and transmitted. They also determine whether the dialed number is accepted or rejected. You can use a dial plan to facilitate dialing or to block certain types of calls such as long distance or international.

If the phone is part of a SPA9000 system, you create a dial plan for the system that is then propagated to the individual phones. In certain cases, you must configure the dial plan on individual phones:

- If the phone is outside the SPA9000 multicast domain (e.g. when the phone is connected to the SPA9000 via a VPN connection).
- For additional extensions on an existing phone (the system wide phone dial plan is propagated only to extension 1 of all phones).

For information on configuring the SPA9000 Dial Plan, see the *SPA9000 Voice System Administration Guide*.

You can configure a dial plan with three interdigit timers:

- Initial timeout (T)—Handset off hook; no digit pressed yet.
- Long timeout (L)—One or more digits pressed, more digits needed to reach a valid number (as per the dial plan).
- Short timeout (S)—Current dialed number is valid, but more digits would also lead to a valid number.

To configure dial plans on specific phone extensions:

-
- STEP 1** Log in to the web administration interface.
- STEP 2** Click **Admin Login** and **advanced**.
- STEP 3** Click the **Ext <number>** tab.
- STEP 4** Under **Dial Plan**, configure the dial plan parameters. See the following table for information:

Parameter	Description
Dial Plan	<p>Dial plan script for the selected extension.</p> <p>The dial plan syntax allows the designation of three parameters for use with a specific gateway:</p> <ul style="list-style-type: none"> uid – the authentication user-id pwd – the authentication password nat – if this parameter is present, use NAT mapping <p>Separate each parameter with a semi-colon (;).</p> <p>See the <i>Cisco SPA9000 Voice System Administration Guide</i> for more detailed information on dial plans.</p>
Caller ID Map	<p>Inbound caller ID numbers can be mapped to a different string. For example, a number that begins with +44xxxxxx can be mapped to 0xxxxxx. This feature has the same syntax as the Dial Plan parameter. With this parameter, you can specify how to map a caller ID number for display on screen and recorded into call logs.</p> <p>NOTE Not applicable to WIP310.</p>
Enable IP Dialing	<p>Enable or disable IP dialing.</p> <p>Defaults to no.</p>
Emergency Number	<p>Enter a comma-separated list of emergency numbers. When one of these numbers is dialed, the unit disables processing of <i>CONF</i>, <i>HOLD</i>, and other similar softkeys or buttons to avoid accidentally putting the current call on hold. The phone also disables hook flash event handling.</p> <p>Only the far end can terminate an emergency call.</p> <p>The phone is restored to normal after the call is terminated and the SPA9X2 is back on-hook.</p> <p>Maximum number length is 63 characters.</p> <p>Defaults to blank (no emergency number).</p> <p>NOTE Not applicable to WIP310.</p>

STEP 5 Click **Submit All Changes**.

Miscellaneous Parameters

This section contains both DTMF parameters and localization parameters:

- [DTMF Notes, page 164](#)
- [Localizing Your IP Phone, page 165](#)
- [Managing the Time and Date, page 167](#)
- [Configuring Daylight Savings Time, page 167](#)
- [Daylight Saving Time Examples, page 168](#)
- [Selecting a Display Language, page 169](#)
- [Creating a Dictionary Server Script, page 171](#)

SPA 9X2

Prefer G729a Code:	*01729	Force G729a Code:	*02729
Miscellaneous			
Set Local Date (mm/dd):		Set Local Time (HH/mm):	
Time Zone:	GMT-08:00	Time Offset (HH/mm):	
Daylight Saving Time Rule:	start=4/1/7;end=10/-1/7;save=1		
Daylight Saving Time Enable:	yes	DTMF Playback Level:	-16
DTMF Playback Length:	.1	Inband DTMF Boost:	12dB
Dictionary Server Script:			
Language Selection:		Language Selection Copy:	
Dictionary Update Status:	255		
Undo All Changes		Submit All Changes	

WIP310

Miscellaneous			
Time Zone:	GMT-05:00	Time Offset (HH/mm):	
Daylight Saving Time Rule:	start=4/1/7;end=10/-1/7;save=1		
DTMF Playback Level:	-16	DTMF Playback Length:	.1
Inband DTMF Boost:	12dB		

DTMF Notes

Dual Tone Multi-Frequency (DTMF) is the system used by touch-tone phones. DTMF assigns a specific frequency (consisting of two separate tones) to each key so that it can easily be identified by a microprocessor.

In-Band and Out-of-Band (RFC 2833): IP phones can relay DTMF digits as out-of-band events to preserve the fidelity of the digits. This can enhance the reliability of DTMF transmission required by many IVR applications such as dial-up banking and airline information.

The following parameters can either help false detection or get better detection by the IVR. In general, the default values are recommended for both IVR functions.

- *DTMF Playback Level*: Local DTMF playback level in decibels per minute, up to one decimal place. Applicable locally when a user presses a digit or when the phone receives an out-of-band (OOB) DTMF signal from the network side. Does not affect DTMF transmission. Defaults to -16.
- *DTMF Playback Length*: Local DTMF playback duration in milliseconds. Affects only OOB. Defaults to .1.
- *Inband DTMF Boost*: Controls the amount of amplification applied to DTMF signals. Affects only tones sent by inband method. Choices are 0, 3, 6, 9, 12, 15, and 18 decibels. Defaults to 12 dB.

To help false detection, avoid inband and use OOB. With OOB, the DTMF Playback Length does not matter. If you use inband, use a smaller DTMF Boost value.

To get better detection by the IVR, avoid inband and use OOB. This way, the DTMF tone is reconstructed by the PSTN gateway or the remote endpoint, and the quality is not subject to distortion from the audio codec. If you use OOB, use a slightly longer DTMF Playback Length.

If you use inband, use a higher Inband DTMF boost.

Localizing Your IP Phone

The following table describes the localization parameters in the *Miscellaneous* section.

Field	Description
Set Local Date (mm/dd)	Enter the local date (<i>mm</i> represents the month and <i>dd</i> represents the day). The year is optional and uses two or four digits. For example, May 1, 2008, can be entered as: 05/01 or 05/01/08 or 05/01/2008
Set Local Time (HH/mm)	Enter the local time (<i>hh</i> represents hours and <i>mm</i> represents minutes). Seconds are optional.
Time Zone	Selects the number of hours to add to GMT to generate the local time for caller ID generation. Choices are GMT-12:00, GMT-11:00,..., GMT, GMT+01:00, GMT+02:00, ..., GMT+13:00. Defaults to GMT-08:00.
Time Offset (HH/mm)	This specifies the offset from GMT to use for the local system time.
Daylight Saving Time Rule	Enter the rule for calculating daylight saving time. See the “Configuring Daylight Savings Time” section on page 167.
Daylight Saving Enable	Select yes to enable or no to disable DST on the SPA9x2. This setting affects all lines (extensions) on the phone.
Dictionary Server Script	Defines the location of the dictionary server, the languages available and the associated dictionary. See the “Creating a Dictionary Server Script” section on page 171.
Language Selection	Specifies the default language. The value must match one of the languages supported by the dictionary server. The script (dx value) is as follows: <Language_Selection ua="na"> </Language_Selection> Defaults to blank; the maximum number of characters is 512. For example: <Language_Selection ua="na"> Spanish </Language_Selection>

Managing the Time and Date

Cisco IP phones obtain the current time information in one of three ways:

- **NTP Server**—You can configure one or two NTP servers on the phone. When the phone first boots up, it tries to contact the first NTP server to get the current time. The phone periodically synchronizes its time with the NTP server. The synchronization period is fixed at 1 hour. In between updates, the phone tracks time with its own internal clock.
- **SIP Messages**—Each SIP message (request or response) sent to the phone may contain a Date header with the current time information. If the header is present, the phone uses it to update its current time.
- **Manual Setup**—The phone also lets you manually enter the current time and date from the phone GUI or the web UI. However, this value is overridden by the NTP time or SIP Message Date whenever they are available to the phone. Manual setup requires you to enter the time in 24-hour format only.

The time served by the NTP Server and the SIP Date Header are expressed in GMT time. The local time is obtained by offsetting the GMT according to the time zone of the region.

The *Time Zone* parameter can be configured from the web page or through provisioning. This time can be further offset by the *Time Offset (HH/mm)* parameter, which must be entered in 24-hour format. This parameter can also be configured from the phone's LCD display.



NOTE: The *Time Zone* and *Time Offset (HH/mm)* offset values are *not* applied to manual time and date setup.

Configuring Daylight Savings Time

The phone supports auto adjustment for daylight saving time. You must set *Daylight Savings Time Enable* to **yes** and enter the DST rule. This option affects the time stamp on *CallerID*.

To enter the rule for calculating DST, include the start, end, and save values separated by semi-colons (;) as follows:

```
Start = start-time; end=end-time; save = save-time
```

For example, the default DST rule is

```
start=4/1/7;end=10/-1/7;save=1.
```

The *start-time* and *end-time* values specify the start and end dates and times of daylight saving time. Each value is in this format: *month/day/weekday[/HH:mm:ss]*

The *month* value equals any value in the range 1-12 (January-December).

The *day* value equals any + or - value in the range 1-31. If day is 1, it means the weekday on or before the end of the month (in other words the last occurrence of weekday in that month).

The *weekday* value equals any value in the range 1-7 (Monday-Sunday). It can also equal 0. If the weekday value is 0, this means that the date to start or end daylight saving is exactly the date given. In that case, the day value must not be negative. If the weekday value is not 0 and the day value is positive, then daylight saving starts or ends on the weekday value on or after the date given. If the weekday value is not 0 and the day value is negative, then daylight saving starts or ends on the weekday value on or before the date given.

Optional time values: *HH* represents hours (0-23), *mm* represents minutes (0-59), and *ss* represents seconds (0-59). Optional values inside brackets [] are assumed to be 0 if not specified. Midnight is represented by 0:0:0 of the given date.

The *save-time* value is the number of hours, minutes, and/or seconds to add to the current time during DST. The *save-time* value can be preceded by a plus (+) or minus (-) sign to indicate addition or subtraction.

Daylight Saving Time Examples

The following example configures daylight savings time for the U.S, adding one hour starting at midnight on the first Sunday in April and ending at midnight on the last Sunday of October; add 1 hour (USA, North America):

```
start=4/1/7/0:0:0;end=10/31/7/0:0:0;save=1
start=4/1/7;end=10/-1/7;save=1
start=4/1/7/0;end=10/-1/7/0;save=1
```

The following example configures daylight savings time for Egypt, starting at midnight on the last Sunday in April and ending at midnight on the last Sunday of September:

```
start=4/-1/7;end=9/-1/7;save=1 (Egypt)
```

The following example configures daylight savings time for New Zealand, starting at midnight on the first Sunday of October and ending at midnight on the third Sunday of March. This only applies to countries that recognize daylight savings time.

```
start=10/1/7:3/22/7;save=1 (New Zealand)
```

The following example reflects the new change starting March 2007. DST starts on the second Sunday in March and ends on the first Sunday in November:

```
start=3/8/7/02:0:0;end=11/1/7/02:0:0;save=1
```

Selecting a Display Language

This section describes how to localize the SPA9x2 phone display language. You can define up to nine languages, in addition to English, to be available and host the dictionaries for each of the languages on the HTTP or TFTP provisioning server. Language support follows Cisco dictionary principles.



NOTE: The WIP310 does not support localization; however, daylight savings time adjustment is supported.

Use the Language Selection parameter to select the phone's default display language. The value must match one of the languages supported by the dictionary server. The script (dx value) is as follows:

- `<Language_Selection ua="na">`
- `</Language_Selection>`

Defaults to blank; the maximum number of characters is 512. For example:

```
<Language_Selection ua="na"> Spanish  
</Language_Selection>
```

During startup, the phone checks the selected language and downloads the dictionary from the TFTP/HFTP provisioning server indicated in the phone's configuration. The dictionaries are available at the following website:

<http://www.cisco.com/go/smallbiz>

Currently dictionaries are available for the following languages:

- English
- Spanish
- German
- Dutch
- Italian
- French
- Portuguese
- Danish
- Swedish
- Czech
- Slovak



NOTE: For language selection, the following character sets are supported: Latin2 (Czech, Hungarian, Polish, Romanian, Croatian, Slovak, Slovenian, Serbian), Cyrillic (Russian, Bulgarian, Ukrainian and others), and Latin5 (Turkish). The phones officially support the ISO-8859-1 to 8859-16 character sets, which encompass all Eastern and Western European languages.

The end user can change the language of the phone on the phone by following these steps:

-
- STEP 1** Press the **Setup** button.
- STEP 2** Select **Language**, then press the **Select** soft button.
- STEP 3** Select **Option** to change the language.
- STEP 4** With the desired language selected, press **Save**.
-

Creating a Dictionary Server Script

The Dictionary Server Script defines the location of the dictionary server, the languages available and the associated dictionary. The syntax is as follows:

```
Dictionary_Server_Script ua="na"/Dictionary_Server_Script
```

Defaults to blank; the maximum number of characters is 512. The detailed format is as follows:

```
serv={server ip port and root path};  
d0=language0;x0=dictionary0 filename;  
d1=language1;x1=dictionary1 filename;  
d2=language2;x2=dictionary2 filename;  
d3=language3;x3=dictionary3 filename;  
d4=language4;x4=dictionary4 filename;  
d5=language5;x5=dictionary5 filename;  
d6=language6;x6=dictionary6 filename;  
d7=language3;x7=dictionary7 filename;  
d8=language8;x8=dictionary8 filename;  
d9=language5;x9=dictionary9 filename;
```

For example:

```
Dictionary_Server_Script ua="na"  
serv=tftp://192.168.1.119/  
;d0=English;x0=enS_v101.xml;d1=Spanish;x1=esS_v101.xml /  
Dictionary_Server_Script
```


Configuring the SPA932 Attendant Console

The SPA932 is a 32-button attendant console for the SPA962 or SPA525G. This section applies only to SPA962 and SPA525G phones with SPA932 attendant console attachments.

**NOTE**

The SPA932 cannot be used with a SPA525G if the SPA525G is part of a Cisco Unified Communications 500 Series for Small Business system.

The SPA932 connects to the phone with the attachment arm provided (not shown). It obtains power directly from the phone and does not require a separate power supply. Two SPA932 units can be attached to a single phone to monitor a total of 64 separate lines.

For more information about installing and configuring the SPA932 with a Cisco SPA9000 system, see the *Cisco SPA9000 Voice System Installation and Configuration Guide*.

This chapter contains the following sections:

- **Configuring the SPA9000 for the SPA932, page 176**
- **Configuring the BroadSoft Server for the SPA932, page 176**
- **Configuring the Asterisk Server for the SPA932, page 177**
- **Configuring the SPA932, page 178**
- **Unit/Key Configuration Scripts, page 179**
- **SPA932 Parameter Notes, page 183**
- **Monitoring the SPA932, page 184**

SPA932 Features

SPA932 features include:

- 32 programmable speed-dial or direct station select (DSS) buttons/LEDs
- Three-color LEDs (red, green, and orange)
- Support for BroadSoft Busy Lamp
- Support for Asterisk Line Monitoring
- Illuminated line status monitoring (Idle, Ringing, Busy, Null, or Registration Error)
- One-touch Call Transfer—Incoming calls can be immediately transferred to the target destination by pressing a button/key as assigned on the SPA932.

The following table describes SPA932 ports and LEDs.

Port/LED	Meaning
AUX IN	Connects to the phone.
AUX OUT	Connects to a second SPA932 unit (optional)
Green	Idle
Red	In-use
Blinking Red	Ringing
Orange	Solid: not registered. Blinking: configuration error.
Off	Not configured

Setting Up the SPA932 Attendant Console

To configure the server to use the SPA932, configure each extension that will be monitored by the attendant console. The SPA932 is a SIP subscriber in relation to each SIP proxy server, which allows the SPA932 to receive NOTIFY messages from the SIP proxy that indicate the status of each monitored phone. In general, the SIP proxy is identified by its IP address, or through a hostname if DNS is configured. The way in which each phone and extension is identified is vendor-specific.

To set up the SPA932 attendant console:

- STEP 1** Complete the physical installation of the SPA932 unit, which connects it to the phone. For instructions on installing the SPA932 and an introduction to its use, refer to the *32 Button Attendant Console for the SPA962 IP Phone Quick Installation Guide* or the *Cisco Small Business Pro SPA525G User Guide (SIP)*, available on Cisco.com.
- STEP 2** Configure one of the following SIP proxy servers:
- SPA9000—See [Configuring the SPA9000 for the SPA932, page 176](#) (verify that your version of the SPA9000 supports the SPA932).
 - BroadSoft—See [Configuring the BroadSoft Server for the SPA932, page 176](#).
 - Asterisk—See [Configuring the Asterisk Server for the SPA932, page 177](#).
- STEP 3** Configure the SPA932 using the web UI. The web server must be connected to the phone to which the SPA932 is physically attached.



NOTE: The SPA962 phone must be running Release 5.1.12 or later. (All releases for the SPA525G are supported.) Also, CTI must be enabled on the phone for an attached SPA932 to properly monitor the IP phone's line status when the SIP proxy server type is set to SPA9000. See the [“Configuring SIP” section on page 89](#).

Configuring the SPA9000 for the SPA932

You must still enable Computer Telephony Integration (CTI) on the SPA9000 web UI, as shown in the following procedure:

-
- STEP 1** Connect to the web UI for the SPA9000.
 - STEP 2** Click **Admin Login** on the web UI page.
 - STEP 3** If necessary, enter **admin** and your password, then **Advanced**.
 - STEP 4** Click the **SIP** tab.
 - STEP 5** In the *SIP Parameters* section, select **yes** from the *CTI Enable* drop-down list.
 - STEP 6** Click **Submit All Changes**.
-

Configuring the BroadSoft Server for the SPA932

The BroadSoft server requires you to assign IP phone users to the Busy Lamp (BLF) Monitor Users List. The BroadSoft server sends updates on the status of each phone on this list to the SPA932, which subscribes for this service.

To configure the BroadSoft server to support the SPA932:

-
- STEP 1** On the BroadSoft server, in the user profile used by the phone to register with the BroadSoft server, select **Assign Services**.
 - STEP 2** On the Assign Services page, from the Available Services column, move **Busy Lamp** to the User Services column.
 - STEP 3** Define the List URI used by the SPA932 to register for BLF monitoring service with the BroadSoft server. This value must match the value entered following the sub = keyword (for example, cisco_list). Select the domain from the drop-down list to match the Unit Key.



NOTE: If you configure more than one monitored list on the BroadSoft server, use the **vid= keyword** in each unit/keyconfiguration script to specify the phone extension to use for each list.

-
- STEP 4** On the Busy Lamp Field page, move users that need to be monitored from the Available Users column to the Monitored Users column.
 - STEP 5** Click the **Add** (or **Add All**) button to move each user to the Monitored Users column. The Directory Number (DN) associated with each user account when it is created on the BroadSoft Server is shown in parenthesis in the Monitored Users list. You use this DN to identify the specific phone assigned to each key on the SPA932.
 - STEP 6** Save and enable your configuration changes on the BroadSoft server.
-

See also the “[Configuring BroadSoft Busy Lamp Field Auto-Configuration \(SPA525G\)](#)” section on page 182.

Configuring the Asterisk Server for the SPA932

To configure the Asterisk server to allow the SPA932 to register for BLF monitoring:

-
- STEP 1** Add a context in the extensions.conf file.
 - STEP 2** Add a `Subscribecontext` command to point to the context in the sip.conf file.
 - STEP 3** Configure the SPA932 to register with the Asterisk server (see “[Configuring the SPA932](#)” section on page 178).

The following example context uses “home” for extension 3500. This is entered in the file extensions.conf:

```
[home]
exten => 3500,1,Dial(SIP/3500)
exten => 3500,hint,SIP/3500
exten => 3500,2,VoiceMail,u3500
exten => 3500,1,3,hangup
...
```

(continued)

In the following example, extension 3500 is used to add `Subscribecontext` to point to the context. This is entered in the file `sip.conf`:

```
[3500]
type=friend
secret=3500
callerid="spa3500"<3500>
nat=no
context=home
mailbox=3500
Subscribecontext=home
...
...
```

Configuring the SPA932

Complete the configuration required for the extensions on your IP PBX and obtain the following information:

- SPA9000 IP address or other hostname
- Phone extension numbers

To configure the SPA932:



NOTE

Steps 1 through 4 are only required when the *Server Type* field is set to **SPA9000**. These steps are not required for other server types such as BroadSoft and Asterisk.

-
- STEP 1** Connect to the web UI for the phone to which the SPA932 is connected.
 - STEP 2** Click **Admin/Advanced** on the web UI page.
 - STEP 3** Click the **SIP** tab.
 - STEP 4** Select **yes** from the *CTI Enable* drop-down list.
 - STEP 5** Click the **SPA932** or **Attendant Console** tab. (If you do not see the SPA932 or Attendant Console tab, check your firmware version, and upgrade as necessary.)
 - STEP 6** Select **yes** from the *Unit 1 Enable* drop-down list. If you have installed two SPA932 units, also select **yes** from the *Unit 2 Enable* drop-down list.

STEP 7 Select your *Server Type* from the drop-down list:

- SPA9000
- BroadSoft
- Asterisk

STEP 8 Make sure that **no** is selected for *Test Mode Enable*. This option is disabled by default. You cannot complete the software configuration for the SPA932 if this option is enabled. (You can use Test Mode Enable later to test the SPA932.)

STEP 9 Create a configuration script for each target extension or user you want to monitor using the SPA932. Enter this script in the appropriate field for each unit/key. See **“Unit/Key Configuration Scripts” section on page 179**.

STEP 10 Click **Submit All Changes**.

The SPA932 allows only one unit/key entry for each target extension or user. The subscription for the user fails if the user is configured on more than one unit/key, or if the target extension is a shared extension. For example, the following configuration would *fail*. It shows user 101 subscribed for two different functions on two unit/keys; all functions should be put on one unit/key:

```
Unit/key1: func=sd;usr=101@$PROXY;nme=test;vid=2;sub=arupiSSomSok1@$PROXY
Unit/key2: fnc=blf+cp;usr=101@$PROXY;nme=test;vid=2;sub=arupiSSomSok1@$PROXY
```

Unit/Key Configuration Scripts

By default, all LEDs on the SPA932 are assigned to the first configured extension on the phone to which it is connected. Assign the LEDs to any of the other five extensions on the phone that you want to monitor using the SPA932.

The configuration script is composed of the following keywords, followed by an equal sign (=) and separated by semicolons (;):

- **fnc**—defines which of the following functions are enabled for the specified key (separate more than one function with a plus sign (+)):
 - **blf**—busy lamp field function used for monitoring line activity
 - **sd**—speed dial function
 - **cp**—call pickup (if supported by the SIP proxy server). Call pickup (cp) must be supported by the SIP proxy server and be used with blf in the configuration. The syntax is fnc=blf+cp.

- **sub**—Use this keyword to identify the phones to be monitored). Its value and syntax is `stationName@$PROXY`, where system variable `$PROXY` contains the proxy server IP and port (e.g. 192.168.8.101:6060).



NOTE: Unit/key LEDs will not light without the “sub” keyword.

- **usr or ext**—Use one of these keywords to identify the specific users or extensions to be monitored. Its value and syntax is `extensionNumber@$PROXY`, where system variable `$PROXY` contains the proxy server IP and port (e.g. 192.168.8.101:6060).
- **nme (optional)**— Use this field with the SPA9000 to identify any alias that has been assigned to the extension in the SPA9X2 phone configuration. The nme parameter indicates the extension name, which in this case is the same as the station name.
- **vid (optional)**— All LEDs on the SPA932 use phone extensions that they are assigned to. By default, LEDs on the SPA932 are assigned to the first configured extension on the connected phone. You can optionally assign LEDs to any other phone extensions using `vid=keyword`. Use this field to identify the phone extension to use with the monitored list specified by the `sub=` keyword, when more than one BLF monitored list is configured on the SIP proxy server. The possible values are 1 to 6, corresponding to each of the six extensions available on the phone. Only use the `vid=` keyword in the first entry assigned to each phone extension. Subsequent keys will use the same extension. See **“SPA932 Parameter Notes” section on page 183**.

Assigning SPA932 LEDs to Phone Extensions

By default, all 32 keys on the SPA932 are assigned to the first extension on the IP phone (extension 101 based on the default multicast autoconfiguration).

To limit the number of LEDs assigned to the first extension, use the `vid=1` keyword. Then use the `vid=2` keyword to assign the next set of keys to the second extension. The numeric values correspond to the default extensions on the phone as follows:

- `vid=1`—extension 101
- `vid=2`—extension 102
- `vid=3`—extension 103
- `vid=4`—extension 104
- `vid=5`—extension 105
- `vid=6`—extension 106

SPA9000 Syntax

Find more complete information in the *Cisco SPA9000 Voice System Installation and Configuration Guide*.

The following entry enables speed dialing, BLF monitoring, and call pick up on a SPA9000 server with:

- IP address 192.168.1.101
- station name phone1
- extension 101
- SPA932 unit/key using the phone's extension 2

Example:

```
fnc=sd+blf+cp;sub=phone1@$PROXY;usr=101@$PROXY;nme=phone1;vid=2
```

BroadSoft syntax

For example, the following enables speed dialing and BLF monitoring, with a BLF monitoring list URI of marketing, for the user account reception, on a BroadSoft server with the IP address 192.168.100.1:

```
fnc=sd+blf;sub=marketing@192.168.100.1;usr=reception@192.168.100.1
```

The **nme** keyword is not used because the BroadSoft server uses the user account name assigned to the BLF monitoring list.

Note that you can configure a list of BLF subscriptions automatically using a URI (rather than individually configuring each BLF entry). See the “[Configuring BroadSoft Busy Lamp Field Auto-Configuration \(SPA525G\)](#)” section on page 182.

Asterisk syntax

The following is an example entry for a Asterisk server. This entry enables speed dialing, BLF monitoring, and call pickup on a Asterisk server with the IP address 192.168.1.11:

```
fnc=sd+blf+cp;sub=35890@192.168.1.11;nme=35890
```

Configuring BroadSoft Busy Lamp Field Auto-Configuration (SPA525G)

Rather than configuring each BLF key individually, you can enter a single SUBSCRIBE URI and automatically generate configuration scripts for BLF keys for all users on the monitored list.

To configure BLF auto-configuration:

-
- STEP 1** Log in to the web administration interface.
 - STEP 2** Click **Admin Login**.
 - STEP 3** Click **advanced**.
 - STEP 4** Click the **Att Console** tab.

STEP 5 In the BLF List URI field, enter the URI to generate BLF keys:

```
listname@domain.com;vid=2
```

Where:

- listname: Name of the list.
- domain.com: Name of the domain.
- vid: Assigns keys to the extension. By default, the SUBSCRIBE is sent when “Ext 1” is registered. You must set the “vid=” to a different value to change this behavior.

For example:

```
mylist@broadsoft.com;vid=2
```

STEP 6 Click **Submit All Changes**.

SPA932 Parameter Notes

Parameters in the SPA932/Attendant Console web page are described in the following table. For information about Unit/Key syntax, see the previous sections.

Parameter	Description
Subscribe Expires	Specifies how long the subscription remains valid. After the specified period of time, elapses, the SPA932 initiates a new subscription. Defaults to 1800.
Subscribe Retry Interval	Specifies the length of time to wait to try again if subscription fails.
Unit 1 Enable	Enables or disables the first SPA932 unit (each phone can have up to two SPA932 units attached).
Subscribe Delay	Length of delay before attempting to subscribe. Defaults to 1.
Unit 2 Enable	Enables or disables the second SPA932 unit (each phone can have up to two SPA932 units attached).

Parameter	Description
Server Type	Selects the type of server used (SPA9000, BroadSoft, or Asterisk).
Test Mode Enable	Enables or disables test mode. When test mode is enabled, the LEDs are turned on when keys are pressed, going from off to green to red, and back to off. In test mode, when all the buttons on the attendant console are returned to off, all the keys become orange. The phone must be rebooted after the test is completed.
Attendant Console Call Pickup Code	The star code used for picking up a ringing call. Defaults to *98.
BLF List URI	Automatically configures BLF subscriptions for all users on a monitored list.
Unit 1 Key 1-32	Enter a strings that define the extension and other parameters associated with each lighted button on the first SPA932 unit. Keywords and values are case-sensitive. The configuration script is described in “Unit/Key Configuration Scripts” section on page 179 .

Monitoring the SPA932

To display the status of the SPA932, click the SPA932 Status link on the GUI. The status of each attendant console attached (Unit 1 and Unit 2) is shown. Parameters are read-only.

SPA932 Unit Monitoring Notes

The following table describes each parameter; both units display the same parameters.

Parameter	Description
Unit Enable	Displays if the Unit is enabled or disabled.
Subscribe Expires	Displays when the current subscription expires. After the subscription expires, the SPA932 automatically requests a new subscription.
HW Version	Displays the version of the hardware.
Unit Online	Displays whether the unit is connected or not.
Subscribe Retry Interval	Displays the length of time the SPA932 waits to try again if subscription fails.
SW Version	Displays the version of the software currently running on the unit.
Key Name	Displays the name assigned to each key (1-32) on the SPA932 attendant console unit.
Type	Displays the function enabled for each key (1-32) on the SPA932 attendant console unit.
Line	Displays the extension assigned to each key (1-32) on the SPA932 attendant console unit.
Station	Displays the subscribe URI configured for each key (1-32) on the SPA932 attendant console unit.
Subscribe	Displays the subscription status of the unit/key. The value can be Yes, Fail, or No. No indicates that the feature/function (fnc) of that line does not require a subscription (such as speed dial).

Creating an LED Script

LED Script

The LED script describes the color and blinking pattern of a Line Key LED. Each script contains a number of fields separated by a semicolon(;). White spaces are ignored. Each field has the syntax *<field-name> = <field-value>*. The allowed *field-name* and corresponding *field-values* are listed below:

c=o|r|g|a

This field sets the **color** of the LED. The 4 choices are:

- o = off
- r = red
- g = green
- a = amber (orange)

p=n[b]|s[b]|f[b]|d[b]|u[d]

This field sets the blinking **pattern** of the LED. The 4 choices are:

- nb = no blink (steady on or off)
- sb = slow blink (1s on and 1s off)
- fb = fast blink (100ms on and 100ms off)
- ud = user-defined (according to the contents of the u field)

u=on/off/on/off/etc.

This is a user-defined blinking pattern used only when p = ud. It consists of up to 4 pairs of on/off duration in seconds with up to 2 decimal places; each value is separated by a forward slash (/).

LED Script Examples

Example 1

```
c=r;p=sb
```

Color is red and pattern is slow blink.

Example 2

```
c=o
```

LED is off.

Example 3

```
c=g
```

Color is green and pattern is steady on (default).

Example 4

```
c=a;p=ud;u=.1/.1/.1/.1/.1/.9
```

Color is amber (orange) and the blink pattern is: 100ms on, 100ms off, 100ms on, 100ms off, 100ms on, 900ms off

LED Pattern

The administrator can also specify a different color and pattern for each of the following states of the call appearance.

- Idle: This call appearance is not in use. It can be used to make outbound call on this station
- Local Seized: This call appearance has been seized by this station to prepare for an outbound call
- Local Progressing: This station is making an outbound call that is progressing
- Local Active: This station is engaged in a connected call on this call appearance
- Local Ringing: This station is ringing for an incoming call on this call appearance
- Local Held: This station has placed this call appearance on hold

- Remote Seized: This call appearance has been seized by another station to prepare for an outbound call
- Remote Progressing: Another station is making a call on this call appearance and is progressing
- Remote Active: Another station is engaged in a connected call on this call appearance
- Remote Ringing: Another station is ringing for an incoming call to this call appearance
- Remote Held: Another station has placed this call appearance on hold
- Remote Undefined: The share call state is not known (this station is waiting for a notification from the application server)
- Registration Failed: This station has failed to register with the proxy server for the corresponding extension
- Registering: The station is attempting registration with the proxy server for the corresponding extension.
- Disabled: This line key on this station is disabled
- Call Back: A call back (repeat dialing) operation is currently active on this call appearance

SPA and Wireless IP Phone Field Reference

This appendix describes the fields within each section of the following web UI pages:

Voice Tab

- [Info Tab, page 191](#)
- [System Tab, page 197](#)
- [SIP Tab, page 202](#)
- [Regional Tab, page 216](#)
- [Phone Tab, page 236](#)
- [User Tab, page 266](#)
- [SPA932 Status, page 273](#)
- [SPA525G-Specific Tabs, page 274](#)



NOTE: For information about the Provisioning page, see the *Cisco SPA Provisioning Guide*.

Info Tab

This section describes the fields for the following headings on the Info tab:

- [Product Information, page 193](#)
- [Phone Status, page 193](#)



NOTE: The fields on this tab are read-only and cannot be edited.

System Information

Parameter	Description
DHCP	Indicates if DHCP is enabled. Displays Enabled on SPA901, DHCP on other phones. NOTE Not applicable to SPA525G or WIP310.
Connection Type (WIP310/ SPA525G only)	Indicates the type of internet connection for the phone: <ul style="list-style-type: none"> ▪ DHCP ▪ Static IP ▪ PPPoE (not applicable to WIP310)
Current IP	Displays the current IP address assigned to the IP phone.
Host Name	Displays the current host name assigned to the SPA9000 (defaults to SipuraSPA).
Domain	Displays the network domain name of the SPA9000.
Current Netmask	Displays the network mask assigned to the SPA9000.
Current Gateway	Displays the default router assigned to the SPA9000.
Primary DNS	Displays the primary DNS server assigned to the SPA9000.

Parameter	Description
Secondary DNS	Displays the secondary DNS server assigned to the SPA9000.
NTP Enable (SPA525G only)	Shows if Network Time Protocol is enabled.
Primary NTP Server (SPA525G only)	IP Address of the primary NTP server.
Secondary NTP Server (SPA525G only)	IP Address of the secondary NTP server.
TFTP Server (SPA525G only)	Address of the TFTP server for provisioning.
Bluetooth Enabled (SPA525G only)	Shows if Bluetooth is enabled.
Bluetooth Firmware Version (SPA525G only)	Displays the Bluetooth firmware version.
Bluetooth Connected (SPA525G only)	Shows if a Bluetooth device is connected to the phone.
Bluetooth MAC (SPA525G only)	Shows the hardware address of the Bluetooth device.
Connected Device ID (SPA525G only)	Shows the name of the connected Bluetooth device.
Wireless Enabled (SPA525G only)	Shows if Wireless-G is enabled on the phone.
Wireless Connected (SPA525G only)	Shows if the phone is connected to the wireless network.
Wireless MAC (SPA525G only)	Shows the hardware address of the Wireless-G controller.
SSID (SPA525G only)	Shows the SSID, or name of the wireless router to which the phone is connected.
Standard Channel (SPA525G only)	Shows the wireless channel being used in the wireless connection.
Security Mode (SPA525G only)	Shows if wireless security is configured on the phone (yes or no).

Product Information

Parameter	Description
Product Name	Model number of the IP phone.
Serial Number	Serial number of the IP phone.
Software Version	Version number of the IP phone software.
Hardware Version	Version number of the IP phone hardware.
MAC Address	Hardware address of the IP phone.
Client Certificate	Status of the client certificate, which authenticates the IP phone for use in the ITSP network. This field indicates if the client certificate is properly installed in the IP phone.
Customization	For an RC unit, this field indicates whether the unit has been customized or not. Pending indicates a new RC unit that is ready for provisioning. If the unit has already retrieved its customized profile, this field displays the name of the company that provisioned the unit.
Licenses	Indicates any additional licenses that you have installed in the IP phone.

Phone Status

Parameter	Description
Current Time	Current date and time of the system; for example, 10/3/2003 16:43:00.
Elapsed Time	Total time elapsed since the last reboot of the system; for example, 25 days and 18:12:36.
Broadcast Pkts Sent	Total number of broadcast packets sent.
Broadcast Bytes Sent	Total number of broadcast packets received.
Broadcast Pkts Recv	Total number of broadcast bytes sent.

Parameter	Description
Broadcast Bytes Recv	Total number of broadcast bytes received and processed.
Broadcast Pkts Dropped	Total number of broadcast packets received but not processed. Most codecs can handle up to 5% random packet drops as long as the packets are random and not in groups of two or more. Concurrent packet drops result in voice quality issues.
Broadcast Bytes Dropped	Total number of broadcast bytes received but not processed.
RTP Packets Sent	Total number of RTP packets sent (including redundant packets).
RTP Bytes Sent	Total number of RTP packets received (including redundant packets).
RTP Packets Recv	Total number of RTP bytes sent.
RTP Bytes Recv	Total number of RTP bytes received.
SIP Messages Sent	Total number of SIP messages sent (including retransmissions).
SIP Bytes Sent	Total number of SIP messages received (including retransmissions).
SIP Messages Recv	Total number of bytes of SIP messages sent (including retransmissions).
SIP Bytes Recv	Total number of bytes of SIP messages received (including retransmissions).
External IP	External IP address used for NAT mapping.
Operational VLAN ID	ID of the VLAN currently in use if applicable. NOTE Not applicable to WIP310.

Ext Status

The following parameters show for each extension on the phone.

Parameter	Description
Registration State	Shows “Registered” if the phone is registered, “Not Registered” if the phone is not registered to the ITSP.
Last Registration At	Last date and time the line was registered.
Next Registration In	Number of seconds before the next registration renewal.
Message Waiting	Indicates whether the phone user has a new voice mail waiting: Yes or No. This is updated when voice mail notification is received.
Mapped SIP Port	Port number of the SIP port mapped by NAT.

Line/Call Status

The following parameters show for each line and call on the phone.

Parameter	Description
Call State	Status of the call.
Tone	Type of tone used by the call.
Encoder	Codec used for encoding.
Decoder	Codec used for decoding.
Type	Direction of the call.
Remote Hold	Indicates whether the far end has placed the call on hold.
Callback	Indicates whether the call was triggered by a call back request.
Peer Name	Name of the internal phone.

Parameter	Description
Peer Phone	Phone number of the internal phone.
Duration	Duration of the call.
Packets Sent	Number of packets sent.
Packets Recv	Number of packets received.
Bytes Sent	Number of bytes sent.
Bytes Recv	Number of bytes received.
Decode Latency	Number of milliseconds for decoder latency.
Jitter	Number of milliseconds for receiver jitter.
Round Trip Delay	Number of milliseconds for delay.
Packets Lost	Number of packets lost.
Packet Error	Number of invalid packets received.
Mapped RTP Port	The port mapped for Real Time Protocol traffic for the call.
Media Loopback	If the call is a loopback call, displays the loopback mode (source or mirror) and type (media or packet). If the call is not loopback, the field appears blank.

Downloaded Ring Tone

Parameter	Description
Status	Indicates whether the phone is downloading a ring tone (and from where) or if it is idle.
Ring Tone 1	Information about the user downloaded ring tone 1: name, size, and time-stamp of the tone.
Ring Tone 2	Information about the user downloaded ring tone 2: name, size, and time-stamp of the tone.

System Tab

This section describes the fields for the following headings on the System tab:

- [System Configuration, page 197](#)
- [Internet Connection Type and Static IP Settings, page 198](#)
- [PPPoE Settings, page 199](#)
- [Optional Network Configuration, page 199](#)
- [VLAN Settings, page 201](#)
- [Wi-Fi Settings \(SPA525G only\), page 201](#)
- [Bluetooth Settings \(SPA525G only\), page 201](#)

System Configuration

Parameter	Description
Restricted Access Domains	This feature is used when implementing software customization.
Enable Web Server	Enable/disable web server of the IP phone. Defaults to yes.
Web Server Port	Port number of the web user interface. Defaults to 80.
Enable Web Admin Access	Lets you enable or disable local access to the web user interface. Select yes or no from the drop-down menu. Defaults to yes.
Admin Passwd	Password for the administrator. Defaults to no password.
User Password	Password for the user. Defaults to no password.

Parameter	Description
SPA525-protocol (SPA525G only)	<p>Allows you to choose the type of protocol for the phone:</p> <ul style="list-style-type: none"> ▪ SIP—Session Initiation Protocol. Choose if the phone is used with a SIP call control system, such as the SPA9000 or a SIP call control system from another provider such as BroadSoft or Asterisk. ▪ SPCP—Smart Phone Control Protocol. Choose if the phone is used with a Cisco Unified Communications Series server, such as the Cisco Unified Communications 500 Series for Small Business.
SPA525-auto-detect-sccp (SPA525G only)	Choose if the phone should automatically detect the type of protocol used on the network to which it is connected. If set to yes, the phone automatically discovers if it is connected to a call control system using SPCP.

Internet Connection Type and Static IP Settings

Parameter	Description
Internet Connection Type	<p>Choose the type of internet connection:</p> <ul style="list-style-type: none"> ▪ DHCP ▪ Static IP ▪ PPPoE (not applicable to WIP310)
Static IP	If static IP was chosen as the type of internet connection, displays the static IP address assigned to the phone.
Netmask	If static IP was chosen as the type
Gateway	Default router IP address. Blank if DHCP assigned.
LAN MTU	LAN Maximum Transmission Unit size. Default value: 1500.

Parameter	Description
Duplex Mode	Duplex Mode—Choose one of the following to configure the speed/duplex for the phone's Ethernet ports: <ul style="list-style-type: none">▪ Auto▪ 10Mbps/Duplex▪ 10Mbps/Half▪ 100Mbps/Duplex▪ 100Mbps/Half

PPPoE Settings

Parameter	Description
PPPoE Login Name	Specifies the account name assigned by the ISP for connecting on a Point-to-Point Protocol over Ethernet (PPPoE) link.
PPPoE Login Password	Specifies the password assigned by the ISP for connecting on a Point-to-Point Protocol over Ethernet (PPPoE) link.
PPPoE Service Name	Specifies the service name assigned by the ISP for connecting on a Point-to-Point Protocol over Ethernet (PPPoE) link.

Optional Network Configuration

Parameter	Description
Host Name	The host name of the SPA9000.
Domain	The network domain of the SPA9000.

Parameter	Description
Primary DNS	DNS server used by SPA9000 in addition to DHCP supplied DNS servers if DHCP is enabled; when DHCP is disabled, this is the primary DNS server. Defaults to 0.0.0.0.
Secondary DNS	DNS server used by SPA9000 in addition to DHCP supplied DNS servers if DHCP is enabled; when DHCP is disabled, this is the secondary DNS server. Defaults to 0.0.0.0.
DNS Server Order	Specifies the method for selecting the DNS server. The options are Manual, Manual/DHCP, and DHCP/Manual.
DNS Query Mode	Do parallel or sequential DNS Query. With parallel DNS query mode, the SPA9000 sends the same request to all the DNS servers at the same time when doing a DNS lookup, the first incoming reply is accepted by the SPA9000. Defaults to parallel.
Syslog Server	Specify the syslog server name and port. This feature specifies the server for logging IP phone system information and critical events. If both Debug Server and Syslog Server are specified, Syslog messages are also logged to the Debug Server.
Debug Server	The debug server name and port. This feature specifies the server for logging IP phone debug information. The level of detailed output depends on the debug level parameter setting.
Debug Level	The debug level from 0-3. The higher the level, the more debug information is generated. Zero (0) means no debug information is generated. To log SIP messages, you must set the Debug Level to at least 2. Defaults to 0.
Primary NTP Server	IP address or name of primary NTP server.
Secondary NTP Server	IP address or name of secondary NTP server.

VLAN Settings

**NOTE**

Not applicable to the WIP310.

Parameter	Description
Enable VLAN	Enable this parameter if the switch to which the Cisco IP phone is connected uses VLAN tagging.
VLAN ID	Enter the VLAN tag for the VLAN to which the Cisco IP phone is assigned. This should be the same VLAN to which the SPA9000 is assigned.
Enable CDP	Enables or disables Cisco Discovery Protocol (CDP). Defaults to Yes.

Wi-Fi Settings (SPA525G only)

Parameter	Description
SPA525-wifi-on	Set to yes to enable Wireless-G service on the SPA525G.

Bluetooth Settings (SPA525G only)

Parameter	Description
Enable BT	Set to yes to enable support for Bluetooth devices on the SPA525G.
SPA525-readonly	If set to yes , the Wi-Fi settings on the phone are read only. If set to no , the Wi-Fi settings on the phone can be changed by the end user.

SIP Tab

This section describes the fields for the following headings on the SIP tab:

- [SIP Parameters, page 202](#)
- [SIP Timer Values \(sec\), page 205](#)
- [Response Status Code Handling, page 208](#)
- [RTP Parameters, page 209](#)
- [SDP Payload Types, page 211](#)
- [NAT Support Parameters, page 214](#)
- [Linksys Key System Parameters, page 216](#)

SIP Parameters

Parameter	Description
Max Forward	SIP Max Forward value, which can range from 1 to 255. Defaults to 70.
Max Redirection	Number of times an invite can be redirected to avoid an infinite loop. Defaults to 5.
Max Auth	Maximum number of times (from 0 to 255) a request may be challenged. Defaults to 2.
SIP User Agent Name	Used in outbound REGISTER requests. Defaults to \$VERSION. If empty, the header is not included. Macro expansion of \$A to \$D corresponding to GPP_A to GPP_D allowed.
SIP Server Name	Server header used in responses to inbound responses. Defaults to \$VERSION.

Parameter	Description
SIP Reg User Agent Name	<p>User-Agent name to be used in a REGISTER request. If this is not specified, the <SIP User Agent Name> is also used for the REGISTER request.</p> <p>Defaults to blank.</p>
SIP Accept Language	<p>Accept-Language header used. To access, click the SIP tab, and fill in the SIP Accept Language field.</p> <p>There is no default (this indicates SPA9000 does not include this header). If empty, the header is not included.</p>
DTMF Relay MIME Type	<p>MIME Type used in a SIP INFO message to signal a DTMF event. This field must match that of the Service Provider.</p> <p>Defaults to application/dtmf-relay.</p>
Remove Last Reg	<p>Lets you remove the last registration before registering a new one if the value is different. Select yes or no from the drop-down menu.</p> <p>Defaults to no.</p>
Use Compact Header	<p>Lets you use compact SIP headers in outbound SIP messages. Select yes or no from the drop-down menu. If set to yes, the phone uses compact SIP headers in outbound SIP messages. If set to no, the phone uses normal SIP headers. If inbound SIP requests contain compact headers, the phone reuses the same compact headers when generating the response regardless the settings of the <Use Compact Header> parameter. If inbound SIP requests contain normal headers, the phone substitutes those headers with compact headers (if defined by RFC 261) if <Use Compact Header> parameter is set to yes.</p> <p>Default: no</p>
Escape Display Name	<p>Lets you keep the Display Name private. Select yes if you want the IP phone to enclose the string (configured in the Display Name) in a pair of double quotes for outbound SIP messages. Any occurrences of ' or \ in the string is escaped with \ and \\ inside the pair of double quotes. Otherwise, select no.</p> <p>Defaults to yes.</p>
SIP-B Enable	Enables Sylantro call features.

Parameter	Description
Talk Package	Enables support for the BroadSoft Talk Package, which enables a user to answer or resume a call by clicking a button in an external application.
Hold Package	Enables support for the BroadSoft Hold Package, which enables a user to place a call on hold by clicking a button in an external application.
Conference Package	Enables support for the BroadSoft Conference Package, which enables a user to start a conference by clicking a button in an external application.
Notify Conference	If enabled, the unit will send out a NOTIFY with event=conference when starting a conference.
RFC 2543 Call Hold	<p>If set to yes, unit will include c=0.0.0.0 syntax in SDP when sending a SIP re-INVITE to the peer to hold the call. If set to no, unit will not include the c=0.0.0.0 syntax in the SDP. The unit will always include a=sendonly syntax in the SDP in either case.</p> <p>Defaults to yes.</p>
Random REG CID On Reboot	<p>If set to yes, the Cisco IP phone uses a different random call-ID for registration after the next software reboot. If set to no, the Cisco IP phone tries to use the same call-ID for registration after the next software reboot. The Cisco IP phone always uses a new random Call-ID for registration after a power-cycle, regardless of this setting.</p> <p>Defaults to no.</p>
Mark All AVT packets	<p>If set to yes, all audio video transport (AVT) tone packets (encoded for redundancy) have the marker bit set. If set to no, only the first packet has the marker bit set for each DTMF event.</p> <p>Defaults to yes.</p>
SIP TCP Port Min	Specifies the lowest TCP port number that can be used for SIP sessions. Defaults to 5060.
SIP TCP Port Max	Specifies the highest TCP port number that can be used for SIP sessions. Defaults to 5080.

Parameter	Description
CTI Enable	The CTI interface allows a third-party application to control and monitor the state of a phone that has registered with the SPA9000. With this interface, an application can control a phone to initiate an outgoing call or answer an incoming call with a mouse click from a PC.
Caller ID Header	Provides the option to take the caller ID from PAID-RPID-FROM, P-ASSERTEDIDENTITY, REMOTE-PARTY-ID, or FROM header.
SRTP Method	<p>Selects the method to use for SRTP. Two choices are available:</p> <ul style="list-style-type: none"> ▪ x-sipura—legacy SRPT method ▪ s-descriptor—new method compliant with RFC-3711 and RFC-4568 <p>The default value is "x-sipura."</p> <p>NOTE Not applicable to WIP310.</p>
Hold Target Before REFER	<p>Controls whether to hold call leg with transfer target before sending REFER to the transferee when initiating a fully-attended call transfer (where the transfer target has answered). Default value is "no," where the call leg is not held.</p> <p>NOTE Not applicable to WIP310.</p>

SIP Timer Values (sec)

Parameter	Description
SIP T1	RFC 3261 T1 value (RTT estimate), which can range from 0 to 64 seconds. Defaults to .5 seconds.
SIP T2	RFC 3261 T2 value (maximum retransmit interval for non-INVITE requests and INVITE responses), which can range from 0 to 64 seconds. Defaults to 4 seconds.

Parameter	Description
SIP T4	RFC 3261 T4 value (maximum duration a message remains in the network), which can range from 0 to 64 seconds. Defaults to 5 seconds.
SIP Timer B	INVITE time-out value, which can range from 0 to 64 seconds. Defaults to 16 seconds.
SIP Timer F	Non-INVITE time-out value, which can range from 0 to 64 seconds. Defaults to 16 seconds.
SIP Timer H	INVITE final response, time-out value, which can range from 0 to 64 seconds. Defaults to 16 seconds.
SIP Timer D	ACK hang-around time, which can range from 0 to 64 seconds. Defaults to 16 seconds.
SIP Timer J	Non-INVITE response hang-around time, which can range from 0 to 64 seconds. Defaults to 16 seconds.
INVITE Expires	INVITE request Expires header value. If you enter 0, the Expires header is not included in the request. Ranges from 0 to 19999999999999999999999999999999. Defaults to 240 seconds.
ReINVITE Expires	ReINVITE request Expires header value. If you enter 0, the Expires header is not included in the request. Ranges from 0 to 19999999999999999999999999999999. Defaults to 30
Reg Min Expires	Minimum registration expiration time allowed from the proxy in the Expires header or as a Contact header parameter. If the proxy returns a value less than this setting, the minimum value is used. Defaults to 1 second.
Reg Max Expires	Maximum registration expiration time allowed from the proxy in the Min-Expires header. If the value is larger than this setting, the maximum value is used. Defaults to 7200.
Reg Retry Intvl (See note below)	Interval to wait before the SPA9000 retries registration after failing during the last registration. Defaults to 30.

Parameter	Description
Reg Retry Long Intvl (See note below)	<p>When registration fails with a SIP response code that does not match <Retry Reg RSC>, the SPA9000 waits for the specified length of time before retrying. If this interval is 0, the SPA9000 stops trying. This value should be much larger than the Reg Retry Intvl value, which should not be 0.</p> <p>Defaults to 1200.</p>
Reg Retry Random Delay	<p>Random delay range (in seconds) to add to <Register Retry Intvl> when retrying REGISTER after a failure. This feature was added in Release 5.1.</p> <p>Defaults to blank, which disables this feature.</p>
Reg Retry Long Random Delay	<p>Random delay range (in seconds) to add to <Register Retry Long Intvl> when retrying REGSITER after a failure. This feature was added in Release 5.1.</p> <p>Defaults to blank, which disables this feature.</p>
Reg Retry Intvl Cap	<p>The maximum value to cap the exponential back-off retry delay (which starts at <Register Retry Intvl> and doubles on every REGISTER retry after a failure). In other words, the retry interval is always at <Register Retry Intvl> seconds after a failure. If this feature is enabled, <Reg Retry Random Delay> is added on top of the exponential back-off adjusted delay value. This feature was added in Release 5.1.</p> <p>Defaults to blank, which disables the exponential back-off feature.</p>
Sub Min Expires	This value sets the lower limit of the REGISTER expires value returned from the Proxy server.
Sub Max Expires	This value sets the upper limit of the REGISTER min-expires value returned from the Proxy server in the Min-Expires header. Defaults to 7200.
Sub Retry Intvl	This value (in seconds) determines the retry interval when the last Subscribe request fails. Defaults to 10.



NOTE: Cisco IP phones can use a RETRY-AFTER value when received from a SIP proxy server that is too busy to process a request (503 Service Unavailable message). If the response message includes a RETRY-AFTER header, the phone waits for the specified length of time before retrying to REGISTER again. If a RETRY-AFTER header is not present, the phone waits for the value specified in the *Reg Retry Interval* or the *Reg Retry Long Interval* parameter.

Response Status Code Handling

Parameter	Description
SIT1 RSC	SIP response status code for the appropriate Special Information Tone (SIT). For example, if you set the SIT1 RSC to 404, when the user makes a call and a failure code of 404 is returned, the SIT1 tone is played. Reorder or Busy Tone is played by default for all unsuccessful response status code for SIT 1 RSC through SIT 4 RSC. Defaults to blank.
SIT2 RSC	SIP response status code to INVITE on which to play the SIT2 Tone. Defaults to blank.
SIT3 RSC	SIP response status code to INVITE on which to play the SIT3 Tone. Defaults to blank.
SIT4 RSC	SIP response status code to INVITE on which to play the SIT4 Tone. Defaults to blank.
Try Backup RSC	SIP response code that retries a backup server for the current request. Defaults to blank.

Parameter	Description
Retry Reg RSC	Interval to wait before the SPA9000 retries registration after failing during the last registration. Defaults to blank.

RTP Parameters

Parameter	Description
RTP Port Min	Minimum port number for RTP transmission and reception. Minimum port number for RTP transmission and reception. Should define a range that contains at least 10 even number ports (twice the number of lines); for example, 100 – 106. Defaults to 16384.
RTP Port Max	Maximum port number for RTP transmission and reception. Should define a range that contains at least 10 even number ports (twice the number of lines); for example, 100 – 106. Defaults to 16482.
RTP Packet Size	Packet size in seconds, which can range from 0.01 to 0.16. Valid values must be a multiple of 0.01 seconds. Defaults to 0.030.
Max RTP ICMP Err	Number of successive ICMP errors allowed when transmitting RTP packets to the peer before the SPA9000 terminates the call. If value is set to 0, the SPA9000 ignores the limit on ICMP errors. Defaults to 0.

Parameter	Description
RTCP Tx Interval	<p>Interval for sending out RTCP sender reports on an active connection. It can range from 0 to 255 seconds. During an active connection, the SPA9000 can be programmed to send out compound RTCP packet on the connection. Each compound RTP packet except the last one contains a SR (Sender Report) and a SDES (Source Description). The last RTCP packet contains an additional BYE packet. Each SR except the last one contains exactly 1 RR (Receiver Report); the last SR carries no RR. The SDES contains CNAME, NAME, and TOOL identifiers. The CNAME is set to <User ID>@<Proxy>, NAME is set to <Display Name> (or Anonymous if user blocks caller ID), and TOOL is set to the Vendor/Hardware-platform-software-version (such as Cisco/SPA9000-1.0.3 1(b)). The NTP timestamp used in the SR is a snapshot of the SPA9000's local time, not the time reported by an NTP server. If the SPA9000 receives a RR from the peer, it attempts to compute the round trip delay and show it as the <Call Round Trip Delay> value (ms) in the Info section of SPA9000 web page.</p> <p>Defaults to 0.</p>
No UDP Checksum	<p>Select yes if you want the SPA9000 to calculate the UDP header checksum for SIP messages. Otherwise, select no.</p> <p>Defaults to no.</p>
Symmetric RTP	<p>Enable symmetric RTP operation. If enabled, sends RTP packets to the source address and port of the last received valid inbound RTP packet. If disabled (or before the first RTP packet arrives) sends RTP to the destination as indicated in the inbound SDP.</p> <p>Defaults to no.</p>

Parameter	Description
Stats In BYE	<p>Determines whether the IP phone includes the P-RTP-Stat header or response to a BYE message. The header contains the RTP statistics of the current call. Select yes or no from the drop-down menu. The format of the P-RTP-Stat header is:</p> <p>P-RTP-State: PS=<packets sent>,OS=<octets sent>,PR=<packets received>,OR=<octets received>,PL=<packets lost>,JL=<jitter in ms>,LA=<delay in ms>,DU=<call duration in s>,EN=<encoder>,DE=<decoder>.</p> <p>Defaults to no.</p>

SDP Payload Types

The configured dynamic payloads are used for outbound calls only where the SPA9000 presents the SDP offer. For inbound calls with a SDP offer, the SPA9000 follows the caller dynamic payload type assignments.

The SPA9000 uses the configured codec names in its outbound SDP. The SPA9000 ignores the codec names in incoming SDP for standard payload types (0 – 95). For dynamic payload types, the SPA9000 identifies the codec by the configured codec names. Comparison is case-insensitive.

Parameter	Description
AVT Dynamic Payload	<p>AVT dynamic payload type. Ranges from 96-127.</p> <p>Defaults to 101.</p>
INFOREQ Dynamic Payload	<p>INFOREQ dynamic payload type.</p> <p>Defaults to blank.</p>
G726r16 Dynamic Payload	<p>G.726-16 dynamic payload type. Ranges from 96-127.</p> <p>Defaults to 98.</p> <p>NOTE Not applicable to SPA525G/WIP310.</p>

Parameter	Description
G726r24 Dynamic Payload	G.726-24 dynamic payload type. Ranges from 96-127. Defaults to 97. NOTE Not applicable to SPA525G/WIP310.
G726r32 Dynamic Payload	G726r32 dynamic payload type. The default is 2.
G726r40 Dynamic Payload	G.726-40 dynamic payload type. Ranges from 96-127. Defaults to 96. NOTE Not applicable to SPA525G/WIP310.
G729b Dynamic Payload	G729b Dynamic Payload type. Defaults to 99.
EncapRTP Dynamic Payload	EncapRTP Dynamic Payload type. Defaults to 112.
RTP-Start-LoopbackDynamic	RTP-Start-Loopback Dynamic Payload. Defaults to 113.
RTP-Start-Loopback Codec	RTP-Start-Loopback Codec. Select one of following: G711u, G711a, G726-16, G726-24, G726-32, G726-40, G729a, or G723. SPA525G choices: G711u, G711a, G726-32, G729a, G722. Defaults to G711u.
AVT Codec Name	AVT codec name used in SDP. Defaults to telephone-event.
G711u Codec Name	G.711u codec name used in SDP. Defaults to PCMU.
G711a Codec Name	G.711a codec name used in SDP. Defaults to PCMA.

Parameter	Description
G726r16 Codec Name	G.726-16 codec name used in SDP. Defaults to G726-16. NOTE Not applicable to SPA525G/WIP310.
G726r24 Codec Name	G.726-24 codec name used in SDP. Defaults to G726-24. NOTE Not applicable to SPA525G/WIP310.
G726r32 Codec Name	G.726-32 codec name used in SDP. Defaults to G726-32.
G726r40 Codec Name	G.726-40 codec name used in SDP. Defaults to G726-40. NOTE Not applicable to SPA525G/WIP310.
G729a Codec Name	G.729a codec name used in SDP. Defaults to G729a.
G729b Codec Name	G.729b codec name used in SDP. Defaults to G729ab.
G723 Codec Name	G.723 codec name used in SDP. Defaults to G723. NOTE Not supported on the WIP310 or SPA525G.
EncapRTP Codec Name	EncapRTP codec name used in SDP. Defaults to encaprtsp.

NAT Support Parameters

Parameter	Description
Handle VIA received	<p>If you select yes, the phone processes the received parameter in the VIA header (this is inserted by the server in a response to any of its requests). If you select no, the parameter is ignored. Select yes or no from the drop-down menu.</p> <p>Defaults to no.</p>
Handle VIA rport	<p>If you select yes, the SPA9000 processes the rport parameter in the VIA header (this is inserted by the server in a response to any of its requests). If you select no, the parameter is ignored. Select yes or no from the drop-down menu.</p> <p>Defaults to no.</p>
Insert VIA received	<p>Inserts the received parameter into the VIA header of SIP responses if the received-from IP and VIA sent-by IP values differ. Select yes or no from the drop-down menu.</p> <p>Defaults to no.</p>
Insert VIA rport	<p>Inserts the rport parameter into the VIA header of SIP responses if the received-from IP and VIA sent-by IP values differ. Select yes or no from the drop-down menu.</p> <p>Defaults to no.</p>
Substitute VIA Addr	<p>Lets you use NAT-mapped IP:port values in the VIA header. Select yes or no from the drop-down menu.</p> <p>Defaults to no.</p>
Send Resp To Src Port	<p>Sends responses to the request source port instead of the VIA sent-by port. Select yes or no from the drop-down menu.</p> <p>Defaults to no.</p>
STUN Enable	<p>Enables the use of STUN to discover NAT mapping. Select yes or no from the drop-down menu.</p> <p>Defaults to no.</p>

Parameter	Description
STUN Test Enable	<p>If the STUN Enable feature is enabled and a valid STUN server is available, the SPA9000 can perform a NAT-type discovery operation when it powers on. It contacts the configured STUN server, and the result of the discovery is reported in a Warning header in all subsequent REGISTER requests. If the SPA9000 detects symmetric NAT or a symmetric firewall, NAT mapping is disabled.</p> <p>Defaults to no.</p>
STUN Server	<p>IP address or fully-qualified domain name of the STUN server to contact for NAT mapping discovery.</p>
EXT IP	<p>External IP address to substitute for the actual IP address of the SPA9000 in all outgoing SIP messages. If 0.0.0.0 is specified, no IP address substitution is performed.</p> <p>If this parameter is specified, the SPA9000 assumes this IP address when generating SIP messages and SDP (if NAT Mapping is enabled for that line). However, the results of STUN and VIA received parameter processing, if available, supersede this statically configured value.</p> <p>Defaults to blank.</p>
EXT RTP Port Min	<p>External port mapping number of the RTP Port Min. number. If this value is not zero, the RTP port number in all outgoing SIP messages is substituted for the corresponding port value in the external RTP port range.</p> <p>Defaults to blank.</p>
NAT Keep Alive Intvl	<p>Interval between NAT-mapping keep alive messages.</p> <p>Defaults to 15.</p>

Linksys Key System Parameters

Parameter	Description
Linksys Key System	Enable or disable the Linksys Key System on the IP phone. Defaults to yes.
Multicast Address	The multicast address is used by the SPA9000 to communicate with the SPA IP phones. Defaults to 224.168.168.168:6061.
Force LAN Codec	The choices are: none, G.711u, or G.711a. Defaults to none.

Provisioning Tab

For information about the Provisioning page, see the *Cisco SPA Provisioning Guide*.

Regional Tab

This section describes the fields for the following headings on the Regional tab:

- **Call Progress Tones, page 217**
- **Distinctive Ring Patterns, page 219**
- **Control Timer Values (sec), page 220**
- **Vertical Service Activation Codes, page 221**
- **Vertical Service Activation Codes, page 221**
- **Outbound Call Codec Selection Codes, page 227**
- **Time (SPA525G Only), page 230**

- [Language \(SPA525G only\), page 230](#)
- [Miscellaneous, page 231](#)

Call Progress Tones

Parameter	Description
Dial Tone	Prompts the user to enter a phone number. Defaults to 350@-19,440@-19;10(*0/1+2).
Bluetooth Dial Tone (SPA525G only)	Tone that indicates a bluetooth headset is on and the user can make a call. Defaults to 350@-19,440@-19;1(0/*0);10(*0/1+2).
Outside Dial Tone	Alternative to the Dial Tone. It prompts the user to enter an external phone number, as opposed to an internal extension. It is triggered by a, (comma) character encountered in the dial plan. Defaults to 420@-16;10(*0/1).
Prompt Tone	Prompts the user to enter a call forwarding phone number. Defaults to 520@-19,620@-19;10(*0/1+2).
Busy Tone	Played when a 486 RSC is received for an outbound call. Defaults to 480@-19,620@-19;10(.5/.5/1+2).
Reorder Tone	Played when an outbound call has failed or after the far end hangs up during an established call. Reorder Tone is played automatically when <Dial Tone> or any of its alternatives times out. Defaults to 480@-19,620@-19;10(.25/.25/1+2).
Off Hook Warning Tone	Played when the caller has not properly placed the handset on the cradle. Off Hook Warning Tone is played when Reorder Tone times out. Defaults to 480@10,620@0;10(.125/.125/1+2).

Parameter	Description
Ring Back Tone	<p>Played during an outbound call when the far end is ringing.</p> <p>Defaults to 440@-19,480@-19;*(2/4/1+2).</p>
Call Waiting Tone	<p>Played when a call is waiting. Defaults to 440@-10;30(.3/9.7/1)</p>
Confirm Tone	<p>Brief tone to notify the user that the last input value has been accepted.</p> <p>Defaults to 600@-16; 1(.25/.25/1).</p>
SIT1 Tone	<p>Alternative to the Reorder Tone played when an error occurs as a caller makes an outbound call. The RSC to trigger this tone is configurable on the SIP screen.</p> <p>Defaults to 985@-16,1428@-16,1777@-16;20(.380/0/1,.380/0/2,.380/0/3,0/4/0).</p>
SIT2 Tone	<p>Alternative to the Reorder Tone played when an error occurs as a caller makes an outbound call. The RSC to trigger this tone is configurable on the SIP screen.</p> <p>Defaults to 914@-16,1371@-16,1777@-16;20(.274/0/1,.274/0/2,.380/0/3,0/4/0).</p>
SIT3 Tone	<p>Alternative to the Reorder Tone played when an error occurs as a caller makes an outbound call. The RSC to trigger this tone is configurable on the SIP screen.</p> <p>Defaults to 914@-16,1371@-16,1777@-16;20(.380/0/1,.380/0/2,.380/0/3,0/4/0)</p>
SIT4 Tone	<p>This is an alternative to the Reorder Tone played when an error occurs as a caller makes an outbound call. The RSC to trigger this tone is configurable on the SIP screen.</p> <p>Defaults to 985@-16,1371@-16,1777@-16;20(.380/0/1,.274/0/2,.380/0/3,0/4/0).</p>
MWI Dial Tone	<p>Played instead of the Dial Tone when there are unheard messages in the caller's mailbox.</p> <p>Defaults to 350@-19,440@-19;2(.1/.1/1+2);10(* /0/1+2).</p>
Cfwd Dial Tone	<p>Played when all calls are forwarded.</p> <p>Defaults to 350@-19,440@-19;2(.2/.2/1+2);10(* /0/1+2).</p>

Parameter	Description
Holding Tone	<p>Notifies the local caller that the far end has placed the call on hold.</p> <p>Defaults to 600@-19*(.1/.1/1,.1/1/1,.1/9.5/1).</p>
Conference Tone	<p>Played to all parties when a three-way conference call is in progress.</p> <p>Defaults to 350@-19;20(.1/1/1,.1/9.7/1).</p>
Secure Call Indication Tone	<p>Played when a call has been successfully switched to secure mode. It should be played only for a short while (less than 30 seconds) and at a reduced level (less than -19 dBm) so it does not interfere with the conversation.</p> <p>Defaults to 397@-19,507@-19;15(0/2/0,.2/1/1,.1/2.1/2).</p>
Page Tone	<p>Specifies the tone transmitted when the paging feature is enabled.</p> <p>Defaults to 600@-16;.3(.05/0.05/1).</p>
Alert Tone	<p>Played when an alert occurs.</p> <p>Defaults to 600@-19;.2(.05/0.05/1).</p>
System Beep	<p>Audible notification tone played when a system error occurs.</p> <p>Defaults to 600@-16;.1(.05/0.05/1).</p> <p>NOTE WIP310 and SPA525G only.</p>

Distinctive Ring Patterns

Parameter	Description
Cadence 1	<p>Cadence script for distinctive ring 1.</p> <p>Defaults to 60(2/4).</p>
Cadence 2	<p>Cadence script for distinctive ring 2.</p> <p>Defaults to 60(.3/.2, 1/2,.3/4).</p>

Parameter	Description
Cadence 3	Cadence script for distinctive ring 3. Defaults to 60(.8/.4,.8/4).
Cadence 4	Cadence script for distinctive ring 4. Defaults to 60(.4/.2,.3/.2,.8/4).
Cadence 5	Cadence script for distinctive ring 5. Defaults to 60(.2/.2,.2/.2,.2/.2,1/4)
Cadence 6	Cadence script for distinctive ring 6. Defaults to 60(.2/.4,.2/.4,.2/4).
Cadence 7	Cadence script for distinctive ring 7. Defaults to 60(4.5/4).
Cadence 8	Cadence script for distinctive ring 8. Defaults to 60(0.25/9.75)
Cadence 9	Cadence script for distinctive ring 9. Defaults to 60(.4/.2,.4/2).

Control Timer Values (sec)

Parameter	Description
Reorder Delay	Delay after far end hangs up before reorder tone is played. 0 = plays immediately, inf = never plays. Range: 0–255 seconds. Defaults to 5.
Call Back Expires	Expiration time in seconds of a call back activation. Range: 0–65535 seconds. Defaults to 1800.

Parameter	Description
Call Back Retry Intvl	Call back retry interval in seconds. Range: 0–255 seconds. Defaults to 30.
Call Back Delay	Delay after receiving the first SIP 18x response before declaring the remote end is ringing. If a busy response is received during this time, the SPA9000 still considers the call as failed and keeps on retrying. Defaults to 0.5.
Interdigit Long Timer	Long timeout between entering digits when dialing. The interdigit timer values are used as defaults when dialing. The Interdigit_Long_Timer is used after any one digit, if all valid matching sequences in the dial plan are incomplete as dialed. Range: 0–64 seconds. Defaults to 10.
Interdigit Short Timer	Short timeout between entering digits when dialing. The Interdigit_Short_Timer is used after any one digit, if at least one matching sequence is complete as dialed, but more dialed digits would match other as yet incomplete sequences. Range: 0–64 seconds. Defaults to 3.

Vertical Service Activation Codes

The following Vertical Service Activation Codes are automatically appended to the dial plan.

Parameter	Description
Call Return Code	This code calls the last caller. Defaults to *69.
Blind Transfer Code	Begins a blind transfer of the current call to the extension specified after the activation code. Defaults to *98.

Parameter	Description
Call Back Act Code	Starts a callback when the last outbound call is not busy. Defaults to *66.
Call Back Deact Code	Cancels a callback. Defaults to *86.
Cfwd All Act Code	Forwards all calls to the extension specified after the activation code. Defaults to *72.
Cfwd All Deact Code	Cancels call forwarding of all calls. Defaults to *73.
Cfwd Busy Act Code	Forwards busy calls to the extension specified after the activation code. Defaults to *90.
Cfwd Busy Deact Code	Cancels call forwarding of busy calls. Defaults to *91.
Cfwd No Ans Act Code	Forwards no-answer calls to the extension specified after the activation code. Defaults to *92.
Cfwd No Ans Deact Code	Cancels call forwarding of no-answer calls. Defaults to *93.
CW Act Code	Enables call waiting on all calls. Defaults to *56.
CW Deact Code	Disables call waiting on all calls. Defaults to *57.
CW Per Call Act Code	Enables call waiting for the next call. Defaults to *71.
CW Per Call Deact Code	Disables call waiting for the next call. Defaults to *70.

Parameter	Description
Block CID Act Code	Blocks caller ID on all outbound calls. Defaults to *67.
Block CID Deact Code	Removes caller ID blocking on all outbound calls. Defaults to *68.
Block CID Per Call Act Code	Blocks caller ID on the next outbound call. Defaults to *81.
Block CID Per Call Deact Code	Removes caller ID blocking on the next inbound call. Defaults to *82.
Block ANC Act Code	Blocks all anonymous calls. Defaults to *77.
Block ANC Deact Code	Removes blocking of all anonymous calls. Defaults to *87.
DND Act Code	Enables the do not disturb feature. Defaults to *78.
DND Deact Code	Disables the do not disturb feature. Defaults to *79.
Secure All Call Act Code	Makes all outbound calls secure. Defaults to *16.
Secure No Call Act Code	Makes all outbound calls not secure. Defaults to *17.
Secure One Call Act Code	Makes the next outbound call secure. (It is redundant if all outbound calls are secure by default.) Defaults to *18.
Secure One Call Deact Code	Makes the next outbound call not secure. (It is redundant if all outbound calls are not secure by default.) Defaults to *19.

Parameter	Description
Paging Code	The star code used for paging the other clients in the group. Defaults to *96.
Call Park Code	The star code used for parking the current call. Defaults to *38.
Call Pickup Code	The star code used for picking up a ringing call. Defaults to *36.
Call UnPark Code	The star code used for picking up a call from the call park. Defaults to *39.
Group Call Pickup Code	The star code used for picking up a group call. Defaults to *37.
Media Loopback Code	The star code used for media loopback. Defaults to *03.

Parameter	Description
Referral Services Codes	<p>These codes tell the SPA9000 what to do when the user places the current call on hold and is listening to the second dial tone.</p> <p>One or more *code can be configured into this parameter, such as *98, or *97!*98!*123, etc. Max total length is 79 chars. This parameter applies when the user places the current call on hold (by Hook Flash) and is listening to second dial tone. Each *code (and the following valid target number according to current dial plan) entered on the second dial-tone triggers the SPA9000 to perform a blind transfer to a target number that is prepended by the service *code.</p> <p>For example, after the user dials *98, the SPA9000 plays a special dial tone called the Prompt Tone while waiting for the user to enter a target number (which is checked according to dial plan as in normal dialing). When a complete number is entered, the SPA9000 sends a blind REFER to the holding party with the Refer-To target equals to *98<target_number>. This feature allows the SPA9000 to hand off a call to an application server to perform further processing, such as call park.</p> <p>The *codes should not conflict with any of the other vertical service codes internally processed by the SPA9000. You can empty the corresponding *code that you do not want to SPA9000 to process.</p>

Parameter	Description
Feature Dial Services Codes	<p>These codes tell the SPA9000 what to do when the user is listening to the first or second dial tone.</p> <p>One or more *code can be configured into this parameter, such as *72, or *72!*74!*67!*82, etc. Max total length is 79 chars. This parameter applies when the user has a dial tone (first or second dial tone). Enter *code (and the following target number according to current dial plan) entered at the dial tone triggers the SPA9000 to call the target number prepended by the *code. For example, after user dials *72, the SPA9000 plays a prompt tone awaiting the user to enter a valid target number. When a complete number is entered, the SPA9000 sends a INVITE to *72<target_number> as in a normal call. This feature allows the proxy to process features like call forward (*72) or BLock Caller ID (*67).</p> <p>The *codes should not conflict with any of the other vertical service codes internally processed by the SPA9000. You can empty the corresponding *code that you do not want to SPA9000 to process.</p> <p>You can add a parameter to each *code in Features Dial Services Codes to indicate what tone to play after the *code is entered, such as *72*c!*67*p'. Below are a list of allowed tone parameters (note the use of back quotes surrounding the parameter w/o spaces)</p> <ul style="list-style-type: none"> ▪ c = Cfwd Dial Tone ▪ d = Dial Tone ▪ m = MWI Dial Tone ▪ o = Outside Dial Tone ▪ p = Prompt Dial Tone ▪ s = Second Dial Tone ▪ x = No tones are place, x is any digit not used above
	<p>If no tone parameter is specified, the SPA9000 plays Prompt tone by default.</p> <p>If the *code is not to be followed by a phone number, such as *73 to cancel call forwarding, do not include it in this parameter. In that case, simply add that *code in the dial plan and the SPA9000 send INVITE *73@..... as usual when user dials *73.</p>

Vertical Service Announcement Codes

- Service Annc (Announcement) Base Number: Defaults to blank.
- Service Annc (Announcement) Extension Codes: Defaults to blank.

Outbound Call Codec Selection Codes

These codes automatically appended to the dial-plan. You do not need to include them in the dial-plan.

Parameter	Description
Prefer G711u Code	Makes this codec the preferred codec for the associated call. Defaults to *017110.
Force G711u Code	Makes this codec the only codec that can be used for the associated call. Defaults to *027110.
Prefer G711a Code	Makes this codec the preferred codec for the associated call. Defaults to *017111
Force G711a Code	Makes this codec the only codec that can be used for the associated call. Defaults to *027111.
Prefer G722 Code	Makes this codec the preferred codec for the associated call. Defaults to *01722. NOTE SPA525G only. Only one G.722 call at a time is allowed. If a conference call is placed, a SIP re-invite message is sent to switch the calls to narrowband audio.

Parameter	Description
Force G722 Code	<p>Makes this codec the only codec that can be used for the associated call.</p> <p>Defaults to *02722.</p> <p>NOTE SPA525G only. Only one G.722 call at a time is allowed. If a conference call is placed, a SIP re-invite message is sent to switch the calls to narrowband audio.</p>
Prefer L16 Code	<p>Makes this codec the only codec that can be used for the associated call.</p> <p>Defaults to *01016.</p>
Force L16 Code	<p>Makes this codec the only codec that can be used for the associated call.</p> <p>Defaults to *02016.</p>
Prefer G723 Code	<p>Makes this codec the preferred codec for the associated call.</p> <p>Defaults to *01723.</p> <p>NOTE Not applicable to WIP310 or SPA525G.</p>
Force G723 Code	<p>Makes this codec the only codec that can be used for the associated call.</p> <p>Defaults to *02723.</p> <p>NOTE Not applicable to WIP310 or SPA525G.</p>
Prefer G726r16 Code	<p>Makes this codec the preferred codec for the associated call.</p> <p>Defaults to *0172616.</p> <p>NOTE Not applicable to WIP310 or SPA525G.</p>
Force G726r16 Code	<p>Makes this codec the only codec that can be used for the associated call.</p> <p>Defaults to *0272616.</p> <p>NOTE Not applicable to WIP310 or SPA525G.</p>

Parameter	Description
Prefer G726r24 Code	<p>Makes this codec the preferred codec for the associated call.</p> <p>Defaults to *0172624.</p> <p>NOTE Not applicable to WIP310 or SPA525G.</p>
Force G726r24 Code	<p>Makes this codec the only codec that can be used for the associated call.</p> <p>Defaults to *0272624.</p> <p>NOTE Not applicable to WIP310 or SPA525G.</p>
Prefer G726r32 Code	<p>Makes this codec the preferred codec for the associated call.</p> <p>Defaults to *0172632.</p>
Force G726r32 Code	<p>Makes this codec the only codec that can be used for the associated call.</p> <p>Defaults to *0272632.</p>
Prefer G726r40 Code	<p>Makes this codec the preferred codec for the associated call.</p> <p>Defaults to *0172640.</p> <p>NOTE Not applicable to WIP310 or SPA525G.</p>
Force G726r40 Code	<p>Makes this codec the only codec that can be used for the associated call.</p> <p>Defaults to *0272640.</p> <p>NOTE Not applicable to WIP310 or SPA525G.</p>
Prefer G729a Code	<p>Makes this codec the preferred codec for the associated call.</p> <p>Defaults to *01729.</p>
Force G729a Code	<p>Makes this codec the only codec that can be used for the associated call.</p> <p>Defaults to *02729.</p>

Time (SPA525G Only)

Parameter	Description
Time Zone	Selects the number of hours to add to GMT to generate the local time for caller ID generation. Choices are GMT-12:00, GMT-11:00,..., GMT, GMT+01:00, GMT+02:00, ..., GMT+13:00. Defaults to GMT-08:00.
Time Offset	This specifies the offset from GMT to use for the local system time.
Daylight Saving Time Rule	See “Daylight Saving Time Rule” in Miscellaneous, page 231 .
Daylight Saving Time Enable	Select yes to enable Daylight Saving Time.

Language (SPA525G only)

Parameter	Description
Dictionary Server Script.	See “Dictionary Server Script” in Miscellaneous, page 231 .
Language Selection	See “Language Selection” in Miscellaneous, page 231 .
Language Selection Copy	Choose a subset of the language.
Dictionary Update Status	Status of the dictionary loading.

Miscellaneous

Parameter	Description
Set Local Date (mm/dd)	<p>Sets the local date (mm represents the month and dd represents the day). The year is optional and uses two or four digits.</p> <p>NOTE Not applicable to the SPA525G.</p>
Set Local Time (HH/mm)	<p>Sets the local time (hh represents hours and mm represents minutes). Seconds are optional.</p> <p>NOTE Not applicable to the SPA525G.</p>
Time Zone	<p>Selects the number of hours to add to GMT to generate the local time for caller ID generation. Choices are GMT-12:00, GMT-11:00,..., GMT, GMT+01:00, GMT+02:00, ..., GMT+13:00.</p> <p>Defaults to GMT-08:00.</p> <p>NOTE Found in the Time section for the SPA525G.</p>
Time Offset (HH/mm)	<p>This specifies the offset from GMT to use for the local system time.</p> <p>NOTE Found in the Time section for the SPA525G.</p>

Parameter	Description
Daylight Saving Time Rule	<p>Enter the rule for calculating daylight saving time; it should include the start, end, and save values. This rule is comprised of three fields. Each field is separated by ; (a semicolon) as shown below. Optional values inside [] (the brackets) are assumed to be 0 if they are not specified. Midnight is represented by 0:0:0 of the given date.</p> <p>This is the format of the rule: Start = <start-time>; end=<end-time>; save = <save-time>.</p> <p>The <start-time> and <end-time> values specify the start and end dates and times of daylight saving time. Each value is in this format: <month> /<day> / <weekday>[/HH:[mm[:ss]]]</p> <p>The <save-time> value is the number of hours, minutes, and/or seconds to add to the current time during daylight saving time. The <save-time> value can be preceded by a negative (-) sign if subtraction is desired instead of addition. The <save-time> value is in this format: [/ [+ -]HH:[mm[:ss]]]</p> <p>The <month> value equals any value in the range 1-12 (January-December).</p> <p>The <day> value equals [+ -] any value in the range 1-31.</p> <p>If <day> is 1, it means the <weekday> on or before the end of the month (in other words the last occurrence of < weekday> in that month).</p>

Parameter	Description
	<p>The <weekday> value equals any value in the range 1-7 (Monday-Sunday). It can also equal 0. If the <weekday> value is 0, this means that the date to start or end daylight saving is exactly the date given. In that case, the <day> value must not be negative. If the <weekday> value is not 0 and the <day> value is positive, then daylight saving starts or ends on the <weekday> value on or after the date given. If the <weekday> value is not 0 and the <day> value is negative, then daylight saving starts or ends on the <weekday> value on or before the date given.</p> <p>The abbreviation HH stands for hours (0-23).</p> <p>The abbreviation mm stands for minutes (0-59).</p> <p>The abbreviation ss stands for seconds (0-59).</p> <p>The default Daylight Saving Time Rule is start=4/1/7;end=10/-1/7;save=1.</p> <p>NOTE Found in the Time section for the SPA525G.</p>
Daylight Savings Time Enable	<p>Select yes to enable Daylight Saving Time.</p> <p>NOTE Found in the Time section for the SPA525G.</p>
DTMF Playback Level	<p>Local DTMF playback level in dBm, up to one decimal place.</p> <p>Defaults to -16.</p>
DTMF Playback Length	<p>Local DTMF playback duration in milliseconds.</p> <p>Defaults to .1.</p>
Inband DTMF Boost	<p>Controls the amount of amplification applied DTMF signals.</p> <p>Choices are 0dB, 3dB, 6dB, 9dB, 12dB, 15dB, or 18dB.</p> <p>Defaults to 12dB.</p>

Parameter	Description
Dictionary Server Script/ SCCP Dictionary Server Script (SPA525G)	<p>Defines the location of the dictionary server, the languages available and the associated dictionary. The syntax is as follows:</p> <pre><Dictionary_Server_Script ua="na"> </Dictionary_Server_Script></pre> <p>Defaults to blank and the maximum number of characters is 512. The detailed format is as follows:</p> <pre>serv={server ip port and root path}; d0=<language0>;x0=<dictionary0 filename>; d1=<language1>;x1=<dictionary1 filename>; d2=<language2>;x2=<dictionary2 filename>; d3=<language3>;x3=<dictionary3 filename>; d4=<language4>;x4=<dictionary4 filename>; d5=<language5>;x5=<dictionary5 filename>; d6=<language6>;x6=<dictionary6 filename>; d7=<language3>;x7=<dictionary7 filename>; d8=<language8>;x8=<dictionary8 filename>; d9=<language5>;x9=<dictionary9 filename>;</pre>

Parameter	Description
	<p>The following is an example value:</p> <pre><Dictionary_Server_Script ua="na"> serv=tftp://192.168.1.119/ ;d0=English;x0=enS_v101.xml;d1=Spanish ;x1=esS_v101.xml </ Dictionary_Server_Script></pre> <p>NOTE Not applicable to the WIP310.</p>
SCCP Dictionary Server Copy (SPA525G only)	Choose a subset of the dictionary server.
Language Selection/SCCP Language Selection (SPA525G)	<p>Specifies the default language. The value needs to match one of the languages supported by the dictionary server. The script (dx value) is as follows:</p> <pre><Language_Selection ua="na"> </Language_Selection></pre> <p>Defaults to blank and the maximum number of characters is 512. The following is an example:</p> <pre><Language_Selection ua="na"> Spanish </Language_Selection></pre> <p>NOTE Not applicable to the WIP310.</p>
SCCP Language Selection Copy (SPA525G)	Choose a subset of the language.
Dictionary Update Status (SPA525G)	Status of the dictionary loading.

Phone Tab

This section describes the fields for the following headings on the Phone tab:

- **General, page 236**
- **Line Key, page 239**
- **Miscellaneous Line Key Settings, page 240**
- **Line Key LED Pattern, page 241**
- **Supplementary Services, page 243**
- **Ring Tone (SPA9X2/SPA525G), page 245**
- **Ring Tone (WIP310), page 246**
- **Auto Input Gain (dB), page 246**
- **Extension Mobility, page 247**
- **BroadSoft Settings (SPA525G), page 247**
- **Lightweight Directory Access Protocol (LDAP) Corporate Directory Search, page 249**

General

Parameter	Description
Station Name	Name to identify this station (reserved for future use).
Voice Mail Number	Phone number or URL to check voice mail. Note that The service provider often hosts a voice mail service. The advantages of hosted voice mail include: <ul style="list-style-type: none">▪ Advanced features such as voice mail to email conversion.▪ Calls can go to voice mail when the broadband connection is down.

Parameter	Description
Text Logo	<p>Text logo to display when the phone boots up. A service provider, for example, can enter logo text as follows:</p> <ul style="list-style-type: none"> Up to 2 lines of text Each line must be fewer than 32 characters Insert a new line character (\n) between lines Insert escape code %0a <p>For example, “Super\n%0aTelecom” will display:</p> <p>Super</p> <p>Telecom</p> <p>NOTE Not applicable to the WIP310. On the SPA525G, this setting is located in the User tab. See Screen (SPA525G), page 269.</p>
BMP Picture Download URL	<p>URL locating the bitmap (.BMP) file to display on the LCD background.</p> <p>For more information, see the “Configuring Phone Information and Display Settings” section on page 47.</p> <p>NOTE Not applicable to the WIP310. On the SPA525G, this setting is located in the User tab. See Screen (SPA525G), page 269.</p>
Select Logo	<p>Select from Default, BMP Picture, Text Logo, or None.</p> <p>Defaults to Default.</p> <p>NOTE Not applicable to the WIP310. On the SPA525G, this setting is located in the User tab. See Screen (SPA525G), page 269.</p>
Select Background Picture	<p>Select from Default, BMP Picture, or None.</p> <p>Defaults to Default.</p> <p>NOTE Not applicable to the WIP310. On the SPA525G, this setting is located in the User tab. See Screen (SPA525G), page 269.</p>

Parameter	Description
Screen Saver Enable	<p>Enables a screen saver on the phone's LCD. When the phone is idle for a specified time, it enters screen saver mode. (Users can set up screen savers directly using phone Setup button.)</p> <p>Any button press or on/off hook event triggers the phone to return to its normal mode. (The screen shows "Press any key to unlock your phone.") If a user password is set, the user must enter it to exit screen saver mode.</p> <p>NOTE Not applicable to the WIP310. Screen saver settings are found in the User tab on the SPA525G.</p>
Screen Saver Wait	<p>Amount of idle time before screen saver displays.</p> <p>NOTE Not applicable to the WIP310. Screen saver settings are found in the User tab on the SPA525G.</p>
Screen Saver Icon	<p>In screen saver mode, the phone LCD can display:</p> <p>A background picture.</p> <p>The station time in the middle of the screen.</p> <p>A moving padlock icon. When the phone is locked, the status line displays a scrolling message "Press any key to unlock your phone."</p> <p>A moving phone icon.</p> <p>The station date and time in the middle of the screen.</p> <p>NOTE Not applicable to the WIP310. Screen saver settings are found in the User tab on the SPA525G.</p>
JPEG Logo Download URL (SPA525G)	URL from which to download a .jpg file for the phone logo display.
JPEG Wallpaper Download URL (SPA525G)	URL from which to download a .jpg file for the phone wallpaper.
Enable SMS	<p>Enables sending and receiving of SMS text messages on the phone.</p> <p>NOTE WIP310 only.</p>

Line Key

When used in the configuration profile, parameters in this section must be appended with *n*, where *n* represents line 1, 2, 3, 4, 5 or 6. For more information on these parameters, see the “[Configuring Lines and Extensions](#)” section on [page 33](#).



NOTE Does not apply to the WIP310.

Parameter	Description
Extension	Extension number of the line key.
Short Name	A short label shown on the LCD display for line key 1 through line key 6.
Share Call Appearance	Yes indicates that Line Key 1/2/3/4/5/6 is a shared call appearance. Otherwise this call appearance is not shared (in other words it is private). Defaults to no.
Extended Function	Use to assign Busy Lamp Field, Call Pickup, and Speed Dial Functions to Idle Lines on the SPA9X2 or SPA525G. Syntax is: <code>fnc=type;sub=stationname@\$PROXY;ext=extension#@\$PROXY</code> where: <ul style="list-style-type: none">fnc: functionblf: busy lamp fieldcp: call pickupsub: station nameext: extension

Parameter	Description
Subscribe Expires	Specifies how long the subscription remains valid. After the specified period of time, elapses, the IP phone initiates a new subscription. Defaults to 1800.
Subscribe Retry Interval	Specifies the length of time to wait to try again if subscription fails.
Subscribe Delay	Length of delay before attempting to subscribe. Defaults to 1.
Server Type	Selects the type of server used (SPA9000, BroadSoft, or Asterisk).

Miscellaneous Line Key Settings

**NOTE**

Does not apply to the WIP310.

Parameter	Description
SCA Line ID Mapping	<p>Specifies the shared call appearance line ID mapping. Choose Vertical First or Horizontal First. Each LED can hold two calls and the first call on an LED makes it light up. Horizontal first means the second call makes the same LED flash. Vertical first means the second call lights up the next LED.</p> <p>For example, if Extension 101 is assigned to two LEDs, and Vertical First is selected, the second call on Extension 101 lights up the second LED. The third call makes the first LED flash, and the fourth call makes the second LED flash.</p> <p>If Horizontal First is selected, the second call on Extension 101 makes the first LED flash. The third call lights up the second LED, and the fourth call makes the second LED flash.</p>

Parameter	Description
SCA Barge-In Enable	Enables the SCA Barge-In. Defaults to no.

Line Key LED Pattern

**NOTE**

Does not apply to the WIP310.

Parameter	Description
Idle LED	LED pattern during the Idle state, where the call appearance is not in use and is available to make a new call. Leaving this entry blank indicates the default value of c=g.
Remote Undefined LED	LED pattern during the Remote Undefined state, where the shared call state is undefined (the station is still waiting for the state information from the application server). Not applicable if the call appearance is not shared. Leaving this entry blank indicates the default value of c=r;p=d.
Local Seized LED	LED pattern during the Local Seized state, where this station has seized the call appearance to prepare for a new outbound call. Leaving this entry blank indicates the default value of c=r.
Remote Seized LED	LED pattern during the Remote Seized state, where the shared call appearance is seized by another station. Not applicable if the call appearance is not shared. Leaving this entry blank indicates the default value of c=r;p=d.
Local Progressing LED	LED pattern during the Local Progressing state, where this station is attempting on this call appearance an outgoing call that is in proceeding (i.e. the called number is ringing). Leaving this entry blank indicates the default value of c=r.

Parameter	Description
Remote Progressing LED	LED pattern during the Remote Progressing state, where another station is attempting on this shared call appearance an outbound call that is progressing. Not applicable if the call appearance is not shared. Leaving this entry blank indicates the default value of c=r;p=d.
Local Ringing LED	LED pattern during the Local Ringing state, when the call appearance is ringing. Leaving this entry blank indicates the default value of c=r;p=f.
Remote Ringing LED	LED pattern during the Remote Ringing state, where the shared call appearance is in ringing on another station. Not applicable if the call appearance is not shared. Leaving this entry blank indicates the default value of c=r;p=d.
Local Active LED	LED pattern during the Local Active state, where the call appearance is engaged in an active call. Leaving this entry blank indicates the default value of c=r.
Remote Active LED	LED pattern during the Remote Active state, where another station is engaged in an active call on this shared call appearance. Not applicable if this call appearance is not shared. Leaving this entry blank indicates the default value of c=r;p=d.
Local Held LED	LED pattern during the Local Held state, where the call appearance is held by this station. Leaving this entry blank indicates the default value of c=r;p=s.
Remote Held LED	LED pattern during the Remote Held state, where another station has placed this call appearance on hold. Not applicable if the call appearance is not shared. Leaving this entry blank indicates the default value of c=4,p=s.
Register Failed LED	LED pattern when the corresponding extension has failed to register with the proxy server. Leaving this entry blank indicates the default value of c=a.
Disabled LED	LED pattern when the Call Appearance is disabled (not available for any incoming or outgoing call). Leaving this entry blank indicates the default value of c=o.
Registering LED	LED Pattern when the corresponding extension is trying to register with the proxy server. Leaving this entry blank indicates the default value of c=r;p=s.

Parameter	Description
Call Back Active LED	Call Back operation is currently active on this call appearance is not shared. Leaving this entry blank indicates the default value of c=r;p=s.

Supplementary Services

Enable or disable the corresponding supplementary services on the phone. A value of “yes” indicates enabled; “no” indicates disabled.

Parameter	Description
Conference Serv	Enable/disable Three way conference service. Defaults to yes.
Attn Transfer Serv	Enable/disable attended-call-transfer service. Defaults to yes.
Blind Transfer Serv	Enable/disable blind-call-transfer service. Defaults to yes.
DND Serv	Enable/disable do-not-disturb service. Defaults to yes.
Block ANC Serv	Enable/disable block-anonymous-call service. Defaults to yes.
Call Back Serv	Enable/disable call-back (a.k.a. repeating dialing) service. Defaults to yes.
Block CID Serv	Enable/disable blocking outbound Caller-ID service. Defaults to yes.
Secure Call Serv	Enable/disable secure-call service. Defaults to yes.

Parameter	Description
Cfwd All Serv	Enable/disable call-forward-all service. Defaults to yes.
Cfwd Busy Serv	Enable/disable call-forward-on-busy service. Defaults to yes.
Cfwd On No Ans Serv	Enable/disable call-forward-on-no-answer service. Defaults to yes.
Paging Serv	Enable/disable the paging service. Defaults to yes.
Call Park Serv	Enable/disable the call park service. Defaults to yes.
Call Pick Up Serv	Enable/disable the call pickup service. Defaults to yes.
ACD Login Serv	Enable/disable the ACD Login Service, used for call centers. Typically enabled with the <SIP-B> parameter. Defaults to no.
Group Call Pick Up Serv	Enable/disable the group call pickup service. Defaults to yes.
Group Call Pick Up Serv	Enable/disable the group call pickup service. Defaults to yes.
ACD Ext	The extension used for handling ACD calls. Select from 1, 2, 3, 4, 5, or 6. Defaults to 1.
Service Annc Serv	Enable/disable sending announcement requests to a customer-supplied announcement server. Defaults to no.
Web Serv	Enable/disable the web server. Defaults to yes.

Ring Tone (SPA9X2/SPA525G)

Each entry defines a ring tone to be used on the phone, with an ID between 1 and 10. The ID can be used in a DirEntry to indicate which ring tone to use when the corresponding caller calls.

Parameter	Description
Ring1	Ring tone script for ring 1. Defaults to n=Classic-1;w=3;c=1.
Ring2	Ring tone script for ring 2. Defaults to n=Classic-2;w=3;c=2.
Ring3	Ring tone script for ring 3. Defaults to n=Classic-3;w=3;c=3.
Ring4	Ring tone script for ring 4. Defaults to n=Classic-4;w=3;c=4.
Ring5	Ring tone script for ring 5. Defaults to n=Simple-1;w=2;c=1.
Ring6	Ring tone script for ring 6. Defaults to n=Simple-2;w=2;c=2.
Ring7	Ring tone script for ring 7. Defaults to n=Simple-3;w=2;c=3.
Ring8	Ring tone script for ring 8. Defaults to n=Simple-4;w=2;c=4.
RingRing9	Ring tone script for ring 9. Defaults to n=Simple-5;w=2;c=5.
Ring10	Ring tone script for ring 10. Defaults to n=Office;w=4;c=1.

Ring Tone (WIP310)

Parameter	Description
Keypad Tone	Select yes to enable the keypad tone to be played when a key on the keypad is pressed. Select no to silence the keypad.
Keypad Tone Volume	Corresponds to the volume of the keypad tone. Default is 5.

Auto Input Gain (dB)

**NOTE**

Does not apply to the WIP310.

Parameter	Description
Handset Input Gain	The amount of amplification to apply to the audio input signal for the handset. Defaults to zero.
Headset Input Gain	The amount of amplification to apply to the audio input signal for the headset. Defaults to zero.
Speakerphone Input Gain	The amount of amplification to apply to the audio input signal for the speakerphone. Defaults to zero.
Handset Additional Input Gain	Applies additional input gain to the handset.
Headset Additional Input Gain	Applies additional input gain to the headset.
Speakerphone Additional Input Gain	Applies additional input gain to the speakerphone.

Extension Mobility

You can use extension mobility currently with BroadSoft. For more information, see [Configuring Extension Mobility with a BroadSoft Server, page 84](#).

**NOTE**

Does not apply to the WIP310.

Parameter	Description
Extension Mobility	Enable or disable extension mobility. Defaults to no (disabled).
EM User Domain	The user domain for extension mobility. Defaults to blank.

BroadSoft Settings (SPA525G)

The Cisco SPA525G supports the BroadSoft directory feature and synchronization of Do Not Disturb and Call Forward. The following configuration fields are available:

Parameter	Description
Directory Enable	Set to yes to enable BroadSoft directory for the phone user. Defaults to no.
XSI Host Server	Enter the name of the server; for example, xsp.xdp.broadsoft.com.
Directory Name	Name of the directory. Displays on the user's phone as a directory choice.

Parameter	Description
Directory Type	Select the type of BroadSoft directory: <ul style="list-style-type: none">Enterprise (default): Allows users to search on last name, first name, user or group ID, phone number, extension, department, or email address.Group: Allows users to search on last name, first name, user ID, phone number, extension, department, or email address.Personal: Allows users to search on last name, first name, or telephone number.
Directory UserID	BroadSoft User ID of the phone user; for example, johndoe@xdp.broadsoft.com.
Directory Password	Alphanumeric password associated with the User ID.
Call Feature Sync Ext	<p>Allows the phone to synchronize with the call server so that if Do Not Disturb or Call Forwarding settings are changed on the phone, changes are also made on the server; if changes are made on the server, they are propagated to the phone.</p> <p>This feature is disabled by default.</p> <p>Choose the extension (1 through 5) that is registered to the BroadSoft server.</p>

XML Services (SPA525G)

The Cisco SPA525G supports XML services, such as an XML Directory Service or other XML applications. The following configuration fields are available:

Parameter	Description
XML Directory Service Name	Name of the XML Directory. Displays on the user's phone as a directory choice.
XML Directory Service URL	URL where the XML Directory is located.
XML Application Service	Name of the XML application. Displays on the user's phone as a web application choice.
XML Application Service URL	URL where the XML application is located.

Lightweight Directory Access Protocol (LDAP) Corporate Directory Search



NOTE

Does not apply to the WIP310.

If using Active Directory with authentication set to MD5, you must first configure the following:

- Click the **System** tab. In the **Optional Network Configuration** section, under **Primary DNS**, enter the IP address of the DNS server.
- In the **Optional Network Configuration** section, under **Domain**, enter the LDAP domain.

Parameter	Description
LDAP Dir Enable	Choose yes to enable LDAP.
LDAP Corp Dir Name	Enter a free-form text name, such as “Corporate Directory.”
LDAP Server	Enter a fully qualified domain name or IP address of LDAP server, in the following format: nnn . nnn . nnn . nnn
LDAP Auth Method	Select the authentication method that the LDAP server requires. Choices are: <ul style="list-style-type: none">▪ None—No authentication is used between the client and the server.▪ Simple—The client sends its fully-qualified domain name and password to the LDAP server. May present security issues.▪ Digest-MD5—The LDAP server sends authentication options and a token to the client. The client returns an encrypted response that is decrypted and verified by the server.

Parameter	Description
LDAP Client DN	<p>Enter the distinguished name domain components [dc] ; for example:</p> <p><code>dc=cv2bu,dc=com</code></p> <p>If using the default Active Directory schema (Name(cn)->Users->Domain), an example of the client DN follows:</p> <p><code>cn="David Lee",dc=users,dc=cv2bu,dc=com</code></p>
LDAP Username	Enter the username for a credentialed user on the LDAP server.
LDAP Password	Enter the password for the LDAP username.
LDAP Search Base	<p>Specify a starting point in the directory tree from which to search.</p> <p>Separate domain components [dc] with a comma. For example:</p> <p><code>dc=cv2bu,dc=com</code></p>
LDAP Last Name Filter	This defines the search for surnames [sn], known as last name in some parts of the world. For example, <code>sn:(sn=*\$VALUE*)</code> . This search allows the provided text to appear anywhere in a name, beginning, middle, or end.
LDAP First Name Filter	This defines the search for the common name [cn]. For example, <code>cn:(cn=*\$VALUE*)</code> . This search allows the provided text to appear anywhere in a name, beginning, middle, or end.
LDAP Search Item 3	Additional customized search item. Can be blank if not needed.
LDAP Item 3 Filter	Customized filter for the searched item. Can be blank if not needed.
LDAP Search Item 4	Additional customized search item. Can be blank if not needed.
LDAP Item 4 Filter	Customized filter for the searched item. Can be blank if not needed.

Parameter	Description
LDAP Display Attrs	<p>Format of LDAP results display on phone where:</p> <ul style="list-style-type: none">▪ a—Attribute name▪ cn—Common name▪ sn—Surname (last name)▪ telephoneNumber—phone number▪ n—Display name▪ t—type▪ p—phone number
LDAP Number Mapping	<p>Can be blank if not needed.</p> <p>NOTE With the LDAP number mapping you can manipulate the number that was retrieved from the LDAP server. For example, you can append 9 to the number if your dial plan requires a user to enter 9 before dialing. If you do not manipulate the number in this fashion, a user can use the Edit Dial feature to edit the number before dialing out.</p>

Ext Tab

The Ext tabs vary by phone:

- SPA922 and WIP310: Ext 1 tab
- SPA942: Ext 1, 2, 3, and 4 tabs
- SPA962: Ext 1, 2, 3, 4, 5, and 6 tabs
- SPA525G: Ext 1, 2, 3, 4, and 5 tabs

This section describes the fields for the following Ext tab headings:

- [Share Line Appearance, page 253](#)
- [NAT Settings, page 253](#)
- [Network Settings, page 254](#)
- [SIP Settings, page 255](#)
- [Call Feature Settings, page 258](#)
- [Proxy and Registration, page 260](#)
- [Subscriber Information, page 262](#)
- [Audio Configuration, page 263](#)
- [Dial Plan, page 265](#)

In a configuration profile, the Line parameters must be appended with the appropriate numeral to indicate the line to which the setting applies. For example:

[1] to specify line one
[2] to specify line two

General

Line Enable: To enable this line for service, select yes. Otherwise, select no.

Defaults to yes.

Share Line Appearance

Parameter	Description
Share Ext	Indicates whether this extension is to be shared with other stations or private. If the extension is not shared, then a call appearance assigned to this extension is not shared, regardless the setting of <Share Call Appearance> for that call appearance. If the extension is shared, then whether or not a call appearance assigned to this extension is shared follows the setting of <Share Call Appearance> for that call appearance. The choices are shared or private. Defaults to shared.
Shared User ID	The user identified assigned to the shared line appearance.
Subscription Expires	The subscription time with the SIP proxy (SPA9000). Defaults to 60 seconds.

NAT Settings

Parameter	Description
NAT Mapping Enable	To use externally mapped IP addresses and SIP/RTP ports in SIP messages, select yes. Otherwise, select no. Defaults to no.
NAT Keep Alive Enable	To send the configured NAT keep alive message periodically, select yes. Otherwise, select no. Defaults to no.

Parameter	Description
NAT Keep Alive Msg	Enter the keep alive message that should be sent periodically to maintain the current NAT mapping. If the value is \$NOTIFY, a NOTIFY message is sent. If the value is \$REGISTER, a REGISTER message without contact is sent. Defaults to \$NOTIFY.
NAT Keep Alive Dest	Destination that should receive NAT keep alive messages. If the value is \$PROXY, the messages are sent to the current or outbound proxy. Defaults to \$PROXY.

Network Settings

Parameter	Description
SIP TOS/DiffServ Value	TOS/DiffServ field value in UDP IP packets carrying a SIP message. Defaults to 0x68.
SIP CoS Value	CoS value for SIP messages. Defaults to 3.
RTP TOS/DiffServ Value	ToS/DiffServ field value in UDP IP packets carrying RTP data. Defaults to 0xb8.
RTP CoS Value	CoS value for RTP data. Defaults to 6.

Parameter	Description
Network Jitter Level	<p>Determines how jitter buffer size is adjusted by the SPA9000. Jitter buffer size is adjusted dynamically. The minimum jitter buffer size is 30 milliseconds or (10 milliseconds + current RTP frame size), whichever is larger, for all jitter level settings. However, the starting jitter buffer size value is larger for higher jitter levels. This setting controls the rate at which the jitter buffer size is adjusted to reach the minimum. Select the appropriate setting: low, medium, high, very high, or extremely high.</p> <p>Defaults to high.</p>
Jitter Buffer Adjustment	<p>Controls how the jitter buffer should be adjusted. Select the appropriate setting: up and down, up only, down only, or disable.</p> <p>Defaults to up and down.</p>

SIP Settings

Parameter	Description
SIP Transport	<p>Select from UDP, TCP, or TLS.</p> <p>Defaults to UDP.</p>
SIP Port	<p>Port number of the SIP message listening and transmission port.</p> <p>Defaults to 5060.</p>
SIP 100REL Enable	<p>To enable the support of 100REL SIP extension for reliable transmission of provisional responses (18x) and use of PRACK requests, select yes. Otherwise, select no.</p> <p>Defaults to no.</p>
EXT SIP Port	<p>The external SIP port number.</p>

Parameter	Description
Auth Resync-Reboot	<p>If this feature is enabled, the SPA9000 authenticates the sender when it receives the NOTIFY resync reboot (RFC 2617) message. To use this feature, select yes. Otherwise, select no.</p> <p>Defaults to yes.</p>
SIP Proxy-Require	<p>The SIP proxy can support a specific extension or behavior when it sees this header from the user agent. If this field is configured and the proxy does not support it, it responds with the message, unsupported. Enter the appropriate header in the field provided.</p>
SIP Remote-Party-ID	<p>To use the Remote-Party-ID header instead of the From header, select yes. Otherwise, select no.</p> <p>Defaults to yes.</p>
Referor Bye Delay	<p>Controls when the SPA9000 sends BYE to terminate stale call legs upon completion of call transfers. Multiple delay settings (Referor, Refer Target, Referee, and Refer-To Target) are configured on this screen. For the Referor Bye Delay, enter the appropriate period of time in seconds.</p> <p>Defaults to 4.</p>
Refer-To Target Contact	<p>To contact the refer-to target, select yes. Otherwise, select no.</p> <p>Default: no</p>
Referee Bye Delay	<p>For the Referee Bye Delay, enter the appropriate period of time in seconds.</p> <p>Defaults to 0.</p>

Parameter	Description
SIP Debug Option	<p>SIP messages are received at or sent from the proxy listen port. This feature controls which SIP messages to log. Choices are as follows:</p> <ul style="list-style-type: none"> ▪ none—No logging. ▪ 1-line—Logs the start-line only for all messages. ▪ 1-line excl. OPT—Logs the start-line only for all messages except OPTIONS requests/responses. ▪ 1-line excl. NTFY—Logs the start-line only for all messages except NOTIFY requests/responses. ▪ 1-line excl. REG—Logs the start-line only for all messages except REGISTER requests/responses. ▪ 1-line excl. OPTINTFYIREG—Logs the start-line only for all messages except OPTIONS, NOTIFY, and REGISTER requests/responses. ▪ full—Logs all SIP messages in full text. ▪ full excl. OPT—Logs all SIP messages in full text except OPTIONS requests/responses. ▪ full excl. NTFY—Logs all SIP messages in full text except NOTIFY requests/responses. ▪ full excl. REG—Logs all SIP messages in full text except REGISTER requests/responses. ▪ full excl. OPTINTFYIREG—Logs all SIP messages in full text except for OPTIONS, NOTIFY, and REGISTER requests/responses. <p>Defaults to none.</p>
Refer Target Bye Delay	<p>For the Refer Target Bye Delay, enter the appropriate period of time in seconds.</p> <p>Defaults to 0.</p>
Sticky 183	<p>If this feature is enabled, the IP telephony ignores further 180 SIP responses after receiving the first 183 SIP response for an outbound INVITE. To enable this feature, select yes. Otherwise, select no.</p> <p>Defaults to no.</p>

Parameter	Description
Auth INVITE	When enabled, authorization is required for initial incoming INVITE requests from the SIP proxy.
Ntfy Refer On 1xx-To-Inv	<p>If set to yes, as a transferee, the phone will send a NOTIFY with Event:Refer to the transferor for any 1xx response returned by the transfer target, on the transfer call leg.</p> <p>If set to no, the phone will only send a NOTIFY for final responses (200 and higher).</p> <p>NOTE: Not applicable to the WIP310.</p>
Use Anonymous With RPID	<p>This parameter applies only if <SIP Remote-Party-ID> is set to yes; otherwise, it is ignored.</p> <p>If the parameter is set to yes, the FROM header's display-name and user-id fields are set to anonymous when the caller blocks his caller-id. If the parameter is set to no, the FROM header's display-name and user-id are not masked. The Remote-Party-ID header indicates privacy=full when the caller wishes to block his caller-id.</p> <p>Default: yes.</p> <p>NOTE: Not applicable to the WIP310.</p>
Set G729annexb	<p>Configure G.729 Annex B settings.</p> <p>NOTE Not applicable to SPA525G.</p>

Call Feature Settings

Parameter	Description
Blind Attn-Xfer Enable	<p>Enables the IP phone to perform an attended transfer operation by ending the current call leg and performing a blind transfer of the other call leg. If this feature is disabled, the IP phone performs an attended transfer operation by referring the other call leg to the current call leg while maintaining both call legs. To use this feature, select yes. Otherwise, select no.</p> <p>Defaults to no.</p>

Parameter	Description
MOH Server	<p>User ID or URL of the auto-answering streaming audio server. When only a user ID is specified, the current or outbound proxy is contacted. Music-on-hold is disabled if the MOH Server is not specified.</p> <p>Defaults to imusic when used with a SPA9000 IP PBX.</p>
Message Waiting	<p>Indicates whether the Message Waiting Indicator on the phone is lit. This parameter is toggled by a message from the SIP proxy to indicate if a message is waiting. You can manually modify it to clear or set the flag in the Ext 1-6 tab.</p> <p>Setting this value to Yes can activate stutter tone and VMWI signal. This parameter is stored in long-term memory and survives after reboot or power cycle.</p> <p>Defaults to No.</p>
Auth Page	<p>Specifies whether to authenticate the invite before auto answering a page.</p> <p>Defaults to No.</p>
Default Ring	<p>Type of ring heard. This corresponds to the Ring Tone on the Phone tab. Choose from No Ring, 1 through 10, User 1, or User 2.</p> <p>Defaults to 1.</p>
Auth Page Realm	<p>Identifies the Realm part of the Auth that is accepted when the Auth Page parameter is set to Yes. This parameter accepts alphanumeric characters.</p> <p>Defaults to blank.</p>
Conference Bridge URL	<p>This is the URL used to join into a conference call, generally in the form of the word "conference" or "<u>user@IPaddress:port</u>".</p> <p>Defaults to blank.</p>
Auth Page Password	<p>Identifies the password used when the Auth Page parameter is set to Yes. This parameter accepts alphanumeric characters.</p> <p>Defaults to blank.</p>
Mailbox ID	<p>Identifies the voice mailbox number/ID for the phone.</p> <p>Defaults to blank.</p>

Parameter	Description
Voice Mail Server	Identifies the SpecVM server for the phone, generally the IP address and port number of the VM server.
State Agent	Reserved feature.
CFWD Notify Serv	Specifies whether to enable a SIP-B feature regarding the sending of a Notify to the phone when a call is forwarded elsewhere. Defaults to No.
CFWD Notifier	Typically, this field is configured with the SIP proxy information.

Proxy and Registration

Parameter	Description
Proxy	SIP proxy server and port number set by the Service Provider for all outbound requests. For example: 192.168.2.100:6060.
Use Outbound Proxy	Enable the use of <Outbound Proxy>. If set to no, <Outbound Proxy> and <Use OB Proxy in Dialog> is ignored. Defaults to no.
Outbound Proxy	SIP Outbound Proxy Server where all outbound requests are sent as the first hop.
Use OB Proxy In Dialog	Whether to force SIP requests to be sent to the outbound proxy within a dialog. Ignored if <Use Outbound Proxy> is no or <Outbound Proxy> is empty. Defaults to yes.
Register	Enable periodic registration with the <Proxy>. This parameter is ignored if <Proxy> is not specified. Defaults to yes.

Parameter	Description
Make Call Without Reg	<p>Allow making outbound calls without successful (dynamic) registration by the unit. If no, the dial tone will not play unless registration is successful.</p> <p>Defaults to no.</p>
Register Expires	<p>Allow answering inbound calls without successful (dynamic) registration by the unit. If proxy responded to REGISTER with a smaller Expires value, the phone will renew registration based on this smaller value instead of the configured value. If registration failed with an Expires too brief error response, the phone will retry with the value given in the Min-Expires header in the error response.</p> <p>Defaults to 60.</p>
Ans Call Without Reg	<p>If enabled, the user does not have to be registered with the proxy to answer calls.</p> <p>Defaults to no.</p>
Use DNS SRV	<p>Whether to use DNS SRV lookup for Proxy and Outbound Proxy.</p> <p>Defaults to no.</p>
DNS SRV Auto Prefix	<p>If enabled, the phone will automatically prepend the Proxy or Outbound Proxy name with _sip._udp when performing a DNS SRV lookup on that name.</p> <p>Defaults to no.</p>
Proxy Fallback Intvl	<p>This parameter sets the delay (sec) after which the phone will retry from the highest priority proxy (or outbound proxy) servers after it has failed over to a lower priority server. This parameter is useful only if the primary and backup proxy server list is provided to the phone via DNS SRV record lookup on the server name. (Using multiple DNS A record per server name does not allow the notion of priority and so all hosts will be considered at the same priority and the phone will not attempt to fall back after a fail over).</p> <p>Defaults to 3600</p>

Parameter	Description
Proxy Redundancy Method	<p>Select Normal or Based on SRV port. The phone creates an internal list of proxies returned in the DNS SRV records.</p> <p>If you select Normal, the list contains proxies ranked by weight and priority.</p> <p>If you select Based on SRV, the phone uses normal, then inspects the port number based on the first listed proxy port.</p> <p>Defaults to Normal.</p>

Subscriber Information

Parameter	Description
Display Name	Display name for caller ID.
User ID	Extension number for this line.
Password	<p>Password for this line.</p> <p>Defaults to blank.</p>
Use Auth ID	<p>To use the authentication ID and password for SIP authentication, select yes. Otherwise, select no to use the user ID and password.</p> <p>Defaults to no.</p>
Auth ID	<p>Authentication ID for SIP authentication.</p> <p>Defaults to blank.</p>
Mini Certificate	<p>Base64 encoded of Mini-Certificate concatenated with the 1024-bit public key of the CA signing the MC of all subscribers in the group.</p> <p>Defaults to blank.</p>
SRTP Private Key	<p>Base64 encoded of the 512-bit private key per subscriber for establishment of a secure call.</p> <p>Defaults to blank.</p>

Audio Configuration

A codec resource is considered as allocated if it has been included in the SDP codec list of an active call, even though it eventually may not be the one chosen for the connection. So, if the G.729a codec is enabled and included in the codec list, that resource is tied up until the end of the call whether or not the call actually uses G.729a. If the G.729a resource is already allocated and since only one G.729a resource is allowed per device, no other low-bit-rate codec may be allocated for subsequent calls; the only choices are G711a and G711u. On the other hand, two G.723.1/G.726 resources are available per device.

Therefore it is important to disable the use of G.729a in order to guarantee the support of two simultaneous uses of the G.723/G.726 codecs.

Parameter	Description
Preferred Codec	<p>Preferred codec for all calls. (The actual codec used in a call still depends on the outcome of the codec negotiation protocol.) Select one of the following: G711u, G711a, G726-16, G726-24, G726-32, G726-40, G729a, or G723.</p> <p>NOTE SPA525G choices are: G711u, G711a, G726-32, G729a, and G722.</p> <p>Defaults to G711u.</p>
Use Pref Codec Only	<p>To use only the preferred codec for all calls, select yes. (The call fails if the far end does not support this codec.) Otherwise, select no.</p> <p>Defaults to no.</p>
Second Preferred Codec	<p>The second preferred codec when the preferred codec cannot be used. If <i>Use Pref Codec Only</i> is enabled (set to yes), this parameter is not used.</p> <p>Defaults to Unspecified.</p>
Third Preferred Codec	<p>The third preferred codec when the preferred codec and second preferred codec cannot be used. If <i>Use Pref Codec Only</i> is enabled (set to yes), this parameter is not used.</p> <p>Defaults to Unspecified.</p>

Parameter	Description
G729a Enable	To enable the use of the G.729a codec at 8 kbps, select yes. Otherwise, select no. Defaults to yes.
G723 Enable	To enable the use of the G.723a codec at 6.3 kbps, select yes. Otherwise, select no. Defaults to yes. NOTE G.723.1 is not supported on the SPA525G or WIP310.
G726-16 Enable	To enable the use of the G.726 codec at 16 kbps, select yes. Otherwise, select no. Defaults to yes.
G726-24 Enable	To enable the use of the G.726 codec at 24 kbps, select yes. Otherwise, select no. Defaults to yes.
G726-32 Enable	To enable the use of the G.726 codec at 32 kbps, select yes. Otherwise, select no. Defaults to yes.
FAX CNG Detect Enable	To enable detection of the fax Calling Tone (CNG), select yes. Otherwise, select no. The default is yes.
G726-40 Enable	To enable the use of the G.726 codec at 40 kbps, select yes. Otherwise, select no. Defaults to yes.
FAX Passthru Codec	Select the codec for fax passthrough, G711u or G711a. The default is G711u.
FAX Passthru Method	Select the fax passthrough method: None, NSE, or ReINVITE. The default is NSE.

Parameter	Description
Silence Supp Enable	To enable silence suppression so that silent audio frames are not transmitted, select yes. Otherwise, select no. Defaults to no.
DTMF Tx Method	Select the method to transmit DTMF signals to the far end: InBand, AVT, INFO, Auto, InBand+INFO, or AVT+INFO. InBand sends DTMF using the audio path. AVT sends DTMF as AVT events. INFO uses the SIP INFO method. Auto uses InBand or AVT based on the outcome of codec negotiation. Defaults to Auto.

A codec resource is considered allocated if it has been included in the SDP codec list of an active call, even though it eventually might not be chosen for the connection. If the G.729a codec is enabled and included in the codec list, that resource is tied up until the end of the call whether or not the call actually uses G.729a. If the G.729a resource is already allocated (and since only one G.729a resource is allowed per phone), no other low-bit-rate codec can be allocated for subsequent calls. The only choices are G.711a and G.711u.

Since two G.723.1/G.726 resources are available per SPA9X2, you should disable the use of G.729a to guarantee support for two simultaneous G.723/G.726 codecs.

Dial Plan

The default dial plan script for each line is as follows: (*xx[3469]110|00|[2-9]xxxxxx1xxx[2-9]xxxxxx|xxxxxxxxxxxxxx.).

Parameter	Description
Dial Plan	<p>Dial plan script for this line.</p> <p>The default is (<9:>xx.)</p> <p>(*xx[3469]11000[2-9]xxxxxx1xxx[2-9]xxxxxxS0xxxxxxxxxxxxxx.)</p> <p>The dial plan syntax is expanded in the SPA to allow the designation of three parameters to be used with a specific gateway:</p> <ul style="list-style-type: none"> ▪ uid – the authentication user-id ▪ pwd – the authentication password ▪ nat – if this parameter is present, use NAT mapping <p>Each parameter is separated by a semi-colon (;).</p>
Enable IP Dialing	<p>Enable or disable IP dialing.</p> <p>Defaults to no.</p>

User Tab

This section describes the fields for the following headings on the User tab:

- **Call Forward, page 267**
- **Speed Dial, page 267**
- **Supplementary Services, page 267**
- **Web Information Service Settings (SPA962/SPA525G), page 268**
- **Traffic Service Information Settings (SPA962), page 268**
- **Audio Volume, page 269**
- **Screen (SPA525G), page 269**
- **Phone GUI Menu Color Settings (SPA962 only), page 271**

Call Forward

Parameter	Description
Cfwd All Dest	Enter the extensions to forward calls to.
Cfwd Busy Dest	Enter the extensions to forward calls to when the line is busy. Defaults to voice mail.
Cfwd No Ans Dest	Enter the extension to forward calls to when the call is not answered. Defaults to voice mail.
Cfwd No Ans Delay	Enter the time delay in seconds to wait before forwarding a call that is not answered. Defaults to 20 seconds.

See **Vertical Service Activation Codes, page 221** for more information on call forwarding parameters.

Speed Dial

Speed Dial 2 through 9: Target phone number (or URL) assigned to speed dial 2, 3, 4, 5, 6, 7, 8, or 9.

Defaults to blank.



NOTE

Speed dial configuration has its own tab on the SPA525G and do not appear in this section on the WIP310. Speed dial configuration from the WIP310 is done on the phone.

Supplementary Services

The SPA9000 provides native support of a large set of enhanced or supplementary services. All of these services are optional. Most supplementary service parameters are listed in **Supplementary Services, page 243**.

The user can enable or disable supplementary services and other settings in this section.

A supplementary service should be disabled if the user has not subscribed for it, or the service provider intends to support similar service using other means.

For more star code or supplementary service information, see [Configuring Supplementary Services \(Star Codes\)](#), page 153.

Web Information Service Settings (SPA962/SPA525G)



NOTE: These parameters apply only to the SPA962/SPA525G.

For information about a user setting RSS newsfeeds, see “[Configuring RSS Newsfeeds on the SPA962/SPA525G Phone](#)” section on page 70.

Traffic Service Information Settings (SPA962)



NOTE: These parameters apply only to the SPA962.

Users can change the defaults for local state, city, street, and zip code for local traffic information.

Location defaults to CA, Cisco, San Jose, 95134.



NOTE: Yahoo! Traffic Alerts provides this information to U.S. customers only.

Audio Volume

**NOTE**

Does not apply to the WIP310.

Parameter	Description
Ringer Volume	Sets the default volume for the ringer.
Speaker Volume	Sets the default volume for the full-duplex speakerphone.
Handset Volume	Sets the default volume for the handset.
Headset Volume	Sets the default volume for the headset.

Screen (SPA525G)

Parameter	Description
Screen Saver Enable	Enables a screen saver on the phone's LCD. When the phone is idle for a specified time, it enters screen saver mode. (Users can set up screen savers directly using phone Setup button.) Any button press or on/off hook event triggers the phone to return to its normal mode. (The screen shows "Press any key to unlock your phone.") If a user password is set, the user must enter it to exit screen saver mode.
Screen Saver Type	Choose the type of screen saver: <ul style="list-style-type: none"> ▪ Black Background—Displays a black screen. ▪ Gray Background—Displays a gray screen. ▪ Black/Gray Rotation—The screen incrementally cycles from black to gray. ▪ Picture Rotation—The screen rotates through available pictures on the phone. ▪ Digital Frame—Shows the background picture.

Parameter	Description
Screen Saver Trigger Time	Number of seconds that the phone remains idle before the screen saver turns on.
Screen Saver Refresh Time	Number of seconds before the screen saver should refresh (if, for example, you chose a rotation of pictures).
Text Logo	<p>Text logo to display when the phone boots up. A service provider, for example, can enter logo text as follows:</p> <ul style="list-style-type: none"> Up to 2 lines of text Each line must be fewer than 32 characters Insert a new line character (\n) between lines Insert escape code %0a <p>For example, "Super\n%0aTelecom" will display:</p> <pre>Super Telecom</pre> <p>For more information, see the “Configuring Phone Information and Display Settings” section on page 47.</p>
BMP Picture Download URL	<p>URL locating the bitmap (.BMP) file to display on the LCD background.</p> <p>For more information, see the “Configuring Phone Information and Display Settings” section on page 47.</p>
Logo Type	<p>Select from Default, Download BMP Picture, or Text Logo.</p> <p>Defaults to Default.</p> <p>For more information, see the “Configuring Phone Information and Display Settings” section on page 47.</p>
Background Picture Type	<p>Select from Default, Download BMP Picture, or None.</p> <p>Defaults to Default.</p> <p>For more information, see the “Configuring Phone Information and Display Settings” section on page 47.</p>
LCD Contrast	Enter a number value from 1 to 30. The higher the number, the greater the contrast on the screen.
Back Light Enable	Select yes to enable the screen back light.

Parameter	Description
Back Light Timer (sec)	Enter the number of seconds before the back light should turn off.

Phone GUI Menu Color Settings (SPA962 only)



NOTE: The following parameters apply only to the SPA962.

Choose from:

- Light Blue
- Light Green
- Pink
- Silver

Defaults to Light Blue.

932 Tab (SPA962/SPA525G only)

This tab includes the following sections:

- **General, page 272**
- **Unit 2, page 273**

General

Parameter	Description
Subscribe Expires	Specifies how long the subscription remains valid. After the specified period of time, elapses, the SPA932 initiates a new subscription. Defaults to 1800.
Subscribe Retry Interval	Specifies the length of time to wait to try again if subscription fails.
Unit 1 Enable	Enables or disables the first SPA932 unit (each IP phone can have up to two SPA932s attached).
Subscribe Delay	Length of delay before attempting to subscribe. Defaults to 1.
Unit 2 Enable	Enables or disables the second SPA932 unit (each IP phone can have up to two SPA932s attached).
Server Type	Selects the type of server used (SPA9000, BroadSoft, or Asterisk).
Test Mode Enable	Enables or disables test mode. When test mode is enabled, the LEDs are turned on when keys are pressed, going from off to green to red, and back to off. In test mode, when all the buttons on the SPA932 are returned to off, all the keys become orange. The SPA962 must be rebooted after the test is completed.
Attendant Console Call Pickup Code	The star code used for picking up a ringing call. Defaults to *98.
BLF List URI	Automatically configures BLF subscriptions for all users on a monitored list. See Configuring BroadSoft Busy Lamp Field Auto-Configuration (SPA525G) , page 182.
Unit 1 Key 1-32	Enter a strings that define the extension and other parameters associated with each lighted button on the first SPA932 unit. Keywords and values are case-sensitive. The configuration script is described in “Unit/Key Configuration Scripts” section on page 179 .

For more information, see **“Setting Up the SPA932 Attendant Console” section on page 175.**

Unit 2

See the description for Unit 1 above. Enter a strings that define the extension and other parameters associated with each lighted button on the second SPA932 unit. Keywords and values are case-sensitive. The configuration script is described in **“Setting Up the SPA932 Attendant Console” section on page 175.**

SPA932 Status

This page provides two tabs to display the status of up to two SPA932 units that are supported by a single SPA962:

- Unit 1—Displays information about the first SPA932 unit.
- Unit 2—Displays information about the second SPA932 unit.

Each tab provides the read-only fields described in the following table:

Parameter	Description
Unit Enable	Displays if the Unit is enabled or disabled.
Subscribe Expires	Displays when the current subscription expires. After the subscription expires, the SPA932 automatically requests a new subscription.
HW Version	Displays the version of the hardware.
Unit Online	Displays whether the unit is powered on and connected or not.
Subscribe Retry Interval	Displays the length of time the SPA932 waits to try again if subscription fails.
SW Version	Displays the version of the software currently running on the unit.
Key Name	Displays the name assigned to each key (1-32) on the SPA932 unit.


Parameter	Description
Type	Displays the function enabled for each key (1-32) on the SPA932 unit.
Line	Displays the extension assigned to each key (1-32) on the SPA932 unit.
Station	Displays the subscribe URI configured for each key (1-32) on the SPA932 unit.

SPA525G-Specific Tabs

The following tabs appear on the SPA525G.

Wi-Fi

Enable or disable the Wireless-G device from this tab.

Parameter	Description
Wireless Enable	Click On to enable the wireless controller.
Wi-Fi Device	<p>Choose the method of wireless setup:</p> <ul style="list-style-type: none">▪ Wi-Fi Profile—Create a wireless profile by manually entering information.▪ Wi-Fi Protected Setup—If your router has a WPS button, you can use Wi-Fi Protected Setup to add a new wireless network profile. 
Wireless Status	Contains information about the wireless network.
Wi-Fi Profile	Contains up to 3 wireless profiles for the phone. Includes a wireless profile for the Cisco Unified Communications Server by default.

Bluetooth (SPA525G)

Enable Bluetooth on the SPA525G by clicking **On**.

Personal Address Book

Address book for the phone. For more information, see the *Cisco Small Business Pro IP Phone SPA525G User Guide*.

Call History

Displays the call history for the phone. To change the information displayed, select the type of call history from the drop-down list:

- All Calls
- Received Calls
- Placed Calls
- Missed Calls

Speed Dials

See [Speed Dial](#), page 267.

Firmware Upgrade

Used to upgrade the firmware for the SPA525G. See [Upgrading Firmware](#), page 20.



Where to Go From Here

Cisco provides a wide range of resources to help you obtain the full benefits of the Cisco SPA and Wireless IP Phones.

Product Resources

Website addresses in this document are listed without **http://** in front of the address because most current web browsers do not require it. If you use an older web browser, you may have to add **http://** in front of the web address.

Resource	Location
Technical Documentation	www.cisco.com/en/US/products/ps10033/tsd_products_support_series_home.html
Firmware Downloads	Go to www.tools.cisco.com/support/downloads , and enter the model number in the Software Search box.
Cisco Community Central > Small Business Support Community	www.myciscocommunity.com/community/smallbizsupport/voiceandconferencing/voice
Phone Support	www.cisco.com/en/US/support/tsd_cisco_small_business_support_center_contacts.html
Warranty and End-User License Agreement	www.cisco.com/go/warranty
Open Source License Notices	www.cisco.com/go/osln

Resource	Location
Regulatory Compliance and Safety Information	www.cisco.com/en/US/products/ps10033/tsd_products_support_series_home.html
Cisco Partner Central (Login Required)	www.cisco.com/web/partners/sell/smb
Cisco Small Business Home	www.cisco.com/smb

Related Documentation

See document...	To learn how to...
<i>SPA9000 Voice System Installation and Configuration Guide</i>	For VARs and Service Providers, describes: <ul style="list-style-type: none">▪ Network design considerations and site preparation▪ Switch configuration▪ Initial installation and configuration of the SPA9000 components▪ SPA9000, SPA400, SPA9X2 series IP phones
<i>SPA9000 Voice System Administration Guide</i>	For VARs and Service Providers. describes: <ul style="list-style-type: none">▪ SPA9000 voice system management▪ SPA9000 IP PBX configuration▪ SPA400 phone gateway and voice mail server configuration
<i>SPA9X2 IP Phone User Guide</i>	For VARs and end phone users, describes: <ul style="list-style-type: none">▪ SPA922, SPA942, SPA962 and SPA932 attendant console▪ Phone set up▪ Phone features

See document...	To learn how to...
<i>Cisco Small Business Pro IP Phone SPA525G (SIP) User Guide</i>	For VARs and end phone users, describes the installation, configuration, and use of the Cisco Small Business Pro IP Phone SPA525G for SIP call control systems.
<i>Cisco Unified Communications Manager Express for Cisco Unified IP Phone SPA525G User Guide</i>	For VARs and end phone users, describes the installation, configuration, and use of the Cisco Small Business Pro IP Phone SPA525G for Cisco Unified Communications call control systems.
<i>Cisco WIP310 Wireless-G IP Phone User Guide</i>	For VARs and end phone users, describes: <ul style="list-style-type: none">▪ WIP310▪ Phone set up▪ Phone features
<i>ATA Administration Guide</i>	For VARs, system administrators, and Service Providers, describes: <ul style="list-style-type: none">▪ Administration and use of ATAs▪ PAP2T, SPA2102, SPA3102, SPA8000, AG310, RTP300, WRP400, and W RTP54G
<i>SPA Provisioning Guide</i>	For Cisco service providers: <ul style="list-style-type: none">▪ Describes automated provisioning features▪ Requires partner login