Linksys SPA9x2 Phone Administration Guide

Version 5.2

ADMINISTRATION GUIDE   SPA922, SPA932, SPA942, SPA962

LINKSYS®
A Division of Cisco

CISCO™

Interim version released June 2008; to be updated August 2008

Introducing Linksys SPA9x2 Phones

This manual is for Linksys partners, value added resellers (VARs) and Internet service providers. Use this manual to configure Linksys 9x2 series phones and users.

NOTE: This manual assumes that your SPA9x2 phones are installed and connected to the network. If not, see "Related Documentation" section on page 2 for installation information.

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 Text in a PDF

By default, the Find toolbar is open. If it has been closed, choose **Edit > Find**.

Use Find to search for text in an open PDF.
1. Enter your search terms in the *Find* box on the toolbar.

2. Optionally click the arrow next to the Find text box to refine your search (such as Whole words only).

3. Press **Enter**. Acrobat jumps to the first instance of the search term. Pressing **Enter** again continues to more instances of the term.

Finding Text in Multiple PDFs

The *Search* window lets you search for terms in multiple PDFs. The PDFs do not need to be open. Either:

- Choose **Edit** > **Search**

    or

- Click the arrow next to the *Find* box and choose Open Full Acrobat Search. The *Search* window appears

In the *Search* window:

1. Enter the text you want to find.

2. Choose **All PDF Documents in.**

Download from Www.Somanuals.com. All Manuals Search And Download.

3. From the drop-down box, choose **Browse for Location**.

4. Choose the location you want to search, either on your computer or on a network, then click **OK**.

5. If you want to specify additional search criteria, click **Advanced Search Options**, and choose the options you want.

6. Click **Search.**

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

Related Documentation

Refer to--and search--the following documents for additional information.

See document...	To Learn how to...
Linksys Voice System Installation and Configuration Guide	For VARs and Service Providers, describes: • Network design considerations and site preparation • Switch configuration • Initial installation and configuration of the LVS components • SPA9000, SPA400, SPA900 series IP phones
Linksys Voice System Administration Guide	For VARs and Service Providers. describes: • LVS management • SPA9000 IP PBX configuration • SPA400 phone gateway and voice mail server configuration
Linksys SPA9x2 Phone User Guide	For VARs and end phone users, describes: • SPA922, SPA942, SPA962 and SPA932 sidecar • Phone set up • Phone features
Linksys ATA Administration Guide	For VARs, system administrators, and Service Providers, describes: • Administration and use of Linksys ATAs • PAP2T, SPA2102, SPA3102, SPA8000, AG310, RTP300, WRP400, and WRTP54G
Linksys SPA Provisioning Guide	For Linksys service providers: • Describes automated provisioning features • Requires partner login

For additional documents and tools related to SPA9x2 phones, see linksys.com and its partner sections. For Europe, Middle East and Africa, see linksys-voip.eu (VARs) and linksys-itsp.com (service providers).

For troubleshooting information, go to linksys.com and click on **Support** > **Technical Support.**

Getting Started

Before you can configure a SPA9x2 phone, you need to install it by connecting the phone to a network (Ethernet) port. For detailed installation information, see the *Linksys SPA9x2 Phone User Guide.*

To get started viewing and configuring phones, see:

- "Basic Phone Features" section on page 3
- "Additional Phone Features" section on page 4
- "SPA9x2 Phone Configuration Scenarios" section on page 6
- "SPA9000 IP PBX System" section on page 7
- "Session Initiation Protocol and SPA9x2 Phones" section on page 9
- "Using SPA9x2 Phones with a Firewall or Router" section on page 10
- "Network Address Translation and SPA9x2 Phones" section on page 11
- "Simple Traversal of UDP Through NATs (STUN)" section on page 13
- "SIP-NAT Interoperation" section on page 13
- "Using SPA9x2 Phones in a VLAN" section on page 14

Basic Phone Features

All SPA9x2 phones support:

- Power over Ethernet (PoE) (802.3af-compliant)
- Two 100 BaseT Ethernet LAN ports (one PC and one WAN)
- Quality of service (QoS)
- Phone power consumption: 6 Watts (varies)

NOTE: If you are not using a PoE switch, you can use the PA100 power supply.

Phone model differences are shown in the following table:

SPA Model	Voice Lines	High-Resolution Graphical Display
SPA922	1	Backlit 128 x 64 pixels with screen saver and customizable logo and background picture (128 x 48 pixels available for background picture).
SPA 942	4	Backlit 128 x 64 pixels with screen saver and customizable logo and background picture (128 x 48 pixels available for background picture).
SPA962	6	Color 320 x 240 pixels with screen saver and customizable logo and background picture (320 x 240 pixels available for background picture). Also provides customizable photo album.

SPS932 Sidecar Attendant for the SPA962

The SPA932 Sidecar for the SPA962 phone 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 sidecars to a SPA962, for 64 extensions. For more information, see Chapter 10, "Using the SPA932 Sidecar."

Additional Phone Features

For a complete list of SPA9x2 features, see the phone data sheets. Included here are some additional features:

- Media Loopback: service providers can use media loopback to measure the voice quality experienced by the end user. One device acts as the audio transmitter and receiver, while the other device acts as the audio mirror. The audio mirror transmits the audio packets received back to the transmitter for testing.

- Remote Provisioning from Phone Keypad: after the user enters the IP address of the provisioning server, the phone resynchronizes to a known path name. This feature enables service providers to have VARs install and provision Linksys phones. See Chapter 5, "Provisioning Basics."

- Extension Mobility, which lets users log in to a phone. The login configures the phone with the user's information, including user name and caller ID.

  Note: This feature is currently only integrated with Broadsoft Client Management System (CMS).

- Star codes/supplementary services

- Service announcement codes

- Codec priority order of up to three preferred codecs

- Full-duplex speakerphone:

- Switching of lower microphone gain to reduce saturation.

- Noise floor estimation using voice activity detection (VAD).

- Phone LCD features:

    - The screen can display the header of the 18X call progress message instead of displaying "Called Party Ringing."

    - A screen saver can appear when the phone is in an idle state for a period of time.

    - A customized service provider logo can be downloaded to display during boot-up.

    - A customized background picture can be downloaded to display on SPA942 and SPA962 phones.

    - The SPA962 can display photos in a "photo album effect" using the HTTP REFRESH feature.

- Emergency Numbers

    - An administrator can enter a list of emergency numbers in the Emergency Number parameter. When one of these numbers is dialed on an IP phone, the unit disables processing of "CONF", "HOLD", and other similar softkeys or buttons to avoid accidentally putting the current call on hold. In addition, only the far end can terminate an emergency call. See .

- GUI Localization: SPA9x2 phone menus can display in languages other than English.

    - Additional Character Sets Supported on Phones include Latin2 (Czech, Hungarian, Polish, Romanian, Croatian, Slovak, Slovenian, Sorbian), 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.

- CALEA Compliance : To ensure compliance with the Communications Assistance for Law Enforcement Act (CALEA) in the United States, privacy setting parameters have been modified for the SPA9x2. If a user's privacy settings are activated, the "From" value is not changed to Anonymous. Instead, the switch equipment masks the caller ID, and the calls can be tracked as required by US government regulations.

Features Requring SIP Support

The following telephony features are provided by the different models of SPA9x2 phones. An asterisk (*) indicates that the feature requires support by the SIP server.

- Reception of Text Messages via SIP: SPA9x2 phones can display text messages according to RFC3428. The *Text Message* and *Text Message From 3rd Party* parameters enable and disable the display of messages up to 255 characters in length. When this feature is enabled, a received message appears on the phone display with the date and time. For more information, see .

- The SPA9x2 phones can support and display up to 50 characters for *Display Name, Password, Auth ID,* and *User ID* parameters.

- Shared Line Appearance

    – SPA922: Two call appearances

    – SPA942: Four call appearances

    – SPA962: Six call appearances

- Call Pick Up - Selective and Group

- Call Park and UnPark

- Music on Hold

- Intercom

- Group Paging

For a list of phone features available to the end user, see the *Linksys SPA9x2 Phone User Guide*.

SPA9x2 Phone Configuration Scenarios

You can configure SPA9x2 phones in different ways:

- Phones connected directly to a service provider (IP Centrex)

- Phones connected to an IP PBX such as the SPA9000

Your configuration can contain:

- Optional SPA400 for voicemail and a landline connection (if using LVS/SPA9000)

- WBP54 wireless bridge for wireless networks

- Linksys ATA, such as the PAP2T (if not using SP9000)

The following diagram shows a possible configuration scenario.
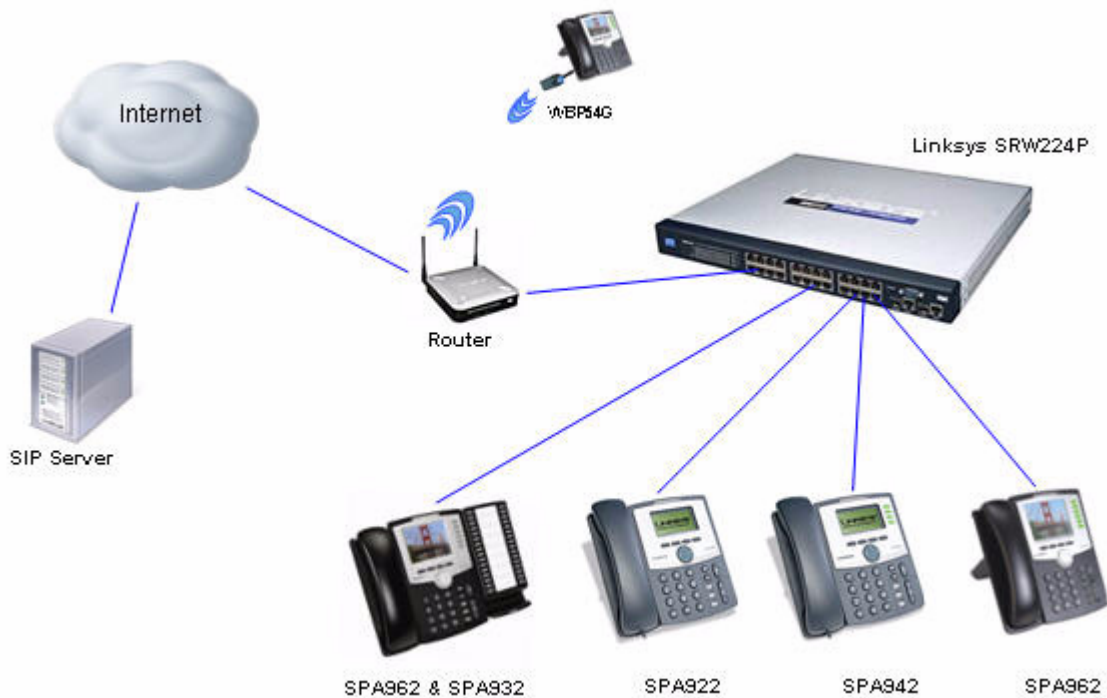
SPA9000 IP PBX System

The SPA9000 IP PBX System (VoIP side), along with the SPA400, which is Internet Telephony Gateway, provides for an analog line and voicemail. When you use SPA9x2 phones with the LVS SPA9000 IP PBX 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 perform SPA9000 administrative tasks using an Interactive Voice Response (IVR) system, the LVS Wizard, or a built-in web server. For more information, see the *Linksys Voice System Administration Guide.*
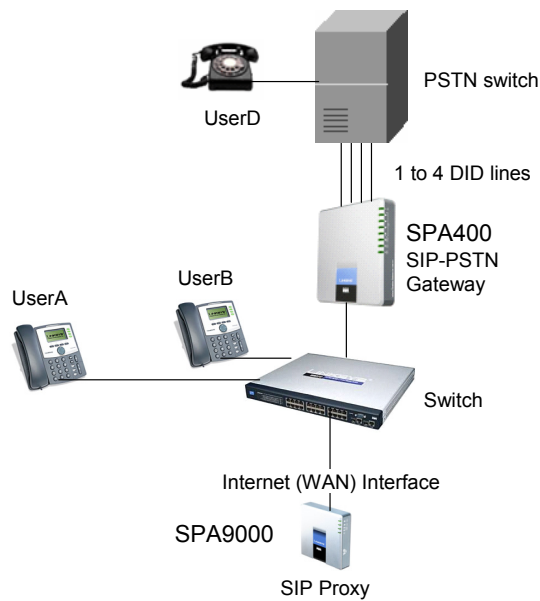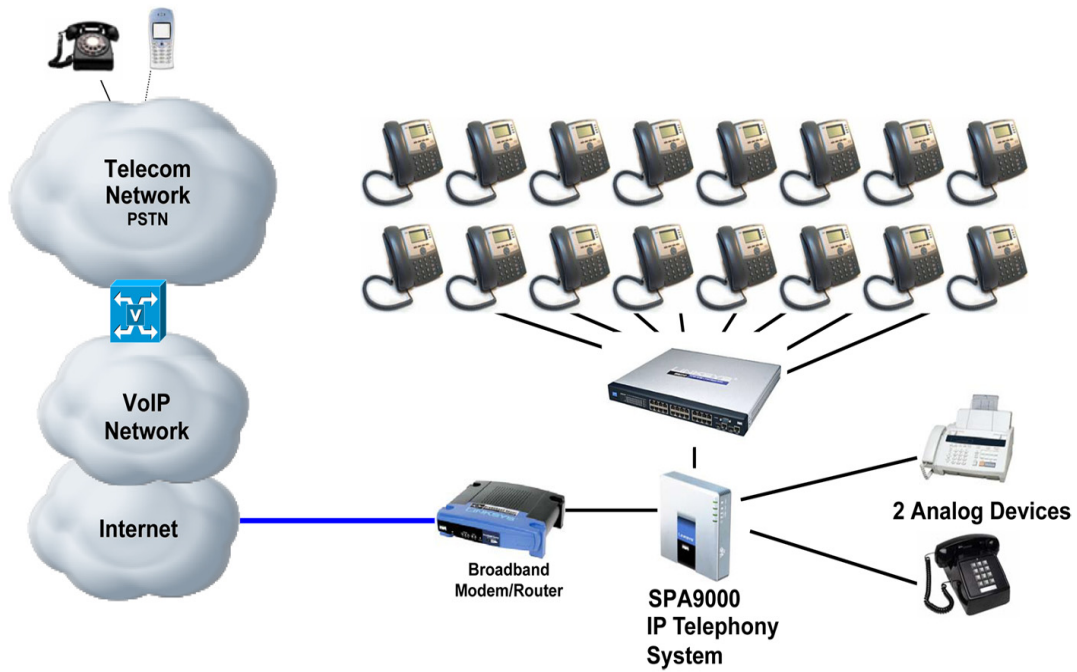
Session Initiation Protocol and SPA9x2 Phones

Linksys SPA9x2 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).
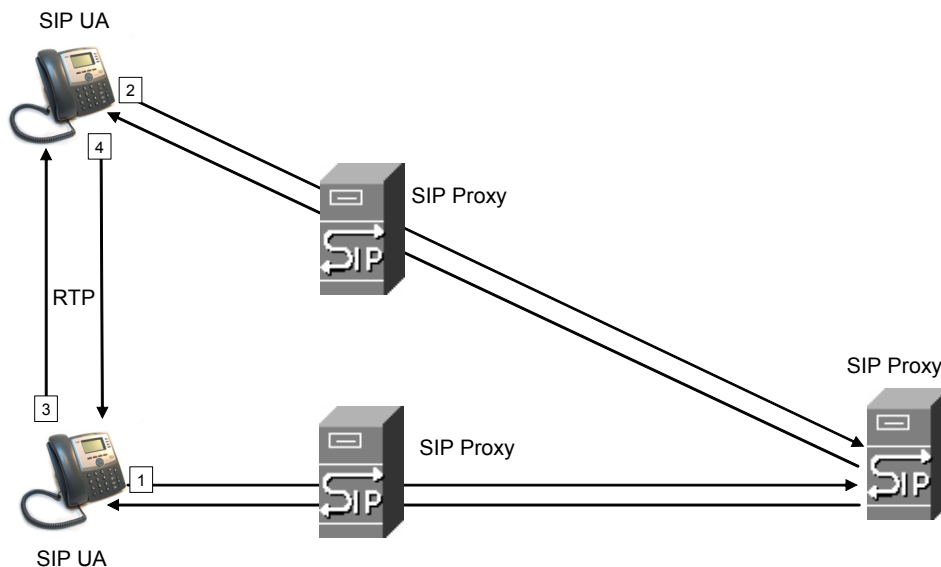
SIP message routing is dynamic. If a SIP proxy receives a request from a UAS for a connection but cannot locate the UAC, the proxy forwards the message to another SIP proxy in the network. When the UAC is located, the response is routed back to the UAS, and a direct peer-to-peer session is established 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 "Simple Traversal of UDP Through NATs (STUN)" section on page 13.

SIP Over TCP

To guarantee state-oriented communications, SPA9x2 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. Linksys 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 SPA9x2 phone.

A simple way to support proxy redundancy is to configure a SIP proxy server in the SPA9x2 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 SPA9x2 phone tries to contact the hosts in the order of their priority.

If the SPA9x2 currently uses a lower-priority proxy server, the phone periodically probes the higher-priority proxy and switches to the higher-priority proxy when available.

The dynamic nature of SIP message routing makes the use of a static list of proxy servers inadequate in some scenarios. In deployments where user agents are served by different domains, for instance, it would not be feasible to configure one static list of proxy servers per covered domain into every SPA9x2 phone.

One solution to this situation is through the use of DNS SRV records. SPA9x2 phones can be instructed to contact a SIP proxy server in a domain named in SIP messages. The SPA9x2 phone consults the DNS server to get a list of hosts in the given domain that provides SIP services. If an entry exists, the DNS server returns a service (SRV) record that contains a list of SIP proxy servers for the domain, with their host names, priority, listening ports, and so on. The SPA9x2 phone tries to contact the list of hosts in the order of their stated priority.

If the SPA9x2 phone is currently using a lower priority proxy server, it periodically probes the higher priority proxy to see whether it is back on line, and attempts to switch back to the higher priority proxy whenever possible.

See for more information.

Using SPA9x2 Phones with a Firewall or Router

When using a SPA9x2 behind a firewall or router, make sure the following ports are not blocked:

- SIP ports—By default, UDP port 5060 through 5064

- RTP ports—16384 to 16482

If security is not a concern, you can disable stateful packet inspection (SPI) on your firewall (if you have it). SPI allows a firewall to be aware of a packet's state; only recently-sent requests are allowed into the network.
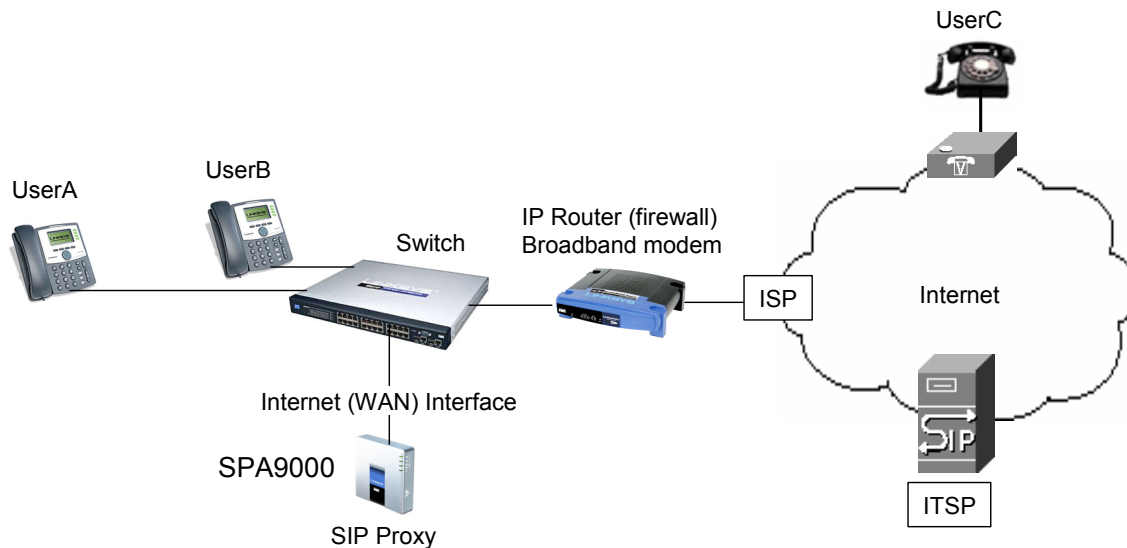
Network Address Translation and SPA9x2 Phones

In a typical application of network address translation (NAT), all devices in a subscriber network access the Internet through a router with a single public IP address. The IP address is assigned by a service provider. The IP header of the packets sent from the private network to the public network is substituted by NAT with the public IP address and a port assigned by the router. The receiver of the packets on the public network sees the packets as coming from the external address instead of the private address of the device.

You can implement NAT in three ways:

- Full cone NAT (one-to-one NAT)— All requests from the same internal IP address and port are mapped to the same external IP address and port. An external host can send a packet to the internal host by sending a packet to the mapped external address

- Restricted cone NAT—All requests from the same internal IP address and port are mapped to the same external IP address and port. Unlike a full cone NAT, an external host cannot send a packet to the internal host unless the internal host previously sent a packet to it.

- Port-restricted cone NAT (symmetric NAT)—Similar to restricted cone NAT, but the restriction includes port numbers. An external host can send a packet to a particular port on the internal host only if the internal host previously sent a user datagram protocol (UDP) packet from that port to the external host. UDP is a connectionless messaging protocol for delivery of data packets.

See the following topics:

-

- "Session Border Controllers" section on page 12

- "Simple Traversal of UDP Through NATs (STUN)" section on page 13

- "SIP-NAT Interoperation" section on page 13

- "Determining the Type of NAT Used on Your Router" section on page 14

Routers and Service Provider Support of NAT

NAT allows multiple devices to share the same public, routable, IP address for establishing connections over the Internet. A router forwards packets between the Internet and the internal, private network.

NAT mapping is the association between a private address and port and a public address and port . The mapping is maintained for a short period of time, varying from a few seconds to several minutes. The mapping time is extended when a source device sends a packet.

Session Border Controllers

The service provider can support NAT mapping using a session border controller. With a session border controller, you do not have to manage NAT on the SPA9x2.

If the service provider does not support session border control, use the NAT Support Parameters provided by the SPA9x2, such as:

- Outbound Proxy (see "Proxy and Registration Parameters" section on page 100)

- STUN Server Enable (see "NAT Support Parameters" section on page 52)

- EXT IP
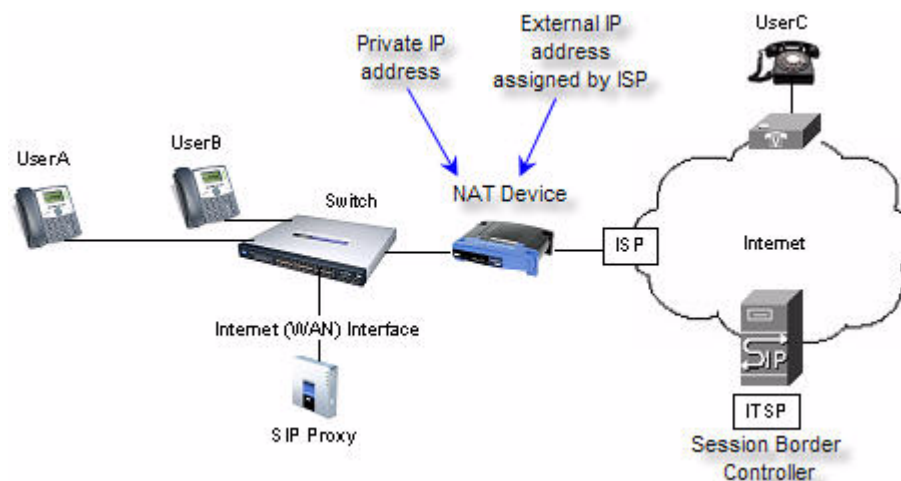
Simple Traversal of UDP Through NATs (STUN)

Simple Traversal of User Datagram Protocol (UDP) through Network Address Translators (NATs), or STUN, is a protocol defined by RFC 3489 that allows a client behind a NAT device to determine the:

- presence and types of NATs and firewalls between them and the public Internet

- public IP addresses allocated to them by the NAT

- port associations

STUN works with many existing NATs and allows for a wide variety of applications to work through existing NAT infrastructure.This information helps set up UDP communication between two hosts that are both behind NAT routers.

NOTE: STUN does not work with a symmetric NAT router.

To determine the type of router you have, see "Determining the Type of NAT Used on Your Router" section on page 14.

You can get open-source STUN software at the following website:

http://www.voip-info.org/wiki-STUN > "Public STUN Servers" section.

Use any public STUN server from the link above when configuring SIP > NAT Support Parameters > STUN Server. See "NAT Support Parameter Notes" section on page 53.

SIP-NAT Interoperation

The addresses where messages/data are sent to a SPA9x2 system are embedded in the SIP messages sent by the device. If the SPA9x2 is located behind a NAT device, its private IP address is not usable for communications with SIP devices outside the private network.

NOTE: A service provider can offer an outbound NAT-Aware proxy, which discovers the public IP address from the remote endpoint, eliminating the need to modify the SIP message from the UAC.
To communicate with a specific public peer address/port, the SPA9x2 must substitute the private address/port with the external IP address/port. The SPA9x2 performs the following:

- Discovers the NAT mappings used to communicate with the peer.

  A STUN server responds to a special NAT-Mapping-Discovery request by sending back a message to the source IP address/port. The SPA9x2 can send this request when it first tries to communicate with a SIP device over the Internet. It then stores the mapping discovery results returned by the server.

- Communicates the NAT mapping information to the external SIP devices.

  - If the device is a SIP Registrar, the information is carried in the Contact header that overwrites the private address/port information.

- If the device is another SIP UA, the information is carried in the Contact header and the Session Description Protocol (SDP) embedded in SIP message bodies. The VIA header in outbound SIP requests might also need to be substituted with the public address if the UAS relies on it to route back responses.

  NOTE: SDP defines a text-based format for describing streaming media sessions and multicast transmissions. SDP is not a transport protocol but a method of describing the details of the transmission. For example, an SDP file contains information about the format, timing and authorship of the transmission, name and purpose of the session, any media, protocols or codec formats, the version number, contact information and broadcast times.

- Extends the discovered NAT mappings by sending keep-alive packets. Because the mapping is alive only for a short period, the SPA9x2 continues to send periodic keep-alive packets through the mapping, as needed.

Determining the Type of NAT Used on Your Router

To determine the type of NAT your router uses, enable debugging on the SPA9x2:

1. Make sure you do not have firewall running on your PC that could block the syslog port (by default this is 514).

2. Log on to the phone's web UI. For information about this, see

3. Click Admin **Login > advanced**.

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 SPA9x2. This port number appears on the output file name. The default port number is 514.

5. Set *Debug Level* to **3**. Do not change the value of the *Syslog Server* parameter.

6. To capture SIP signaling messages, click the **Ext** tab.

7. Set *SIP Debug Option* to **Full**. The default output is named syslog.514.log (if not port number was specified).

8. To collect information about what type of NAT your router uses click the **SIP** tab and scroll to NAT Support Parameters.

9. Back in the *System* 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."

Using SPA9x2 Phones in a VLAN

If you use a VLAN your SPA9x2 voice packets are tagged with the VLAN ID.

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 SPA9x2 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. To enter VLAN settings, see "Configuring VLAN Settings" section on page 44.

Getting Started

Read this chapter to learn about:

From there you can begin using the web UI to view and change phone, extension, and user information.

Determining SPA9x2 Firmware Version

Perform the following steps to determine the phone's current firmware version:

1. Press the **Menu** button on the phone.

2. Press **10**, or scroll down until **Product Info** is highlighted and then press the **select** soft key.

3. Press **3** from the *Product Info* menu or scroll down to highlight **Software Version** and press the **select** soft key.

Downloading the Firmware

To download the firmware, you need a computer with an active Internet connection.

To download new firmware for a SPA9x2 phone, perform the following steps:

1. To begin, direct your browser to the following URL: http://www.linksys.com.

From the website homepage menu, select **Support** > **Downloads**.

2. Select your device from the drop-down list and click **Downloads For This Product**.

3. Click **Firmware** to go to the firmware download page.

4. Click **Download** to download the firmware file.

If you are using Windows XP Service Pack 2 (SP2) and Internet Explorer, a "Pop-up blocked" message might appear. If so, click the information bar and select **Temporarily Allow Pop-ups**. Then click **Download** again.

5. Click **Save** in the *File Download* dialog box that appears.

6. In the *Save As* dialog box, choose a location for the file and click **Save**.

7. When the download is complete, if prompted, click **Close**.

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

8. 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.

To extract the firmware file from the archive, use a utility such as WinZip, or use the built-in decompression features of Windows XP.

After downloading and uncompressing the firmware, upgrade the phone firmware by following the steps in the following section.

Upgrading SPA9x2 Firmware

NOTE: If you are using the SPA900 Series phones with a SPA9000, see the *Linksys Voice System Administration Guide* for instructions to upgrade your phones using the LVS Wizard.
To upgrade a phone's firmware:

1. After extracting the file, run the executable file to upgrade the firmware.

2. When the *Firmware Upgrade* warning window appears, click **Continue**.

3. Enter the IP address of the device in the field provided.
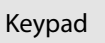
NOTE: For information about remotely upgrading a SPA9x2 phone, see "Provisioning Configuration from Phone Keypad" section on page 55 and the *Linksys SPA Provisioning Guide* (service provider login required).

Linksys SPA9x2 Administration Guide   17

Download from Www.Somanuals.com. All Manuals Search And Download.
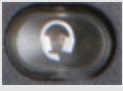
Using the Phone Display and Buttons

All SPA9x2 phones have the same basic buttons. Only the SPA922 has no line buttons, because it has only one voice line.

You can use the following graphic and table to identify buttons and features on your phone.

Phone Feature	Description
Message Waiting Indicator	Indicates an incoming call or new voice message. Can also notify you of error status. • Red: new voice message • Flashing red: Incoming call
LCD Screen	Configurable to display: • Date and time • Station name • Line extensions • Softkey options • Startup logo, screen saver, background photo, and photo album (SPA962)

Line keys	Indicate status of different voice lines. (Not applicable to SPA922). Defaults are:
	• Green: idle
	• Red:
	    • Steady--active/in-use
	    • Blinking--on hold
	• Orange: unregistered
	You can configure up to 16 different line states. For more information, see Appendix A "Creating an LED Script".

Softkey buttons	Each activates a softkey option displayed on your LCD screen.
Navigation button	Allows you to scroll through menu items and soft buttons.
Keypad	Lets you dial phone numbers, enter letters, and choose menu items.
Messages button	Auto-dials your voice message service (varies by service).
Setup button	Lets you configure various phone features and system settings.
Mute button	Toggles the microphone on or off. When the microphone is muted, the button is lit.

Phone Feature	Description
Headset button	Toggles the headset on or off. When the headset is on, the button is lit.
Speaker button	Toggles the speakerphone on or off. When the speakerphone is on, the button is lit.
Hold button	Toggles the call on and off hold.
Volume button	Controls volume for the: • Handset • Headset • Full-duplex speaker (off-hook) • Ringer (on-hook)

Using the LCD Screen

You can configure many settings directly from the phone.

1. Press the **Setup** button. The LCD shows your options.

2. Use the **Navigation** button to scroll up and down the list of options.

3. Choose an option by either pressing the **Select** softkey button or by pressing the number shown next to the option (for example, press **3** on your keypad to access Call History).

4. Change your options as needed, then press the **Change** or **Save** softkey button. To cancel any changes, press the **Cancel** softkey button.

As an administrator, you will use the web UI to configure phones and users.

Using the Web User Interface

To access the SPA9x2 phone administration web user interface (UI):

1. Launch a web browser on a computer that can reach the SPA9x2 phone on the network.

2. Direct the browser to the IP address of the SPA9x2 phone. To determine the IP address, either:

a. Press the **Setup** button, then press **9** or scroll to option **9** and press **Select**.

b. Option 2 shows the phone's current IP address.

3. Enter the IP address in your web browser address bar. For example:

   http://192.168.1.8

NOTE: If you are connected to a VPN, you must first disconnect.

4. Various tabs--Info, System, Phone, User, and so on--appear based on your account privileges and current view.

5. If you make changes, use the buttons at the bottom of the screen to either:

   • **Submit All Changes** and save them. Some saved changes cause the SPA9x2 phone to reboot.
   or
   • **Undo All Changes** button to undo your changes.

NOTE: If you have trouble accessing the web UIweb UI, perform the following steps on the SPA9x2:

1. Press the **Setup** button on the phone.

2. Scroll to **N9 Network** using **Navigation** button and select it.

3. Scroll to **7 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**.

4. Press **OK**, then press **Save**.

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

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.

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

The following links appear in the upper right corner:

- *Admin Login/ User Login*: toggles between views. The Admin view shows more configurable parameters.

- *Basic/ advanced*: toggles between views. The advanced view shows more configurable parameters.

- *Personal Directory*: lets you enter your regularly called numbers and names. You can also access your directory from the phone by pressing **Setup > 1** (see "Using the Personal Directory" section on page 23)

- *Call History*: shows your redial list, answered calls, and missed calls. You can also access call history from the phone by pressing **Setup > 3**. (see "Viewing Call History" section on page 24)

- *SPA932 Status* (for SPA962 only)--displays status of SPA932 lines if you have one or two SPA932 sidecars attached to your SPA962. For more information about the SPA932, see Chapter 10, "Using the SPA932 Sidecar."

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.

Administrator Privileges

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

NOTE: Provisioning information is available only via a service provider login on linksys.com.

To directly access the Administrator account, enter:

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

If a password has been set for the Admin Login, you are prompted to enter the password. An admin can change between User Login and Admin Login. Enter passwords as needed to log in.

Admin Login Web Pages

The following tabs and web pages are available when you use the *Admin Login*:

- Info

- System
- SIP
- Provisioning
- Regional
- Phone
- Extension (1-4 on the SPA942, 1-6 on the SPA962)
- User
- SPA932 (when configuring a SPA962; this tab appears regardless of whether a SPA932 sidecar is actually attached)

User Login Web Pages

The following tabs and web pages are available when you use the *User Login*:

- Info
- System
- Phone
- User

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 Personal Directory

A directory entry consists of the following information:

- Name
- Phone number or (SIP) URL
- Ring Tone

Up to 100 entries can be stored in the SPA9x2 phone. An entry can be added or edited from the phone GUI or on the web UI. To view or edit the directory, click **Personal Directory** in the upper right corner of the web UI page. Alternately, you can access it from the phone selecting **Setup > Directory** or pressing the **dir** (**Directory**) soft button.

To add a new entry to the directory on the web UI, type the entry on the first available line. Use the following format:

> n=*Person's Name*;p=*phonenumber*;r=*optional_ringtone*, for example:
> n=Eva Lee;p=08003311345

Prepend the number with any necessary steering or outside line digits, such as 9 or 0.

Click **Submit Changes** to save any changes to the personal directory. The SPA9x2 reboots.

NOTE: When viewing the directory, entries that use the default ringtone do not display a value for "r." A ringtone value of r=# indicates that a custom ring is used for that directory entry.

Viewing Call History

Three call logs are maintained for each phone:

- Redial List—a redial entry is added when dialing is completed, regardless if the number is correct or the call is successful.
- Answered Calls—an answered entry is logged when the incoming call is answered.
- Missed Calls—a missed entry is logged for each incoming call that rings but is not answered.

To view or edit the directory, click **Call History** in the upper right corner of the web UI page. You can also access it from the phone by pressing
Setup > 3 Call History.

The SPA9x2 phone keeps up to 60 entries per log in reverse chronological order. This history is saved in the phone's non-volatile memory.

Each entry is shown on the web page with the name, extension or phone number, and the date-time of the call. If a name is not available, it does not appear.

Caller and Called Name Matching

When making an outgoing call, the SPA9x2 phone tries to find the dialed number in the personal directory first, then from the missed call log, and finally from the answered call log. If a match is found and the name field is present in the matched entry, it is shown on the call screen as the called peer name.

For incoming calls, the phone tries to find the caller phone number in the personal directory. If a match is found and the name field is present in the matched entry, it replaces the current caller ID name on the call screen. The same name also goes into the incoming call log. If a match is not found, or the name field is not present in the matched entry, the current caller ID name is used (if it exists).

Web Admin Tabs

The following graphic shows all Admin Login *advanced* tabs that could appear on your web UI pages. When viewing in User Login or basic view, fewer tabs are visible (as previously discussed).

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 Chapter 11, "SPA9x2 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 ...
View phone phone, extension, and line/call information such, including: • DHCP, current IP address, DNS addresses • software and hardware versions • broadcast, RTP, and SIP information • registration state • packets sent, received, lost, and so on	**Info** tab See "Obtaining Phone Information (Info Tab)" section on page 27.
Configure system-level parameters, including network and debug parameters. To: • 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	**System** tab See Chapter 3, "Setting System Features."
Configure parameters to adjust SIP stack and protocols. To enable: • CTI • SIP-B • STUN	**SIP** tab See Chapter 4, "Setting SIP Parameters."

To perform these tasks...	On the web UI, click the ...
Configure provisioning parameters. To: • enable remote provisioning • enable firmware upgrades • set general purpose parameters	**Provisioning** tab The *Provisioning* tab is viewable by Admin logins only. See Chapter 5, "Provisioning Basics." For additional information about provisioning, see the *Linksys SPA Provisioning Guide* (for Linksys service providers)
Configure parameters that depend on country or region, including: • call progress tones • ring patterns • star codes/vertical service activation codes • vertical service announcement codes • local date/time and language	**Regional** tab See Chapter 6, "Setting Regional Parameters."
Configure General phone station info, which applies to all extensions configured for the phone, including: • 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 7, "Customizing SPA9x2 Phones."
Customize individual extension parameters, including: • 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 so on • audio settings • dial plan settings	**Ext** tab (1-6, depending on SPA9x2 phone model) See Chapter 8, "Customizing SPA9x2 Extensions."
Customize user-level parameters, including: • call forward • speed dial • supplementary services • web information (RSS newsfeeds) • traffic information settings • audio volume • phone GUI settings	**User** tab See Chapter 9, "Customizing Users."

To perform these tasks...	On the web UI, click the ...
View and change parameters for Unit 1 and Unit 2 (applicable only to SPA962 phones with one or two SPA932 sidecars attached)	**932** tab See Chapter 10, "Using the SPA932 Sidecar."

Obtaining Phone Information (Info Tab)

Once you log on to the web UI (see "Using the Web User Interface" section on page 20), you can check the current status of a SPA9x2 by clicking the **Info** tab (appears by default):
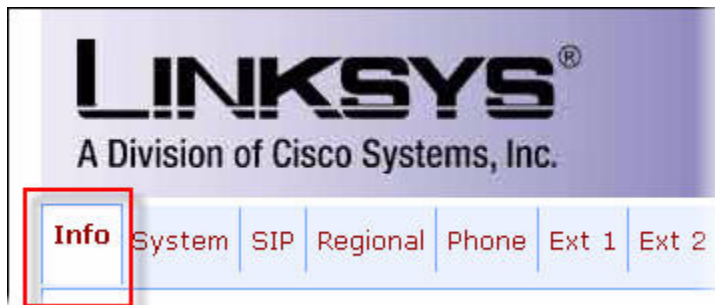
The info tab shows all current phone and extension information, including phone statistics and the registration status of all phone extensions. All fields are read-only.

To perform these tasks...	Click the **Info** tab > scroll to this section...
View phone system and product information such as: • DHCP • current IP address • DNS addresses • software and hardware versions	Info > System Information Info > Product Information See "System and Product Information" section on page 28.
View phone status information such as: • current time • broadcast, RTP, and SIP information • operational VLAN	Info > Phone Status "Phone Status Notes" section on page 29.
View extension status for each extension on the phone, such as: • registration state • mapped SIP port • message waiting status	Info > Ext Status "Extension Status Information" section on page 31.

To perform these tasks...	Click the **Info** tab > scroll to this section...
View line/call status for each line on the phone, such as: • current call state • packets sent, received, and lost, and so on • downloaded ring tones	Info > Line/Call Status "Line and Call Status Information" section on page 32.
View downloaded ring tones	Info > Line/Call Status "Downloaded Ring Tones" section on page 33.

The following figures show all fields contained in the Info tab. The information shown is the same for all logins (Admin and User) and views (basic and advanced).

System and Product Information

Check the Info tab for the phone's current IP address, current software version, and the registration status of a phone's extensions, among many other things.

Product Information Notes

- *Client Certificate*: Status of the client certificate, which authenticates the SPA9x2 phone for use in the ITSP network. This field indicates if the client certificate is properly installed in the SPA9x2 phone.

- *Customization*: For a remote customization (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 it.

Phone Status Information

Phone Status Notes

- *Elapsed Time*: Time elapsed since the last reboot of the system; for example, 20 days and 03:15:06.

- *Broadcast Pkts Dropped*: 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*: Number of broadcast bytes received but not processed.

- *RTP Packets Sent*: Number of RTP packets sent, including redundant packets.

- *RTP Bytes Sent*: Number of RTP bytes sent, including redundant packets.

- *SIP Messages Sent*: Number of SIP messages sent, including retransmissions.

- *SIP Bytes Sent*: Number of SIP bytes sent, including retransmissions.

- *SIP Messages Recv*: Number of SIP messages received, including retransmissions.

- *SIP Bytes Recv*: 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.

Understanding Lines, Calls, and Shared Call Appearances

Each SPA9x2 has a different number of lines:

- SPA922: one line

- SPA942: four lines

- SPA962: six lines

- SPA932 sidecar for the SPA962: 32 lines

Each line corresponds to a phone number (or extension) used for calls. The number of extensions you see listed in the SPA9x2's Extension Status depends on how many lines the phone has. A "call appearance" physically corresponds to a line key on a station, so a call appearance is an instance of an extension.

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

When the same extension is assigned to line keys on more than one station, it is called a shared line appearance. Extension 1 is referred to as the primary extension. Some features can be activated only on the primary extension, such as call forwarding.

One extension can be assigned to multiple line keys on a phone. In fact, all four call appearances can be instances of the same extension or phone number. An extension is not a shared line appearance unless it is also assigned to a line key on another station.

Understanding Extensions

Extensions contain core account information and control parameters.

Although the account information is usually the same for a shared extension on different stations, the other parameters can differ. For example, the dial plan or the preferred codec to use when making a call on an extension could be different for two different stations sharing the same extension.

Lines can be in the following states:

- Disabled: The line key is disabled

- Idle (Ready): Ready for use

- Dialing: Collecting digits from the user to be dialed out from this line

- Calling: Waiting for the called party to respond

- Proceeding (Progressing): Called party station is ringing

- Ringing (Alerting): Station is ringing with an Incoming call

- Connected: Connected with remote party

- Held: Remote party is on hold

- Invalid: Remote party hangs up or an error occurs while attempting an outbound call

- Busy: the line is being used by another station (shared lines only)

Shared Line (or Call) Appearances

SPA9x2 phones support shared line appearances in association with a Broadsoft application server. An extension can be shared by two or more stations. All call appearances on a shared extension are shared.

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.

All sharing stations ring on an inbound call to the shared call appearance extension. Whoever picks up the call first takes the call. When a call is placed on hold by one station, it can be resumed from another station sharing the same call appearance.

Line Key LEDs

This section describes line key LEDS on SPA9x2 phones with multiple line keys.

A line key corresponds to a call appearance. Call appearances for the same extension are numbered in ascending order of their line key position starting at 1. The line key LED color and blinking pattern are fully programmable.

The line key LED can be in one of four conditions:

- Red (r): Steady indicates active (in use); Blinking indicates on hold

- Green (g): Idle

- Orange: Unregistered

- Off (o): Disabled

SPA9x2 phones have the following built- in blinking patterns defined:

- No Blink (nb): Steady on (or off)

- Slow Blink (sb): 500ms on, 500ms off

- Fast Blink (fb): 100ms on, 100ms off

- Double Blink (db): 100ms on, 100ms off, 100ms on, 700ms off

The administrator can also define blinking patterns by modifying the syntax of the LedScript. See Appendix A "Creating an LED Script".

Extension Status Information

The line and call status appear below the Extension status on the Info web page.

Extension Status Notes

- *Next Registration In*: Number of seconds before the next automatic registration renewal.

- *Message Waiting*: Indicates whether you have new voice mail waiting (Yes or No). This is updated whenever a voice message is received.

- *Mapped SIP Port*: Port number of the SIP port mapped by NAT. Note that this field can be helpful in troubleshooting one-way audio calls.

For information about setting extension parameters, see Chapter 8, "Customizing SPA9x2 Extensions."

Line and Call Status Information

Line and Call Status Notes

- *Call state*: see "Understanding Extensions" section on page 30.

- *Tone*: Type of ringtone used by the call.

- *Encoder/Decoder*: Codecs used for encoding and decoding.

- *Call 1/2 FAX*: Status of the fax pass-through mode.

- *Type*: Direction of the call, either inbound or outbound.

- *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 and Peer Phone*: Name and phone number of the other side of the call.

- *Duration*: Duration of the call.

- *Decode Latency*: Number of milliseconds for decoder latency.

- *Jitter*: Number of milliseconds for receiver jitter.

- *Round Trip Delay*: Number of milliseconds for delay.

- *Packet Error*: Number of invalid packets received. Any number other than 0 indicates a problem, most likely out of order packets.

- *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.

For more line and extension information see, Chapter 8, "Customizing SPA9x2 Extensions.".

Downloaded Ring Tones
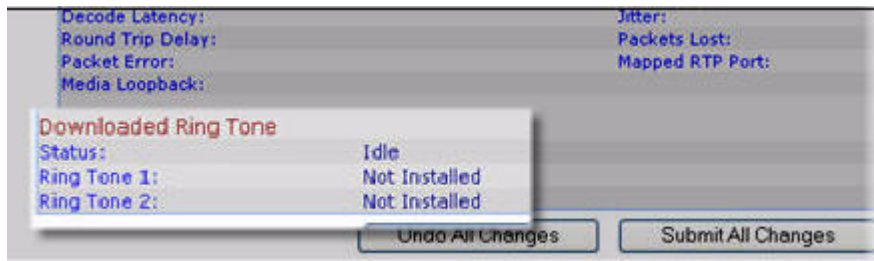
Ring Tone Notes

- *Status*: Indicates if the phone is currently downloading a ring tone (and from where) or if it is idle.

- *Ring Tone 1*: Information about downloaded ring tone 1: name, size, and time-stamp of the tone.

- *Ring Tone 2*: Information about downloaded ring tone 2: name, size, and time-stamp of the tone.

For more information about downloading ring tones, see:

www.linksys.com/support

What's Next?

Next you'll begin configuring the system, provisioning users, and customizing phones and extensions. The admininstration web server lets you perform all necessary tasks remotely or on site.

Although you can perform some tasks directly on the SPA9x2 LCD, this guide focuses on using the web UI. For information about using phone LCD options, see the *Linksys SPA9x2 User Guide.*

Basic provisioning information is contained in this guide. If you are a service provider, see the *Linksys SPA Provisioning Guide.*

Setting System Features

Use the System tab to enable web server access, set security features, the debug server, and VLAN settings. See the following sections:

Setting Security Features

The following features help ensure that your SPA9x2 calls are secure and authenticated.

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.

SPA9x2 phones use UDP as a standard for SIP transport, but they also support SIP over TLS for added security.

To enable TLS for a SPA9x2 phone:

1. Log in to the SPA9x2 phone's administration web browser.

2. Click **Ext 1**, then scroll to the *SIP Settings* section.

3. Select **TLS** from the *SIP Transport* drop-down box.

4. Click **Submt 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.

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 (see "Subscriber Information Parameters" section on page 102).

Secure Call Service is defined in the *Phone* tab (see "Supplementary Services Parameters" section on page 89). Secure Call Service activates Linksys secure encryption of RTP streams between the two endpoints. You can disable this if the other endpoint (or gateway) does not support this Linksys proprietary method.

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, press **6** from the Product Info menu, or scroll down to the highlight **Client Cert**, and press the **select** soft button.

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. Linksys SPA9x2 phones support several codecs. See:

Supported Codecs

Negotiation of the optimal voice codec sometimes depends on the ability of SPA9x2 phone to "match" a codec name with the far-end device/gateway codec name. SPA9x2 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 SPA9x2 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 SPA9x2 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

Codec (Voice Compression Algorithm)	Complexity and Description	MOS Score
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, SPA9x2 phones can support dynamic payloads for G.726	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. SPA9x2 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.	3.8

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.

Linksys recommends using the SPA9x2 with QoS-capable networking equipment that can prioritize the VoIP application traffic. QoS features are available on many Linksys 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

Codec	Est. Bandwidth Budget	2 Calls	4 Calls	6 Calls	8 Calls
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. SPA9x2 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.

- 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—SPA9x2 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 SPA9x2 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. SPA9x2 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. SPA9x2 phones have a near-end echo canceller with at least 8 ms tail length to compensate for impedance match. SPA9x2 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 Linksys 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. SPA9x2 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.

Setting System Features

See the following sections:

Setting System Configuration Information

System Configuration Notes

- *Restricted Access Domains*: Enter fully-qualified domain names (FQDNs) for each SIP server you want the phone to respond to. Separate with semicolons. If you enter domains, the SPA9x2 will respond to SIP messages *only* from the entered servers.

- *Enable Web Server*: Enables or disables the web administration server for the phone. Disable web server functionality for a phone to prevent unauthorized access to the phone's web server pages and information.

- *Web Server Port*: Default to port 80 for the web UI.

- *Enable Web Admin Access*: Lets you enable or disable local access to the **Admin Login** of the web UI. Defaults to **yes** (enabled). For more information about the Admin and User Logins, see *"Using the Web User Interface" section on page 20*.

- *Admin Passwd*: Set this if you want the system administrator to log on to the web UI with a password. The password prompt will appear when an admininstrator clicks **Admin Login**. The maximum password length is 32 characters.

- *User Password*: Set this 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

Setting Internet, IP and PPPoE Information

DHCP and PPPoE Notes

- *DHCP Renewal on Timeout*: SPA9x2 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.

- *PPPoE settings*: 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.

Setting a Static IP Address on the IP Phone

As with many system and phone settings, you can use the phone softkeys as an alternative to the web GUI:

1. On the SPA9x2, press the **Setup** button.

2. Press **9** or scroll to *9 Network* using the **Navigation** button.

3. With *1 WAN Connection Type* highlighted, press the **Edit** soft button.

4. Press the **Option** soft button until *Static IP* displays. Press **OK.**

5. Scroll to *Non-DHCP IP Address*. Press **Edit**, then enter the static IP address of the phone using the asterisk (*) in place of periods. For example, the IP address 192.168.1.100 would be entered as 192*168*1*100. Press **OK**.

6. Scroll to **9 Non-DHCP Subnet Mask**. Press **Edit**, then enter the subnet mask of the phone using the asterisk (*) in place of periods. For example, the IP address 255.255.255.0 would be entered as 255*255*255*0. Press **OK**.

7. Scroll to **10 Non-DHCP Default Route**. Press **Edit**, then enter the default gateway of the phone using asterisks as shown above. Press **OK**.

8. Scroll to **11 Non-DHCP DNS 1**. Press **Edit**, then enter the DNS1 of the phone using asterisks as shown above. Press **OK**. Optionally scroll to **12 Non-DHCP DNS2**.

9. Press the **Save** soft button.

NOTE: If you do not press **Save**, your changes are not applied. To cancel your changes, press the **Cancel** softkey.
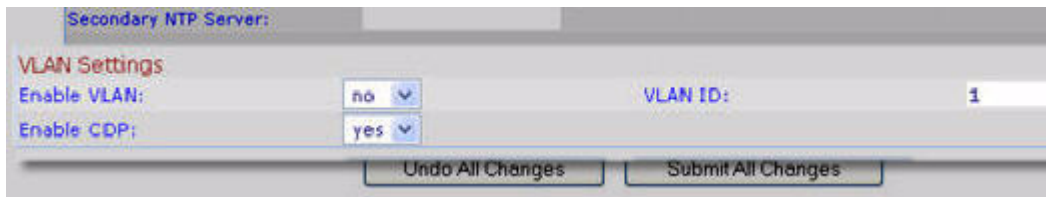
Setting Optional Network Parameters

Optional Network Configuration Notes

- *HostName*: Enter the host name of the SPA9x2 phone.

- *Syslog Server*: Enter the syslog server name and port. The syslog server logs SPA9x2 system information and critical events. If you enter both a *Debug Server* and a *Syslog Server*, syslog messages are also logged to the debug server.

- *Debug Server*: Enter the debug server name and port. This feature specifies the server for logging SPA9x2 debug information. The level of detailed output depends on the debug level parameter setting (below).

- *Debug Level*: Enter a value from 0-3. The highest level, 3, is recommended because the higher levels provide more debug information. To log SIP messages, you must set this value to 2 or 3. To limit debug information, set this parameter to a low value such as 0 or 1. Defaults to 0.

- *Primary NTP Server* and *Secondary NTP Serve*r: Enter an IP address or hostname for the NTP Server. The phones use these servers to synchronize its time. Defaults to blank.

43

Configuring VLAN Settings

VLAN Settings Notes

- *Enable CDP* only if you are using a Cisco switch. CDP is negotiation based and determines which VLAN the SPA9x2 resides in.

- If you use a VLAN without CDP (VLAN enabled and CDP disabled), enter a *VLAN ID* for the SPA9x2. Note that only voice packets are tagged with the VLAN ID.

44

Setting SIP Parameters

Use the *SIP* tab to set SIP, RTP, SDP, NAT and Linksys key system parameters. This chapter discusses:

SIP Parameters

Use this section to configure various SIP parameters, including enabling CTI.

SIP Parameter Notes

The following table describes all 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 a request can be challenged. Ranges from 0-255. Defaults to 2.
SIP User Agent Name	Used in outbound REGISTER requests. If empty, the header is not included. Macro expansion of $A to $D corresponding to GPP_A to GPP_D allowed. Defaults to $VERSION.
SIP Server Name	Server header used in responses to inbound responses. Defaults to $VERSION.
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 SPA9x2 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 SPA9x2 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 (see "Subscriber Information Parameters" section on page 102) 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 Sylantro call features. See www.sylantro.com for more information. Defaults to no.
Talk Package *need info*	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 *need info*	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 *need info*	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 *need info*	If set to yes, the SPA9x2 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 SPA9x2 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 SPA9x2 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, theSPA9x2 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.
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.
CTI Enable	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 SPA9x2 and, for example, initiate or answer a call by clicking a mouse on a PC, **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 10, "Using the SPA932 Sidecar." Defaults to no.

Parameter	Description
Caller ID Header	Select where the SPA9x2 gets its caller ID from: PAID-RPID-FROM P-ASSERTED-IDENTITY REMOTE-PARTY-ID FROM header Defaults to PAID-RPID-FROM

SIP Timer Values

SIP timer values control transmittal, time-out, delay intervals and so on for SPA9x2 data. Each value is displayed in seconds.

SIP Timer Notes

Register Retry Enhancements: this feature adds flexibility to the delay timers that are activated when the SIP REGISTER of a device fails. Timers are selected based on the failure response code sent by the SIP proxy. Depending on the type of registration failure, either a short timer or a long timer is activated.

All SIP timer parameters are included in the following table.

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.

Parameter	Description
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.
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 1999999999999999999999999999999. 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 1999999999999999999999999999999. 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 SPA9x2 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 SPA9x2 waits for this length of time before retrying. If this interval is 0, the SPA9x2 stops trying. This value should be much larger than the Reg Retry Intvl value. Defaults to 1200 seconds.

Parameter	Description
Reg Retry Random Delay	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. Defaults to 0, which disables this feature.
Reg Retry Long Random Delay	Random delay add ed to Register Retry Long Intvl value when retrying REGISTER after a failure. Reg_Retry_Long_Random_Delay—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. Defaults to 0, which disables this feature.
Reg Retry Intvl Cap	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). 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.
Sub Min Expires	The lower limit of the REGISTER (suscribe) expires value returned from the proxy server. Defaults to 10 seconds.
Sub Max Expires	The upper limit of the REGISTER (subscribe) min-expires value returned from the proxy server in the Min-Expires header. Defaults to 7200 seconds.
Sub Retry Intvl	The retry interval when the last Subscribe request fails. Defaults to 10 seconds.

NOTE: SPA9x2 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

Response Status Notes

- *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.

- *Try Backup RSC*: SIP response code that retries a backup server for the current request. Defaults to blank.

- *Retry Reg RSC* : Interval the SPA9x2 waits before re-trying registration after a failed registration. Defaults to blank.

RTP Parameters

RTP Notes

- *Max RTP ICMP Err*: Number of successive ICMP errors allowed when transmitting RTP packets to the peer before the SPA9x2 terminates the call. If the value is set to 0 (the default), the SPA9x2 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 SPA9x2 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 reciever report (RR); the last SR carries no RR. The SDES 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 Linksys/SPA9x2-5.2.2(SCb)).

  – The NTP timestamp used in the SR is a snapshot of the SPA9x2's local time, not the time reported by an NTP server.

  – If the SPA9x2 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 SPA9x2 web page. It can range from 0 to 255 seconds. Defaults to 0 (recommended).

- No UDP Checksum: Select yes if you want the SPA9x2 to calculate the UDP header checksum for SIP messages. Since this involves computation load, you should keep the default value (no) to disable it.

SDP Payload Types

Configured dynamic payloads are used for outbound calls only when the SPA9x2 presents an SDP offer. For inbound calls with an SDP offer, the phone follows the caller's assigned dynamic payload type.

SPA9x2s use the configured codec names in outbound SDP. For incoming SDP with standard payload types of 0-95, the SPA9x2 ignores the codec names. For dynamic payload types, the SPA9x2 identifies the codec by the configured codec names (comparison is case-sensitive).

SDP Payload Parameters

- *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.

NAT Support Parameters

NAT Support Parameter Notes

- *STUN Enable*: Enables the use of STUN to discover NAT mapping. Select yes or no from the drop-down menu. Defaults to no.

- *STUN Test Enable*: If the STUN Enable feature is enabled and a valid STUN server is available, the SPA9x2 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 SPA9x2 detects symmetric NAT or a symmetric firewall, NAT mapping is disabled.
  Defaults to no.

- *STUN Server*: IP address or fully-qualified domain name of the STUN server to contact for NAT mapping discovery. You can get open-source STUN software at the following website:

  http://www.voip-info.org/wiki-STUN > "Public STUN Servers" section.

  Use any public STUN server from the link above for the STUN Server parameter.

- *EXT IP*: External IP address to substitute for the actual IP address of the SPA9x2 in all outgoing SIP messages. If 0.0.0.0 is specified, no IP address substitution is performed. If this parameter is specified, the SPA9x2 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, supercede this statically configured value.
  Defaults to blank.

Linksys Key System Parameters

Linksys Key System Notes

- *Linksys Key System*: enables or disables the Linksys Key System for use with the LVS SPA9000. Defaults to yes. See the *LVS Administration Guide* for more details.

- *Multicast Address*: used by the SPA9000 to communicate with SPA9x2 phones. Defaults to 224.168.168.168:6061.

- *Force LAN Codec*: for use with the LVS. Choices are G.711u or G.711a. Defaults to none.

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:

-

-

-

-

-

-

-

-

-

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

- *Linksys Provisioning Guide* (service provider login required)

- *Linksys Voice System Administration Guide*

- Service provider documentation

Provisioning Capabilities

The SPA9x2 provides for secure provisioning and remote upgrade. Provisioning is achieved through configuration profiles transferred to the device via TFTP, HTTP, or HTTPS.

The SPA9x2 can be configured to automatically resync its 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 SPA9x2 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 Linksys. The SPA9x2 supports up to 256-bit symmetric key encryption of profiles. For the initial transfer of the profile encryption key (initial provisioning stage), the SPA9x2 can receive a profile from an encrypted channel (HTTPS with client authentication), or it can resync to a binary profile generated by the Linksys SIP profile compiler. In the latter case, the SIP profile compiler can encrypt the profile specifically for the target SPA9x2, without requiring an explicit key exchange.

Remote firmware upgrade is achieved via TFTP or HTTP or HTTPS. Remote upgrades are controlled by configuring the desired firmware image URL into the SPA9x2 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 Linksys phones.

To provision from the phone:

1. Press **Setup**, then scroll to **22 Profile Rule**.

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). hen the address can be entered in and press Resync.

Note that remote customization blink patterns use the SPA9x2 *Mute* button LED as follows:

- 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

SPA9x2 Configuration Profiles

The SPA9x2 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 SPA9x2 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 SPA9x2 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 Linksys 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 Linksys 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.

Refer to the *LVS SPA Provisioning Guide* for further details.

Upgrading, Resyncing, and Rebooting Phones

The Administrator can upgrade, reboot, or resync SPA9x2s using the web interface.

Upgrading Firmware on a Phone

Use the upgrade URL to upgrade firmware on the SPA9x2. You can upgrade from either a TFTP or HTTP server.

NOTE: The Upgrade Enable parameter on the Provisioning web page must be set to Yes: SPA9x2 web UI > **Provisioning** > **Firmware Upgrade** > **Upgrade Enable: yes**
.
Use the following syntax to upgrade firmware on a phone:

http://*spa9x2-ip-addr*/admin/upgrade?*protocol://server-name[:port]]/firmware-path*

- Protocol defaults to TFTP.

- Server name defaults to the host requesting the URL.

- Port defaults to 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). The firmware-pathname is typically the file name of the binary located in a directory on the TFTP or HTTP server.

Resyncing a Phone

You can resync a SPA9x2 phone to a specific remote profile. The configuration of the phone you resync will match the configuration of the remote phone. The SPA9x2 can be configured to resync its internal configuration state to a remote profile periodically and on power up.

NOTE: The SPA9x2 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://*spa9x2-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 SPA9x2 phone if needed.

Use the following syntax to reboot a phone:

http://*spa9x2-ip-addr*/admin/reboot

SPA9x2 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 SPA9x2 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 SPA9x2 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 SPA9x2 is idle, because this may trigger a software reboot.

General purpose parameters are provided to help service providers manage the provisioning process. Each SPA9x2 can be configured to periodically contact a normal provisioning server (NPS). Communication wit 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.

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 SPA9x2 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 SPA9x2 continues to process A-records until the first server responds. If no server associated with the A-records responds, the SPA9x2 logs an error to the syslog server.

Retail Provisioning

The SPA9x2 firmware includes an web UI that displays SPA9x2 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 Linksys 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 SPA9x2 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/linksys-init/1234abcd

In this example, 1234abcd is the PIN number of the new account. The remote provisioning server is configured to associate the SPA9x2 that is performing the resync request with the new account, based on the URL and the supplied PIN. Through this initial resync operation, the SPA9x2 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/linksys-init

For both initial and permanent access, the provisioning server relies on the SPA9x2 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 SPA9x2 supports a more convenient mechanism for in-house preprovisioning. With the factory default configuration, a SPA9x2 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 SPA9x2 to a LAN environment configured to preprovision SPAs. Any new SPA9x2 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 SPA9x2 provisioning server.

Subsequently, when a new customer signs up for service, the preprovisioned SPA9x2 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, possibly through a router. On power-up the SPA9x2 already knows the server to contact for its periodic resync update.

Configuration Access Control

Besides configuration parameters that control resync and upgrade behavior, the SPA9x2 provides mechanisms for restricting end-user access to various parameters.

The SPA9x2 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 SPA9x2, 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 SPA, on a parameter-by-parameter basis. Optionally, user access to the SPA9x2 web UI can be totally disabled.

The Internet domains accessed by the SPA9x2 for resync, upgrades, and SIP registration for Line 1 can be restricted.

Using HTTPS

The SPA9x2 provides a reliable and secure provisioning strategy based on HTTPS requests from the SPA9x2 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 Linksys SPA9x2 phones, you must generate a Certificate Signing Request (CSR) and submit it to Linksys. Linksys generates a certificate for installation on the provisioning server that is accepted by SPA9x2 phones when they seek to establish an HTTPS connection with the provisioning server.

How HTTPS Works

HTTPS encrypts the communication between the client and the server, protecting the message contents from other intervening network devices. The encryption method for the body of the communication between client and server is based on symmetric key cryptography. With symmetric key cryptography, a single secret key is shared by the client and the server over a secure channel protected by Public/Private key encryption.

Messages encrypted by the secret key can only be decrypted using the same key. HTTPS supports a wide range of symmetric encryption algorithms. The SPA9x2 implements up to 256-bit symmetric encryption, using the American Encryption Standard (AES), in addition to 128-bit RC4.

HTTPS also provides for the authentication of the server and the client engaged in a secure transaction. This feature ensures that the provisioning server and an individual client cannot be spoofed by other devices on the network. This is an essential capability in the context of remote endpoint provisioning.Server and client authentication is performed using public/private key encryption, using certificates containing the public key. Text encrypted with a public key can be decrypted only by its corresponding private key (and vice versa). The SPA9x2 supports the Rivest, Shamir, and Adelman (RSA) algorithm for public/private key cryptography.

Certificates are authenticated in the context of a certificate chain. A certificate authority lies at the root of the chain, with all other certificates depending on the root authority for authority.

Server Certificates

Each secure provisioning server is issued an secure sockets layer (SSL) server certificate, directly signed by Linksys. The firmware running on the SPA9x2 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 Linksys.

This mechanism protects the service provider from unauthorized access to the SPA9x2 endpoint, or any attempt to spoof the provisioning server. This might allow the attacker to reprovision the SPA9x2, 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 Linksys SPA9x2.

Client Certificates

In addition to a direct attack on the SPA, an attacker might attempt to contact a provisioning server using a standard web browser, or other HTTPS client, to obtain the SPA9x2 configuration profile from the provisioning server. To prevent this kind of attack, each SPA9x2 also carries a unique client certificate, also signed by Linksys, 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.

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).
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.

General Purpose Parameters

The following table defines the function and usage of each parameter in the General Purpose Parameters section of the Provisioning tab.

- *GPP_A through GPP_P*: General purpose provisioning parameters. These parameter 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 that 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 *Linksys SPA Provisioning Guide.*

Setting Regional Parameters
Advanced Scripting for Cadences, Call Progress Tones, and Ring Tones

Setting Regional Parameters

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" section on page 63

- "Call Progress Tones" section on page 67

- "Distinctive Ring Patterns" section on page 68

- "Control Timer Values (sec)" section on page 68

- "Miscellaneous Parameters" section on page 76

- "Localizing Your SPA9x2 Phone" section on page 78

- "Selecting a Language" section on page 80

SPA9x2 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 an 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[;S2]

  where Si=Di(oni,1/offi,1[,oni,2/offi,2[,oni,3/offi,3[,oni,4/offi,4[,oni,5/offi,5[,oni,6/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.

- Di 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.

Linksys SPA9x2 Phone Administration Guide

Download from Www.Somanuals.com. All Manuals Search And Download.

Setting Regional Parameters
Advanced Scripting for Cadences, Call Progress Tones, and Ring Tones

Setting Regional Parameters

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" section on page 63

- "Call Progress Tones" section on page 67

- "Distinctive Ring Patterns" section on page 68

- "Control Timer Values (sec)" section on page 68

- "Miscellaneous Parameters" section on page 76

- "Localizing Your SPA9x2 Phone" section on page 78

- "Selecting a Language" section on page 80

SPA9x2 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 an 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[;S2]

  where Si=Di(oni,1/offi,1[,oni,2/offi,2[,oni,3/offi,3[,oni,4/offi,4[,oni,5/offi,5[,oni,6/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.

- Di 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.

Linksys SPA9x2 Phone Administration Guide   63

Download from Www.Somanuals.com. All Manuals Search And Download.

- **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:

- FreqScript;Z_1[;Z2]. The section Zi is similar to the Si section in a CadScript except that each on/off segment is followed by a frequency components parameter: Zi = Di(oni,1/offi,1/fi,1[,oni,2/offi,2/fi,2 [,oni,3/offi,3/fi,3 [,oni,4/offi,4/fi,4 [,oni,5/offi,5/fi,5 [,oni,6/offi,6/fi,6]]]]])

    where fi,j = n1[+n2]+n3[+n4[+n5[+n6]]]]] and 1 < nk < 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

Call Progress Tones parameters define the level, frequency and cadence of the dial, ringing and call progress tones played by the phones. Each country has a defined standard level, frequency and cadence for majority of the tones.

Call Progress Tones Notes

The default values provided correspond to US tones standard. For definitions of all call progress tones, see .

Distinctive Ring Patterns

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

Ring Pattern Notes

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

Control Timer Values (sec)

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 255.
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.

Field	Description
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 SPA9x2 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. 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. 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 (Star Codes)

SPA9x2 phones support service activation codes, also known as star codes. The caller 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.

This feature supports all services that can be activated on the phone.

NOTE: Some service providers choose to disable star codes. See "Disabling/Deactivating Services" section on page 73.

Interdigit Long Timer:	10		Interdigit Short Timer:	3
Vertical Service Activation Codes				
Call Return Code:	*69		Blind Transfer Code:	*98
Call Back Act Code:	*66		Call Back Deact Code:	*86
Cfwd All Act Code:	*72		Cfwd All Deact Code:	*73
Cfwd Busy Act Code:	*90		Cfwd Busy Deact Code:	*91
Cfwd No Ans Act Code:	*92		Cfwd No Ans Deact Code:	*93
CW Act Code:	*56		CW Deact Code:	*57
CW Per Call Act Code:	*71		CW Per Call Deact Code:	*70
Block CID Act Code:	*67		Block CID Deact Code:	*68
Block CID Per Call Act Code:	*81		Block CID Per Call Deact Code:	*82
Block ANC Act Code:	*77		Block ANC Deact Code:	*87
DND Act Code:	*78		DND Deact Code:	*79
Secure All Call Act Code:	*16		Secure No Call Act Code:	*17
Secure One Call Act Code:	*18		Secure One Call Deact Code:	*19
Paging Code:	*96		Call Park Code:	*38
Call Pickup Code:	*36		Call UnPark Code:	*39
Group Call Pickup Code:	*37		Media Loopback Code:	*03
Referral Services Codes:				
Feature Dial Services Codes:				
Service Annc Base Number:				

Vertical Service Activation Code Notes

- You can change default star code values for your area or region.

- *Call Back*: Similar to the call back service offered by the PSTN, the caller can activate this service on a busy number to recieve a call back as soon as the busy number becomes available. The SPA9x2 phone repeatedly dials the busy number until the called party rings or answers. Call back is also referred to as repeat dialing and works as follows:

  a. The caller selects a line key and enters ***66** (call back activation code).

  b. The phone uses this call appearance to call the last called number. The retry period and expiration time are configurable.

  c. While the call back service is active, the corresponding line key flashes green. The line can still be used: the call back service will pause temporarily until the call appearance is idle again. To cancel the service, the caller can pick up any line and enter ***86**.

  d. When the call back number rings or answers, the call appearance rings like a normal incoming call. If the call back party answers the call before the caller does, the phone sends a holding tone to the call back party. If the caller picks up the call back first, the phone rings like a regular outbound call.

- *Secure All Calls*: Activates sRTP in all extensions. Note that you must install a set of certificates to get sRTP in both secure ends.

- *Media Loopback*: An SP 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. For more information, see

- *Referral Services Codes*: One or more * code can b* codee 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 SPA9x2 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 SPA9x2 plays a special prompt tone while waiting for the user the enter a target number (which is validated according to the dial plan as in normal dialing).

  b. When a complete number is entered, the SPA9x2 sends a blind REFER to the holding party with the Refer-To target equals to *98 *target_number*. This feature allows the SPA9x2 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 SPA9x2. You can empty the any * code that you do not want to SPA9x2 to process.

- *Feature Dial Services Codes*: Tell the SPA9x2 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 SPA9x2 to call the target number prepended by the * code.
  For example:

  a. After the user dials *72, the SPA9x2 plays a special prompt tone while waiting for the user the enter a target number (which is validated according to the dial plan as in normal dialing).

  b. When a complete number is entered, the SPA9x2 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 SPA9x2. You can empty any * code you do not want to SPA9x2 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 place, x is any digit not used above

If no tone parameter is specified, the SPA9x2 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.

Using Star Codes to Activate/Deactivate Services

SPA9X2 phones accept star codes to activate or deactivate certain services on the phone.

To activate a star code, either off hook the phone and dial the * code or, if you do not know the code, scroll through the LCD menu options to select the desired star code.

When off-hook dialing, the star code is processed by the phone as soon as it is entered. When on-hook dialing, you can pre-dial multiple star codes before the target number. The star codes are handled in the order they are entered, before the target number is dialed. Star codes can also be included in a directory entry or redial list.

The following list briefly defines the supplementary services accessible with star codes (listed on the web page):

- Call Return (*69)—Calls the last caller, regardless which extension.
- Call Back(*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.
- Cancel Call Back(*86)—Cancels the last call back operation.
- Call Forward All (*72)—Call forwards all inbound calls. Applies to primary extension only
- Cancel Call Forward All (*73)—Cancels call forward all. Applies to primary extension only
- Call Forward Busy (*90)—Call forwards on busy. Applies to primary extension only:
- Cancel Call Forward Busy (*91)—Cancels call forward on busy. Applies to primary extension only.
- Call Forward No Answer (*92)—Call forwards if no answer. Applies to primary extension only.
- Cancel Call Forward No Answer (*93)—Cancels call forward no answer. Applies to primary extension only.
- Block Caller ID Always (*67)—Blocks caller ID on all outbound calls. Applies to all extensions.
- Block Caller ID Per Call (*81)—Blocks caller ID on the next outbound call (on the current call appearance only).
- Unblock Caller ID Always (*68)—Unblocks caller ID on all outbound calls. Applies to all extensions.
- Unblock Caller ID Per Call (*82)—Unblocks caller ID on the next outbound call (on the current call appearance only).
- Secure All Calls (*16)—Defaults to prefer to use encrypted media (voice codecs).

- Secure No Calls (*17)—Defaults to prefer to use unencrypted media for all outbound calls. Applies to all extensions.
- Secure Next Call (*18)—Prefers to use encrypted media for the next outbound call (on this call appearance only).
- Do Not Secure Next Call (*19)—Prefers to use unencrypted media for the next outbound call (on this call appearance only).
- Do Not Disturb (*78)—Applies to all extensions.
- Cancel Do Not Disturb (*79)—Applies to all extensions.
- Block Anonymous Calls—Applies to all extensions.
- Cancel Block Anonymous Calls—Applies to all extensions.
- Blind Transfer (*98)—This allows you to transfer a call to another person without talking to that person:

  1. Put the call on hold.

  2. Get a dial tone on another line.

  3. Call the person to whom you are transferring the call by pressing *98, followed by their extension.

  4. Hang up the receiver.

Disabling/Deactivating Services

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

- Clear the star code in the *Vertical Service Activation* section above

- Disable the service in the *Phone* tab "Supplementary Services Parameters" section on page 89 section

Note that 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 (!) .

Vertical Service Announcement Codes

Vertical Service Announcement Codes allow the network to play a service announcement when a Supplementary (Vertical) service is activated by the phone (for example, when a user activates call forwarding in the phone, the phone calls a specific number for the announcement server to play a message "All your calls will be forwarded to the number 555-555-555").

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.

Vertical Service Announcement Notes

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

> [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
>
> Example:
> <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: "1234*00*@$proxy".

When the user *disables* the call waiting service, SPA9x2 connects to "1234*01*@$proxy".

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

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.

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 "Supported Codecs" section on page 37.

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 "Audio Configuration Parameters" section on page 102)

- 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 "Audio Configuration Parameters" section on page 102).

Miscellaneous Parameters

This section contains both DTMF parameters and localization parameters:

- "DTMF Notes"

- "Localizing Your SPA9x2 Phone"

- "Managing the Time and Date"

- "Configuring Daylight Savings Time"

- "Daylight Saving Time Examples"

- "Selecting a Language"

- "Creating a Dictionary Server Script"

DTMF Notes

DTMF, short for dual tone multi-frequency, 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): SPA9x2 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 iformation.

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 whne 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 miliseconds. Affects only OOB. Defaults to .1.

- I*nbound 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 Inbound DTMF boost.

Localizing Your SPA9x2 Phone

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

Field	Description
Set Local Date (mm/dd)	Enter the local date (*mm* represents the month and *dd* 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 (*hh* represents hours and *mm* 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. "Configuring Daylight Savings Time" section on page 79.
Daylight Saving Time 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. "Creating a Dictionary Server Script" section on page 81.
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

SPA9x2 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 SPA9x2 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.

78

- Manual Setup—The SPA9x2 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 SPA9x2 phone supports auto adjustment for daylight saving time (DST). 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/-1/5;end=9/-1/4;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 applie 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 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: For complete information on selecting a language, see the *LVS Installation and Configuration Guide.*

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.linksys-itsp.com

NOTE: You must have an ID, Username, and Password for this site.

Dictionaries are found in the *LVS* area of the *FW and Tools* section. 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, Sorbian), 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. For exceptions, see For exceptions, see
http://wwwin.cisco.com/linksys/smb/files/LVS52_RN_B-WEB_20080123_LB.pdf

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

1. Press the **Setup** button.

2. Select **18 Language**, then press the **Select** soft button.

3. Select **Option** to change the language.

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

Customizing SPA9x2 Phones

Use the *Phone* tab parameters to configure general parameters on the SPA9x2 that affect all extensions, such as:

General Phone Information

General Phone Notes

The following table describes the general phone fields.

Parameter	Description
Station Name	Name to identify this station.
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 voicemail include: • Advanced features such as voice mail to email conversion. • Calls can go to voice mail when the broadband connection is down.
Text Logo	Text logo to display when the phone boots up. A service provider, for example, can enter logo text as follows: • 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 For example, "Super\n%0aTelecom" will display: Super Telecom
BMP Picture Download URL	URL locating the bitmap (.BMP) file to display on the LCD background. For more information, see "Changing the Display Background (SPA942/962)" section on page 85
Select Logo	Select from Default, BMP Picture, Text Logo, or None. Defaults to Default.
Select Background Picture	Select from Default, BMP Picture, or None. Defaults to Default.
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 Wait	Amount of idle time before screen saver displays.

Parameter	Description
Screen Saver Icon	In screen saver mode, the phone LCD can display: 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 screeen.

Changing the Display Background (SPA942/962)

You can use a logo and picture to customize the background on your SPA942 and SPA962 LCD displays:

- SPA942: use 128 x 48 black and white image (1-bit color).

- SPA962: 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.

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

2. Log on to the web UI for the phone.

3. Click the **Phone** tab and refer to the General settings at the top of the page.

4. Select **BMP Picture** from the *Select Background Picture* drop-down list.

5. 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*
6. 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 selected in the *Select Background Picture* field is set to **BMP Picture).**

Using the Photo Album Feature on the SPA962

The photo album feature provides a refreshable display of photos on SPA962 phones. 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 re-gets the picture after the specified number of seconds.

Line Key Information

For each line (or extenison) on the SPA942 and SPA962, you can configure:

- Extension number or disabled

- A short name

- Shared call call appearance or private

Line Key Notes

- *Extension*: select an extension to use for this line key:

  – SPA922: 1 or disabled.

  – SPA942: 1, 2, 3, 4, or disabled.

  – SPA962: 1, 2, 3, 4, 5, 6, or disabled.

  Defaults to 1.

- *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. Phones share incoming calls and ring simultaneously. The first person to answer the call ends the call appearance.
If you select private, the call appearance is private and not shared with any other phone.
Defaults to private. See the next section.

Configuring Shared Line Extension on SPA9x2 Phones

Shared line appearance (SLA) allows multiple stations to share an extension number and manage a call as a group. When the active station places the call on a shared line key on hold, the call can be resumed from any of the sharing client stations simply by pressing the corresponding line key from another station.

To configure a group of client stations to share the same extension, you must set parameters in the *EXT* tab and the *Phone* tab:

1. Select one of the extensions on the client station to configure the share line.

NOTE: As a convention, 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**.

2. Under **Share Line Appearance**, set **Share Ext** to **shared**. Then under **Shared User ID**, set the user ID which is being shared. Set up the rest of the account information just like the primary EXT on the same station (User ID, Password, Proxy, and so on).

3. Click the **Phone** tab. Assign one or more line key on the client station to the shared extension. Set **Share Call Appearance** of the line key to **shared**.

NOTE: You can think of a shared line appearance as a physical line key on the client station.
    •   Line Key Parameters

You can configure shared call appearance (SCA) settings and line key LED patterns.
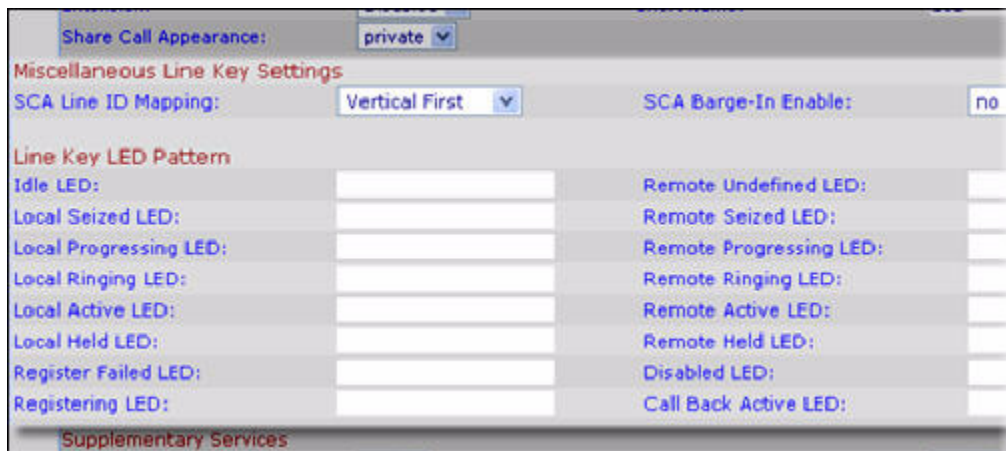
Miscellaneous Line Key Settings Notes

•   *SCA Line ID Mapping*: Specifies shared call appearance (SCA) line ID 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.
    If you select Vertical first, the next LED on the phone flashes with the second incoming call.
    If you select horizontal first, the same LED to flashes with the second incoming call.
    Defaults to Vertical first.

•   *SCA Barge-In Enable*: Enables SCA barge-in, which lets users haring call appearances add themselves to a call on a shared line. Defaults to no.

Line Key LED Pattern Notes

The following table describes all LED states.

NOTE: 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
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).
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).
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).

For more information on LEDs, see .

Supplementary Services Parameters

Supplementary services parameters define conference calls, blind call transfers, blocking caller IDs and so on for the phone. Most supplementary services are enabled by default.

- Select "yes" to enable a service on the phone. Except where noted, all parameters default to "yes."

- Select "no" to disable the service on the phone.

- "Serv" refers to "service."

Supplementary Services Notes

Abbreviations are spelled out in the following table.

Parameter Name	Enables/Disables
Conf	Three-way conference.
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.
DND	Do not disturb.
Block ANC	Blocks anonymous calls.
Call Back	Call back service, which is dialing the last received call.
Block CID	Blocks outbound caller ID.
Secure Call	Secure calls on the phone. See "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." section on page 36.
Cfwd All	Forwards all calls.
Cfwd Busy	Forwards calls only if the line is busy.
Cfwd On No Ans	Forwards calls only if the line is not answered.
Paging	Paging feature (see"Using Group Paging" section on page 91.)
Call Park	Call parking feature (see "Using Call Parking" section on page 91.)
Call Pick Up	Call pick up feature (see "Using Call Pick Up and Group Call Pick Up" section on page 91.)
ACD Login	Typically used for call centers, automatic call distributor (ACD) handles incoming calls and manages them based on a database of instructions. You can enable this with the SIP B parameter ("SIP Parameter Notes" section on page 46). Defaults to no (disabled)
Group Call Pick Up	Group call pick up feature (see "Using Call Pick Up and Group Call Pick Up" section on page 91).
ACD Ext	Extension used for handling ACD calls. Select 1-6, depending on your phone model. Defaults to 1.

Parameter Name	Enables/Disables
Serv Annc	Service announcement requests can be sent to a customer-supplied announcement server. Defaults to no (disabled).
Web Serv	Use of the web UI.

Using Group Paging

Group Paging lets the user page all the client stations at once. If the client station is on a call while a group page starts, the call is automatically placed on hold. 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 start a group page, make a call by dialing **pagegroup**. For convenience, this special number is included in the corporate directory. A speed dial or personal directory entry can also be defined locally on a specific phone.

Using Call Parking

If call parking is enabled, users can store a call so that they or someone else can retrieve it from another phone (for example, a phone at a co-worker's desk or in a conference room).

NOTE: This feature is supported only on teh PBS (SIP trunking) system. If the SPA9x2 is on a user agent (UA) , call parking is not available.

To park a call, perform the following steps when a call comes in on a SPA9x2:

1. Press the **park** soft button or enter ***38.**

2. Enter any number (for example, 1234). The call is parked.

3. To retrieve the call from the same phone or another SPA9x2, press **unpark** or enter ***39** .

4. Enter the number previously entered when you parked the call (for example, 1234). This unparks the call.

Using Call Pick Up and Group Call Pick Up

If *Call Pick Up* is enabled a user can answer a call that rings on a co-worker's phone by redirecting the call to their own phone:

1. Determine which station is ringing (by listening, for example). This is the target station.

2. Press the **Pickup** soft button or enter ***36**.

3. When *Enter Pick up number* appears, enter one of the following:

   – extension number of the target station, or

   – *Station Name* of the target station.

4. The phone LCD displays a list of calls currently ringing on the target station. Select the call to pick up from the list.

If *Group Call Pick Up* is enabled a user can answer a call that rings on *any* co-worker's phone by redirecting the call to their own phone:

1. Determine whether *any* client station is ringing (by listening, for example). It could be more than one.

2. Press the **GrPickup** soft button or enter ***37**.

3. The phone LCD lists the station names that are currently ringing. Select the target station you want to answer.

NOTE: The Station Name parameters must be defined for these stations; otherwise, the list is empty and pressing the Dial soft button reboots the phone. To define station names, see "General Phone Notes" section on page 83.

4. The phone LCD displays a menu of calls currently ringing on the target station. Select the call you want to pick up.

Ring Tone Parameters

You can define up to ten ring tones for a phone. You can define:

- The default ring tone for the extension

- Specific ring tones assigned to individual callsers in the personal directory. These override the default ring tone.

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.

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

To convert a file for use as a ring tone, obtain a .wav file less than 8 seconds in length and convert it using the Upload Ringtone utility available from the following website:

http://www.linksys.com/support

Audio Input Gain (dB) and Extension Mobility Settings

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

- 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).

Using Extension Mobility

If your SPA9x2 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.

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.

Extension Mobility Notes

- *Extension Mobility*: enables or disables EM. Defaults to no (disabled).

- *EM User Domain*: enter the Broadsoft domain for the phone.

- At the user end, a *Login* prompt appears on the phone display when EM is enabled. The user must enter their *User ID* and *Password*.

For more information, see http://www.broadsoft.com.

Customizing SPA9x2 Extensions

Use the *Extension* tabs to configure the following for each extension on your SPA9x2 phone:

- "General, Shared Line Appearance, and NAT Settings" section on page 95

- "Network Settings and SIP Settings" section on page 97

- "SIP Settings Notes" section on page 97

- "Call Feature Settings" section on page 99

- "Proxy and Registration Parameters" section on page 100

- "Configuring a SIP Proxy Server" section on page 101

- "Subscriber Information Parameters" section on page 102

- "Audio Configuration Parameters" section on page 102

- "Dial Plan Parameters" section on page 105

General, Shared Line Appearance, and NAT Settings

General Parameter Notes

To enable this line for service, select yes (the default). Otherwise, select no.

Share Line Appearance Notes

The following table describes Share Line Appearance parameters.

Parameter	Description
Share Line Appearance	Indicates whether this extension is shared with other stations or private. 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. LCDs of all phones that share line appearances show when a shared extension is in use. Defaults to private.
Shared User ID	The user assigned to the shared line. Defaults to blank.
Subscription Expires	The subscription expiration time with the SIP proxy. Defaults to 3600 seconds.

NAT Settings Notes

Parameter	Description
NAT Mapping Enable	Select yes to use externally mapped IP addresses and SIP/RTP ports in SIP messages. Otherwise, select no. Defaults to no.
NAT Keep Alive Enable	Select yes to send the configured NAT keep alive message periodically. Otherwise, select no. Defaults to no.
NAT Keep Alive Msg	Enter the keep alive message that is 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 receives NAT keep alive messages. If the value is $PROXY, keep alive messages are sent to the current or outbound proxy. Defaults to $PROXY.

Network Settings and SIP Settings

Network Settings Notes

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.
RTP CoS Value [0-7]	CoS value for RTP data. Defaults to 6.

SIP Settings Notes

You can configure the following SIP settings for each extension on a phone.

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.
Auth Resync-Reboot	If this feature is enabled, the SPA9x2 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.
Referor Bye Delay	Controls when the SPA9x2 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. Defaults to 4.
Refer Target Bye Delay	For the Refer Target Bye Delay, enter the appropriate period of time in seconds. Defaults to 0.
Sticky 183	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. Defaults to no.
Auth INVITE	When enabled, authorization is required for initial incoming INVITE requests from the SIP proxy.
Referee Bye Delay	For the Referee Bye Delay, enter the appropriate period of time in seconds. Defaults to 0.

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 for extension 1 through 5065 for extension 6.
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.
Auth Resync-Reboot	If this feature is enabled, the SPA9x2 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.

Call Feature Settings

You can set various features on the extension, such as message waiting, default ring, mailbox ID, and so on.

Call Feature Notes

- If the Message Waiting Indicator at the top of your SPA9x2 is not lighting when voicemail is left, set the *Message Waiting* field to **yes**.

- *MOH Server*: User ID or URL of the music-on-hold (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 SPA9000 IP PBX, defaults to imusic. For more information, see the *LVS Administration Guide*.

- *Default Ring*: Type of ring, which corresponds to the Ring Tone on the Phone tab. Choose from No Ring, 1 through 10, User 1, or User 2.
  Defaults to 1.

- *Conference Bridge URL*: URL used to join into a conference call, generally in the form of the word "conference" or *user@IPaddress:port*.
  Defaults to blank.

- *Mailbox ID*: Enter the voice mail line number and phone extension. For example, 2101 indicates line 2, phone extension 101.

Proxy and Registration Parameters

The following table describes proxy and registration 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: • Use Outbound Proxy is set to no or • 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.

Parameter	Description
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.
Ans Call Without Reg	If enabled, the user does not have to be registered with the proxy to answer calls. Defaults to no.
Use DNS SRV	Enables DNS SRV lookup for the proxy and outbound proxy. Defaults to no.
DNS SRV Auto Prefix	Enables the phone to automatically prepend the proxy or outbound proxy name with _sip._udp when performing a DNS SRV lookup on that name. Defaults to no.
Proxy Fallback Intvl	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. 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. Defaults to 3600 seconds.
Proxy Redundancy Method	Select Normal or Based on SRV port. The phone creates an internal list of proxies returned in the DNS SRV records. If you select Normal, the list contains proxies ranked by weight and priority. If you select Based on SRV, the phone uses normal, then inspects the port number based on the first-listed proxy port. Defaults to Normal.

Configuring a SIP Proxy Server

You can configure SIP proxy servers in the Proxy and Registration section of the EXT tabs:

1. Select **Admin > Advanced**.

2. Enter your service provider name in the *Proxy* field.

3. Select **yes** in the *Use DNS SRV* field.

4. Select **yes** in the *DNS SRV Auto Prefix* field.

The phone tries to register and the server sends a list of IP addresses in order of priority. The phone contacts the highest priority proxy server whenever possible.

Subscriber Information Parameters

Subscriber Information Notes

The following table describes all subcriber information parameters.

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.

Audio Configuration Parameters

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 G729a 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 G711a and G711u.

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.

103

Audio Configuration Notes

The following tables discusses all audio configuration parameters.

Parameter	Description
Preferred Codec	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. See "Supported Codecs" section on page 37. Defaults to G711u.
Use Pref Codec Only	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. Defaults to no.
Second Preferred Codec	If the first codec fails, this codec is tried. Defaults to unspecified.
Third Preferred Codec	If the second codec fails, this codec is tried. Defaults to unspecified.
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.
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.
G726-40 Enable	To enable the use of the G.726 codec at 40 kbps, select yes. Otherwise, select no. Defaults to yes.
Release Unused Codec	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. Defaults to yes.

Parameter	Description
DTMF Process AVT	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. Defaults to yes.
Silence Supp Enable	To enable silence suppression so that silent audio frames are not transmitted, select yes. Otherwise, select no. See "Factors Affecting Voice Quality" section on page 39. Defaults to no.
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.

Dial Plan Parameters

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.

Dial Plan Notes

This table describes all Dial Plan parameters:

Parameter	Description

Dial Plan	Dial plan script for the selected extension.
	
	The dial plan syntax allows the designation of three parameters for use with a specific gateway:
	• uid – the authentication user-id
	• pwd – the authentication password
	• nat – if this parameter is present, use NAT mapping
	Separate each parameter with a semi-colon (;).
	Example 1:
	`*1xxxxxxxxxx<:@fwdnat.pulver.com:5082;uid=jsmith;pwd=xyz`
	
	Example 2:
	`*1xxxxxxxxxx<:@fwd.pulver.com;nat;uid=jsmith;pwd=xyz`
	
	Example 3:
	`[39]11<:@gw0>`
	
	The phone recognizes locally configured PSTN gateways (represented by "gw0" in the above example).
	See Appendix B "Configuring a Dial Plan" for more detailed information.

Caller ID Map	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.
Enable IP Dialing	Enable or disable IP dialing. Defaults to no.

Parameter	Description
Emergency Number	Enter a comma-separated list of emergency numbers. When one of these numbers is dialed, the unit disables processing of *CONF, HOLD*, and other similar softkeys or buttons to avoid accidentally putting the current call on hold. The phone also disables hook flash event handling.
	Only the far end can terminate an emergency call.
	The phone is restored to normal after the call is terminated and the SPA9x2 is back on-hook.
	Maximum number length is 63 characters.
	Defaults to blank (no emergency number).

Customizing Users

Use the *User* tab to configure the following features for each extension on your SPA9x2 phone. Many of these features are designed for the end-user to customize. Several also be configured directly on the phone. See:

- *"Call Forward and Speed Dial Parameters"* section on page 108

- *"Supplementary Services"* section on page 109

- *"Setting RSS Newsfeeds on the SPA962 Phone"* section on page 110

- *"Web Information Service Settings (SPA962 Only)"* section on page 110

- *"Audio Volume Settings"* section on page 111

- *"Phone GUI Color Settings (SPA962 Only)"* section on page 112

Call Forward and Speed Dial Parameters

Call Forward Notes

See *"Vertical Service Activation Codes (Star Codes)"* section on page 69 for information on call forwarding parameters.

Speed Dial Notes

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.

Supplementary Services

Supplementary Services Notes

The SPA9x2 provides native support of a large set of enhanced or supplementary services (also known as star codes). All of these services are optional. In this section, you can enable or disable specific supplementary services for users.

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 SPA9x2.

Most supplementary services are described in *"Vertical Service Activation Codes (Star Codes)" section on page 69*.

- *Text Message*: SPA9x2 phones can receive and display text messages via SIP according to RFC3428. Select **yes** to enable receipt of SIP MESSAGE messages and display of text from the PROXY.

- *Text Message From 3rd Party*: select **yes** to enable receipt and display of text messages from any third party directly, without proxy involvement.

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

109

Setting RSS Newsfeeds on the SPA962 Phone

The SPA962 provides the option to view RSS newsfeeds. You can enable newsfeeds on the web UI or on the phone LCD:

1. Press the **Setup** button, then scroll to option **21 Web Service** (or press 21 on the keypad). Press the **Select** soft button.

2. Select one of the following options:

   - 1 Stock Quote (default to CSCO)

   - 2 Traffic (defaults to CA, Cisco, San Jose, 95134)

   - 3 Local News (defaults to URL http://rss.cnn.com/rss/cnn_us.rss)

   - 4 World News (defaults to URL http://newsrss.bbc.co.uk/rss/newsonline_uk_edition/ world/rss.xml)

   - 5 Finance News (defaults to URL http://finance.yahoo.com/rss/topstories)

   - 6 Sports News (defaults to URL http://rss.news.yahoo.com/rss/sports

   - 7 Politics News (defaults to URL http://rss.news.yahoo.com/rss/politics)

You can change options 3 through 7 on the user tab, as shown in the following section.

Web Information Service Settings (SPA962 Only)

You can also configure web services on the web UI.

Linksys SPA9x2 Phone Administration Guide 110

Download from Www.Somanuals.com. All Manuals Search And Download.

Web Information Service Notes

Parameter	Description
RSS Feed URLs 1-5	URLs for Local and World news, Finance, Sports, and Politics. For more information, including default URLs, see "Setting RSS Newsfeeds on the SPA962 Phone" section on page 110
Feed Data Update Interval	Interval at which the news feed is updated Defaults to 15 minutes.
Scrolling Timer	Defines how fast the information scrolls on the phone LCD. Information scrolls when it does not all fit on the LCD screen. Defaults to 0.8 seconds.

Traffic Information Settings

Change the defaults to your local state, city, street, and zip code for local traffic information.

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

Audio Volume Settings

The default audio volumes entered here can be modified using the volume control button on the SPA9x2, then pressing the **Save** soft button.

Audio Volume Notes

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.

Phone LCD Notes

- *LCD Contrast*: Users can brighten or darken the SPA9x2 LCD contrast. Values range from 1-10.

- *Back Light Timer*: Displays the time afer which the LCD turns off with non-use. Defaults to 10 seconds. This field does not apply to the SPA922

Phone GUI Color Settings (SPA962 Only)
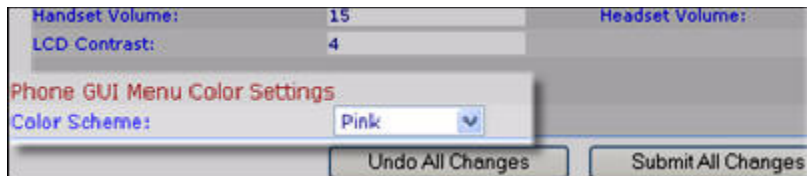
NOTE: The following parameters apply only to the SPA962.
Chose from:

- Light Blue

- Light Green

- Pink

- Silver

Defaults to Light Blue.

Using the SPA932 Sidecar

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

The SPA932 connects to the SPA962 with the attachment arm provided (not shown). It obtains power directly from the SPA962 and does not require a separate power supply. Two SPA932 units can be attached to a single SPA962 to monitor a total of 64 separate lines.

For more information about installing and configuring the SPA932 with LVS, see the *Linksys Voice System Installation and Configuration Guide.*

SPA932 features include:

- 32 programmable speed-dial or direct station select (DSS) buttons/LEDs

- Three-color LEDs (red, green, and amber)

- 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 LED colors.

.

Port/LED	Meaning
AUX IN	Connects to the SPA962
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 Sidecar

To configure the server to use the SPA932, configure each extension that will be monitored by the SPA932. 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 sidecar:

1. Complete the physical installation of the SPA932 unit, which connects it to the SPA962. 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.*

2. Configure one of the following SIP proxy servers:

   – SPA9000: see Configuring the SPA9000 for the SPA932 (verify that your version of the SPA9000 supports the SPA932)

   – Broadsoft: see Configuring the Broadsoft Server for the SPA932

   – Asterisk: see Configuring the Asterisk Server for the SPA932

3. Configure the SPA932 using the web UI. The web server must be connected to the SPA962 to which the SPA932 is physically attached.

Note The SPA962 must be running Release 5.1.12 or later. Also, CTI must be enabled on the SPA962 for an attached SPA932 to properly monitor the SPA962's line status. See "SIP Parameter Notes" section on page 46.

Configuring the SPA9000 for the SPA932

The default multicast autoconfiguration assigns each phone with an extension starting with 101.

Even if you use the default autoconfiguration performed by the SPA9000, you must still enable Computer Telephony Integration (CTI) on the SPA9000 web UI, as shown in the following procedure:

1. Connect to the web UI for the SPA9000.

2. Click **Admin Login** on the web UI page.

3. If necessary, enter **admin** and your password, then **Advanced**.

4. Click the **SIP** tab.

5. In the *SIP Parameters* section, select **yes** from the *CTI Enable* drop-down list.

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) User Monitor 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:

1. On the Broadsoft server, in the user profile used by the SPA962 to register with the Broadsoft server, select **Assign Services**.

2. On the Assign Services page, from the Available Services column, move **Busy Lamp** and **Phone Status Monitoring** to the User Services column.

3. Define the List URI used by the SPA932 to register for BLF monitoring service with the Broadsoft server. This value must be entered following the **sub=** keyword, when configuring each key on the 932 (see "Monitoring the SPA932" section on page 121).

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 SPA962 extension to use for each list.

4. On the Phone Status Monitoring page, move users that need to be monitored from the Available User columnn to the Monitored User column.

5. Click the **Add** (or **Add All**) button to move each user to the Monitored Users column.

6. 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.

7. Save and enable your configuration changes on the Broadsoft server.

Configuring the Asterisk Server for the SPA932

To configure the Asterisk server to allow the SPA932 to register for BLF monitoring:
1. Add a context in the extensions.conf file.

2. Add a Subscribecontext command to point to the context in the sip.conf file.

3. Configure the SPA932 to register with the Asterisk server (see "Configuring the SPA932" section on page 117.

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

```
[home]
exten => 3500,1,Dial(SIP/3500/20)
exten => 3500,hint,SIP/3500
exten => 3500,2,Voicemail,u3500
exten =>3500,1,3,hangup
...
```

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:

1. Connect to the web UI *for the SPA962* to which the SPA932 is connected.

2. Click **Admin**/**Advanced** on the web UI page.

3. Click the **SIP** tab.

4. Select **yes** from the *CTI Enable* drop-down list.

5. Click the **SPA932** tab. (If you do not see the SPA932 tab, check your firmware version, and upgrade as necessary.)

The system displays the SPA932 web page.

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.

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

    – SPA9000

    – Broadsoft

    – Asterisk

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.)

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 118.

10. Click **Submit All Changes**.

NOTE: 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. 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 SPA962 to which it is connected. Assign the LEDs to any of the other five extensions on the SPA962 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—Use this keyword 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 SPA962 extensions that they are assigned to. By default, LEDs on the SPA932 are assignd to the first configured extension on the connected SPA962. You can optionaly assign LEDs to any other SPA962 extensions using vid=keyword. Use this field to identify the SPA962 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 SPA962.
  Only use the vid= keyword in the first entry assigned to each SPA962 extension.

Subsequent keys will use the same extension. See "SPA932 Parameter Notes" section on page 120.

Assign SPA932 LEDs to SPA962 extensions.

By default, all 32 keys on the SPA932 are assigned to the first extension on the 962 (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 SPA962 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 *Linksys 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 SPA962's extension 2

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@;**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.

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

SPA932 Parameter Notes

Parameters in the SPA932 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 SPA962 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 SPA962 can have up to two SPA932s attached).
Server Type	Selects the type of server used (SPA9000, Broadsoft, or Askterisk).
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 sidecar are returned to off, all the keys become amber. The SPA962 must be rebooted after the test is completed.
Call Pickup Code	The star code used for picking up a ringing call. Defaults to *98.

Parameter	Description
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 118.*

Monitoring the SPA932

To display the status of the SPA932, click the SPA932 Status link on the SPA932 tab. The status of each sidecar attached (Unit 1 and Unit 2) is shown. Parameters are read-only.
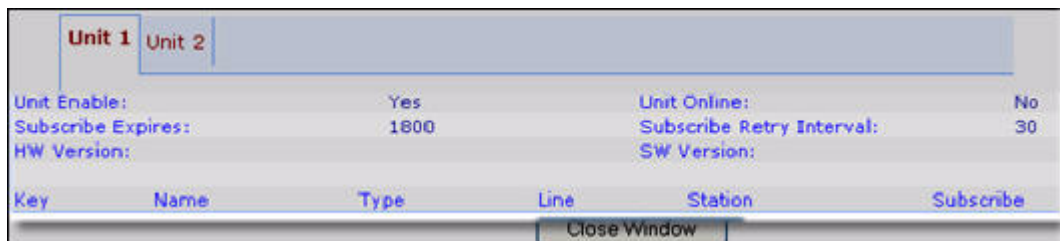
SPA932 Unit Monitoring Notes

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 sidecar unit.
Type	Displays the function enabled for each key (1-32) on the SPA932 sidecar unit.

Parameter	Description
Line	Displays the extension assigned to each key (1-32) on the SPA932 sidecar unit.
Station	Displays the subscribe URI configured for each key (1-32) on the SPA932 sidecar 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).

SPA9x2 Phone Field Reference

This chapter describes the fields within each section of the following web UI pages:

✏️

Note   For information about the Provisioning page, see the *Linksys SPA Provisioning Guide*.

Info Tab

This section describes the fields for the following headings on the Info tab:

✏️

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.
Current IP	Displays the current IP address assigned to the SPA9x2.

Parameter	Description
Host Name	Displays the current host name assigned to the SPA9x2 (defaults to SipuraSPA).
Domain	Displays the network domain name of the SPA9x2.
Current Netmask	Displays the network mask assigned to the SPA9x2.
Current Gateway	Displays the default router assigned to the SPA9x2.
Primary DNS	Displays the primary DNS server assigned to the SPA9x2.
Secondary DNS	Displays the secondary DNS server assigned to the SPA9x2.

Product Information

Parameter	Description
Product Name	Model number of the SPA9x2.
Serial Number	Serial number of the SPA9x2.
Software Version	Version number of the SPA9x2 software.
Hardware Version	Version number of the SPA9x2 hardware.
MAC Address	MAC address of the SPA9x2.
Client Certificate	Status of the client certificate, which authenticates the SPA9x2 for use in the ITSP network. This field indicates if the client certificate is properly installed in the SPA9x2.
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 SPA9x2.

Phone Status

Parameter	Description
Current Time	Current date and time of the system; for example, 10/3/2003 16:43:00.

Parameter	Description
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.
Broadcast Bytes Recv	Total number of broadcast bytes received and processed.
Broadcast Pkts Dropped	Total number of broadcast packets received but not processed.
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.

Ext Status

Parameter	Description
Registration State	Shows "Registered" if the phone is registered, "Not Registered" if the phone is not registered.
Last Registration At	Last date and time the line was registered.

Parameter	Description
Next Registration In	Number of seconds before the next registration renewal.
Message Waiting	Indicates whether you have new voicemail waiting: Yes or No. This is updated when voicemail notification is received.
Mapped SIP Port	Port number of the SIP port mapped by NAT.

)

Line /Call Status

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.
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.
Rount Trip Dealy	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.

Parameter	Description
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

Parameter	Description
Restricted Access Domains	This feature is used when implementing software customization.
Enable Web Server	Enable/disable web server of SPA9x2 This feature should only be used on firmware version 1.0.9 or later. Defaults to yes.
Web Server Port	Port number of the SPA9x2 web UI. Defaults to 80.

Parameter	Description
Enable Web Admin Access	Lets you enable or disable local access to the web UIweb UI. 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.

Internet Connection Type and Static IP Settings

Parameter	Description
DHCP	Enable or disable DHCP. Defaults to DHCP.
Static IP	IP address. Blank of DHCP assigned address is used.
Netmask	Network mask. Blank if DHCP assigned.
Gateway	Default router IP address. Blank if DHCP assigned.

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 SPA9x2.
Domain	The network domain of the SPA9x2.
Primary DNS	DNS server used by SPA9x2 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 SPA9x2 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 SPA9x2 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 SPA9x2. Defaults to parallel.
Syslog Server	Specify the syslog server name and port. This feature specifies the server for logging SPA9x2 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 SPA9x2 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.

Parameter	Description
Secondary NTP Server	IP address or name of secondary NTP server.

VLAN Settings

Parameter	Description
Enable VLAN	Enable this parameter if the switch to which the Linksys IP phone is connected uses VLAN tagging.
VLAN ID	Enter the VLAN tag for the VLAN to which the Linksys 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.

SIP Tab

This section describes the fields for the following headings on the SIP tab:

- SIP Parameters, page -130

- SIP Timer Values (sec), page -133

- Response Status Code Handling, page -134

- RTP Parameters, page -135

- SDP Payload Types, page -137

- NAT Support Parameters, page -139

- Linksys Key System Parameters, page -140

SIP Parameters

Parameter	Description
Max Forward	SIP Max Forward value, which can range from 1 to 255. Defaults to 70.

Parameter	Description
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.
SIP Reg User Agent Name	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. Defaults to blank.
SIP Accept Language	Accept-Language header used. To access, click the SIP tab, and fill in the SIP Accept Language field. There is no default (this indicates SPA9x2 does not include this header). If empty, the header is not included.
DTMF Relay MIME Type	MIME Type used in a SIP INFO message to signal a DTMF event. This field must match that of the Service Provider. Defaults to application/dtmf-relay.
Remove Last Reg	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. Defaults to no.
Escape Display Name	Lets you keep the Display Name private. Select yes if you want the SPA9x2 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. Defaults to yes.
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	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. Defaults to yes.
Random REG CID On Reboot	If set to yes, the Linksys IP phone uses a different random call-ID for registration after the next software reboot. If set to no, the Linksys IP phone tries to use the same call-ID for registration after the next software reboot. The Linksys IP phone always uses a new random Call-ID for registration after a power-cycle, regardless of this setting. Defaults to no.
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.
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.

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.
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.
SIP Timer J	Non-INVITE response hang-around time, which can range from 0 to 64 seconds. Defaults to 16 seconds.
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	Interval to wait before the SPA9x2 retries registration after failing during the last registration. Defaults to 30.
Reg Retry Long Intvl	When registration fails with a SIP response code that does not match<Retry Reg RSC>, the SPA9x2 waits for the specified length of time before retrying. If this interval is 0, the SPA9x2 stops trying. This value should be much larger than the Reg Retry Intvl value, which should not be 0. Defaults to 1200.

Parameter	Description
Reg Retry Random Delay	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. Defaults to blank, which disables this feature.
Reg Retry Long Random Delay	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. Defaults to blank, which disables this feature.
Reg Retry Intvl Cap	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. Defaults to blank, which disables the exponential back-off feature.
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.

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.

Parameter	Description
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.
Retry Reg RSC	Interval to wait before the SPA9x2 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. <RTP Port Min> and <RTP Port Max> should define a range that contains at least 4 even number ports, such as 100 – 106. Defaults to 16384.
RTP Port Max	Maximum port number for RTP transmission and reception. Defaults to 16482.
RTP Packet SizeSPA901	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 SPA9x2 terminates the call. If value is set to 0, the SPA9x2 ignores the limit on ICMP errors. Defaults to 0.

135

Parameter	Description
RTCP Tx Interval	Interval for sending out RTCP sender reports on an active connection. It can range from 0 to 255 seconds. During an active connection, the SPA9x2 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 Linksys/SPA9x2-1.0.31(b)). The NTP timestamp used in the SR is a snapshot of the SPA9x2's local time, not the time reported by an NTP server. If the SPA9x2 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 SPA9x2 web page. Defaults to 0.
No UDP Checksum	Select yes if you want the SPA9x2 to calculate the UDP header checksum for SIP messages. Otherwise, select no. Defaults to no.
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 yes.
Stats In BYE	Determines whether the SPA9x2 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.

SDP Payload Types

The configured dynamic payloads are used for outbound calls only where the SPA9x2 presents the SDP offer. For inbound calls with a SDP offer, the SPA9x2 follow the caller dynamic payload type assignments.

The SPA9x2 uses the configured codec names in its outbound SDP. The SPA9x2 ignores the codec names in incoming SDP for standard payload types (0 – 95). For dynamic payload types, the SPA9x2 identifies the codec by the configured codec names. Comparison is case-insensitive.

Parameter	Description
AVT Dynamic Payload	AVT dynamic payload type. Ranges from 96-127. Defaults to 101.
INFOREQ Dynamic Payload	INFOREQ dynamic payload type. Defaults to blank.
G726r16 Dynamic Payload	G.726-16 dynamic payload type. Ranges from 96-127. Defaults to 98.
G726r24 Dynamic Payload	G.726-24 dynamic payload type. Ranges from 96-127. Defaults to 97.
G726r32 Dynamic Payload	G726r32 dynamic payload type. Defaults to 2.
G726r40 Dynamic Payload	G.726-40 dynamic payload type. Ranges from 96-127. Defaults to 96.
EncapRTP Dynamic Payload	EncapRTP Dynamic Payload type. Defaults to 112.
RTP-Start-Loopback Dynamic	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.

Parameter	Description
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.
G726r24 Codec Name	G.726-24 codec name used in SDP. Defaults to G726-24.
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.
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.
EncapRTP Codec Name	EncapRTP codec name used in SDP. Defaults to encaprtp.

NAT Support Parameters

Parameter	Description
Handle VIA received	If you select yes, the SPA9x2 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. Defaults to no.
Handle VIA rport	If you select yes, the SPA9x2 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. Defaults to no.
Insert VIA received	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. Defaults to no.
Insert VIA rport	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. Defaults to no.
Substitute VIA Addr	Lets you use NAT-mapped IP:port values in the VIA header. Select yes or no from the drop-down menu. Defaults to no.
Send Resp To Src Port	Sends responses to the request source port instead of the VIA sent-by port. Select yes or no from the drop-down menu. Defaults to no.
STUN Enable	Enables the use of STUN to discover NAT mapping. Select yes or no from the drop-down menu. Defaults to no.

Parameter	Description
STUN Test Enable	If the STUN Enable feature is enabled and a valid STUN server is available, the SPA9x2 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 SPA9x2 detects symmetric NAT or a symmetric firewall, NAT mapping is disabled. Defaults to no.
STUN Server	IP address or fully-qualified domain name of the STUN server to contact for NAT mapping discovery.
EXT IP	External IP address to substitute for the actual IP address of the SPA9x2 in all outgoing SIP messages. If 0.0.0.0 is specified, no IP address substitution is performed. If this parameter is specified, the SPA9x2 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. Defaults to blank.
EXT RTP Port Min	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. Defaults to blank.
NAT Keep Alive Intvl	Interval between NAT-mapping keep alive messages. Defaults to 15.

Linksys Key System Parameters

Parameter	Description
Linksys Key System	Enable or disable the Linksys Key System on the IP phone. Defaults to yes.

Parameter	Description
Multicast Address	The multicast address is used by the SPA9000 to communicate with the Linksys IP 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 *Linksys SPA Provisioning Guide.*

Regional Tab

This section describes the fields for the following headings on the Regional tab:

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).
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).

Parameter	Description
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).
Ring Back Tone	Played during an outbound call when the far end is ringing. Defaults to 440@-19,480@-19;*(2/4/1+2).
Call Waiting Tone	Played when a call is waiting. Defaults to 440@-10;30(.3/9.7/1)
Confirm Tone	Brief tone to notify the user that the last input value has been accepted. Defaults to 600@-16; 1(.25/.25/1).
SIT1 Tone	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. Defaults to 985@-16,1428@-16,1777@-16;20(.380/0/1,.380/0/2,.380/0/3,0/4/0).
SIT2 Tone	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. Defaults to 914@-16,1371@-16,1777@-16;20(.274/0/1,.274/0/2,.380/0/3,0/4/0).

Parameter	Description
SIT3 Tone	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. Defaults to 914@-16,1371@-16,1777@-16;20(.380/0/1,.380/0/2,.380/0/3,0/4/0)
SIT4 Tone	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. Defaults to 985@-16,1371@-16,1777@-16;20(.380/0/1,.274/0/2,.380/0/3,0/4/0).
MWI Dial Tone	Played instead of the Dial Tone when there are unheard messages in the caller's mailbox. Defaults to 350@-19,440@-19;2(.1/.1/1+2);10(*/0/1+2).
Cfwd Dial Tone	Played when all calls are forwarded. Defaults to 350@-19,440@-19;2(.2/.2/1+2);10(*/0/1+2).
Holding Tone	Informs the local caller that the far end has placed the call on hold. Defaults to 600@-19*(.1/.1/1,.1/.1/1,.1/9.5/1).
Conference Tone	Played to all parties when a three-way conference call is in progress. Defaults to 350@-19;20(.1/.1/1,.1/9.7/1).
Secure Call Indication Tone	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. Defaults to 397@-19,507@-19;15(0/2/0,.2/.1/1,.1/2.1/2).
Page Tone	Specifies the tone transmitted when the paging feature is enabled. Defaults to 600@-16;.3(.05/0.05/1).
Alert Tone	Played when an alert occurs. Defaults to 600@-19;.2(.05/0.05/1).

Distinctive Ring Patterns

Parameter	Description
Cadence 1	Cadence script for distinctive ring 1. Defaults to 60(2/4).
Cadence 2	Cadence script for distinctive ring 2. Defaults to 60(.3/.2, 1/.2,.3/4.
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 to60(.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 255.
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 SPA9x2 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.
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.

Parameter	Description
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.
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.

Parameter	Description
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.
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	These codes tell the SPA9x2 what to do when the user places the current call on hold and is listening to the second dial tone. 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 SPA9x2 to perform a blind transfer to a target number that is prepended by the service *code. For example, after the user dials *98, the SPA9x2 plays a special dial tone called the Prompt Tone while waiting for the user the enter a target number (which is checked according to dial plan as in normal dialing). When a complete number is entered, the SPA9x2 sends a blind REFER to the holding party with the Refer-To target equals to *98<target_number>. This feature allows the SPA9x2 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 SPA9x2. You can empty the corresponding *code that you do not want to SPA9x2 to process.

Parameter	Description
Feature Dial Services Codes	These codes tell the SPA9x2 what to do when the user is listening to the first or second dial tone. 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 SPA9x2 to call the target number prepended by the *code. For example, after user dials *72, the SPA9x2 plays a prompt tone awaiting the user to enter a valid target number. When a complete number is entered, the SPA9x2 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). The *codes should not conflict with any of the other vertical service codes internally processed by the SPA9x2. You can empty the corresponding *code that you do not want to SPA9x2 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'. Below are a list of allowed tone parameters (note the use of back quotes surrounding the parameter w/o 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 place, x is any digit not used above If no tone parameter is specified, the SPA9x2 plays 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, simple add that *code in the dial plan and the SPA9x2 send INVITE *73@..... as usual when user dials *73.

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 G723 Code	Makes this codec the preferred codec for the associated call. Defaults to *01723.
Force G723 Code	Makes this codec the only codec that can be used for the associated call. Defaults to *02723.
Prefer G726r16 Code	Makes this codec the preferred codec for the associated call. Defaults to *0172616.

Parameter	Description
Force G726r16 Code	Makes this codec the only codec that can be used for the associated call. Defaults to *0272616.
Prefer G726r24 Code	Makes this codec the preferred codec for the associated call. Defaults to *0172624.
Force G726r24 Code	Makes this codec the only codec that can be used for the associated call. Defaults to *0272624.
Prefer G726r32 Code	Makes this codec the preferred codec for the associated call. Defaults to *0172632.
Force G726r32 Code	Makes this codec the only codec that can be used for the associated call. Defaults to *0272632.
Prefer G726r40 Code	Makes this codec the preferred codec for the associated call. Defaults to *0172640.
Force G726r40 Code	Makes this codec the only codec that can be used for the associated call. Defaults to *0272640.
Prefer G729a Code	Makes this codec the preferred codec for the associated call. Defaults to *01729.
Force G729a Code	Makes this codec the only codec that can be used for the associated call. Defaults to *02729.

Miscellaneous

Parameter	Description
Set Local Date (mm/dd)	Sets the local date (mm represents the month and dd represents the day). The year is optional and uses two or four digits.
Set Local Time (HH/mm)	Sets the local time (hh represents hours and mm 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.

153

Parameter	Description
Daylight Saving Time Rule	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.
	This is the format of the rule: Start = <start-time>; end=<end-time>; save = <save-time>.
	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 <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]]]
	The <month> value equals any value in the range 1-12 (January-December).
	The <day> value equals [+|-] any 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.
	The abbreviation HH stands for hours (0-23).
	The abbreviation mm stands for minutes (0-59).
	The abbreviation ss stands for seconds (0-59).
	The default Daylight Saving Time Rule is start=4/1/7;end=10/-1/7;save=1.
Daylight Savings Time Enable	
DTMF Playback Level	Local DTMF playback level in dBm, up to one decimal place. Defaults to -16.
DTMF Playback Length	Local DTMF playback duration in milliseconds. Defaults to .1.

Parameter	Description
Inband DTMF Boost	Controls the amount of amplification applied DTMF signals. Choices are OdB, 3dB, 6dB, 9dB, 12dB, 15dB, or 18dB. Defaults to 12dB.

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 and 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>;`
	
	The following is an example value:
	
	`<Dictionary_Server_Script ua="na">`
	`serv=tftp://192.168.1.119/`
	`;d0=English;x0=enS_v101.xml;d1=Spanish;x1=esS_v101.xm`
	`l </Dictionary_Server_Script>`

Language Selection	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:
	
	`<Language_Selection ua="na">`
	`</Language_Selection>`
	
	Defaults to blank and the maximum number of characters is 512. The following is an example:
	
	`<Language_Selection ua="na"> Spanish`
	`</Language_Selection>`

Phone Tab

This section describes the fields for the following headings on the Phone tab:

-

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 voicemail include: • Advanced features such as voice mail to email conversion. • Calls can go to voice mail when the broadband connection is down.

Text Logo	Text logo to display when the phone boots up. A service provider, for example, can enter logo text as follows:
	
	• 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
	For example, "Super\n%0aTelecom" will display:
	    Super
	    Telecom

BMP Picture Download URL	URL locating the bitmap (.BMP) file to display on the LCD background. For more information, see "Changing the Display Background (SPA942/962)" section on page 85
Select Logo	Select from Default, BMP Picture, Text Logo, or None. Defaults to Default.
Select Background Picture	Select from Default, BMP Picture, or None. Defaults to Default.

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 Wait	Amount of idle time before screen saver displays.
Screen Saver Icon	In screen saver mode, the phone LCD can display: 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 screeen.

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 .

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.

Miscellaneous Line Key Settings

Parameter	Description
SCA Line ID Mapping	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. 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. 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.
SCA Barge-In Enable	Enables the SCA Barge-In. Defaults to no.

Line Key LED Pattern

Parameter	Description
Idle LED	LED pattern during the Idle state, where the call appearance is not is 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 application if the call appearance is not shared. Leaving this entry blank indicates the default value of c=r;p=d.

Parameter	Description
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.
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 is 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.

Parameter	Description
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.
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.
Cfwd All Serv	Enable/disable call-forward-all service. Defaults to yes.

Parameter	Description
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

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.
Ring1	Ring tone script for ring 1. Defaults to n=Classic-1;w=3;c=1.

Auto Input Gain (dB)

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.

Parameter	Description
Speakerphone Input Gain	The amount of amplification to apply to the audio input signal for the speakerphone. Defaults to zero.

Extension Mobility

You can use extension mobility currently with Broadsoft CMS. For more information, see .

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.

Ext Tab

The Ext tabs vary by phone:

- SPA922: Ext 1 tab

- SPA942: Ext 1, 2, 3,and 4 tabs

- SPA962: Ext 1, 2, 3, 4, 5, and 6 tabs

This section describes the fields for the following Ext tab headings:

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.
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	Determines how jitter buffer size is adjusted by the SPA9x2. 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.

SIP Settings

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.
Auth Resync-Reboot	If this feature is enabled, the SPA9x2 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.

Parameter	Description
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.
Referor Bye Delay	Controls when the SPA9x2 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. Defaults to 4.
Refer-To Target Contact	
Referee Bye Delay	For the Referee Bye Delay, enter the appropriate period of time in seconds. Defaults to 0.

Parameter	Description
SIP Debug Option	SIP messages are received at or sent from the proxy listen port. This feature controls which SIP messages to log. Choices are as follows: • 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. OPT|NTFY|REG—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. OPT|NTFY|REG—Logs all SIP messages in full text except for OPTIONS, NOTIFY, and REGISTER requests/responses. Defaults to none.
Refer Target Bye Delay	For the Refer Target Bye Delay, enter the appropriate period of time in seconds. Defaults to 0.
Sticky 183	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. Defaults to no.

Parameter	Description
Auth INVITE	When enabled, authorization is required for initial incoming INVITE requests from the SIP proxy.
Ntfy Refer On 1xx-To-Inv	
Use Anonymous With RPID	

Call Feature Settings

Parameter	Description
Blind Attn-Xfer Enable	Enables the SPA9x2 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 SPA9x2 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. Defaults to no.
MOH Server	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. Defaults to imusic when used with a SPA9000 IP PBX.
Message Waiting	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. 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. Defaults to No.
Auth Page	Specifies whether to authenticate the invite before auto answering a page. Defaults to No.

Parameter	Description
Default Ring	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. Defaults to 1.
Auth Page Realm	Identifies the Realm part of the Auth that is accepted when the Auth Page parameter is set to Yes. This parameter accepts alphanumeric characters. Defaults to blank.
Conference Bridge URL	This is the URL used to join into a conference call, generally in the form of the word "conference" or "user@IPaddress:port". Defaults to blank.
Auth Page Password	Identifies the password used when the Auth Page parameter is set to Yes. This parameter accepts alphanumeric characters. Defaults to blank.
Mailbox ID	Identifies the voicemail box number/ID for the phone. Defaults to blank.
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.
Make Call Without Reg	Allow making outbound calls without successful (dynamic) registration by the unit. If no, the dial tone will not play unless registration is successful. Defaults to no.
Register Expires	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. Defaults to 60.
Ans Call Without Reg	If enabled, the user does not have to be registered with the proxy to answer calls. Defaults to no.
Use DNS SRV	Whether to use DNS SRV lookup for Proxy and Outbound Proxy. Defaults to no.
DNS SRV Auto Prefix	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. Defaults to no.

Parameter	Description
Proxy Fallback Intvl	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). Defaults to 3600
Proxy Redundancy Method	Select Normal or Based on SRV port. The phone creates an internal list of proxies returned in the DNS SRV records. If you select Normal, the list contains proxies ranked by weight and priority. If you select Based on SRV, the phone uses normal, then inspects the port number based on the first listed proxy port. Defaults to Normal.

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.

Parameter	Description
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.

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	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. Defaults to G711u.
Use Pref Codec Only	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. Defaults to no.
Second Preferred Codec	The second preferred codec when the preferred codec cannot be used. If *Use Pref Codec Only i*s enabled (set to **yes**), this parameter is not used. Defaults to Unspecified.
Third Preferred Codec	The third preferred codec when the preferred codec and second preferred codec cannot be used. If *Use Pref Codec Only is* enabled (set to **yes**), this parameter is not used. Defaults to Unspecified.

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.
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.
G726-40 Enable	To enable the use of the G.726 codec at 40 kbps, select yes. Otherwise, select no. Defaults to yes.
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 G729a 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 G711a and G711u.

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]11|0|00|[2-9]xxxxxx|1xxx[2-9]xxxxxx|xxxxxxxxxxxx.). For more information, see Appendix B "Configuring a Dial Plan".

Parameter	Description

Dial Plan	Dial plan script for this line.
	
	The dial plan syntax is expanded in the SPA to allow the designation of three parameters to be used with a specific gateway:
	
	  • uid – the authentication user-id
	
	  • pwd – the authentication password
	
	  • nat – if this parameter is present, use NAT mapping
	
	Each parameter is separated by a semi-colon (;).
	
	Furthermore, it recognizes gw0, gw1, …, gw4 as the locally configured gateways, where gw0 represents the local PSTN gateway in the same unit.
	
	Example 1:
	
	`*1xxxxxxxxxx<:@fwdnat.pulver.com:5082;uid=jsmith;pwd=xyz`
	
	Example 2:
	
	`*1xxxxxxxxxx<:@fwd.pulver.com;nat;uid=jsmith;pwd=xyz`
	
	Example 3:
	
	`[39]11<:@gw0>`

Enable IP Dialing	Enable or disable IP dialing. Defaults to no.

User Tab

This section describes the fields for the following headings on the User tab:

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 -145 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.

Supplementary Services

The SPA9x2 provides native support of a large set of enhanced or supplementary services. All of these services are optional. Most supplementary service paramters are listed in Supplementary Services, page -160.

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 Supplementary Services Parameters, page -89.

Web Information Service Settings (SPA962)

✎
Note These parameters apply only to the SPA962.

For information about a user setting RSS newsfeeds, see "Setting RSS Newsfeeds on the SPA962 Phone" section on page 110.

Traffic Service Information Settings (SPA962)

✎
Note Traffic 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! Traffice Alerts provides this information to U.S. customers only.

Audio Volume

.

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.

Phone GUI Menu Color Settings (SPA962 only)

✎
Note The following parameters apply only to the SPA962.

Chose from:

- Light Blue

- Light Green

- Pink

- Silver

Defaults to Light Blue.

932 Tab (SPA962 only)

This tab includes the following sections:

- •Unit 2, page -179General, page -178

- General, page -179

- Unit 2, page -179General

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 SPA962 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 SPA962 can have up to two SPA932s attached).
Server Type	Selects the type of server used (SPA9000, Broadsoft, or Askterisk).
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 sidecar are returned to off, all the keys become amber. The SPA962 must be rebooted after the test is completed.
Call Pickup Code	The star code used for picking up a ringing call. Defaults to *98.
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 118..

For more information, see "Setting Up the SPA932 Sidecar" section on page 114.

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 "Unit/Key Configuration Scripts" section on page 118.

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 sidecar unit.

- Unit 2—Displays information about the second SPA932 sidecar 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 sidecar unit.
Type	Displays the function enabled for each key (1-32) on the SPA932 sidecar unit.
Line	Displays the extension assigned to each key (1-32) on the SPA932 sidecar unit.
Station	Displays the subscribe URI configured for each key (1-32) on the SPA932 sidecar unit.

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

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 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

Configuring a Dial Plan

The Linksys IP phone allows each line (extension) to be configured with a distinct dial plan. The dial plan specifies how to interpret digit sequences dialed by the user, and how to convert those sequences into an outbound dial string.

The SPA9x2 phone syntax for the dial plan closely resembles the corresponding syntax specified by MGCP and MEGACO. Some extensions are added that are useful in an end-point.

Dial Plan Digit Sequences

The plans contain a series of digit sequences, separated by the | character. The collection of sequences is enclosed in parentheses.

When a user dials a series of digits, each sequence in the dial plan is tested as a possible match. The matching sequences form a set of candidate digit sequences. As more digits are entered by the user, the set of candidates diminishes until only one or none are valid.

Any one of a set of terminating events triggers the SPA9x2 phone to either accept the user-dialed sequence and transmit it to initiate a call, or else to reject it as invalid. The terminating events are as follows:

- No candidate sequences remain—The number is rejected.
- Only one candidate sequence remains, and it has been matched completely—The number is accepted and transmitted after any transformations indicated by the dial plan, unless the sequence is barred by the dial plan, in which case the number is rejected.
- A timeout occurs—The digit sequence is accepted and transmitted as dialed if incomplete, or transformed as per the dial plan if complete.
- An explicit "send" (user presses the # key)—The digit sequence is accepted and transmitted as dialed if incomplete, or transformed as according to the dial plan if complete.

The time-out duration depends on the matching state. If no candidate sequences are as yet complete (as dialed), the *Interdigit_Long_Timeout* value applies. If a candidate sequence is complete, but there exists one or more incomplete candidates, the *Interdigit_Short_Timeout* value applies.

The following table shows the entries to use when programming the dial plan.

Dial Plan Entries

Dial Plan Entry	Function
*xx	Allows arbitrary 2-digit star code
[3469]11	Allows x11 sequences (for example, 311, 411, 611, 911)
0	Dials operator
00	Dials international operator

Dial Plan Entry	Function
[2-9]xxxxxx	Dials 7-digit US local number
1xxx[2-9]xxxxxx	Dials US 1 + 10-digit long distance number
xxxxxxxxxx.	Dials all other numbers, including international long distance

Dial Plan Rules

This section describes the rules that apply to configuring and interpreting dial plans.

NOTE: White space in a dial plan script is ignored, but it may be used for readability.

Digit Sequence Syntax

Each digit sequence within the dial plan consists of a series of elements, which are individually matched to the keys pressed by the user. Elements can be one of the following:

- Individual keys 0, 1, 2 . . . 9, *, #.
- The letter x matches any one numeric digit (0 .. 9)
- A subset of keys within brackets (allows ranges): for example, [389] means 3 or 8 or 9)
  - Numeric ranges (n-n) are allowed within the brackets: for example, [2-9] means any digit from 2 through 9)
  - Ranges can be combined with other keys: e.g. [235-8*] means 2 or 3 or 5 or 6 or 7 or 8 or *.

Element Repetition

Any element can be repeated zero or more times by appending a period (.) to the element. Thus, "01." matches "0", "01", "011", "0111", … and so on.

Sub-sequence Substitution

A sub-sequence of keys (possibly empty) can be automatically replaced with a different sub-sequence using an angle bracket notation: < dialed-subsequence : transmitted-subsequence >. So, for example, "<8:1650>xxxxxxx" would match "85551212" and transmit "16505551212".

Intersequence Tones

An "outside line" dial tone can be generated within a sequence by appending a comma (,) between digits. Thus, the sequence "9, 1xxxxxxxxxx" sounds an "outside line" dial tone after the user presses 9, until the 1 is pressed.

Number Barring

A sequence can be barred (rejected) by placing a ! character at the end of the sequence. Thus, "1900xxxxxxx!" automatically rejects all 900 area code numbers from being dialed.

Interdigit Timer Master Override

The long and short interdigit timers can be changed in the dial plan (affecting a specific line) by preceding the entire plan with the following syntax:

• Long interdigit timer: L : *delay-value*

• Short interdigit timer: S : *delay-value*

Thus, "L:8,(. . .)" sets the interdigit long timeout to 8 seconds for the line associated with this dial plan. And, "L:8,S:4,(. . .)" overrides both the long and the short time-out values.

Local Timer Overrides

The long and short timeout values can be changed for a particular sequence starting at a particular point in the sequence. The syntax for long timer override is:

   L delay-value<space>
Note the terminating space character. The specified delay-value is measured in seconds. Similarly, to change the short timer override, use: S delay-value<space>.

Pause

A sequence may require an explicit pause of some duration before continuing to dial digits, in order for the sequence to match. The syntax for this is similar to the timer override syntax:

   P delay-value <space>
The delay-value is measured in seconds.

This syntax allows for the implementation of Hot-Line and Warm-Line services. To achieve this, one sequence in the plan must start with a pause, with a 0 delay for a Hot Line, and a non-zero delay for a Warm Line.

Implicit Sequences

The SPA9x2 phone implicitly appends the vertical code sequences entered in the web UI Regional parameter settings to the end of the dial plan for both Line 1 and Line 2. Likewise, if *Enable_IP_Dialing* is enabled, IP dialing is also accepted on the associated line.

Dial Plan Examples

The following dial plan accepts only US-style 1 + area-code + local-number, with no restrictions on the area code and number:

```
( 1 xxx xxxxxxx )
```

The following also allows 7-digit US-style dialing, and automatically inserts a 1 + 212 (local area code) in the transmitted number.

```
( 1 xxx xxxxxxx | <:1212> xxxxxxx )
```

For an office environment, the following plan requires a user to dial 8 as a prefix for local calls and 9 as a prefix for long distance. In either case, an "outside line" tone is played after the initial 8 or 9, and neither prefix is transmitted when initiating the call.

```
( <9,:> 1 xxx xxxxxxx | <8,:1212> xxxxxxx )
```

The following allows only placing international calls (011 call), with an arbitrary number of digits past a required 5 digit minimum, and also allows calling an international call operator (00). In addition, it lengthens the default short interdigit timeout to 4 seconds.

```
S:4, ( 00 | 011 xxxxx x. )
```

The following allows only US-style 1 + area-code + local-number, but disallows area codes and local numbers starting with 0 or 1. It also allows 411, 911, and operator calls (0).

```
( 0 | [49]11 | 1 [2-9]xx [2-9]xxxxxx )
```

The following allows US-style long distance, but blocks 9xx area codes:

```
( 1 [2-8]xx [2-9]xxxxxx )
```

The following allows arbitrary long distance dialing, but explicitly blocks the 947 area code.

```
( 1 947 xxxxxxx ! | 1 xxx xxxxxxx )
```

The following implements a hot line phone, which automatically calls 1 212 5551234 after a delay of 0 seconds.

```
( S0 <:12125551234> )
```

The following provides a warm line to a local office operator (1000) after five seconds, unless a four-digit extension is dialed by the user.

```
( P5 <:1000> | xxxx )
```

Dial Plan Timers

The dial plan functionality is regulated by the following configurable parameters:

- *Interdigit_Long_Timer* (defaults to 10)
- *Interdigit_Short_Timer* (defaults to 3)
- *Dial_Plan* (1 and 2)

Interdigit Long Timer

The *Interdigit_Long_Timer* specifies the default maximum time (in seconds) allowed between dialed digits, when no candidate digit sequence is as yet complete. (See the discussion of the *Dial_Plan* parameters for an explanation of candidate digit sequences).

Interdigit Short Timer

The *Interdigit_Short_Timer* specifies the default maximum time (in seconds) allowed between dialed digits, when at least one candidate digit sequence is complete as dialed. (See the following discussion of *Dial_Plan* parameters for an explanation of candidate digit sequences).

Dial Plans

Dial Plan parameters contain the actual dial plan scripts for each line (extension) n, where n is a number from 1 to 4 on the SPA942 and 1 to 6 on the SPA962..
The Dial_Plan[n] for Each Line n defaults to:
(*xx | [3469]11 | 0 | 00 | <:1408>[2-9]xxxxxx | 1[2-9]xx[2-9]xxxxxx | 011x.)

The default dial plan script for each line is as follows:

(*xx|[3469]11|0|00|[2-9]xxxxxx|1xxx[2-9]xxxxxx|xxxxxxxxxxxx.).

If IP dialing is enabled, one can dial [user-id@]a.b.c.d[:port], where '@', '.', and ':' are dialed by entering *, user-id must be numeric (like a phone number) and a, b, c, d must be between 0 and 255, and port must be larger than 255.

If port is not given, 5060 is used. Port and User-Id are optional.

If the user-id portion matches a pattern in the dial plan, then it is interpreted as a regular phone number according to the dial plan. The INVITE message, however, is still sent to the outbound proxy if it is enabled.

Contacts

For additional information, search all related PDF documents (see). Support is also available by phone or online.

US/Canada Contacts

24-Hour Technical Support

   866 606 1866

RMA (Return Merchandise Authorization)

   http://www.linksys.com/warranty

Website

   http://www.linksys.com

FTP Site

   ftp://ftp.linksys.com

Support

   http://www.linksys.com/support

Sales Information

   800-546-5797 (800-LINKSYS)

EU Contacts

Website

   http://www.linksys.com/international

Product Registration

   http://www.linksys.com/registration

LINKSYS®
A Division of Cisco

8060952NC-EW

CISCO

Free Manuals Download Website

http://myh66.com

http://usermanuals.us

http://www.somanuals.com

http://www.4manuals.cc

http://www.manual-lib.com

http://www.404manual.com

http://www.luxmanual.com

http://aubethermostatmanual.com

Golf course search by state

http://golfingnear.com

Email search by domain

http://emailbydomain.com

Auto manuals search

http://auto.somanuals.com

TV manuals search

http://tv.somanuals.com