



ADMINISTRATION GUIDE

BUSINESS SERIES

Linksys Voice System SPA 900 Series IP Phones

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Preface

This guide describes administration and use of the Linksys Voice System (LVS) IP phones. This preface describes:

- [Document Audience, page ix](#)
- [How This Document is Organized, page x](#)
- [Related Documentation, page x](#)
- [Technical Support, page x](#)

Document Audience

This document is written for the following audience:

- Internet Telephony Service Providers (ITSPs, abbreviated to SPs) offering services using LVS products
- Value-Added Resellers (VARs) and resellers who need LVS configuration references
- System administrators or anyone who performs LVS installation and administration

**Note**

This guide does not provide the configuration information required by specific SPs. Please consult with your SP for specific parameters.

How This Document is Organized

To do this ...	Refer to
Get an overview of Linksys IP phones and their functions	Chapter 1, “Introducing Linksys IP Phones”
Use the different administration and configuration tools provided for managing Linksys 900 Series IP phone	Chapter 2, “Getting Started”
Configure and monitor a Linksys 900 Series IP phone	Chapter 3, “Managing Linksys IP Phones”
Find configuration and monitoring options available from the LCD display on SPA900 Series IP phones	Chapter 4, “SPA900 Series LCD Command Reference”
Find the function and usage for each field or parameter on the Linksys 900 Series IP phone administration web server pages	Chapter 5, “Linksys IP Phone Field Reference”

Related Documentation

The following documentation provides additional information about Linksys IP phones:

- *SPA IP Phone User Guide*
- *SPA IP Phone Quick Reference Guide*
- *SPA Provisioning Guide* (Linksys Partner Login required)
- *Linksys Voice System Installation Guide*
- *Linksys Voice over IP Product Guide: SIP CPE for Massive Scale Deployment*
- *SPA 2.0 Analog Telephone Adapter Administrator Guide*

Technical Support

If you are an end user of LVS products and need technical support, contact the VAR or SP that supplied the equipment.

Technical support contact information for authorized Linksys Voice System partners is as follows:

- LVS Phone Support (requires an authorized partner PIN)
888 333-0244 Hours: 4am-6pm PST, 7 days a week
- E-mail support
voipsupport@linksys.com

Introducing Linksys IP Phones

This chapter introduces the functionality of the Linksys 900 Series IP phones and includes the following:

- [Overview, page 1-1](#)
- [Linksys IP Phone Features, page 1-2](#)
- [Feature Descriptions, page 1-5](#)
- [Technology Background, page 1-8](#)

Overview

[Table 1-1](#) summarizes the ports and features provided by the Linksys 900 Series IP phones described in this document.

Table 1-1 *Linksys IP Phones*

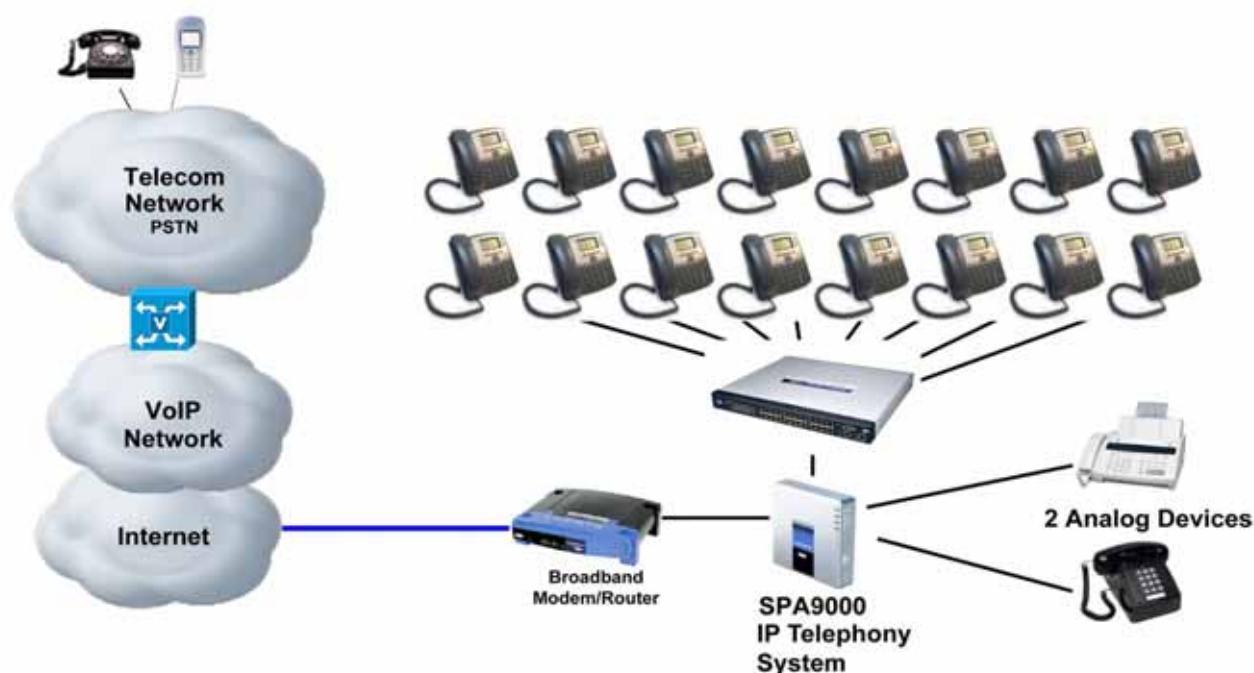
Product Name	RJ-45	Voice Lines	Additional Features/Notes
SPA901	One (1)	One (1)	No display.
SPA921	One (1)	One (1)	1 display.
SPA922	Two (2)	One (1)	Monochrome display, Power over Ethernet (PoE) support, and an extra Ethernet port for connecting another device to the LAN.
SPA932	—	—	Attendant sidecar for SPA962 with 32 LEDs/buttons for monitoring and call transfer. Support for Broadsoft Busy Lamp Field and Asterisk Line Monitoring.
SPA941	One (1)	Four (4)	Monochrome display.
SPA942	Two (2)	Four (4)	Monochrome display, Power over Ethernet (PoE) support, and an extra Ethernet port for connecting another device to the LAN
SPA962	Two (2)	Six (6)	High-resolution color display, Power over Ethernet (PoE) support, and an extra Ethernet port for connecting another device to the LAN



Note PoE units (SPA922, SPA942, and SPA962) do not come with an external power adapter. The PA100 power supply must be ordered separately if you are not using a PoE switch.

[Figure 1-1](#) illustrates how the IP phones are connected in a VoIP network.

Figure 1-1 Linksys IP Phones in a VoIP Network



Linksys IP Phone Features

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

:

- Shared Line Appearance *
 - SPA901: Two Call Appearances Accessed Via Flash Key or Hook-Flash
 - WIP310, SPA921, and SPA922: Two call appearances
 - SPA941 and SPA942: Four call appearances
 - SPA962: Six call appearances
- Line Status Indicators
- Call Hold
- Music on Hold *
- Call Waiting
- Outbound Caller ID Blocking
- Call Transfer - Attended and Blind
- Call Conferencing

- Call Pick Up - Selective and Group *
- Call Park and UnPark *
- Call Swap
- Call Back on Busy
- Call Blocking - Anonymous and Selective
- Call Forwarding - Unconditional, No Answer, On Busy
- Hot Line and Warm Line Automatic Calling
- Call Logs (60 entries each): Made, Answered, and Missed Calls
- Do Not Disturb (callers hear line busy tone)
- URI (IP) Dialing Support (Vanity Numbers)
- Date and Time with Intelligent Daylight Savings Support
- Call Duration and Start Time Stored in Call Logs
- Ten-User Downloadable Ring Tones - Ring Tone Generator Free from www.linksys.com
- Speed Dialing
- Automatic Redial
- Configurable Dial/Numbering Plan Support - per Line
- Intercom *
- Group Paging *
- DNS SRV and Multiple A Records for Proxy Lookup and Proxy Redundancy
- Syslog, Debug, Report Generation, and Event Logging
- Secure Call Encrypted Voice Communication Support
- Built-in Web Server for Administration and Configuration with Multiple Security Levels
- Automated Provisioning, Multiple Methods. Up to 256-Bit Encryption: (HTTP, HTTPS, TFTP)
- Optionally Require Admin Password to Reset Unit to Factory Defaults
- NAT Traversal
- Set Preferred CODEC, Per Call, All Calls
- Call Return - Redial Last Caller
- Configurable Dial/Numbering Plan Support
- Support Linksys Voice System Automatic Configuration

SPA901 Features

The SPA901 provides the following features that are not needed with the Linksys IP phones that provide an LCD display:

- Built-in Interactive Voice Response (IVR) system to check status and change configuration
- Ringer and Handset Volume Controls
- Handset Input Gain Adjustment

SPA92x, SPA94x, SPA962

The SPA921, SPA922, SPA941, SPA942, and SPA962 provide an LCD display and additional features that are not provided with the SPA901, including the following:

- Line Status Indicators: Active Line, Name, and Number
- Menu-Driven User Interface
- Digits Dialed with Number Auto-Completion
- Caller ID Name and Number and Outbound Caller ID Blocking
- On-Hook Dialing
- Redial from Call Logs
- Personal Directory with Auto-dial (100 entries)
- On Hook Default Audio Configuration (Speakerphone and Headset)
- Called Number with Directory Name Matching
- Call Number using Name - Directory Matching or via Caller ID
- Subsequent Incoming Calls with Calling Name and Number
- Name and Identity (Text) Displayed at Start Up
- Distinctive Ringing Based on Calling and Called Number

Ensuring Voice Quality

Voice quality perceived by the subscribers of the IP Telephony service should be indistinguishable from that of the PSTN. Voice quality can be measured with such methods as Perceptual Speech Quality Measurement (PSQM), with a scale of 1–5, in which lower is better; and Mean Opinion Score (MOS), with a scale of 1–5, in which higher is better.

[Table 1-2](#) displays speech quality metrics associated with various audio compression algorithms. For information about bandwidth requirements for each supported codec, refer to [Table 2-1 on page 2-10](#).

Table 1-2 *Speech Quality Metrics*

Algorithm	Complexity	MOS Score
G.711	Very low	4.5
G.726	Low	4.1 (32 kbps)
G.729a	Low-medium	4
G.729	Medium	4
G.723.1	High	3.8



Note

Linksys IP phones support all the above voice coding algorithms.

Feature Descriptions

Linksys IP phones are full featured, fully programmable IP phones that can be custom provisioned within a wide range of configuration parameters. This chapter contains a high-level overview of features to provide a basic understanding of the feature breadth and capabilities of Linksys IP phones.

- [SIP Proxy Redundancy, page 1-5](#)
- [Supported Codecs, page 1-5](#)
- [Other Features, page 1-7](#)

SIP Proxy Redundancy

In typical commercial IP Telephony deployments, all calls are established through a SIP proxy server. An average SIP proxy server may handle tens of thousands of subscribers. It is important that a backup server be available so that an active server can be temporarily switched out for maintenance. Linksys IP phones support the use of backup SIP proxy servers so that service disruption should be nearly eliminated.

A simple way to support proxy redundancy is to configure a SIP proxy server in the Linksys IP phone configuration profile.

- In Linksys SPA Configuration menu web GUI, enter your service provider name in the Proxy field. The system
- In the DNS SRV Auto Prefix field, enter Yes.
- In the User DNS SRV field, enter Yes.
- The phone tries to register and the server sends a list of IP addresses in order of priority. (automatically)

where the list is arranged in order of priority. The Linksys IP phone attempts to contact the highest priority proxy server whenever possible.

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 Linksys IP phone. One solution to this situation is through the use of DNS SRV records. Linksys IP phones can be instructed to contact a SIP proxy server in a domain named in SIP messages. The Linksys IP 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 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 Linksys IP phone tries to contact the list of hosts in the order of their stated priority.

If the Linksys IP 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.

Supported Codecs

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

The administrator can select the low-bit-rate codec used for each line. G.711a and G.711u are always enabled. [Table 1-3](#) describes the codecs supported by the Linksys IP phones.

Table 1-3 Codecs Supported by Linksys IP Phones

Codec (Voice Compression Algorithm)	Description
G.711 (A-law and μ -law)	This very low complexity codec 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.
G.726	This low complexity codec supports compressed 16, 24, 32, and 40 kbps digitized voice transmission at one through ten 10 ms voice frames per packet. This codec provides high voice quality.
G.729A	The ITU G.729 voice coding algorithm is used to compress digitized speech. Linksys supports G.729. G.729A is a reduced complexity version of G.729. It requires about half the processing power to code G.729. The G.729 and G.729A bit streams are compatible and interoperable, but not identical.
G.723.1	Linksys IP phones support the use of ITU G.723.1 audio codec at 6.4 kbps. Up to two channels of G.723.1 can be used simultaneously. For example, Line 1 and Line 2 can be using G.723.1 simultaneously, or Line 1 or Line 2 can initiate a three-way conference with both call legs using G.723.1.

When no static payload value is assigned per RFC 1890, Linksys IP phones can support dynamic payloads for G.726.

Other Features

Table 1-4 summarizes the features provided by Linksys IP Phones.

Table 1-4 Linksys Phone Features

Feature	Description
Music On Hold	On a connected call, Linksys IP phones may place the remote party on call. If the remote party indicates that they can still receive audio while the call is holding, the MOH server sends streaming audio.
Secure Calls	A user (if enabled by service provider or administrator) has the option to make an outbound call secure in the sense that the audio packets in both directions are encrypted.
Adjustable Audio Frames Per Packet	This feature 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.
DTMF	In-Band and Out-of-Band (RFC 2833) (SIP INFO) Linksys IP phones may relay DTMF digits as out-of-band events to preserve the fidelity of the digits. This can enhance the reliability of DTMF transmission required by many IVR applications such as dial-up banking and airline information.
Call Progress Tone Generation	Linksys IP phones have configurable call progress tones. Parameters for each type of tone may include number of frequency components, frequency and amplitude of each component, and cadence information.
Call Progress Tone Pass Through	This feature allows you to hear the call progress tones (such as ringing) that are generated from the far-end network.
Jitter Buffer—Dynamic (Adaptive)	<p>Linksys IP phones can buffer incoming voice packets to minimize out-of-order packet arrival. This process is known as jitter buffering. The jitter buffer size proactively adjusts or adapts in size, depending on changing network conditions.</p> <p>Linksys IP phones have a Network Jitter Level control setting for each line of service. The jitter level decides how aggressively Linksys IP phones try to shrink the jitter buffer over time to achieve a lower overall delay. If the jitter level is higher, it shrinks more gradually. If jitter level is lower, it shrinks more quickly.</p>

Table 1-4 Linksys Phone Features (continued)

Feature	Description
Voice Activity Detection with Silence Suppression and Comfort Noise Generation	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. VAD uses a very sophisticated algorithm to distinguish between speech and non-speech signals. Based on the current and past statistics, the VAD algorithm decides whether or not speech is present. If the VAD algorithm decides speech is not present, the silence suppression and comfort noise generation is activated. This is accomplished by removing and not transmitting the natural silence that occurs in normal two-way connection. The IP bandwidth is used only when someone is speaking. During the silent periods of a telephone call, additional bandwidth is available for other voice calls or data traffic because the silence packets are not being transmitted across the network. Comfort Noise Generation provides artificially-generated background white noise (sounds), designed to reassure callers that their calls are still connected during silent periods. If Comfort Noise Generation is not used, the caller may think the call has been disconnected because of the “dead silence” periods created by the VAD and Silence Suppression feature.
Configurable Dial Plan with Interdigit Timers	Linksys IP phones have three configurable interdigit timers: <ul style="list-style-type: none"> 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.
Report Generation and Event Logging	Linksys IP phones report a variety of status and error reports to assist service providers in diagnosing problems and evaluating the performance of their services. The information can be queried by an authorized agent, using HTTP with message-digest authentication, for instance. The information may be organized as an XML page or HTML page.
Syslog and Debug Server Records	Linksys IP phones support detailed logging of all activities for further debugging. The debug information may be sent to a configured Syslog server. Linksys IP phones provide configuration settings that determine the type of activity/events that should be logged as, for example, a debug level setting.
Dynamic Payload	When no static payload value is assigned per RFC 1890, Linksys IP phones can support dynamic payloads for G.726.
Call Statistics and Reporting	The statistics collected by Linksys IP phones during normal operation statistics are available in the Info tab. Line status is reported for each line (1 and 2). Each line maintains up to 2 calls: Call 1 and 2.

Technology Background

This section provides background information about the technology and protocols used by the phone:

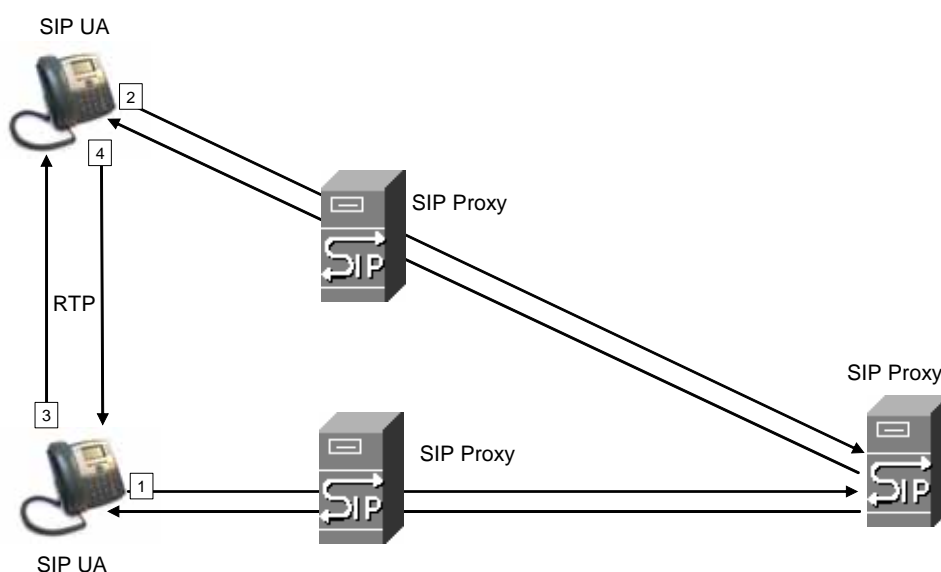
- [Session Initiation Protocol, page 1-9](#)

- [Using Linksys IP Phones with a Firewall or Router, page 1-9](#)
- [Using Linksys IP Phones with a Firewall or Router, page 1-9](#)

Session Initiation Protocol

Linksys 900 Series IP phones are implemented using open standards, such as Session Initiation Protocol (SIP), allowing interoperation with all ITSPs supporting SIP. [Figure 1-2](#) illustrates a SIP request for connection to another subscriber in the network. The requestor is called the user agent server (UAS), while the recipient is called the user agent client (UAC).

Figure 1-2 SIP Requests and Responses



In a SIP VoIP network, when the SIP proxy receives a request from a UAS for a connection and it does not know the location of the UAC, it forwards the message to another SIP proxy in the network. Once the UAC is located and the response is routed back to the UAS, a direct peer-to-peer session is established between the two UAs. The actual voice traffic is transmitted between UAs over dynamically assigned ports using the Real-time Protocol (RTP).

Using Linksys IP Phones with a Firewall or Router

When using a Linksys IP phone behind a firewall or router, make sure that the following ports are not blocked:

- SIP ports—By default, UDP port 5060 and 5061
- RTP ports—16384 to 16482

If security is not a concern in your environment, you can consider disabling SPI, if this function exists on your firewall.

Network Address Translation

This section describes issues that arise when using the LVS system on a network behind a network address translation (NAT) device. It includes the following topics:

- [NAT Overview, page 1-10](#)
- [NAT Types, page 1-11](#)
- [Simple Traversal of UDP Through NAT \(STUN\), page 1-11](#)
- [SIP-NAT Interoperation, page 1-12](#)

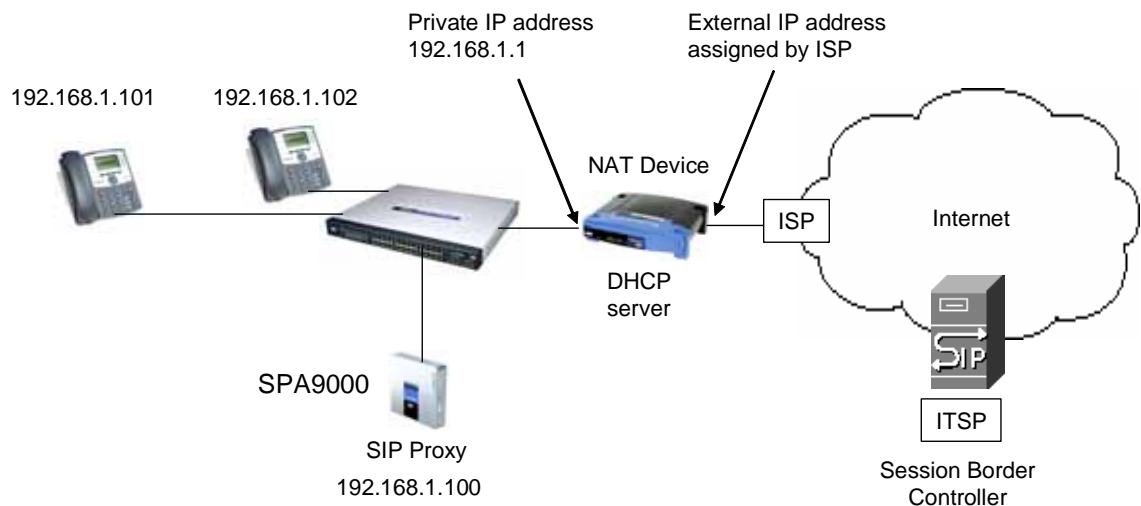
NAT Overview

Network Address Translation (NAT) allows multiple devices to share the same public, routable, IP address for establishing connections over the Internet. NAT is typically performed by a router that forwards packets between the Internet and the internal, private network.

The association between a private address and port and a public address and port is called a NAT *mapping*. This mapping is maintained for a short period of time, that varies from a few seconds to several minutes. The expiration time is extended whenever the mapping is used to send a packet from the source device.

The ITSP may support NAT mapping using a Session Border Controller (see [Figure 1-3](#)).

Figure 1-3 NAT Support with Session Border Controller Provided by ITSP



This is the preferred option because it eliminates the need for managing NAT on the Linksys IP phone. If this is not available, you need to discuss with the ITSP how to use the NAT Support Parameters provided by the Linksys IP phone, such as <Outbound Proxy> and <STUN Server Enable>.

A typical application of NAT allows all the devices in a subscriber network to access the Internet through a router with a single public IP address assigned by an ISP. 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.

NAT Types

The ways that NAT is implemented can be divided into the following categories:

- Full cone NAT—Also known as 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 can send a packet to the internal host only if the internal host had previously sent a packet to it.
- Port restricted cone NAT/symmetric NAT—Port restricted cone NAT or symmetric NAT is like a restricted cone NAT, but the restriction includes port numbers. Specifically, an external host can send a packet to a particular port on the internal host only if the internal host had previously sent a packet from that port to the external host.

With symmetric NAT, all requests from the same internal IP address and port to a specific destination IP address and port are mapped to a unique external source IP address and port. If the same internal host sends a packet with the same source address and port to a different destination, a different mapping is used. Only an external host that receives a packet can send a UDP packet back to the internal host.

Simple Traversal of UDP Through NAT (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 its public address, the type of NAT implemented, and the port associated on the Internet connection with a particular local port. This information is used to set up UDP communication between two hosts that are both behind NAT routers. Open source STUN software can be obtained at the following website:

<http://www.voip-info.org/wiki-Open+Source+VOIP+Software>

STUN does not work with a symmetric NAT router. To determine the type of NAT your router uses, complete the following steps:

Enable debugging on the Linksys IP phone:

1. Make sure you do not have firewall running on your PC that could block the syslog port (by default this is 514).
2. On the administration web server, System tab, 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 Linksys IP phone.
3. Set <Debug level> to **3**, but do not change the value of the <syslog server> parameter.
4. To capture SIP signaling messages, under the Line tab, set <SIP Debug Option> to **Full**. The output is named syslog.514.log.
5. To determine the type of NAT your router is using set <STUN Test Enable> to **yes**.
6. View the debug messages to determine if your network uses symmetric NAT.

SIP-NAT Interoperation

In the case of SIP, the addresses where messages/data should be sent to a Linksys IP phone system are embedded in the SIP messages sent by the device. If the Linksys IP phone system is located behind a NAT device, the private IP address assigned to it is not usable for communications with the SIP entities outside the private network.

**Note**

The ITSP might offer an outbound NAT-Aware proxy, which discovers the public IP address from the remote endpoint and eliminates the need to modify the SIP message from the UAC.

The Linksys IP phone system must substitute the private IP address information with the proper external IP address/port in the mapping chosen by the underlying NAT to communicate with a particular public peer address/port. For this, the Linksys IP phone system must perform the following tasks:

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

This can be done with the help of an external device, such as a STUN server. A STUN server responds to a special NAT-Mapping-Discovery request by sending back a message to the source IP address/port of the request, where the message contains the source IP address/port of the original request. The Linksys IP phone system can send this request when it first attempts to communicate with a SIP entity over the Internet. It then stores the mapping discovery results returned by the server.

- Communicate the NAT mapping information to the external SIP entities.

If the entity is a SIP Registrar, the information should be carried in the Contact header that overwrites the private address/port information. If the entity is another SIP UA when establishing a call, the information should be carried in the Contact header as well as in the 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.

- Extend the discovered NAT mappings by sending keep-alive packets.

Because the mapping is alive only for a short period, the Linksys IP phone system continues to send periodic keep-alive packets through the mapping to extend its validity as necessary.

2

Getting Started

This chapter describes the tools and utilities available for administering Linksys phones. It includes the following sections:

- [Linksys 900 Series IP Phones, page 2-1](#)
- [SPA9000 IP PBX System, page 2-9](#)
- [Establishing Connectivity, page 2-9](#)
- [Using the Administration Web Server, page 2-12](#)
- [Web Interface URLs, page 2-14](#)
- [Provisioning, page 2-15](#)
- [Using the Interactive Voice Response Interface, page 2-16](#)

**Note**

If the Linksys IP phone is supplied or sponsored by an Internet Telephone Service Provider (ITSP), certain network and service settings may be preconfigured. Depending on the configuration policy, access by an end user to specific configuration settings may be restricted or blocked.

Linksys 900 Series IP Phones

The Linksys provides fully-featured VoIP phones that integrate with the Linksys SPA9000 to provide connectivity to other local stations, and through an ITSP to IP phones over the Internet. In addition, the optional SPA400 integrates with the SPA9000 and provides connectivity between SPA900 IP phones and the PSTN. This section summarizes the ports and hardware features provided by each device. It includes the following topics:

- [Caring for Your Hardware, page 2-2](#)
- [SPA901, page 2-2](#)
- [SPA92x, SPA94x, and SPA962 Hardware Features, page 2-3](#)
- [SPA92x, SPA94x, and SPA962 Hardware Features, page 2-3](#)
- [SPA922, page 2-5](#)
- [SPA932, page 2-5](#)
- [SPA941, page 2-6](#)
- [SPA942, page 2-7](#)
- [SPA962, page 2-8](#)

Caring for Your Hardware

The Linksys 900 Series IP phones are electronic devices that should not be exposed to excessive heat, sun, cold or water. To clean the equipment, use a slightly moistened paper or cloth towel. Do not spray or pour cleaning solution directly onto the hardware unit.

SPA901

The SPA901 provides an entry-level IP phone that can be wall mounted (see [Figure 2-1](#)). The following are the hardware features provided by the SPA901:

- Voice Mail Message Waiting Indicator Light
- Redial Button
- Dedicated Flash Button
- Volume Control Button Cycles Through Volume Levels. Controls Ringer and Handset Volume.
- Standard 12-Button Dialing Pad
- High Quality Handset and Cradle
- Ethernet LAN – 10BaseT RJ-45
- 5-volt DC Universal (100-240 Volt) Switching Power Adaptor

Figure 2-1 SPA901



The following tables describe the status indicators and controls on the front of the device and the ports on the back panel of the device.

Front Panel and Side of Phone

Feature	Function
Message waiting light	On: you have a voicemail message.
STATUS	Off: make a call. On: make a call; line is shared. Flashing: shared line on hold. On red: line in use. Flashing red: local call on hold. On orange: phone not registered.
FLASH	Press to pickup a second incoming call, transfer a call, or setup a three-way conference call.
REDIAL	Press the redial the last number called.

Back Panel

Port	Function
Phone jack	Connect to the handset.
Ethernet port	Connect to the SPA9000 through a local switch.
Power	Connect to the 5-volt power supply.

SPA92x, SPA94x, and SPA962 Hardware Features

The SPA941, SPA942, and SPA962 provide LCD displays and other additional hardware features, including the following:

- Four Illuminated Call Appearance Line Buttons with Tricolor LED's (six LED's on the SPA962)
- LED Indicates Line State – Active, Idle, On-Hold, Unregistered
- Line LED Configurable to 13 Different States (See [“Line Key LED Pattern”](#) section on page 5-31).
- Dedicated Illuminated Buttons for:
 - Audio Mute On/Off
 - Headset On/Off
 - Speakerphone On/Off
- Four Soft Key Buttons
- Four Way Rocking Directional Knob for Menu Navigation
- Voice Mail Message Waiting Indicator Light
- Voice Mail Message Retrieval Button
- Dedicated Hold Button
- Settings Button for Access to Feature, Set-up, and Configuration Menus
- Volume Control Rocking Up/Down Knob Controls Handset, Headset, Speaker, Ringer
- Standard 12-Button Dialing Pad
- High Quality Handset and Cradle

- Built-In High Quality Microphone and Speaker
- Headset Jack – 2.5 millimeter
- Two Ethernet LAN ports (100BaseT RJ-45)
- 802.3af Compliant Power over Ethernet (PoE) (not available on SPA9x1 models)
- Optional 5 volt DC Universal (100-240 Volt) Switching Power Adaptor - Power Supply is Ordered Separately

SPA921

The SPA921 has one RJ-11 phone port and one 100BaseT RJ-45 port for connecting the phone to the LAN (see [Figure 2-2](#)).

Figure 2-2 SPA921



The following tables describe the status indicators and controls on the front of the device and the ports on the back panel of the device.

Front Panel

Feature	Function
LCD display	Lists device status and configuration options.
Telephone keypad	Enters numeric digits for initiating a call or for entering configuration information.
Navigation button	Scrolls between display and configuration options in the LCD display.
Soft keys 1-4	Selects options on the LCD display

Back Panel

Port	Function
Phone jack	Connects to the handset.
Ethernet ports	Connects to the SPA9000 through a local switch. The SPA922 has two ports. Use the other port to connect to a PC or other LAN device.
Power	Connect to the 5-volt power supply.

SPA922

The SPA922 is similar to the SPA921, but provides Power over Ethernet (PoE) and an extra Ethernet port for connecting another device to the LAN (see [Figure 2-3](#)). The SPA100 power supply must be ordered separately if you are not using a PoE switch.

Figure 2-3 SPA922



SPA932

The SPA932 is a 32-button attendant console for the SPA962, providing 32 three-color (red, green, and amber) programmable LEDs, with support for Broadsoft Busy Lamp Field and Asterisk Line Monitoring (see [Figure 2-4](#)). The SPA932 attaches 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.

Figure 2-4 SPA932 (Front View with SPA962)



Each SPA932 unit provides 32 programmable speed-dial or direct station select (DSS) buttons. Each lighted button indicates line status (idle, ringing, busy, or null) using a busy lamp field (BLF). Incoming calls can be immediately transferred to the proper location by pushing a button assigned to the extension on the SPA932.

The following table summarizes the meaning of each light color and pattern.

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

SPA941

The SPA941 provides four lines (see [Figure 2-5](#)).

Figure 2-5 SPA941



The following tables describe the status indicators and controls on the front of the device and the ports on the back panel of the device.

Front Panel

Feature	Function
LCD display	Lists device status and configuration options.
Telephone keypad	Enters numeric digits for initiating a call or for entering configuration information.
Navigation button	Scrolls between display and configuration options in the LCD display.
Soft keys 1-4	Selects options on the LCD display.
Line status indicators 1-4	Displays status of each extension.

Back Panel

Port	Function
Phone jack	Connects to the handset.
Ethernet ports	Connects to the SPA9000 through a local switch.
Power	Connects to the 5-volt power supply.

SPA942

The SPA942 is similar to the SPA941, but provides two Ethernet ports for connecting to the LAN and supports Power over Ethernet (see [Figure 2-6](#)). The PA100 power supply must be ordered separately if you are not using a PoE switch. See the table for SPA941.

Figure 2-6 SPA942



SPA962

The SPA962 has a 320 x 240 true color, four-inch, LCD, provides up to six telephone extensions, and supports PoE (see [Figure 2-7](#)). The PA100 power supply must be ordered separately if you are not using a PoE switch.

Figure 2-7 SPA962



The following tables describe the status indicators and controls on the front of the device and the ports on the back panel of the device.

Front Panel

Feature	Function
Full-color LCD display	Lists device status and configuration options.
Telephone keypad	Enters numeric digits for initiating a call or for entering configuration information.
Navigation button	Scrolls between display and configuration options in the LCD display.
Soft keys 1-4	Selects options on the LCD display.
Line status indicators 1-6	Displays status of each extension.

Back Panel

Port	Function
Phone jack	Connects to the handset.
Ethernet ports (2)	Connects to the SPA9000 through a local switch. Use the second port to connect to a PC or other LAN device.
Power	Connects to the 5-volt power supply.
AUX RJ-11 port	Connects to optional SPA932

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.

The system provides auto attendant features for multiple extensions. Features include:

- Auto Attendant
- Shared line appearances
- Configurable call routing
- Multiple DID numbers per VoIP line
- Call hunting (sequential, round robin, random)
- Call transfer and Call forward
- Call hold and Call waiting

You can perform SPA9000 administrative tasks using an Interactive Voice Response (IVR) system or a built-in Web Server.

Establishing Connectivity

This section describes how to connect the SPA900 Series IP phone hardware. It includes the following topics:

- [Bandwidth Requirements](#)¹, page 2-10

- [Installing the SPA900 Series IP Phone, page 2-10](#)

Bandwidth Requirements1

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 Linksys IP phone 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. [Table 2-1](#) illustrates the bandwidth budget using different codecs.

Table 2-1 Ethernet Bandwidth Budget for Off-Net VoIP Calling

Codec	Approximate bandwidth budget for each side of conversation	2 calls	4 calls	6 calls	8 calls
G.711	110 kbps	220 kbps	440 kbps	660 kbps	880 kbps
G.726-40	87 kbps	174 kbps	348 kbps	522 kbps	696 kbps
G.726-32	79 kbps	158 kbps	316 kbps	474 kbps	632 kbps
G.726-24	71 kbps	142 kbps	284 kbps	426 kbps	568 kbps
G.726-16	63 kbps	126 kbps	252 kbps	378 kbps	504 kbps
G.729	55 kbps	110 kbps	220 kbps	330 kbps	440 kbps

This table is based on the following assumptions:

- Bandwidth Calculated with No Silence Suppression
- 20 Millisecond of payload per RTP packet



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>

Installing the SPA900 Series IP Phone

This section describes how to install and verify operation of SPA900 Series IP phones.

Check to make sure that you have the following package contents:

- Linksys 900 Series phone
- Ethernet cable
- Power adapter (must be ordered separately for PoE units: SPA922, SPA942, and SPA962)

Assembling the Phone and Connecting to the Network

Perform the following steps to connect Linksys IP phone.

-
- Step 1** Find the end of the coiled phone cord that has the longer straightened end. Plug that end into the back of the phone base (phone symbol).
- Step 2** Plug in the short straightened end of the handset cord into the jack at the bottom of the handset.
- Step 3** If you are using an external power source, push the power cord into the power supply, and plug the power supply into the phone base unit. Use only the power supply that came with the phone. SPA922, SPA942 and SPA962 can be configured without external power, if connecting to an IEEE powered Ethernet switch. See Linksys.com for details.
- Step 4** Plug the Ethernet cable into the back of the base station.
- Step 5** Plug the other end of the Ethernet cable into your already prepared network connection. If the phone is configured using an optional PoE adapter, see the PoE device documentation for setup instructions. With the SPA922, SPA942, and SPA962, the second Ethernet port can be used to connect a PC. The AUX RJ-11 port on the SPA962 is reserved for SPA 932 use.



Note Do NOT plug a telephone line into the AUX port.

Attaching the Desk Stand

This and the following sections apply to a SPA900 Series IP phone with an LCD display. For information regarding the SPA901 IP phone, refer to the *SPA901 QuickStart Guide*.

-
- Step 1** Line up the tabs on the desk stand with the slots on the back of the phone.
- Step 2** Slide the bottom tabs into the slots
- Step 3** Lightly press down on the top of the desk stand. It should easily slide into the top slots. Do not force.
-

Mounting the Phone to the Wall



Note When mounting the phone to a wall, do not attach the desk stand.

-
- Step 1** Attach two appropriate screws to the wall. Leave 1/4" distance from the wall. This allows you to slide the mounting brackets on to the screws.
- Step 2** Push down slightly to lock the phone in place.
-

You can purchase an optional wall mount kit (MB100) from your retailer or Linksys.com.

Turning on the Phone

If you are using an external power source, instead of Power over Ethernet (PoE), plug the adapter to an electrical outlet. Use only the adapter supplied for the specific phone.

- The phone executes a boot-up sequence when the power source is connected.
- During this sequence, the display first shows LINKSYS® or a customized message and then “Initializing Network.”
- All of the lights on the phone flash.
- After the sequence, which should take about five seconds, the phone displays the standard menu for the phone.
- The date, name, and number of the phone will display on the top line of the LCD screen.

At the bottom of the LCD screen, **redial**, **dir**, **cfwd**, and **dnd** are displayed over their associated soft buttons. A small arrow appears next to **dnd** indicating that more menus can be accessed by pushing the right side of the navigation button.

If the phone does not display the appropriate screen, verify the installation and connectivity. If these are correct, try unplugging the phone and plugging it back in again. If you still do not see the display, refer to the [“Troubleshooting FAQ” section on page 3-42](#).

Using the Administration Web Server

This section describes how to use the administration web server to configure the Linksys IP phone. It includes the following topics:

- [Connecting to the Administration Web Server, page 2-12](#)
- [Administrator Account Privileges, page 2-13](#)

Connecting to the Administration Web Server

To access the Linksys IP phone administration web server, perform the following steps.

Step 1 Launch a web browser on a computer connected to the Linksys IP phone .

Step 2 Determine the address of the administration web server.

For a SPA900 Series phone with an LCD display, press Menu, 9, 2, or following these steps:

- a. Press the Menu button.
- b. Scroll down to **Network** and press **select**.
- c. View the Current IP (Option 2).

For a SPA901 phone follow these steps:

- a. Press ******** on the keypad to access the IVR menu.
- b. Press **110#** to determine the Internet (WAN) IP address.

Step 3 Direct the browser to the IP address of the Linksys IP phone.

Changing between the tabs on the Voice page does not discard the unsubmitted changes. You can wait until completing all your changes on the Voice pages before submitting them.



Note Changing between the Router and Voice pages discards any unsubmitted changes to either page.

Step 4 Click the **Voice** tab.

Step 5 Click **Admin** and **Advanced**.

The Administrator account name is **admin**, and the User account name is **user**. These account names cannot be changed.

The system prompts for the Administrator account password if it has been set. If prompted, type the password provided by the ITSP and press Enter.

Step 6 To view the status information for the phone, click **Status**.

Enter the appropriate login information. Two views of the administration web server are available. Click **basic** to view basic settings, or click **advanced** to view advanced settings.

Step 7 If you make changes on a screen, click **Submit All Changes** to save the changes,

Click **Undo All Changes** button to undo your changes. When changes are saved, the Linksys IP phone may reboot.

Administrator Account Privileges

The Linksys IP phone support two levels of administration privileges: Administrator and User. Both privileges can be password protected.



Note By default, there are no passwords assigned for either the Administrator account or the User account.

The Administrator account has the privilege to modify all the web profile parameters and can also modify the passwords of both Administrator and User account. The User account only has the privilege to access part of the web profile parameters. The parameters that the User account can access are specified using the Administrator account on the Provisioning page of the administration web server.

To directly access the Administrator account level privilege, use the following URL:

<http://ipaddress/admin/voice>

If the password has been set for the Administrator account, the browser prompts for authentication. The User account name and the Administrator account name cannot be changed.

When browsing pages with the Administrator account privilege, you can switch to User account privilege by clicking the **User Login** link.

If the User account password is set, the browser prompts for authentication when you click the **User Login** link). From the User account, you can switch to the Administrator account by clicking the **Admin Login** link. Authentication is required if the Administrator account password has been set.



Note Switching between User and Administrator accounts or between basic and advanced views discards any uncommitted changes that have already been made on the web pages.

Web Interface URLs

The Linksys IP phone web interface supports several functions through special URLs:

- Upgrade
- Reboot
- Resync

Administrator account privilege is needed for these functions.

Upgrade URL

The Upgrade URL lets you upgrade the Linksys IP phone to the firmware specified by the URL, which can identify either a TFTP or HTTP server.



Note

If the value of the <Upgrade Enable> parameter in the Provisioning page is **No**, you cannot upgrade the Linksys IP phone even if the web page indicates otherwise.

The syntax of the Upgrade URL is as follows:

- `http://spa-ip-addr/admin/upgrade?[protocol://][server-name[:port]][/firmware-pathname]`

Both HTTP and TFTP are supported for the upgrade operation.

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 specified, the default port of the protocol is used. (69 for TFTP or 80 for HTTP)

The *firmware-pathname* is typically the file name of the binary located in a directory on the TFTP or HTTP server. If no *firmware-pathname* is specified, /spa.bin is assumed, as in the following example:
`http://192.168.2.217/admin/upgrade?tftp://192.168.2.251/spa.bin`

Resync URL

The Resync URL lets you force the Linksys IP phone to do a resync to a profile specified in the URL, which can identify either a TFTP, HTTP, or HTTPS server.



Note

The SPA resyncs only when it is idle.

The syntax of the Resync URL is as follows:

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

If no parameter follows **/resync?**, the Profile Rule setting from the Provisioning page is used.

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).

The profile-path is the path to the new profile with which to resync, for example:

`http://192.168.2.217admin/resync?tftp://192.168.2.251/spaconf.cfg`

Reboot URL

The Reboot URL lets you reboot the Linksys IP phone.



Note

The Linksys IP phone reboots only when it is idle.

The Reboot URL is `http://spa-ip-addr/admin/reboot`.

Provisioning

This section describes the provisioning functionality of the Linksys IP phone. This section includes the following topics:

- [Provisioning Capabilities, page 2-15](#)
- [Configuration Profile, page 2-15](#)

For detailed information about provisioning the Linksys IP phone, refer to the *Linksys SPA Provisioning Guide*.

Provisioning Capabilities

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

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

Remote firmware upgrade is achieved via TFTP or HTTP (firmware upgrades using HTTPS are not supported). Remote upgrades are controlled by configuring the desired firmware image URL into the Linksys IP phone via a remote profile resync.

For further information about remote provisioning refer to the *LVS SPA Provisioning Guide*.

Configuration Profile

The Linksys IP phone configuration profile can be either an XML file or a binary file with a proprietary format.

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>
```

Binary format profiles contain Linksys IP phone parameter values and user access permissions for the parameters. By convention, the profile uses the extension .cfg (for example, spa2000.cfg). The Linksys Profile Compiler (SPC) tool compiles a plain-text file containing parameter-value pairs into a properly formatted and encrypted .cfg file. The SPC tool is available from Linksys for the Win32 environment and Linux-i386-elf environment. Requests for SPC tools compiled on other platforms are evaluated on a case-by-case basis. Please contact your Linksys sales representative for further information about obtaining the SPC tool.

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";
```

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

The names of parameters in XML profiles can generally be inferred from the Linksys IP phone 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.

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.

Using the Interactive Voice Response Interface

This section describes how to read or write basic network configuration settings using the keypad on the SPA901 IP Phone. To configure the other SPA phones, use the LCD display (see the “[Using the 900 Series LCD Display](#)” section on page 3-1) or the administration web server (see the “[Using the Administration Web Server](#)” section on page 2-12).

This section includes the following topics:

- [Using the IVR Menu on a Linksys SPA901 Phone, page 2-16](#)
- [IVR Options, page 2-17](#)
- [Entering a Password through the IVR, page 2-19](#)

By default, there is no password required for any of the IVR options. If the Administrator account password is set, password authentication is required for some options.

The interactive voice response (IVR) features that are available depend on your system configuration. A convenient quick-reference for the IVR is available at the following website:

<http://www.linksys.com>

Select the Products > Business > IPCommunications > Linksys Voice System > SPA9000 > IVR Quick Reference Card link.

Using the IVR Menu on a Linksys SPA901 Phone

To use the IVR menu:

Step 1 On a SPA901rt of the Linksys IP phone.

Step 2 Press **** (quickly press the star key four times).
Wait until you hear “Linksys configuration menu.”



Note You cannot access the IVR from a phone that is currently connected to a call.

Step 3 Refer to [Table 2-2](#) to identify the option required.

Step 4 Enter the required option followed by the # (pound) key.

To enter a period, use the star key (*).

When entering a value, such as an IP address, to exit without entering any changes, press the * (star) key twice within half a second. Otherwise, the * is treated as a decimal point.

After entering a value, such as an IP address, press the # (pound) key to indicate you have finished your selection. To save the new setting, press **1**. To review the new setting, press **2**. To re-enter the new setting, press **3**. To cancel your entry and return to the main menu, press * (star).

For example, to enter the IP address *191.168.1.105* by keypad, press these keys: **191*168*1*105**. Press the # (pound) key to indicate that you have finished entering the IP address. Then press **1** to save the IP address or press the * (star) key to cancel your entry and return to the main menu.

If the menu is inactive for more than one minute, the Linksys IP phone times out. You need to re-enter the menu by pressing ****.

Step 5 To exit the menu, hang up the telephone.

The settings you have saved take effect after you hang up the telephone. The Linksys IP phone may reboot at this time.

IVR Options

[Table 2-2](#) summarizes the options provided by the IVR.

Table 2-2 IVR Options

IVR Action	IVR Menu Choice	Parameter(s)	Notes
Enter IVR Menu	* * * *	None	Ignore SIT or other tones until you hear, “Linksys configuration menu. Enter option followed by the pound key or hang-up to exit.”
Exit IVR Menu	3948	None	
Check DHCP	100	None	IVR announces whether DHCP is enabled or disabled.
Enable/Disable DHCP	101	Enter 1 to enable Enter 0 to disable	Requires password
Check WAN IP Address	110	None	IVR announces the current IP address of the WAN port.

Table 2-2 IVR Options (continued)

Set Static IP Address	111	Enter IP address using numbers on the telephone key pad. Use the * (star) key when entering a decimal point.	DHCP must be “Disabled,” otherwise you hear, “Invalid Option,” if you try to set this value. Requires password
Check Network Mask	120	None	IVR announces the current network mask of SPA.
Set Network Mask	121	Enter value using numbers on the telephone key pad. Use the * (star) key when entering a decimal point.	DHCP must be “Disabled,” otherwise you hear, “Invalid Option,” if you try to set this value. Requires password
Check Static Gateway IP Address	130	None	IVR announces the current gateway IP address of SPA.
Set Static Gateway IP Address	131	Enter IP address using numbers on the telephone key pad. Use the * (star) key when entering a decimal point.	DHCP must be “Disabled,” otherwise you hear, “Invalid Option,” if you try to set this value. Requires password
Check MAC Address	140	None	IVR announces the MAC address of SPA in hex string format.
Check Firmware Version	150	None	IVR announces the version of the firmware running on the SPA.
Check Primary DNS Server Setting	160	None	IVR announces the current setting in the <Primary DNS> parameter.
Set Primary DNS Server	161	Enter IP address using numbers on the telephone key pad. Use the * (star) key when entering a decimal point.	Requires password
Check administration web server port	170	None	IVR announces the port that the web server is listening on. (Default is 80)
Check LAN IP Address	210	None	IVR announces the current IP address of the LAN port.
Check PBX multicast address	180	None	IVR announces the current value.
Set PBX multicast address	181	Enter IP address and port. Use * key for entering a dot. For example, 224.168.168.169:8089 is 224*168*168*169*8089.	Enter a * between the IP address and the Port fields. Requires password
Enable/Disable administration web server	7932	Enter 1 to enable Enter 0 to disable	Requires password

Table 2-2 IVR Options (continued)

Manual Reboot of Unit	732668	None	After you hear “Option Successful,” hang up. Unit reboots automatically.
User Factory Reset of Unit WARNING: ALL “User-Changeable” NON-DEFAULT SETTINGS WILL BE LOST! This might include network and service provider data.	877778	Enter 1 to confirm Enter *(star) to cancel operation	SPA prompts for confirmation. After confirming, you hear “Option Successful.” Hang up. Unit reboots and all “User Changeable” configuration parameters are reset to factory default values.
Factory Reset of Unit WARNING: ALL NON-DEFAULT SETTINGS WILL BE LOST! This includes network and service provider data.	73738	Enter 1 to confirm Enter * (star) to cancel operation	SPA prompts for confirmation. After confirming, you hear “Option Successful.” Hang up. Unit reboots and all configuration parameters are reset to factory default values.


Note

The items marked with “Requires Password” require a password only if the Administrator password is set.

Entering a Password through the IVR

To input the password using the phone keypad, the following translation convention applies:

- To input: A, B, C, a, b, c—press “**2**”
- To input: D, E, F, d, e, f—press “**3**”
- To input: G, H, I, g, h, i—press “**4**”
- To input: J, K, L, j, k, l—press “**5**”
- To input: M, N, O, m, n, o—press “**6**”
- To input: P, Q, R, S, p, q, r, s—press “**7**”
- To input: T, U, V, t, u, v—press “**8**”
- To input: W, X, Y, Z, w, x, y, z—press “**9**”
- To input all other characters in the Administrator account password, press “**0**”


Note

This translation convention applies only to the password input.

For example, to input password **test#@1234** by phone keypad, you need to press the following sequence of digits: **8378001234**.

1. After entering a value, press the # (pound) key to indicate end of input.
 - To save value, press **1**.
 - To review the value, press **2**.
 - To re-enter the value, press **3**.
 - To cancel the value entry and return to the main configuration menu, press * (star).

Notes:

- The final # key is not included in the password value.
 - Saved settings take effect when the telephone is hung up, and if necessary, the Linksys IP phone automatically reboots.
2. After one minute of inactivity, the unit times out. The user needs to re-enter the configuration menu from the beginning by pressing * * * *.

3

Managing Linksys IP Phones

This chapter describes how to configure Linksys IP phones and includes the following sections:

- [Using the 900 Series LCD Display, page 3-1](#)
- [Localization, page 3-5](#)
- [Changing the Display Background \(SPA942/962\), page 3-7](#)
- [Using the SPA932 \(Sidecar\) with the SPA962, page 3-8](#)
- [Configuring the Web Service, page 3-14](#)
- [Call Appearances and Extensions, page 3-21](#)
- [Line Key LEDs, page 3-22](#)
- [Using Call Features, page 3-24](#)
- [Configuring a Dial Plan, page 3-33](#)
- [System Administration, page 3-37](#)
- [Troubleshooting FAQ, page 3-42](#)

Using the 900 Series LCD Display

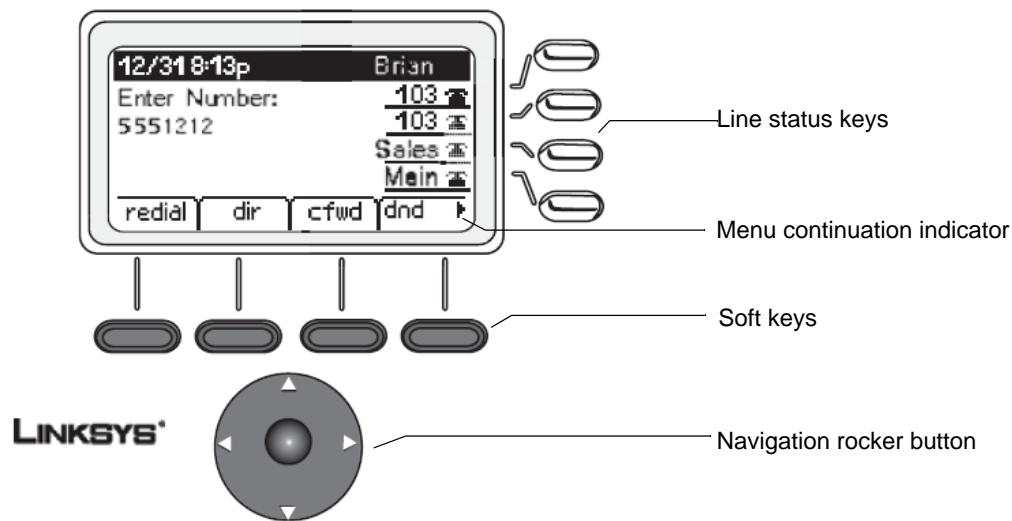
All Linksys IP phone models provide an LCD display except for the SPA901, which you configure using the Interactive Voice Response (IVR) interface, described in the [“Using the Administration Web Server” section on page 2-12](#). This section describes the basic functions of the LCD display and includes the following topics:

- [SPA900 Series LCD Display Controls, page 3-1](#)
- [Entering and Saving Settings, page 3-4](#)
- [Call Appearances and Extensions, page 3-21](#)
- [Using Soft Keys, page 3-3](#)

SPA900 Series LCD Display Controls

The LCD display on the SPA900 Series IP phones provides menu options for operating, configuring and monitoring the IP phone. For information about each of these options, refer to [Chapter 4, “SPA900 Series LCD Command Reference”](#). [Figure 3-1](#) illustrates the main controls on the LCD display.

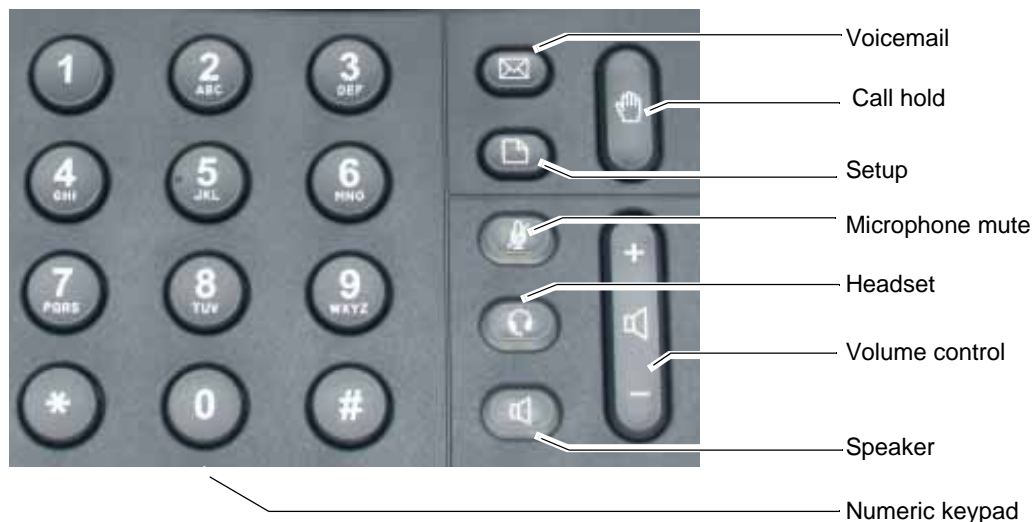
Figure 3-1 SPA900 Series LCD Controls



The function of the four soft keys depends on the option appearing on the LCD display. When more than four functions are available, a small left or right-facing triangle (menu continuation indicator) appears at the bottom of the LCD display. This indicates that more soft key functions can be accessed by pressing the **Navigation** button left or right.

Figure 3-2 summarizes the function of the other buttons on the telephone keypad.

Figure 3-2 Telephone Keypad



To access an option, press the **Setup** button and press the **Navigation** button to select the option. Then push the left soft key beneath the **Select** option to select the option. To use the shortcut to an option, press the **Setup** button and type the number assigned to the option. The option does not have to appear on the LCD to use the shortcut. To select an option with a two-digit shortcut, press the second number within two seconds of the first number.

Using Soft Keys

The LCD can display up to four options at one time. You can scroll through a longer list of options by pressing the **Navigation** button up or down. Pressing down or up when at the bottom or top of a list displays the first or last item in the list.

Typically, when you select monitoring options, only the **Cancel** soft key is available. [Table 3-1](#) summarizes the general soft key labels that appear on the LCD display.

**Note**

The following table shows the soft key labels for the SPA962 in parentheses (). This manual generally shows procedures using the non-SPA962 labels.

Table 3-1 General LCD Display Soft Key Labels

Label (SPA962 Label)	Description
cfwd (Forward)	Forwards calls, with one of four options: All, When the line is busy, When there is no answer, or When there is no answer after a time delay.
dir (Directory)	Accesses the phone directory.
dnd (DND)	(Do not disturb) Sends incoming calls to voicemail, or gives a busy signal if voicemail is not available.
grPick (GrPickup)	Picks up a call from an extension outside the group.
lcr (Call Rtn)	(Last call return) Dials the last call received.
pickup (Pickup)	Picks up a call from another phone in the group.
redial (Redial)	Displays a list of recently called numbers.
unpark (Unpark)	Picks up a call from where it was parked.

When selecting an option that allows modifying values, the LCD displays soft key labels indicating the functions available, such as **add**, **paste**, **edit**, and **delete**. To save changes, press the **save** soft key; to discard changes, press the **cancel** soft key.

[Table 3-2](#) summarizes the function of each of the soft key labels that appear after pressing the **Setup** button. This manual generally shows procedures using the non-SPA962 labels.

**Note**

The following table shows the soft key labels for the SPA962 in parentheses (). This manual generally shows procedures using the non-SPA962 labels.

Table 3-2 Setup Soft Keys

Label (SPA962 Label)	Description
<< for more options to the left or >> for more options to the right	Moves left or right through entry without deleting characters.
add (Add)	Adds an entry.
alpha, IP, num (Alpha, IP, Num)	Used with directory and speed dial to enter a URL, IP address or number. Press the soft key repeatedly to change selection.
cancel (Cancel)	Cancels change before pressing save or ok .

Table 3-2 Setup Soft Keys

Label (SPA962 Label)	Description
change (Change)	Changes the ring tone feature.
clear (Clear)	Clears the entire entry.
copy (Copy)	In a directory, use the Navigation button to highlight an entry, then press to copy an entry.
delChr (DeleteChr)	Deletes the last character entered.
delete (Delete)	Deletes an entry.
dial (Dial)	Places a call to the number highlighted on the LCD screen.
edit (Edit)	Edits part of an entry.
ok (OK)	Confirms an operation.
paste (Paste)	Use the Navigation button to highlight an entry in the personal directory and press paste to place a copied entry
play (Play)	Plays ring tones.
save (Save)	Saves changes.
select (Select)	Chooses an item to change or view.
y/n (Y/N)	Toggles between yes and no settings.

Entering and Saving Settings

The SPA900 Series phone GUI lets you enter alphanumeric and numeric inputs. You can enter the following data types into the phone:

- Numeric string, such as a phone number when dialing
- Alphanumeric string, such a name when adding a directory entry
- IP address, such as a DNS server IP address
- Boolean (y/n), such as enabling or disabling the DHCP option or selecting a preferred audio device or ring tone

When more than one data type applies to an entry, the phone shows a soft key for switching input modes among the types: you can scroll through the available types, such as num, alpha, or IP. The cursor also changes according to [Table 3-3](#), to indicate your current input mode:

Table 3-3 Input Modes

Input Mode	Soft Key	Cursor (appending)	Cursor (inserting)
Numeric	num	Blinking underscore _	Blinking vertical line
Alphanumeric	alpha	Blinking full-height block ¶	Blinking open rectangle .
IP Address	IP	Blinking half-height block .	Blinking half rectangle .

When entering digits in numeric mode, enter the corresponding digits from the numeric key pad. When entering letters or symbols during alphanumeric mode, the phone shows a template of two or more choices as you press each digit key. Scroll through the choices by pressing the same key multiple times. To accept a symbol as input, do one of the following:

- Stop pressing the digit key for 1.5 seconds
- Pressing another digit key
- Navigate to another input field (if applicable) by pressing the UP/DOWN key, or
- Save the value by pressing the proper soft key, such **save** or **ok**

When entering an IP address, pressing the * digit inserts a dot (.), or a colon (:) if there are 3 dots entered already.

When entering numeric, alphanumeric, or IP address values, the following additional soft keys are available for easier editing:

- <<—Move cursor to the left by one character
- >>—Move cursor to the right by one character
- **erase**<—Erase the character to the left of the cursor
- **clear**—Clear the entire entry

Holding down the <<, >>, or **erase**<, soft key or any numeric keypad keys automatically repeats the entry. If the input value is wider than the LCD can display, the SPA900 Series phone automatically scrolls the input to the left so that you can continue to enter input. On the other hand, the SPA900 Series phone stops accepting further input for a given field if the maximum allowed input length has been reached for that field. For Boolean input, the SPA900 Series phone displays (y/n); pressing the soft key toggles the current input between yes and no.

When the SPA900 Series phone displays an option, pressing the soft key either changes the input field to the next available choice in round-robin fashion, or displays a menu of choices. For most settings, the input value is saved once you press **save** or **ok** before exiting the edit screen for that setting.

However, there are a few exceptions where you must press **save** again when all the required changes have been made to the settings so that they are saved into the configuration at the same time. These exceptions are:

- Preferences
- Call Forward
- Network
- Speed Dialing

Localization

This section applies to the following SPA900 Series IP Phones:

- SPA922
- SPA942
- SPA962

This section describes how to localize the IP phone display language for these models, referred to in this section as SPA9x2 phones.

SPA9x2 phones support setting an alternative language on the user display. The administrator or service provider defines up to nine languages, in addition to English, to be made available and hosts the dictionaries for each of the languages on the HTTP or TFTP provisioning server. Language support follows Cisco dictionary principles.

The paragraphs describe parameters that can be defined in the SPA9x2 configuration file.

- Dictionary server script—This parameter is used to define 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>
```

The default is 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.xml
</Dictionary_Server_Script>
```

- Language Selection—This parameter is used to select the default language. Its 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>
```

The default is blank and the maximum number of characters is 512. The following is an example:

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

These parameters can also be modified from the administration web server, on the Regional settings tab, as shown in [Figure 3-3](#).

Figure 3-3 Dictionary Server Configuration Parameters

The screenshot shows the 'Regional' settings tab in the Linksys IP Phone administration web server. The 'Dictionary Server Script' field is set to 'serv=tftp://192.168.1.119/;d0=English;x0=english4.xml;d1=Spanish;x1=s' and the 'Language Selection' field is set to 'Spanish'. These two fields are highlighted with a red rectangular box. Other visible settings include 'Set Local Date (mm/dd):', 'Time Zone: GMT+01:00', 'Daylight Saving Time Rule: start=4/1/7;end=10/-1/7;save=1', 'DTMF Playback Level: -16', 'Inband DTMF Boost: 12dB', 'Set Local Time (HH/mm):', 'Time Offset (HH/mm):', and 'DTMF Playback Length: .1'.

During startup, the phone checks the selected/preferred language and downloads the dictionary from the TFTP/HFTP provisioning server indicated in the device 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.

The 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

The end user can change the language of the phone from the user menu, by following these steps:

1. Press the **Setup** button.
2. Select **Language** (option 9).
3. Select **Edit** to change language.
4. Select **Option** to change the language.
5. Select **OK** and **Save**.

Changing the Display Background (SPA942/962)

This section applies to the SPA 942 and SPA962, which provide the option to create a customized background. For the SPA962, you can use a 320 x 240 pixel bitmap (.bmp) image (1, 4, 8, 24, or 32-bit color) for the background image that appears on the LCD display. For the SPA942, the image should be 128 x 48 black and white (1-bit color).

You can include the definition of the required parameters in the configuration profile for the SPA 942 or SPA962, or use the administration web server.

To use the administration web server to configure a customized image perform the following steps:

- Step 1** Copy the bitmap (.bmp) image you want to use to a TFTP server that is accessible from the SPA962.
- Step 2** Connect a browser to the administration web server for the SPA 942 or SPA962.
- Step 3** Click the **Phone** tab and refer to the **General** tab at the top of the page (see [Figure 3-4](#)).

Figure 3-4 Background Picture Parameters

General			
Station Name:	cti101	Voice Mail Number:	vmm
Text Logo:			
BMP Picture Download URL:			
Select Logo:	Default	Select Background Picture:	Default

- Step 4** Choose one of the following options from the **Select Logo** pull-down selection list to control what displays when the phone is starting:
- Default—Uses the default Linksys logo.
 - BMP Picture—Uses the .bmp file from the URL in the BMP Picture Download URL field.
 - None—No logo used.
- Step 5** Choose one of the following options from the **Select Background Picture** pull-down selection list to control the background used:
- Default—Uses the default Linksys background.
 - BMP Picture—Uses the .bmp file from the URL in the BMP Picture Download URL field.
 - Text Logo—Uses the text background in the Text Logo field (SPA942 only).
 - None—No background used.

If you select **Text Logo**, enter up to 63 characters in the Text Logo field.

If you select **BMP Picture**, enter the URL to the server where the BMP picture is accessible. For example:

`tftp://mytftpserver.mydomain.com/pictures/background1.bmp`

The URL can use HTTP, HTTPS, or TFTP, with a maximum length of 255 characters.

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

Using the SPA932 (Sidecar) with the SPA962

The SPA932 is a 32-button attendant console for the SPA962. Each SPA932 unit provides 32 programmable speed-dial or direct station select (DSS) buttons. Each lighted button indicates line status (idle, ringing, busy, or null) using a busy lamp field (BLF). Incoming calls can be immediately transferred to the proper location by pushing a button assigned to the extension on the SPA932. Two SPA932 units can be attached to a single SPA962 to monitor a total of 64 separate lines.

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 possibly through a hostname if DNS is properly configured. The way in which each phone and extension is identified is vendor-specific.

To complete the configuration required to use the SPA932, perform the following steps:

- Step 1** Complete the physical installation of the SPA932 unit, which connects it to the SPA962.
- For instructions regarding the physical installation of the SPA932 and an introduction to its use, refer to the *32 Button Attendant Console for the SPA962 IP Phone Quick Installation Guide*.

- Step 2** Configure the SIP proxy server:
- Step 3** Configure the SPA932 using the administration web server connected to the SPA962 to which the SPA932 is physically attached.

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 administration web server, as shown in the following procedure.

- Step 1** Connect to the administration web server for the SPA9000.
- To use the SPA962 with a SPA932, it must be running Release 5.1.12 or later.
- Step 2** Click **Admin Login** on the administration web server page.
- Step 3** If necessary, enter **admin** and your password.
- Step 4** Click **Advanced**.
- Step 5** Click the **SIP** tab.

In the PBX Phone Parameters section, select **yes** from the CTI Enable pull-down selection list.

Figure 3-5 CTI Enable Parameter

PBX Phone Parameters

Next Auto User ID: 253 Phone Ext Password:

Phone Upgrade Rule:

Phone Dial Plan: (9,[3469]11S0|9,<:1408>[2-9]xxxxxx|9,<:1>[2-9]xxxxxxxxxS0|9,1[2-9]x

Phone Config XML:

Use LVS_PROXY: no CTI Enable: no

Undo All Changes Submit All Changes

- Step 6** Click **Submit All Changes** at the bottom of the page.

Configuring the Broadsoft Server for the SPA932

The Broadsoft server requires you to assign IP phone users to the Busy Lamp Field (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, complete the following steps:

-
- Step 1** On the Broadsoft server, in the user profile used by the SPA962 to register with the Broadsoft server, select **Assign Services**.
- Step 2** On the Assign Services page, from the Available Services column, move **Busy Lamp Field** and **Phone Status Monitoring** to the User Services column.
- Step 3** Define the List URI used by the SPA932 to register for BLF monitoring service with the Broadsoft server.

This value must be entered following the **sub=** keyword, when configuring each key on the 932 (see [“Monitoring the SPA932” section on page 3-14](#)).



Note If you configure more than one monitored list on the Broadsoft server, use the **vid=** keyword to specify the extension to use for each list.

- Step 4** On the Phone Status Monitoring page, move users that need to be monitored from the Available User column to the Monitored User column.
- Click the **Add** (or **Add All**) button to move each user to the Monitored Users column.
- The Directory Number (DN) associated with each user account when it is created on the Broadsoft Server is shown in parenthesis in the Monitored Users list. You use this DN to identify the specific phone assigned to each key on the SPA932.
- Step 5** 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, perform the following steps:

-
- Step 1** Add a context in the extensions.conf file.
- Step 2** Add a Subscribecontext command to point to the context in the sip.conf file.
- Step 3** Configure the SPA932 to register with the Asterisk server (see
-

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

Example 3-1 Asterisk Context for SPA932

```
[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, exten 3500 is used to add "Subscribecontext" to point to the context. This is entered in the file sip.conf.

Example 3-2 Asterisk Subscribecontext for SPA932

```
[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 SPA9000 IP PBX and obtain the following information, which is required to configure the SPA932:

- SPA9000 IP address (or hostname if supported)
- Phone extension numbers

By default, all LEDs on the SPA932 are assigned to the first configured extension on the SPA962 to which it is connected. You need to assign the LEDs to any of the other five extensions on the SPA962 that you want to monitor using the SPA932.

To configure the SPA932, perform the following steps:

-
- Step 1** Connect to the administration web server for the SPA962 to which the SPA932 is connected.
To use the SPA962 with a SPA932, it must be running Release 5.1.12 or later.
- Step 2** Click **Admin/Advanced** on the administration web server page.
- Step 3** Click the SIP tab.
- Step 4** Select **yes** from the CTI Enable pull-down selection list.
- Step 5** (optional) Change the SPA962 station name.
Click the **Phone** tab and edit the Station Name parameter in the General section.
- Step 6** Click the **SPA932** tab.

If you do not see the SPA932 tab, check your firmware version, and upgrade as necessary.
The system displays the page shown in [Figure 3-6](#).

Figure 3-6 SPA932 Configuration

Step 7 Select **yes** from the Unit 1 Enable pull-down selection list.

If you have installed two SPA932 units, also select **yes** from the Unit 2 Enable pull-down selection list.

Step 8 Make sure that **no** is selected on the Test Mode Enable selection list.

This option, which is useful for testing the SPA932, is disabled by default. You cannot complete the software configuration for the SPA932 if this option is enabled.

- SPA9000
- Broadsoft
- Asterisk

Step 9 For each extension or user that you want to monitor using the SPA932, enter the configuration script in the appropriate field for each key.



Note You can identify the user or extension number in the space provided next to each key before proceeding. This will help ensure that you enter the correct information in each field.

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)
- sub—Use this keyword to identify the phones to be monitored.
- —Use this keyword to identify the specific users or extensions to be monitored.
- nme—(optional) Use this field with the SPA9000 to identify any alias that has been assigned to the extension in the Linksys IP phone configuration.
- vid—(optional) 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.

**Note**

You only need to use the `vid=` keyword in the first entry assigned to each SPA962 extension. Subsequent keys will use the same extension.

- `n` with the extension on the SPA962 assigned to the LED on the SPA932

The `nme` parameter indicates the extension name, which in this case is the same as the station name.

Broadsoft syntax

The following is an example entry for a Broadsoft server:

fnc=enabled_functions;sub=blf_list_uri@broadsoft;usr=dn@broadsoft

Replace the variables (in italics), as follows:

- *enabled_functions* with **cp** for call pickup, **sd** for speed dial, and **blf** for busy lamp field; separate multiple functions in any order with a plus sign (+)
- *blf_list_uri* with the URI assigned to the BLF monitoring list on the Broadsoft server.
- *broadsoft* with the IP address or hostname of the Broadsoft server.
- *dn* with the number associated with the user account on the Monitored Users list.

For example, the following enables speed dialing and BLF monitoring for the user account reception, with a BLF monitoring list URI of marketing, on a Broadsoft server with the IP address 192.168.100.1:
fnc=sd+blf;sub=marketing@;usr=reception@

**Note**

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:

fnc=sd+blf+cp;sub=35890@192.168.1.11;nme=35890

This entry enables speed dialing and BLF monitoring on a Asterisk server with the IP address 192.168.1.11:

Step 10 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

Step 11 Repeat Step 9 for each extension or user that you are monitoring using the SPA932.

Step 12 Click **Submit All Changes** at the bottom of the page.

For instructions about how to configure phone extensions using the SPA Installation Wizard or the administration web server, refer to the *SPA9000 Administration Guide*.

Step 13 Repeat Step 9 for each extension or user that you are monitoring using the SPA932.

Step 14 Click **Submit All Changes** at the bottom of the page.

Monitoring the SPA932

To display the status of the SPA932, click the SPA932 Status link on the SPA932 tab. The system displays the screen shown in [Figure 3-7](#).

Figure 3-7 SPA932 Status

Unit 1		Unit 2	
Unit Enable:	Yes	Unit Online:	Yes
Subscribe Expires:	1800	Subscribe Retry Interval:	30
HW Version:	1.0.5	SW Version:	2.0.2

Key	Name	Type	Line	Station	Subscribe

Close Window

For information about each field on this display, refer to [Chapter 5, “Linksys IP Phone Field Reference”](#).

Configuring the Web Service



Note

Only web-enabled phones allow these features.

Web Interface Basic and Advanced Views

You can access configuration parameters using either the Basic or Advanced view. The SPA Provisioning tab is only visible from the Advanced Administrator view of the web interface.



Warning

Switching between the Basic and Advanced views discards the uncommitted changes that have been made on the web pages.

Configuration Parameters

You can configure SPA phone parameters in two ways:

- phone configuration web page
- remote provisioning

In addition, a small subset of "user options" can be set via the phone's GUI under various setup menus. On the configuration web page, the parameters are organized into several groups with each group listed under a tab on the top of the web page.

To see a group of parameters, click on the corresponding tab on the web page. The following groups are defined:

- Info – Status Information (read only). This is the default tab when the web page is first loaded
- System – System level parameters, including network and debug parameters
- SIP – Parameters to adjust the SIP stack behavior
- Provisioning – Parameters that control remote provisioning
- Regional – Parameters that depends on country or region, such as call progress tones and * codes
- Phone – Parameters that apply to all the extensions configured for the phone
- Ext 1 – Parameters that apply to Extension 1
- Ext 2 – Parameters that apply to Extension 2
- Ext 3 – Parameters that apply to Extension 3
- Ext 4 – Parameters that apply to Extension 4
- User – User level parameters (other parameters are administrator level)

Each parameter has a name and corresponding value. A parameter-name is of the form:

`seg[_seg][[n]]`

where `seg` is text segment composed of ascii characters only excluding spaces and underscores

`[]`

indicates optional components

and $n = 1, 2, 3, 4$.

Parameter names are case sensitive. Parameter values can be case sensitive or case insensitive, depending on the parameter. For example:

`Proxy[1], RTP_Packet_Size, DND_Act_Code`

When a parameter name is displayed on the web page, the underscore is replaced by a single space and the `[n]` component, if exists, is hidden. For example:

`Proxy, RTP Packet Size, DND Act Code`

Notes

- *var* (in italics) represents a variable or value to be substituted by the actual value it represents. *var* can appear standalone or anywhere within a longer string.
- `a|b|c` represents choices. The value can be one of the given choices.
- `[optional value]` represents an optional value
- `<Par Name>` represents a configuration parameter name, such as `<Proxy>`. In a profile, the corresponding tag is formed by replacing the space with an underscore "_", such as `<RTP Packet Size>` is replaced with `RTP_Packet_Size`. Some parameters carry an implicit index value if they apply to multiple extensions, such as `<Proxy>` which exists for Extensions 1, 2, 3, and 4. In a profile, "[n]" must be appended to the parameter to make it unambiguous. In this example, $n = 1, 2, 3, 4$ for `Proxy[1]`, `Proxy[2]`, `Proxy[3]`, and `Proxy[4]` respectively.
- An empty default value field implies an empty string ("")
- The SPA continues to use the last configured values for tags that are not present in a given profile.

- Templates are compared in the order given. The first, **not the closest**, match is selected. The parameter name must match exactly.
- If more than one definition for a parameter is given in a configuration file, the the phone uses last definition in the file.
- A parameter specification with an empty parameter value forces the parameter back to its default value. To specify an empty string instead, use the empty string "" as the parameter value.

Data Types

- **Unsn** – Unsigned n-bit value, where n = 8, 16, or 32. It can be specified in decimal or hex format such as 12 or 0x18 as long as the value can fit into n bits.
- **Sign** – Signed n-bit value. It can be specified in decimal or hex format. Negative values must be preceded by a "-" sign. A '+' sign before a positive value is optional.
- **Int** – A generic integer value. The range depends on the parameter.
- **Strn** – A generic string with up to n non-reserved characters.
- **Floatn** – A floating point value with up to n decimal places.
- **Timen** – Time duration in seconds, with up to n decimal places. Extra decimal places specified are ignored.
- **PwrLevel** – Power level expressed in dBm with 1 decimal place, such as -13.5 or 1.5 (dBm).
- **Bool**: Boolean value of either "yes" or "no".
- **{a,b,c,...}** – A choice among a, b, c, ...
- **IP** – IP Address in the form of x.x.x.x, where x between 0 and 255. For example, 10.1.2.100.
- **Port** – TCP/UDP Port number (0-65535). It can be specified in decimal or hex format.
- **UserID** – User ID as appeared in a URL, up to 63 characters.
- **FQDN** – Fully Qualified Domain Name, such as "sip.linksys.com:5060", or "109.12.14.12:12345". It can contain up to 63 characters.
- **Phone** – A phone number string, such as 14081234567, *69, *72, 345678, or a generic URL such as 1234@10.10.10.100:5068, or jsmith@linksys.com. It can contain up to 39 characters.
- **ActCode** – Activation code for a supplementary service, such as *69. It can contain up to 11 characters.
- **PhTmpl** – A phone number template. Each template can contain 1 or more patterns separated by a ",". White space at the beginning of each pattern is ignored. The question mark "?" and asterisk "*" represent wildcard characters. The template can contain up to 39 characters. Examples: "1408*, 1510*", "1408123????, 555?1".
- **RscTmpl** – A template of SIP Response Status Codes, such as "404, 5*", "61?", "407, 408, 487, 481". It can contain up to 39 characters.
- **CadScript** – A mini-script that specifies the cadence parameters of a signal. Up to 127 characters. Syntax:
 - 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.

- **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
- **ToneScript** – A mini-script that specifies the frequency, level and cadence parameters of a call progress tone. May contain up to 127 characters. Syntax:
- FreqScript;Z₁[;Z₂]. The section Z_i is similar to the S_i section in a CadScript except that each on/off segment is followed by a frequency components parameter: Z_i =
 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

- **RingScript** – A script that describes a ring tone.

Syntax: *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

RSS Newsfeeds (SPA962)

The SPA962 provides the option to view RSS newsfeeds. Press the Setup button-21 (Web Service) and select one of the following options on the LCD display:

1. Stock Quote
2. Traffic
 - a. Option 2 shows "Traffic Service Information Settings" at the bottom.
 - b. Enter your state, street, city and zip code to display local traffic information (defaults to CA, Cisco, San Jose, and 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 <http://finance.yahoo.com/rss/topstories>)
6. Sport News (defaults to <http://rss.news.yahoo.com/rss/sports>)
7. Politics News (defaults to <http://rss.news.yahoo.com/rss/politics>)

Option 3 to 7 are configurable using the administration web server, as shown in the following procedure.

-
- Step 1** Connect to the administration web server for the SPA962.
 - Step 2** Click **Admin/Advanced** on the administration web server page.
 - Step 3** Click the **Phone** tab.
 - Step 4** Change the name or URL used for each of the configurable options:
 - Option 3—RSS Feed URL 1:
 - Option 4—RSS Feed URL 2:
 - Option 5—RSS Feed URL 3:
 - Option 6—RSS Feed URL 4:
 - Option 7—RSS Feed URL 5:
 - Step 5** Change the update interval, if desired (defaults to 15)..
 - Step 6** Change the default scrolling timer, if desired (defaults to 0.8).

When the news information is too large to be displayed in one window, the display automatically scrolls. The Scrolling Timer defines the speed of the scrolling; the default is .8 seconds.
 - Step 7** Click **Submit All Changes** at the bottom of the page.
-

Call Appearances and Extensions

A client station is a Linksys IP phone with one or more extensions and call appearances provisioned. An extension in the Linksys IP phone is a VoIP account in an ITSP network or an IP-PBX system. An extension can be identified with a unique user ID (typically, a phone number).

Up to four extensions can be configured on a SPA922 or 942 Series phone, while up to six extensions can be configured on a SPA962 phone. These are referred to as Ext *n*, or *En*, where *n* is the numeric identifier of the extension, such as Ext 1 or E1.

When the same extension is configured on more than one station, it is called a shared extension. Extension 1 is referred to as the primary extension. Some features can be activated only on the primary extension, such as Call Forwarding and Voice Mail Waiting Indicator (VMWI).

A call appearance physically corresponds to a line key on a station. There are four line keys on the SPA922 and 942 Series phones, and six line keys on the SPA962 phone. These are referred to as Line Key *n*, or *Ln*, where *n* is the numeric identifier of the extension, such as Line 1 or L1. Functionally, a call appearance is an instance of an extension. If an extension is assigned to line keys on multiple stations, it is called a shared line appearance.

One extension can be assigned to multiple line keys on a Linksys IP phone. In fact, all four call appearances can be instances of the same extension. An extension is not a Shared Line Appearance unless it is also assigned to a line key on another station. Any of the four line keys can be disabled. Each call appearance supports one call at a time, either active or on hold.

The VoIP Interface (VI) refers to an extension and its associated control parameters that are configured on a particular station. Linksys IP phones include a rich set of configuration parameters to control the operation of the phone. Configuring an extension includes configuring the core account information and VI control parameters.

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

A Call Appearance State refers to the state of a call appearance, which can be one of the following:

- Disabled: The line key is disabled
- Idle (Ready): The Call Appearance is 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 (a.k.a. Progressing): Called party station is ringing
- Ringing (a.k.a. Alerting): Incoming call, station is ringing
- Connected: Connected with remote party
- Held: Remote party is on hold
- Invalid: Remote party hangs up or error while attempting outbound call
- Busy: the line is being used by another station (shared line only)

A Call State is the Call Appearance State applied to a specific call instance. Call States are divided into two general categories:

- Active (Call) State: If the state is Dialing, Calling, Proceeding, Connected, or Invalid. When the state of the Call is Active, it is referred to as an Active Call
- Standby (Call) State: the state is Ringing or Holding. When the state of the Call is Standby, it is referred to as a Standby Call

Line Key LEDs

This section summarizes the behavior of the line key LEDs on Linksys IP phones with multiple line key status indicators. For more information, see [“Line Key LED Pattern” section on page 5-31](#).

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)
- Green (g)
- Amber (a)
- Off (o).

The 900 Series phone has four 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 arbitrary blinking pattern by modifying the syntax of the LedScript.

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...`

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 slow blink

Example 2

`c=o`

LED is off

Example 3

`c=g`

Color is green and steady on

Example 4

`c=a;p=ud;u=.1/.1/.1/.1/.1/.9`

Color is amber and blink with the pattern (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

For more information, see [“Line Key LED Pattern” section on page 5-31](#).

Using Call Features

This section describes how the main call features are used or configured. It includes the following topics:

- [Selecting the Audio I/O Device and Line, page 3-24](#)
- [Making Calls, page 3-25](#)
- [Answering and Ending Calls, page 3-25](#)
- [Hold and Resume, page 3-26](#)
- [Call Waiting, page 3-26](#)
- [Speed Dialing, page 3-26](#)
- [Three-Way Conferencing, page 3-26](#)
- [Attended Call Transfer, page 3-27](#)
- [Blind Call Transfer, page 3-27](#)
- [Call Back, page 3-28](#)
- [Message Waiting Indication \(MWI\), page 3-28](#)
- [Accessing Voicemail, page 3-28](#)
- [Muting Calls, page 3-28](#)
- [Shared Call Appearances, page 3-29](#)
- [Personal Directory, page 3-29](#)
- [Caller and Called Name Matching, page 3-29](#)
- [Dialing Assistance, page 3-30](#)
- [Supplementary Services, page 3-30](#)
- [Call Logs, page 3-30](#)
- [Audio Volume Adjustment, page 3-31](#)
- [Managing Ring Tones, page 3-32](#)

Selecting the Audio I/O Device and Line

The Linksys IP phones provide three audio I/O devices:

- Handset
- Built-in microphone and speaker
- External microphone and headset

Only one audio device can be selected at one time. The speaker is selected by pressing the Speaker button, while the headset is selected by pressing the Headset button. When Speaker or Headset is selected, the corresponding LED is steady green.

The handset is selected whenever it goes from the on-hook to the off-hook position. Selecting or turning off an audio device is equivalent to an off-hook action, while turning on an audio device is equivalent to an on-hook.

When there are no active calls, all audio devices are deselected. When one of the calls becomes active, the handset is selected automatically as the audio device if it is off-hook; otherwise, the speaker or the headset is selected according to configuration. You can configure whether the speaker or the headset device has higher preference.]

A line is selected by pressing the corresponding line key. To make or receive calls, you must select a call appearance (line key) or an audio device. If you select a call appearance, then an audio device is selected automatically according to user preference setting. If you select an audio device, an idle call appearance is selected automatically in the order L1, L2, L3, and L4. You can switch between audio devices while the call in any of the active states.

A new call appearance is selected automatically when the user presses the **conf** (**Conf** on the SPA962) or **xfer** (**Transfer** on the SPA962) soft key. The 900 Series phone tries to pick an idle call appearance that is on the same extension as the last active call. If that call appearance is not available, the phone tries to pick a call appearance whose extension has the same proxy server as the last active call. Finally, if that fails, the first idle call appearance is selected, in numeric order (L1, L2, L3, L4).

Making Calls

There are two steps in making a call:

1. Select an audio device
2. Dials the number

The Linksys IP phone lets you perform these two steps in either order. In other words, the user can select an audio device either before or after dialing. The Linksys IP phone supports two types of dialing: explicit or implicit. Explicit dialing occurs when the user performs one of the following actions:

- Enters the target number digits one by one
- Enters the speed dial assignment of the target number
- Selects an entry from a directory, and presses the dial soft key
- Presses a dedicated key, such as the Voice Mail key

Implicit dialing refers to dialing a highlighted entry from one of the device directories or logs. A number can be implicitly dialing by selecting an audio device while the number is highlighted on the display.

If an audio device is selected prior to dialing, only explicit dialing is supported. Both explicit and implicit dialing are allowed if the audio device is selected after dialing. Explicit dialing before the audio device is selected is called on-hook dialing.

When an audio device is selected before dialing, a call appearance (for making the call) is also indirectly selected. The 900 Series phone picks the first available call appearance, in ascending order, starting with Line Key One. If no call appearances are available, no audio devices are selected. A call appearance can be explicitly selected by pressing the corresponding line key. An audio device is selected automatically based on the device programmed preferences.

Answering and Ending Calls

A line key LED rapidly blinks red when a call is incoming. You can answer the call by:

- Selecting an audio device.

The 900 Series phone answers the ringing call. If there are simultaneous incoming calls, the call ringing on the lowest numbered line key is answered.

- Pressing the corresponding line key. The default audio device is automatically selected.

When a call is in the active state (not on hold or parked), the call can be ended by turning off the currently selected audio device. A call in a standby state cannot be ended; it must be brought back to the active state first.

Hold and Resume

An active call can be placed on hold by explicitly pressing the Hold key, or by pressing another line key (to answer an incoming call, resume a held call, or start a new call). When a call is on hold, the corresponding line key LED slowly blinks red. To resume a call that is on hold, press the corresponding line key. When the phone is hosting a three-way conference, two call appearances are in the connected state. Pressing the Hold key places both calls on hold. The calls can be individually brought back to the connected state by pressing the corresponding line keys.

Call Waiting

The Call Waiting function is activated when a device has a call in the active state and another call is incoming. Linksys IP phones do not support multiple calls on the same line key. Incoming calls are assigned to an unused line key, causing the line key to quickly blink red. (Note that the Voice Mail Waiting Indicator also blinks red whenever there is an incoming call.) The phone does not ring. However, to alert you, the call waiting tone is played into the active audio device. The 900 Series phone allow multiple lines to simultaneously cause Call Waiting. All four call appearances can ring at the same time, after one of the extension keys is answered, the other three call appearances go into the Call Waiting state.

Speed Dialing

The 900 Series phone has eight programmable speed dial numbers. These use the administration web server User tab, or the LCD on phones that provide an LCD display.

To use the LCD to configure speed dialing, press the **Setup** button and then press the Speed Dial soft key. The speed dial numbers are assigned to the digits two through nine. If a slot is not configured, the LCD displays Not Assigned. Highlight the speed dial entry you want to add or modify, press **edit** to make the changes, and press **ok** to store the value.

A speed dial entry can be a phone number or URL. You can also enter a name that matches one of the directory entries. As the value is entered, the 900 Series phone attempts to match the entry to the directory entries and brings up a list of potential matches on the screen. Scroll down to select one of the suggested entries. To dial a speed dial entry, dial the corresponding single digit like a 1-digit phone number. (You can enter # after the speed dial number, but it is not necessary.)

Three-Way Conferencing

There are two ways to initiate a three-way conference on the 900 Series phone. First, during an active call, if the phone has one or more idle call appearances, press the **conf** soft key. This places the active call on hold and selects the next available call appearance for dialing (the dial tone is heard in the active audio device). Dial the third party, the party to be added to the conference call. After the third party line rings or is answered, presses the **conf** soft key again to join the two calls into a conference.

Second, if the phone has a call on hold while another call is either ringing or is connected, the **confLx** soft key appears. If the phone has only one call on hold, pressing the **confLx** soft key joins the call on hold with the active call. If the phone has more than one call on hold, pressing the **confLx** soft key places the active call on hold and prompts you to select the other line key to join with the conference call.

Once the conference starts, the 900 Series phone plays a special brief tone to all three parties to indicate that a conference call is in progress. The initiator of the conference call can terminate it at any time by hanging up. The initiator can leave the conference without ending the call by pressing the **join** soft key; this allows the other two parties to continue the call. This is implemented as a call transfer, which may not work if the other two parties are hosted by different service providers.

If the extension is configured with an external Conference Bridge URL, the conference is not limited to three participants; it can include as many as the conference bridge supports. There are two restrictions that apply when using an external conference bridge:

- The conference can only be initiated (by pressing the **conf** soft key) when the third party has answered, instead of when the phone is either ringing or answered
- Ringing call cannot be brought into a conference by pressing the **confLx** soft key; only calls that are on hold can be brought into the conference with the **confLx** soft key.

Attended Call Transfer

An attended call transfer allows a user to optionally transfer a call to a third party, after having a discussion with that third party. There are two ways to perform an attended call transfer:

- During an active call, press the **xfer** soft key to place the current call on hold and to activate an idle line key.
- Dial the transfer target number on the newly activated line. When the target is either ringing or answered, press the **xfer** soft key again to complete the transfer.

While one or more call appearances is on hold, initiate a call on an idle line key. While the call is either ringing or is connected, press the **xferLx** line key. If only one other call is on hold, the active call is connected with the call on hold. Otherwise, you are prompted to select to which line key to transfer the call.

At the completion of call transfer operation, the holding call peer is the transferee, and the connected or proceeding call peer is the transfer target. The case where a call transfer is completed when the transfer target is ringing is called a semi-attended call transfer. This is different from a blind transfer, which is described in the next subsection.

Blind Call Transfer

A blind call transfer allows you to transfer a call without speaking with the receiver of the transferred call. For the user, the call ends as soon as the transfer target phone number is dialed. The user phone sends the contact information for the transfer target to the phone of the other party on the call. The information is carried in a signaling message. When the message is sent, the user phone is dropped from the call and the other party phone automatically dials the transfer target.

To perform a blind call transfer, press the **bxfer** soft key and then dial the target number. The **bxfer** soft key automatically becomes active during a call, but it may not be visible on the display. If the **bxfer** key is not visible, press the left/right arrows on the **Navigation** button until it scrolls onto the display.

Some SPA phones do not have **bxfer** soft keys. To perform a blind transfer on these devices, during the call, press the **xfer** soft key, or the line key of an unused line. There must be a spare line available to perform the transfer. Once the new dial tone is heard, enter ***98**. The phone prompts the user to enter the phone number of the transfer target. Enter the number to complete the blind transfer. If the user phone has more than one call on-hold on, the last call placed on hold is transferred.

Call Back

This service mimics the call back service offered by the PSTN. You can activate this service on a busy number so that you are called back as soon as the busy number becomes available. The 900 Series phone implements this feature by repeatedly dialing the busy number periodically until the called party rings or answers, or until the service order is canceled by the user or expires. For that reason, it is referred to as the repeat dialing service.

To activate this service, you select a line key and enter *66 (call back activation code). The 900 Series phone then uses this call appearance to call the last called number. The retry period and expiration time are configurable. To cancel the service, you pick up any line and enter *86. When call back service is active, the corresponding Line key flashes green. You can still use this line to make or answer calls. The call back service is temporarily paused until the call appearance is idle again.

When the call back number rings or answers, the call appearance rings like a normal incoming call (the ring to be used is the default ring for the corresponding extension for that Line key). If the call back party answers the call before the local user does, the 900 Series phone sends a holding tone to the call back party. If the local user picks up the call back Line key first, you hear a ring back tone like a regular outbound call.

Message Waiting Indication (MWI)

The 900 Series phone indicates message waiting in two ways:

- Turning on the red MWI light - LED (with an envelope icon right above LED) on the phone.
- Playing stutter tone instead of regular dial tone.

The MWI light only applies to new message indication received on Extension 1. It is not turned on or off due to new message indication received on the other extensions [2 to 4]. On the other hand, the 900 Series phone can be configured to play a stutter tone on the individual call appearance according to the message waiting state of the corresponding Extension X.

Accessing Voicemail

The 900 Series phone has a voicemail retrieval key for quick access to voicemail. If a voicemail access number or URL is not specified, pressing the key brings up a screen where you can enter the appropriate information. This information can also be entered from the web page or downloaded via provisioning. The 900 Series phone attempts to call the voicemail number or URL if there is an idle call appearance available.

Muting Calls

You can mute the audio input into the phone when an audio device is switched on, by pressing the MUTE key. To un-mute, press the MUTE key again. If no audio device is on, pressing the Mute key has no effect. When switching from the speaker phone to the handset, the phone is automatically un-muted.

Shared Call Appearances

The 900 Series phone supports shared call 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 share call appearance to make a call only if the call appearance is not being used by another station. All 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.

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 900 Series phone. An entry can be added or edited from the phone GUI or on the web page or through remote provisioning. To do this from the web page or through remote provisioning, you specify the value as a DirEntry script. To view or edit the personal directory via the phone web interface, use the link http://ip_address/pdir.htm.

You can invoke the Directory menu on the phone in 2 ways:

- Press the **Setup** button and select **Directory**
- Press **dir** when not already in a Setup display

You can add a new entry to the Directory by selecting the New Entry option (the first item) on the Directory menu and press **add**. To modify or delete an existing item, highlight the corresponding entry and press **edit** or **del**. When trying to delete an entry, the phone prompts you for confirmation before proceeding.

You can copy an existing directory entry to create a new entry. To do this, highlight the entry you want to copy and press **copy**. Then highlight the New Entry option and press **paste**. You can edit the new entry and then press **save** to save the entry. You can also save a call log entry into the directory. To do this, select a call log entry and press **save**. Then edit the entry if needed, and press **save** to save the new entry in the directory.

Caller and Called Name Matching

When making an outgoing call, the 900 Series 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, 900 Series 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 and is shown on the call screen as the calling peer name. 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.

Dialing Assistance

Dialing assistance is an option you can enable or disable under the Preferences menu. If the option is enabled, the phone shows up to ten potential matches from the Redial List (most recent first) and from the Personal Directory as you dial the target number. This feature applies to both off-hook (with dial tone) and on-hook (without dial tone) dialing. When the suggestion list appears, you can scroll down the list using the Up/Down key to select the desired target number from the list. When looking for matches from the Redial List and the Directory, the 900 Series phone skips any leading *codes, so that the match is purely based on the target number. If configured on the phone, the *code that is skipped may be any of the following:

- Call forward activation and deactivation codes
- Secure call activation and deactivation codes
- Block CID activation and deactivation codes
- Block anonymous call activation and deactivation codes
- DND activation and deactivation codes
- Prefer codec and Force codec codes
- Referral services codes and Feature Dial Services codes

Supplementary Services

The 900 Series phone supports the following supplementary services:

- Block Caller ID —If enabled, the 900 Series phone attempts to hide the caller-ID on all outbound calls by default. You can also enable or disable this feature on a per call basis by pre dialing a *code before making the call.
- Block Anonymous Call —Rejects all inbound calls with no caller-ID with a 406 response
- Do Not Disturb (DND) —Rejects all inbound calls as if the phone is busy. That is, the phone either responds a 302 if call forward on busy is enabled, or 486 otherwise.
- Secure Call —If enabled, the 900 Series phone tries to negotiate on all outbound calls to use encrypted media (SRTP) by default. You can also enable or disable this feature on a per-call basis by pre dialing a *code before making the call. Note that to use Secure Call on an extension, <Mini Certificate> and <SRTP Private Key> must be configured for that extension. The enabling and disabling of these services applies to calls on all configured extensions on the 900 Series phone

Call Logs

There are three call logs maintained for each VoIP interface:

- Redial List—Each Redial List entry is added when dialing is completed, regardless if the number is correct or not, or if the call is successful or not.
- Answered Calls—An Answered Call is logged when the incoming call is answered.
- Missed Calls—A Missed Call is logged for each incoming call that causes the phone to ring but not answered.

The 900 Series phone keeps up to 60 entries per log in reverse chronological order. The logs are saved in the phone non-volatile memory. All call logs can be viewed on the 900 Series phone web page using the following link:

http://ip_address/calllog.htm

Each log is shown on the web page with the following syntax:

name,ext-id,date-time stamp

Where:

- ext-id is the extension or phone number of the call peer.
- name is the call peer name. If this field is missing, the peer name is not available.
- date-time stamp is the time at which the call log is entered. It has the format mm/dd hh/mm<a|p> (a = am and p = pm).

For example:

Joe Smith,14089991234,10/18 11:15a

No name appears if the name is not available.

Table 3-4 summarizes call log operations..

Table 3-4 Call Log Operations

Operation	Steps
To view a call log	<ol style="list-style-type: none"> 1. Press Menu. 2. Select Call History. 3. Select the log you want to view.
To delete a log	<ol style="list-style-type: none"> 1. Highlight the entry. 2. Press del.
To edit a log	<ol style="list-style-type: none"> 1. Highlight the entry. 2. Press edit. 3. Press save to save the changes, or cancel to abort the change.
To call a log entry	<ol style="list-style-type: none"> 1. Highlight the entry. 2. Press dial, or Off-Hook, or Turn on Speaker, or Turn on Headset, or press a line key.
To save a log entry into Personal Directory (if the directory is not already full)	<ol style="list-style-type: none"> 1. Highlight the entry. 2. Modify the entry as needed. 3. Press save to save the entry in the personal directory.

Audio Volume Adjustment

You can adjust four kinds of audio volume. Note that you must save the volume immediately after adjusting:

- Ringer volume (on the speaker)
- Speaker
- Handset
- Headset

To adjust the volume, press the VOL key to invoke the volume adjustment screen. Use the UP/DOWN keys to increase or decrease the volume, then press the Save softkey. The kind of volume to adjust is by context. When no audio device is selected at the moment, the Ringer volume is adjusted. Each

UP/DOWN key press plays a short burst of the current ring tone. If one of the audio devices is selected, the volume adjustment applies to that device. The screen shows a pictorial representation of the volume level.

Each volume has 16 steps in nonlinear logarithmic scale. Each setting is a 16-bit multiplier in the range 41 to 32764 in Q3.12 format (or 0.01 to 7.999 or 40 to 18 dB). Each step is equal to $58/(16-1)$ or 3.87 dB. Additional volume settings can be added and accessed via the phone setup screens. These volumes are:

- DTMF Echo Volume (use <DTMF Playback Level>)
- Call Progress Tone Volume (setting the ToneScript for each tone)

Managing Ring Tones

The 900 Series phone offers 10 ring tones. The characteristics of each ring tone is configurable using a Ring Tone script. In a Ring Tone script, you can assign a name for the ring tone, and specify two components:

- Waveform—1, 2, 3, or 4
- Cadence—1, 2, 3, or 8

You can assign a default ring tone for each extension. You can assign a ring tone for each personal directory entry (which overrides the default ring tone). Ring tone can be selected from the user interface. In addition, the 900 Series phone allows two user-downloadable ring tones. Therefore a total of 12 ring tone choices are available as the default ring for an extension or assigned to individual callers in the personal directory.

User-downloaded ring tone are labeled User 1 and User 2 in the choices for <Default Ring>. On the phone ring tone menu, the User 1 and 2 choices are replaced by the corresponding name of the ring tone or Not Installed if the user ring tone slots have not been used.

The user ring tone (1 or 2) can be downloaded using the phone web interface with the link:

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

Where <url> syntax is `[tftp://][host[:port]]/<path>`.

Only TFTP is supported. If the host is not specified, the TFTP host is the web client. The default port is 69. To remove the User 1 ring tone from the phone, set the <path> to delete, as follows:

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

The link is case sensitive.

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.

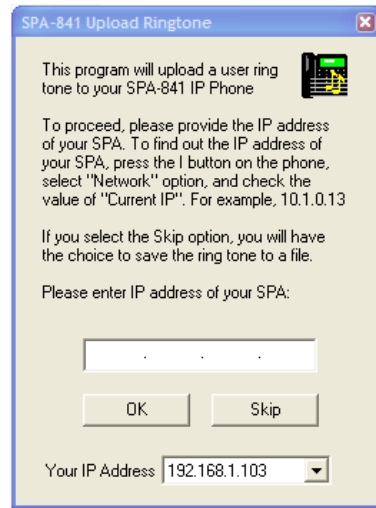
There is no need to reboot the phone after downloading a ring tone. However, if the phone is already in ring tone LCD menu while the new ring tone is downloaded, you need to exit and re-enter the ring tone menu to see the new ring tone name on the menu.

To convert a file for use as a ringtone, 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.sipura.com/support/index.htm>

Scroll down to **SPA Firmware with Upgrade Utility Program > SPA-841, Version 3.0** and click on **Ringtone_Generator**.

Figure 3-8 illustrates the user interface provided by this utility.

Figure 3-8 Upload Ringtone Utility

For input, you can use any .wav file less than eight seconds.

Configuring a Dial Plan

The Linksys IP phone allows each line 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 Linksys IP 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 Linksys IP 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` applies. If a candidate sequence is complete, but there exists one or more incomplete candidates, the `<Interdigit_Short_Timeout>` applies.

If you are using the SPA9000, enter `9 > *69` to callback and call return.

Table 3-5 describes the entries to use when programming the dial plan.

Table 3-5 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
[2-9]xxxxxx	Dials 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>xxxxxx`” 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, 1xxxxxxxxx” 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, “1900xxxxxxxx!” 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 Linksys IP phone implicitly appends the vertical code sequences entered in the administration web server 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 xxxxxxxx )
```

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

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

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 xxxxxxxx | <8,:1212> xxxxxxxx )
```

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 xxxxxxxx ! | 1 xxx xxxxxxxx )
```

The following implements a hot line phone, which automatically calls 1 212 5551234.

```
( 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> | xxxxx )
```

Dial Plan Timers

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

- Interdigit_Long_Timer
- Interdigit_Short_Timer
- Dial_Plan ([1] and [2])

Interdigit Long Timer

ParName	Interdigit_Long_Timer
Default	10

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 parameter for an explanation of candidate digit sequences).

Interdigit Short Timer

ParName	Interdigit_Short_Timer
Default	3

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

ParName	Dial_Plan[n] for Each Line <i>n</i>
Default	(*xx [3469]11 0 00 <:1408>[2-9]xxxxxx 1[2-9]xx[2-9]xxxxxx 011x.)

The Dial_Plan parameters contain the actual dial plan scripts for each line *n*, where *n* is a number from 1 to 4.

System Administration

This section describes features and operations required for administering and maintaining the basic function of the Linksys IP phone. It includes the following topics:

- [Reboot and Restart, page 3-37](#)
- [Factory Reset, page 3-38](#)
- [Password Protection, page 3-38](#)
- [Managing the Time/Date, page 3-38](#)
- [Daylight Saving Time, page 3-38](#)
- [Using Star Codes to Activate/Deactivate Services, page 3-39](#)
- [Disabling Services, page 3-41](#)
- [Error and Log Reporting, page 3-42](#)

Reboot and Restart

- Reboot cold boots the phone and restarts the phone's DHCP client.
- Restart warm boots the phone, without restarting the phone's DHCP client.

You must confirm the chosen option before continuing.

Factory Reset

Factory reset resets all parameters to default value. The personal directory and call logs are also cleared. You must confirm this option before continuing.

Password Protection

You can configure a password on the phone to protect against access to the certain options.

Managing the Time/Date

The Linksys IP phone obtains the current time information in one of the following 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 SPA may contain a Date header with the current time information. If the header is present, the SPA uses it to update its current time.
- **Manual Setup**—The Linksys IP phone also lets you manually enter the current time and date from the phone GUI or from the web page. However, this value is overridden by the NTP time or SIP Message Date whenever they are presented 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 using the <Time Offset (HH/mm)> parameter, which must be entered in 24-hour format. This parameter can also be configured from the LCD display. The <Time Zone> and <Time Offset (HH/mm)> offset values are not applied to the manual time/date setup.

Daylight Saving Time

The Linksys IP phone supports auto adjustment for daylight saving time. To enable this feature, the administrator must configure the <Daylight Saving Time Rule> parameter on the Regional tab. This parameter is a rule with three fields separated by semicolon (;) as shown below:

```
start=<start-time>;end=<end-time>;save=<save-time>
```

Where:

- <start-time> and <end-time> are of the form: <month>/<day>/<weekday>[/HH:[mm[:ss]]]
- <save-time> is of form: [/ [+|-]HH:[mm[:ss]]] The <save-time> value can be preceded by a negative (-) sign if subtraction is desired instead of addition.
- <month> = 1,2,12 (for Jan, Feb, Dec)
- <day> = [+|-]1,2,3,...,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).

- <weekday> = 1, 2, 3,..., 7 (for Mon, Tue, ..., Sun), or 0. If <weekday> is 0, it means the date to start or end daylight saving is at exactly the given date. In that case, the <day> value must not be negative. If <weekday> is not zero, then the daylight saving starts or ends on the <weekday> on or after the given date if <day> is positive, or on or before the given date if <day> is negative.
- HH = hour (0-23)
- mm = minute (0-59)
- ss = second (0-59)

Optional values inside [] are assumed to be 0 if they are not specified. Midnight means 0:0:0 of the given date.

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 applies to countries that recognize daylight savings time.

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

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

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

Using Star Codes to Activate/Deactivate Services

The Linksys IP phone accepts 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 (if you know the star code number)

or

Off hook the phone, press the right arrow to see/ press the button for star codes, scroll through the star code list and select the desired star code (if you do not know the star code number)

Below is a list of services that are accessible with a star code, with the default star code shown in parentheses.

- 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:
 - a. Put the call on hold
 - b. Get a dial tone on another line
 - c. Call the person to whom you are transferring the call by pressing *98, followed by their extension
 - d. Hang up the receiver
- Prefer G.711u (*017110)—Sets preferred codec to G.711u for next outbound call.
- Prefer G.711a (*017111)—Sets preferred codec to G.711a for next outbound call.
- Prefer G.723 (*01723)—Sets preferred codec to G.723 for next outbound call.
- Prefer G.726r16 (*0172616)—Sets preferred codec to G.726r16 for next outbound call.
- Prefer G.726r24 (*0172624)—Sets preferred codec to G.726r24 for next outbound call.
- Prefer G.726r32 (*0172632)—Sets preferred codec to G.726r32 for next outbound call.
- Prefer G.726r40 (*0172640)—Sets preferred codec to G.726r40 for next outbound call.
- Prefer G.729a (*01729)—Sets preferred codec to G.729a for next outbound call.
- Force G.711u (*027110)—Forces G.711u for next outbound call.
- Force G.711a (*027111)—Forces G.711a for next outbound call.
- Force G.723 (*02723)—Forces G.723 for next outbound call.
- Force G.726r16 (*0272616)—Forces G.726r16 for next outbound call.

- Force G.726r24 (*0272624)—Forces G.726r24 for next outbound call.
- Force G.726r32 (*0272632)—Forces G.726r32 for next outbound call.
- Force G.726r40 (*0272640)—Forces G.726r40 for next outbound call.
- Force G.729a (*02729)—Forces G.729a for next outbound call.

When off-hook dialing, the star code is processed by the phone as soon as it is entered and processed by the phone. 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. The ***code** soft key indicates the *codes available. Star codes can also be included in a directory entry or redial list.

In addition there are two special codes:

- Referral Services Codes: A list of star codes separated by a vertical bar (|), such as *199|*09, where entering each member triggers a blind transfer operation. The transfer target number in this case is prepended by the corresponding star code, whereas the blind transfer itself contains the target number only without the star code. Like a normal blind transfer, a referral service code can only be entered when the last call is placed on hold.
- Feature Dial Services Codes: A list of star codes separated by a vertical bar (|), where entering each member causes the next outbound call target number to be prepended by the corresponding star code. May be useful for some Service Providers who have their own star codes. These Service Provider star codes can be listed under the Feature Dial Services Codes.

Disabling Services

Services handled locally by the phone can be disabled by the administrator in one or two ways:

- Directly disabling the service in the configurable profile,
- Emptying out the *code associated with the service.

Services that can be disabled (or enabled) by the first method include the following:

- Block Caller ID (including blocking and unblocking, per call and always)
- Block Anonymous Calls
- Do Not Disturb
- Secure Call (including enabling and disabling, per call and always)
- Call Forward All
- Call Forward Busy
- Call Forward No Answer
- Call Back
- Conference Call (no star code)
- Attended Call Transfer (no star code)
- Blind Call Transfer

Services that can be disabled (or enabled) by the second method include any service that has a star code assignment in the Linksys IP phone configuration profile. Note that if a service is enabled, but the corresponding *code is emptied, the service can still be enabled/disabled by the end- user from the phone LCD or the administration web server.

If a service is disabled, any soft key associated with that service is hidden on the LCD. These are: **dnd**, **-dnd**, **cfwd** and **-cfwd**. Any menu item associated with a disabled service is preceded with an exclamation mark (!) and no soft keys is visible to select that entry.

Error and Log Reporting

The SPA Error Status Code (ESC) is used to indicate the current operation status of the SPA unit. An error state can be a relatively long transient state or a steady state.

The Error Status Code is a 4 digit number. The first digit indicates the error class: 1xxx represents normal operation states while 2xxx—9xxx represent error states that must be fixed for the unit to function properly.

The message waiting light LED has the following Blink Rate Definitions:

- ON—LED remains solid on (indicates voicemail, no errors)
- OFF—LED remains solid off
- LONG (Long On)—3.0 second on, 1 second off continuously
- FAST—0.125 second on, 0.125 second off continuously
- SLOW—0.5 second on, 0.5 second off continuously
- VSLO (Very Slow)—1.0 second on, 1.0 second off continuously
- HB (Heart Beat)—0.1 second on, 0.1 second off, 0.1 second on, 1 second off continuously

Troubleshooting FAQ

This section describes some common problems and solutions. If the suggestions provided here do not solve the problem, contact the appropriate technical support department.

If you are an end user of LVS products and need technical support, contact the reseller or Internet telephony service provider (ITSP) that supplied the equipment.

Technical support contact information for authorized Linksys Voice System partners is as follows:

- LVS Phone Support (requires an authorized partner PIN)
888 333-0244 Hours: 4am-6pm PST, 7 days a week
- E-mail support
voipsupport@linksys.com

When you contact the VoIP network administrator or service provider, have the model, serial number, and telephone number ready for technical support. The serial number is printed on the bottom of the phone.

Q. Why do the line status lights on my Linksys IP phone occasionally flash red?

A. A flashing red line status light on the Linksys IP phone indicates a local LAN issue. If statistics on the switch show a high number of collisions on the ports connected to the Linksys IP phone, you can reduce the speed on the ports to 10mb half duplex to eliminate the collision problems.

Q. Why can I not leave a voicemail message longer than 30 seconds?

A. If you are using 1.0.0.3 firmware, upgrade SPA400 to version 1.0.0.5 or greater. The upgrade will let you leave a message of 60 seconds.

Q. Why does the phone does not turn on or boot?

A. Check all connections into the phone (5 Volt Power supply, RJ 45 Ethernet connection, and handset cord).

- If using PoE, verify that the RJ-45 connector is firmly connected to the switch/wall connection.
- If not using PoE, check the wall socket and confirm that AC power is available. Also, try plugging another electrical device into the socket to confirm power.

Q. Why does the phone not make or receive calls?

A. If the phone does not make or receive calls, first verify that the phone's line LEDs are green. Any other color indicates either a registration or network problem. If the LEDs are not green:

1. Press the **Setup** button, then 11, then 2.

This provides the registration information for Extension 1 of the phone. The screen should display E1 Registered and other information in the format provided below:

```
Text Box: E1 Registered
Last Reg: 08:33 1/3
Next Reg in: 13s
Status cancel cancel
```

There may be a problem if the screen displays Not Registered.

2. Write down whether the phone is Registered or Not Registered.
3. To exit from this option, press the **Setup** button.
4. Get the current IP address of the phone by pressing **Menu, 9, 2**.

The phone displays the current IP address.

5. If there is no address or the address does not appear correct for your network, restart the phone, the SPA9000, and the router or DHCP server.
6. Get the current IP address of the phone by pressing **Menu, 9, 2**.
7. Contact the network administrator or service provider. Be prepared to provide the call log information, the voice quality issue type, the current IP address and registration status (Registered, Not Registered).

Q. Why are voice calls of poor sound quality?

A. The source of this issue is often one of the following two issues:

- The phone or call controller may have been configured improperly.
- The transport network has performance issues.

To troubleshoot this problem, gather the following information:

Identify when and where the issue occurred. Look at the Redial List (calls that the user made) by pressing Menu, 3, 1, and write down the calls that were affected. Then look at the Answered Calls by pressing Menu, 3, 2, and write down the calls that were affected.

Identify the type of voice quality issue (for example, part of the conversation is dropped, excessive echo, unclear or garbled speech, one way conversation, or low volume)

4

SPA900 Series LCD Command Reference

This chapter provides a systematic reference for the configuration and monitoring options available from the LCD display on SPA900 Series IP phones. All models provide an LCD display except for the SPA901. The SPA901 uses the Interactive Voice Response (IVR) interface, described in the [“Using the Interactive Voice Response Interface”](#) section on page 2-16.

This chapter includes the following sections:

- [1 Directory, page 4-2](#)
- [2 Speed Dial, page 4-3](#)
- [3 Call History, page 4-3](#)
- [4 Ring Tone, page 4-4](#)
- [5 Preferences, page 4-4](#)
- [6 Call Forward, page 4-7](#)
- [7 Time/Date, page 4-8](#)
- [8 Voice Mail, page 4-9](#)
- [9 Network, page 4-9](#)
- [10 Product Info, page 4-13](#)
- [11 Status, page 4-14](#)
- [12 Reboot, page 4-16](#)
- [13 Restart, page 4-16](#)
- [14 Factory Reset, page 4-16](#)
- [15 Custom Reset, page 4-16](#)
- [16 Set Password, page 4-17](#)
- [17 Set LCD Contrast, page 4-17](#)
- [18 CallPark Status, page 4-17](#)
- [19 Language \(SPA922, 942, and 962\), page 4-17](#)
- [20 GUI Properties \(SPA962\), page 4-18](#)
- [21 Web Service \(SPA962\), page 4-18](#)

1 Directory

The directory stores the list of the phone numbers and names. This section describes how to enter and save phone numbers and names into the directory. It includes the following sections:

- [Entering Names and Numbers into the Directory, page 4-2](#)
- [Entering Directory Names, Numbers and Ring Default, page 4-2](#)



Note

The following sections show the soft key names for the SPA962 in parentheses ().

Entering Names and Numbers into the Directory

Shortcut: Setup, 1, add, *name*, scroll down, *number*, scroll down or save

To enter the **Directory** option, press the **Setup** button and then press the **1** button. Alternatively, you may press the Setup button and the Directory option is highlighted. Then press the **select** soft key. After pressing select, the menu displays **New Entry** and existing entries in the directory below. To add a **New Entry**, press the add button, (left soft key). To cancel, press the cancel soft key.

After pressing the **add** button, the menu shows the **Name, Number, and Ring Default** with the cursor flashing in the first character of the Name area.

Entering Directory Names, Numbers and Ring Default

After pressing **add**, you are prompted to enter a **Name**. To enter a letter of a name, press the letter on the numeric keypad.

For example, press **2**, and the letter **a** appears. If the cursor is in the first letter of name, it is a capital letter. Every two seconds, the cursor flashes. If the **2** is NOT pressed again before the cursor flashes, the phone accepts the **A** as the entry and it moves to the next position. To obtain a different letter/number associated with the **2** (a, b, c, 2, A, B, C), press **2** again before the cursor flashes. The letters/numbers are displayed at the bottom of the screen and the letter/number that is entered is highlighted.

The first letter in a directory name defaults to a capital letter and subsequent letters in the name defaults to lower case. After entering the last letter, scroll down. The cursor is moved to the **Number** field. Enter the telephone number associated with this name. When you are finished entering the number, scroll down. The cursor is moved to the **Ring** field. Enter the Ring type for this entry, and then press **ok**. The entry is saved in the directory.

Press the **delChr** soft key to correct a mistake. Each time the **delChr** soft key is pressed one letter from the entered string is removed from the display. Press **clear** to clear the whole string.

To enter a phone number into the directory, simply dial the number including the **1** and the Area Code (for example, 14085721111).



Note

Enter 9 first to call an outside line.

The same screen can be used to set the phone to a distinctive ring for each caller. To set the ring, move the cursor to the **Ring Default** area by pressing the **down** arrow button after entering the phone number instead of pressing **save** per above. Use the arrow button to scroll to and highlight the ring for this person and press the **select** soft key. When the entry is complete, press the **save** button (left soft key). After the information has been saved, the phone displays the directory with the name and number.

**Note**

Be sure to press the **save** button after entering information. Otherwise, the information is not saved and will not appear in the directory.

2 Speed Dial

The Speed Dial option allows a single number to be used to automatically dial a telephone, IP address, or URL from the directory. This section provides directions for setting up and using speed dial numbers. To set-up a speed dial entry, press the Setup button and then press **2**. Alternatively, press the Setup button and scroll down until **SpeedDial** is highlighted and then press the **select** soft key.

The Speed dial menu indicates numbers available for assignment as **<Not Assigned>**. To assign an entry to an available number, press the number (or scroll down until it is highlighted), then press the **edit** soft key. The menu displays **Enter Speed Dial 1**. After pressing edit, the menu displays Enter Speed Dial Number 1. To enter a number or a letter, press the appropriate number. There are two options to create a speed dial entry:

- Enter a telephone number, an IP address or a URI into a speed dial memory location. When finished, press **ok** (left soft key). The entry is added to speed dial.
- Enter a name and number from the directory. As you enter letters, the phone displays names in the directory that match the letters displayed. Use the scroll buttons at any time to highlight to the name or type the name in completely. After highlighting a name (or typing in the name), press **ok**. The entry is added to speed dial.

**Note**

To assign a new name to an already assigned number, perform the same steps but clear the old entry before entering the new name.

Press the delChr soft key if a mistake is entered. Pressing the delChr soft key removes one letter from the entered string at a time.

There are two ways to use speed dial entries:

- Dial the speed dial number and then lift the handset.
- Lift the handset, dial the speed dial number, and then #.

3 Call History

Shortcut:: Setup, 3

To access Call History, press the **Setup** button, then press **3**. Alternatively, you can press Setup, scroll down until call history is highlighted, and then press the **select** soft key.

The Call History option provides three options:

- Redial List
- Answered Calls
- Missed Calls

To select an option, press the corresponding number or scroll down until the appropriate option is highlighted and press the **select** soft key. To return to the previous menu use the **cancel** option (right soft key).

Redial List

Shortcut:: Setup, 3, 1

After selecting this option, the phone displays the call entries in the Redial List.

The details of the call are displayed including the name if available, along with the telephone number, the date, and the time of the call. To dial an entry, scroll through the list to highlight the number and press the **dial** soft key. To return to the Setup option, press the **cancel** soft key.

Answered Calls

Shortcut:: Setup, 3, 2

After selecting this option, the phone displays the call entries in the Answered Call List.

The details of the call are displayed, including the name if available, along with the telephone number, the date, and the time of the call. To dial an entry, scroll through the list to highlight the number and press the **dial** soft key. To return to the Setup option, press the **cancel** soft key.

Missed Calls

Shortcut:: Setup, 3, 3

After selecting this option, the phone displays the call entries in the Missed Calls List. The details of the call are displayed, including the name if available, along with the telephone number, the date, and the time of the call. To dial an entry, scroll through the list to highlight the number and press the **dial** soft key. To return to the Setup option, press the **cancel** soft key.

4 Ring Tone

Shortcut:: Setup, 4, select extension, change, select ring tone

This option lets you select a different ring tone for each extension that the phone supports. Phones can have distinctive ring tones, which can be useful in a shared office environment.

To set or change the ring tone for a specific extension, press Setup and then select ring tones by pressing **4**. Alternatively, press the Setup button and scroll down to the ring tone option and press **select** using the left soft key. To exit this option and return to the previous menu, press the **cancel** soft key.

When the Ring Tone option is selected, the ring tone used for each extension is displayed. To change a Ring tone, either enter the corresponding number or scroll down until the selection is highlighted and press the **change** soft key. To return to the previous menu, press the **cancel** soft key.

To select a ring tone, press the corresponding number or highlight the option. To listen to the ring tones, press the **play** soft key. To pick the ring tone desired for this extension, press the **select** soft key. After pressing **select**, the display returns to the menu showing the extensions and the ring tones associated with them.

5 Preferences

This option allows you to set the preferences on the phone including the following:

- 5.1 Block Caller ID
- 5.2 Block Anonymous callers
- 5.3 Do Not Disturb
- 5.4 Secure Call
- 5.5 Dial Assistance
- 5.6 Preferred Audio Device
- 7 Auto Answer Page
- 8 Send Audio To Speaker
- 9 Time Format
- 10 Date Format
- 11 Miss Call Shortcut
- 12 Call Waiting

To select the Preferences option, press the Setup button and then press **5** or scroll down to the Preference option, highlight it, and press the **select** soft key. To select an option from the Preferences menu, press the number of the selection or scroll down to the selection and press the **select** soft key. To enable or disable an option, toggle between Yes and No by pressing the **y/n** soft key.

5.1 Block Caller ID

Shortcut: Setup, 5, edit, y/n, ok

When Block Caller ID is enabled, the calling number does not appear to the called party.

To select Block Caller ID, press the Setup button and then press **5** for preferences, or scroll down until Preferences is highlighted, and press the **select** soft key. With **Block Caller ID No** highlighted, press the **edit** soft key.

Enable or disable the option by pressing the **y/n** soft key and press **ok**. The display returns to the **Preferences Menu**.

5.2 Block Anonymous Call

Shortcut: Setup, 5, select, 2, edit, y/n, ok

When this feature is enabled, calls from callers that do not present their Caller ID are not received.

To select **Block Anonymous Calls**, press the Setup button and then press **5**, or scroll down until the selection is highlighted and press the **select** soft key. Press **2** or scroll down until **Block Anonymous Calls** is highlighted and then press the **edit** soft key.

Enable or disable the option by pressing the **y/n** soft key and press **ok**. The display returns to the **Preferences Menu**.

5.3 Do Not Disturb

Shortcut: Setup, 5, 3, edit, y/n, ok

Do Not Disturb routes the call to voicemail or presents a busy signal if voicemail is not available.

From the Preferences menu, press **3** or scroll down until the selection is highlighted and then press the **edit** soft key. This option is on the second preferences screen so scroll past the Block Caller ID, and Block Anonymous caller options to get to the Do Not Disturb option.

Enable or disable the option by pressing the **y/n** soft key and press **ok**. The display returns to the **Preferences Menu**.

5.4 Secure Call

Shortcut: Setup, 5, 4, edit, y/n, ok

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

From the Preferences menu, press **4** or scroll down until the selection is highlighted and then press **edit**. This option is on the second Preference screen; scroll past the first three preferences.

Enable or disable the option by pressing the **y/n** soft key and press **ok**. The display returns to the **Preferences Menu**.

5.5 Dial Assistance

Shortcut: Setup, 5, 5, edit, y/n, ok

The Dial Assistance option allows the Linksys IP phone to automatically suggest a number on the display based on entries in the directory or call logs. Use the down arrow button to select the number to be dialed. Press **dial** to complete the call

Enable or disable the option by pressing the **y/n** soft key and press **ok**. The display returns to the **Preferences Menu**.

5.6 Preferred Audio Device

Shortcut: Setup, 5, 6, edit, y/n, ok

From the Preferences menu, press **6** for **Preferred Audio Device** or scroll down until the selection is highlighted and then press **edit**.

Use the **option** soft key to toggle between the two options, Speaker or Handset. With the Preferred Audio Device showing the desired selection, press **ok**. The display returns to the Preferences menu.

5.7 Auto Answer Page

Shortcut: Setup, 5, 7, edit, y/n, ok

From the Preferences menu, press **7** for **Auto Answer Page** or scroll down until the selection is highlighted and then press **edit**.

Enable or disable the option by pressing the **y/n** soft key and press **ok**. The display returns to the **Preferences Menu**.

5.8 Send Audio To Speaker

Shortcut: Setup, 5, 8, edit, y/n, ok

From the Preferences menu, press **8** for **Send Audio To Speaker** or scroll down until the selection is highlighted and then press **edit**.

Use the **option** soft key to toggle between the two options, Speaker or Handset. With the desired selection displayed, press **ok**. The display returns to the Preferences menu.

5.9 Time Format

Shortcut: Setup, 5, 9, edit, y/n, ok

From the Preferences menu, press **9** for **Preferred Audio Device** or scroll down until the selection is highlighted and then press **edit**.

Use the **option** soft key to toggle between the two options, 12hr and 24hr. With the desired selection displayed, press **ok**. The display returns to the Preferences menu.

5.10 Date Format

Shortcut: Setup, 5, 10, edit, y/n, ok

From the Preferences menu, press **10** for **Date Format** or scroll down until the selection is highlighted and then press **edit**.

Use the **option** soft key to toggle between the two options, month/day or day/month. With the desired selection displayed, press **ok**. The display returns to the Preferences menu.

5.11 Miss Call Shortcut

Shortcut: Setup, 5, 11, edit, y/n, ok

From the Preferences menu, press **11** for **Miss Call Shortcut** or scroll down until the selection is highlighted and then press **edit**.

Enable or disable the option by pressing the **y/n** soft key and press **ok**. The display returns to the **Preferences Menu**.

5.12 Call Waiting

Shortcut: Setup, 5, 12, edit, y/n, ok

From the Preferences menu, press **12** for **Call Waiting** or scroll down until the selection is highlighted and then press **edit**.

Enable or disable the option by pressing the **y/n** soft key and press **ok**. The display returns to the **Preferences Menu**.

6 Call Forward

To view or change the call forwarding setting, press the Setup button and press the **6** button or use the arrow button to scroll down to **Call Forward** and press the **edit** soft key. The Call Forward menu provides four call forwarding options:

- 6.1 CFWD All Number—Forwards all calls to a single number
- 6.2 CFWD Busy Number—Forwards calls to another number when the phone is busy
- 6.3 CFWD No Ans Number—Forwards calls to another number if there's no answer
- 6.4 CFWD No Ans Delay—Forwards calls if there is no answer after a time delay

6.1 CFWD All Number

Shortcut: Setup, 6, edit, *phone_number*, Ok

This option forwards all calls to the specified number.

Enter the phone number for calls to be forwarded using the same method as dialing a number. For an external number, include the **1** and the area code (for example, **14085551212**). Erase mistaken numbers, one at a time using the **delChr** soft key. After the number is entered correctly, press **ok**.

The forwarding number is displayed on the Call Forward option. Calls are not forwarded if a telephone number is not entered followed by **ok**.

6.2 CFWD Busy Number

Shortcut: Setup, 6, 2, edit, *phone_number*, ok

This option forwards calls to another number when the phone is busy.

To forward all calls to specific telephone number when the phone is busy Press Setup, Press **6**, press **2** or scroll down until **CFWD Busy Number** is highlighted and press the **edit** soft key. Enter the telephone number to which to forward calls when the phone is busy. After the number is entered correctly, press **ok**. The forwarding number is displayed on the **Call Forward** sub menu. Calls are not forwarded unless a telephone number is entered.

6.3 CFWD No Ans Number

Shortcut: Setup, 6, 3, edit, *phone_number*, ok

This option forwards calls to another number if there is no answer.

Press the **edit** soft key and enter the phone number to which calls should be forwarded when the phone is not answered. After the number is entered correctly, press **ok**. The forwarding number is displayed on the **Call Forward** sub menu. Calls are not forwarded if a telephone number is not entered followed by **ok**.

6.4 CFWD No Ans Delay

Shortcut: Setup, 6, 4, edit, *phone_number*, ok

This option sets the time interval after which forwarding occurs.

Press the **edit** soft key and the number of seconds to elapse before the phone forwards the call. Calls are forwarded to the telephone number entered using the **CFWD No Ans** option after the specified time has elapsed.

7 Time/Date

Shortcut: Setup, 7, *date, time, offset*, save

To set the date and time, press **Setup** and then press **7**, or scroll down until the selection is highlighted and then press **select**. Enter the date and time in the following format:

- Date—**m/d/y**
- Time—**H:m:s**

- Offset—+/-**H:m:s**

To enter a / or : use the * (star) key. For offset value, use the * (star) key for + and the # (pound) key for -. After entering the required values, press the **save** soft key, to save the information into the phone memory. Erase mistaken numbers, one at a time using the **delChr** soft key.

8 Voice Mail

Shortcut: Setup, 8, *voicemail_number*, save

To set the number to call voicemail, press **Setup** and then press **8**, or scroll down until the selection is highlighted and press the **select** soft key.

When the display prompts for the Voice Mail Number, enter the number as if dialing the number. Erase mistaken numbers, one at a time using the **delChr** soft key. Then press the **save** button to save the number. After the number is saved, use the mailbox key on the right hand side of the phone (this is the third button up from the Setup button) to dial the mailbox.

9 Network

To view or modify network parameters, press the Setup button and then press **9**, or scroll down until **Network** is highlighted and then press the **select** soft key. This menu displays the IP network configuration settings.

The following options are available from the Network menu:

- 9.1 DHCP
- 9.2 Current IP Address
- 9.3 Host Name
- 9.4 Domain
- 9.5 Current Subnet Mask
- 9.6 Current Gateway
- 9.7 Enable Web Server
- 9.8 Non DHCP IP Address
- 9.9 Non DHCP Subnet Mask
- 9.10 Non DHCP Default Route
- 9.11 Non DHCP DNS 1
- 9.12 Non DHCP DNS 2
- 9.13 Non DHCP NTP Server 1
- 9.14 Non DHCP NTP Server 2
- 9.15 Multicast Address
- 9.16 Enable VLAN
- 9.17 VLAN ID
- 9.18 CDP

9.1 DHCP

Shortcut: Setup, 9, edit, y/n, ok

When DHCP is enabled, the phone obtains its IP address from a DHCP server.

To view DHCP status, press the **Setup** button and press **9**, or scroll down to the **Network** option and press the **select** soft key. The display shows **1 DHCP Enable** with a **Yes** or **No** to indicate the current status.

To enable or disable DHCP, press the **edit** soft key and press the **y/n** soft key to toggle between **Yes** and **No**. After making the selection, press the **ok** button and the display returns to the **Network** menu.

9.2 Current IP Address

Shortcut: Setup, 9, 2 save

To view the current IP address, press **2** on the Network menu, or scroll down until Current IP Address is highlighted and press the **select** soft key. The Current IP Address is set in one of the following ways:

- Receive the address from a DHCP server by enabling DHCP (Option 9.1)
- Manually set the address using the Non DHCP IP Address option (Option 9.8)

9.3 Host Name

Shortcut: Setup, 9, 3 save

The **Host Name** is set by the administrator. The user cannot change the setting. To view the **Host Name**, press **3** from the Network menu, or scroll down to **Host Name** and press the **select** soft key.

9.4 Domain

Shortcut: Setup, 9, 4 save

The Domain Name displays the name of the domain in which the phone is operating and is not user configurable. To view the Domain, press **4** from the Network menu, or scroll down to highlight **Domain** and press **select**.

9.5 Current NetMask

Shortcut: Setup, 9, 5 save

To view the Current NetMask, press **5** from the Network menu, or scroll down to highlight **NetMask** and press the **select** soft key. The Current NetMask is set in one of the following ways:

- Receive the address from a DHCP server by enabling DHCP (option 9.1)
- Manually set the address using the Non DHCP Subnet Mask option (Option 9.9)

9.6 Current Gateway

Shortcut: Setup, 9, 6 save

To view the **Current Gateway**, press **6** on the Network menu, or scroll down to highlight **Current Gateway** and press the **select** soft key.

The Current Gateway is set in one of the following ways:

- Receive the address from a DHCP server (you cannot edit it)
- Manually set the address using the Non DHCP Default Route option (Option 9.10)

9.7 Enable Web Server

Shortcut: Setup, 9, 7, y/n, ok

To view the status of the **Web Server**, press **7** from the Network menu, or scroll down to highlight **Enable Web Server** and press the **select** soft key. To enable or disable the **Web Server**, press the **edit** soft key. Press the **y/n** soft key to toggle between Yes to No and press **ok**.

9.8 Non DHCP IP Address

Shortcut: Setup, 9, 8, ip_address, ok

To manually set the IP address, press **8** from the Network menu, or scroll down to highlight **Non DHCP IP Address** and press the **select** soft key. Then press the **edit** soft key. The display shows the option to enter an IP address. enter the IP address here. This IP address is always used by the phone (it is a static IP address). After entering IP address, press **ok**.

9.9 Non DHCP Subnet Mask

Shortcut: Setup, 9, 9, subnet_mask, ok

To manually set the subnet mask, press **9** from the Network menu, or scroll down to highlight **Non DHCP Subnet Mask** and press the **select** soft key. Then press the **edit** soft key. After entering the subnet mask, press **ok**.

9.10 Non DHCP Default Route

Shortcut: Setup, 9, 10, ip_address, ok

To edit the default gateway (router), press **10** or scroll down to highlight **Non DHCP Default Route** and press the **select** soft key. Then press the **edit** soft key. After entering the default route, press **ok**.

9.11 Non DHCP DNS 1

Shortcut: Setup, 9, 11, ip_address, ok

To edit the primary DNS server used by the phone, press **11** from the Network menu, or scroll down and highlight **Non DHCP DNS 1** and press the **select** soft key. Then press the **edit** soft key. After entering the IP address of the server, press **ok**.

9.12 Non DHCP DNS 2

Shortcut: Setup, 9, 12, ip_address, ok

To edit the secondary DNS server used by the phone, press **12** from the Network menu, or scroll down and highlight **Non DHCP DNS 1** and press the **select** soft key. Then press the **edit** soft key. After entering the IP address of the server, press **ok**.

9.13 Non DHCP NTP Server 1

Shortcut: Setup, 9, 13, *ip_address*, ok

The Network Time Protocol (NTP) server sets the time used by the IP phone. To edit the primary NTP server used by the phone, press **13** from the Network menu, or scroll down and highlight **Non DHCP DNS 1** and press the **select** soft key. Then press the **edit** soft key. After entering the IP address of the server, press **ok**.

To change the phone to use a Non DHCP NTP Server 1, press the **edit** soft key. The display shows the option to enter an IP address. enter the IP Address of the NTP Server1. This IP address is always used by the phone, it is a static IP address. Erase mistaken numbers, one at a time using the **delChr** soft key. After IP address has been entered, press **ok**. The display returns to the Non DHCP NTP Server1 setting. The phone uses the IP address immediately.

9.14 Non DHCP NTP Server 2

Shortcut: Setup, 9, 14, *ip_address*, ok

To edit the secondary NTP server used by the phone, press **14** from the Network menu, or scroll down and highlight **Non DHCP DNS 1** and press the **select** soft key. Then press the **edit** soft key. After entering the IP address of the server, press **ok**.

9.15 Multicast Address

Shortcut: Setup, 9, 15, *multicast_address*, ok

To edit the secondary NTP server used by the phone, press **15** from the Network menu, or scroll down and highlight **Multicast Address** and press the **select** soft key. Then press the **edit** soft key. After entering the address, press **ok**.

9.16 Enable VLAN

Shortcut: Setup, 9, 16, y/n, ok

To enable VLAN, press **16** from the Network menu, or scroll down and highlight **Enable VLAN** and press the **select** soft key. Then press the **edit** soft key and press the **y/n** soft key to toggle between **Yes** and **No**. After making the selection, press the **ok** button and the display returns to the **Network** menu.

9.17 VLAN ID

Shortcut: Setup, 9, 17, *VLAN_ID*, ok

To edit the VLAN ID, press **17** from the Network menu, or scroll down and highlight **VLAN ID** and press the **select** soft key. Then press the **edit** soft key. After entering the ID, press **ok**.

9.18 CDP

Shortcut: Setup, 9, 18, y/n, ok

To enable Cisco Discovery Protocol (CDP), which allows Cisco devices to discover one another automatically, press **18** from the Network menu, or scroll down and highlight **CDP** and press the **select** soft key. Then press the **edit** soft key and press the **y/n** soft key to toggle between **Yes** and **No**. After making the selection, press the **ok** button and the display returns to the **Network** menu.

10 Product Info

To view the parameters in the **Product Info**, press the **Setup** button and then press **10**, or scroll down until **Product Info** is highlighted and then press the **select** soft key. The information displayed cannot be edited. The options on the Product Info menu are as follows:

- 10.1 Product Name
- 10.2 Serial Number
- 10.3 Software Version
- 10.4 Hardware Version
- 10.5 MAC Address
- 10.6 Client Cert
- 10.7 Customization

10.1 Product Name

Shortcut: Setup, 10, select

To view the Product Name, press the Setup button and press **10**, or scroll down to the **Product Info** menu, and press the **select** soft key. The Product Name is also printed on the bottom of the phone.

10.2 Serial Number

Shortcut: Setup, 10, 2, select

To view the Serial Number, press **2** from the Product Info menu, or scroll down to highlight **Serial Number** and press the **select** soft key. The serial number is also printed on the bottom of the phone.

10.3 Software Version

Shortcut: Setup, 10, 3, select

To view the software version, press **3** from the Product Info menu or scroll down to highlight **Software Version** and press the **select** soft key.

10.4 Hardware Version

Shortcut: Setup, 10, 4, select

To view the hardware version, press **4** from the Product Info menu or scroll down to highlight **Hardware Version** and press the **select** soft key.

.10.5 MAC Address

Shortcut: Setup, 10, 5, select

To view the MAC address, press **5** from the Product Info menu or scroll down to highlight **MAC Address** and press the **select** soft key. The MAC address is also printed on the bottom of the phone.

.10.6 Client Cert

Shortcut: Setup, 10, 6, select

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 key.

10.7 Customization

Shortcut: Setup, 10, 7, select

The phone can be configured for customization. Press **7** from the Product Info menu, or scroll down to the highlight **Customization**, and press the **select** soft key. “Open” indicates that the phone can be customized.

11 Status

To view the status of the phone and extension parameters, press the Setup button and press **11**, or scroll down until **Status** is highlighted and then press the **select** soft key. This menu displays read-only information about the status of the phone and provides the following options.



Note

On the SPA962 phone, “EXT” appears as “Ext,” and “Line” appears as “Line Key.”

- **Phone**
- **EXT 1**
- **EXT 2**
- **EXT 3**
- **EXT 4**
- **Line 1**
- **Line 2**
- **Line 3**
- **Line 4**

Phone

Shortcut: Setup, 11, select

To view the phone status, press the Setup button and then **11** or scroll down to **Status** and press the **select** soft key. The display provides the date and time, Up time, Transmitted packets (Tx) and Received Packets (Rx).

Ext 1/2/3/4

Shortcut: Setup, 11, 2 or 3 or 4 or 5 select

To see the settings for an extension, from the Status menu, press **2** for Ext 1, **3** for Ext 2, **4** for Ext 3, and **5** for Ext 4.

The LCD displays **Registered** if the extension is registered, the last registration time and date, and the time in seconds until the next registration (Next Reg). If the extension is not registered, the phone displays **Not Registered** along with the proxy's IP address and port number.

Line 1, 2,3,4

Shortcut: Setup, 11, 6 or 7 or 8 or 9 select

To see the settings for a line, from the Status menu, press **6** for Ext 1, **7** for Ext 2, **8** for Ext 3, and **9** for Ext 4. The display shows the status of the line including if the line is ready and what tone is used.

Provisioning

Shortcut: Setup, 11, 14 select

To see the provisioning status for your phone, from the Status menu, scroll to **14**, then press **select**. The display shows the profile, current provisioning status (Failed or Successful). If Failed, the reason for the failure is displayed.

Call Statistics History

Shortcut: Setup, 11, 15 select

To see the call statistics history for your phone, from the Status menu, scroll to **15**, then press **select**. The display shows the call history. If there is no call history logged, the phone displays "No Item."

12 Reboot

Shortcut: Setup, 12, ok

This option lets you reboot the phone. To select Reboot, press the Setup button and press **12**, or scroll down until the selection is highlighted and then press **select** soft key. The LCD displays **Confirm System Reboot?** You can choose either ok (OK on the SPA962) or Cancel. Select **ok** (**OK** on the SPA962) to reboot the phone. The option is only displayed for 5 seconds. After 5 seconds, the display returns to the Setup option.

13 Restart

Shortcut: Setup, 13, ok

This option lets you restart the phone. To select Restart, press the Setup button and press **13**, or scroll down until the selection is highlighted and then press the **select** soft key.

The menu displays **Confirm System Restart?** You can choose either ok (OK on the SPA962) or Cancel. Select **ok** (**OK** on the SPA962) to restart the phone. This option is only displayed for 5 seconds. After 5 seconds, the display is returned to the Setup option.

14 Factory Reset

Shortcut: Setup, 14, ok

This option lets you reset the parameters to the factory settings. To select Factory Reset, press the Setup button and press 14, or scroll down until the selection is highlighted and press the **select** soft key.

The menu displays **All parameters will be reset to factory default values. Continue?**

You can choose either ok (OK on the SPA962) or Cancel. Select **ok** (**OK** on the SPA962) to to change the settings of the phone to the factory default settings. This option is only displayed for 5 seconds. After 5 seconds, the display is returned to the Setup option.

15 Custom Reset

Shortcut: Setup, 15, ok

The Custom Reset feature works with the Remote Customization (RC) unit. RC units have different configurations, depending on the ITSP. Custom Reset allows a Service Provider to change its original configuration. This is a one-time install; after the RC is configured for a specific Service Provider, it cannot be changed.

When a user receives an RC unit, the unit obtains the correct profile for its ITSP using the MAC and server addresses.

16 Set Password

Shortcut: Setup, 16, *password*, ok

This option lets you set a password to protect the phone from unauthorized use. To select this option, press the Setup button and press **15**, or scroll down until the selection is highlighted and press the **select** soft key.

When the LCD displays **New Password**, enter the password and press the **save** soft key.



Caution

If the password is lost, to recover the phone you must perform a factory reset. A factory reset erases all current settings, including the directory and call logs.

17 Set LCD Contrast

Shortcut: Setup, 17, scroll knob left or right, save

This option lets you adjust the LCD contrast. To use this option, press the Setup button and press **16** scroll down to highlight this option and press the **select** soft key.

Press the Navigation button right or left to adjust the LCD contrast up or down. Press the **save** soft key when finished, or the **cancel** soft key to keep the previously selected LCD contrast.

18 CallPark Status

Shortcut: Setup, 18

This option lets you display the status of the CallPark feature. When this option is selected, the LCD displays the Requesting CallPark Status message.

The LCD then displays the CallPark Status or displays the Request CallPark Status Failed message and the request is retried.

To cancel the request press **Cancel**.

19 Language (SPA922, 942, and 962)

Shortcut: Setup, 19, Edit, Option, OK, Save

This option is available only with Release 5.1.4 or later of the SPA922, SPA942, and SPA962. This option lets you change the language of the phone:

1. Press the **Setup** button.
2. Select **Language** (option 18).
3. Select **Edit** to change language.
4. Select **Option** to change the language.
5. Select **OK** and **Save**.

Dictionaries are currently available for the following languages:

- English

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

For information about configuring the languages available through the LCD menu, refer to the [“Localization” section on page 3-5](#).

20 GUI Properties (SPA962)

The SPA962 lets you change the background on the LCD display. You can change:

- Preferred Background Picture: BMP Picture, Default, or None
- Preferred Menu Color Scheme: Light Blue, Light Green, Pink, Silver

21 Web Service (SPA962)

The SPA962 provides the option to view RSS newsfeeds by selecting one of the following options on the LCD display:

1. Stock Quote.
2. Traffic
3. Local News
4. World News
5. Finance News
6. Sport News
7. Politics News

Option 3 to 7 are configurable using the administration web server, as described in [“Configuring the Web Service” section on page 3-14](#).

5

Linksys IP Phone Field Reference

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

- [Info Tab, page 5-2](#)
- [System Tab, page 5-6](#)
- [SIP Tab, page 5-9](#)
- [Regional Tab, page 5-19](#)
- [Phone Tab, page 5-30](#)
- [Ext Tab, page 5-36](#)
- [User, page 5-48](#)
- [932 Tab \(SPA962 only\), page 5-51](#)
- [SPA932 Status, page 5-54](#)

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:

- [System Information, page 5-2](#)
- [Product Information, page 5-2](#)
- [Phone Status, page 5-3](#)



Note

The fields on this tab are read-only and cannot be edited.

System Information

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

Product Information

Field	Description
Product Name	Model number of the Linksys IP phone.
Serial Number	Serial number of the Linksys IP phone.
Software Version	Version number of the Linksys IP phone software.
Hardware Version	Version number of the Linksys IP phone hardware.
MAC Address	MAC address of the Linksys IP phone.
Client Certificate	Status of the client certificate, which authenticates the Linksys IP phone for use in the ITSP network. This field indicates if the client certificate is properly installed in the Linksys IP phone.
Customization (not applicable to SPA901)	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 Linksys IP phone.

Phone Status

Field	Description
Current Time	Current date and time of the system; for example, 10/3/2003 16:43:00.
Elapsed Time	Total time elapsed since the last reboot of the system; for example, 25 days and 18:12:36.
Broadcast Pkts Sent	Total number of broadcast packets sent.
Broadcast Bytes Sent	Total number of broadcast packets received.
Broadcast Pkts Recv	Total number of broadcast bytes sent.
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 (not applicable to SPA901)	ID of the VLAN currently in use.

Ext 1/2/3/4/5/6 Status

Field	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.

Field	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 1/2/3/4/5/6 Call 1/2 Status

Field	Description
Hazardous Potential	The amount of hazardous potential is displayed.
Foreign Voltage	The amount of foreign voltage is displayed.
Long Resistive Fault	The amount of long resistive fault is displayed.
REN	The Ringer Equivalence Number (REN) is displayed.
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.

Round Trip Delay	Number of milliseconds for delay.
Packets Lost	Number of packets lost.
Packet Error	Number of invalid packets received.
Mapped RTP Port	The port mapped for Real Time Protocol traffic for the call.
Media Loopback (not applicable to SPA901)	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

Field	Description
Status	Indicates whether the phone is downloading a ring tone (and from where) or if it is idle.
Ring Tone 1	Information about the user downloaded ring tone 1: name, size, and time-stamp of the tone.
Ring Tone 2	Information about the user downloaded ring tone 2: name, size, and time-stamp of the tone.

System Tab

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

- [System Configuration, page 5-6](#)
- [Internet Connection Type, page 5-6](#)
- [Optional Network Configuration, page 5-7](#)
- [VLAN Settings, page 5-8](#)

System Configuration

Field	Description
Restricted Access Domains (not applicable to SPA901)	This feature is used when implementing software customization.
Enable Web Server (not applicable to SPA901)	Enable/disable web server of Linksys IP phone This feature should only be used on firmware version 1.0.9 or later. Defaults to yes.
Web Server Port (not applicable to SPA901)	Port number of the Linksys IP phone administration web server. Defaults to 80.
Enable Web Admin Access (not applicable to SPA901)	Lets you enable or disable local access to the administration web server. Select yes or no from the drop-down menu. Defaults to yes.
Admin Passwd (not applicable to SPA901)	Password for the administrator. Defaults to no password.
User Password	Password for the user. Defaults to no password.

Internet Connection Type

Field	Description
DHCP	Enable or disable DHCP. Defaults to DHCP (yes on the SPA901).
Static IP (SPA901 only)	IP address. Blank if DHCP assigned address is used.
Netmask (SPA901 only)	Network mask. Blank if DHCP assigned.

Field	Description
Gateway (SPA901 only)	Default router IP address. Blank if DHCP assigned.

Static IP Settings

**Note**

None of the following parameters apply to the SPA901.

Field	Description
Static IP	IP address. Blank if DHCP assigned address is used.
Netmask	Network mask. Blank if DHCP assigned.
Gateway	Default router IP address. Blank if DHCP assigned.

PPPoE Settings

**Note**

None of the following parameters apply to the SPA901.

Field	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

Field	Description
Host Name	The host name of the Linksys IP phone.
Domain	The network domain of the Linksys IP phone.
Primary DNS	DNS server used by Linksys IP phone in addition to DHCP supplied DNS servers if DHCP is enabled; when DHCP is disabled, this is the primary DNS server. Defaults to 0.0.0.0.

System Tab

Field	Description
Secondary DNS	DNS server used by Linksys IP phone in addition to DHCP supplied DNS servers if DHCP is enabled; when DHCP is disabled, this is the secondary DNS server. Defaults to 0.0.0.0.
DNS Server Order (Not applicable to SPA901)	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 Linksys IP phone sends the same request to all the DNS servers at the same time when doing a DNS lookup, the first incoming reply is accepted by the Linksys IP phone. Defaults to parallel.
Syslog Server	Specify the syslog server name and port. This feature specifies the server for logging Linksys IP phone system information and critical events. If both Debug Server and Syslog Server are specified, Syslog messages are also logged to the Debug Server.
Debug Server	The debug server name and port. This feature specifies the server for logging Linksys IP phone debug information. The level of detailed output depends on the debug level parameter setting.
Debug Level	The debug level from 0-3. The higher the level, the more debug information is generated. Zero (0) means no debug information is generated. To log SIP messages, you must set the Debug Level to at least 2. Defaults to 0.
Primary NTP Server (Not applicable to SPA901)	IP address or name of primary NTP server.
Secondary NTP Server (Not applicable to SPA901)	IP address or name of secondary NTP server.

VLAN Settings



Note None of the following parameters apply to the SPA901.

Field	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 5-9](#)
- [SIP Timer Values \(sec\), page 5-11](#)
- [Response Status Code Handling, page 5-13](#)
- [RTP Parameters, page 5-13](#)
- [SDP Payload Types, page 5-14](#)
- [NAT Support Parameters, page 5-16](#)
- [Linksys Key System Parameters, page 5-17](#)

SIP Parameters



Note

None of the following parameters apply to the SPA901.

Field	Description
Max Forward	SIP Max Forward value, which can range from 1 to 255. Defaults to 70.
Max Redirection	Number of times an invite can be redirected to avoid an infinite loop. Defaults to 5.
Max Auth	Maximum number of times (from 0 to 255) a request may be challenged. Defaults to 2.
SIP User Agent Name	Used in outbound REGISTER requests. Defaults to \$VERSION. If empty, the header is not included. Macro expansion of \$A to \$D corresponding to GPP_A to GPP_D allowed.
SIP Server Name	Server header used in responses to inbound responses. Defaults to \$VERSION.
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 Linksys IP phone 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.

Field	Description
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.
Use Compact Header	Lets you use compact SIP headers in outbound SIP messages. Select yes or no from the drop-down menu. If set to yes, the Linksys IP phone uses compact SIP headers in outbound SIP messages. If set to no, the Linksys IP phone uses normal SIP headers. If inbound SIP requests contain compact headers, Linksys IP phone reuses the same compact headers when generating the response regardless the settings of the <Use Compact Header> parameter. If inbound SIP requests contain normal headers, Linksys IP phone substitutes those headers with compact headers (if defined by RFC 261) if <Use Compact Header> parameter is set to yes. Defaults to no.
Escape Display Name	Lets you keep the Display Name private. Select yes if you want the Linksys IP phone to enclose the string (configured in the Display Name) in a pair of double quotes for outbound SIP messages. Any occurrences of or \ in the string is escaped with \ and \\ inside the pair of double quotes. Otherwise, select no. Defaults to yes.
SIP-B Enable	Enables Sylantro call features.
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	Enables or disables the Computer Telephone Interface feature provided by some servers.

SIP Timer Values (sec)



Note

None of the following parameters apply to the SPA901.

Field	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.
INVITE Expires	INVITE request Expires header value. If you enter 0, the Expires header is not included in the request. Ranges from 0 to 19999999999999999999999999999999. Defaults to 240.
ReINVITE Expires	ReINVITE request Expires header value. If you enter 0, the Expires header is not included in the request. Ranges from 0 to 19999999999999999999999999999999. Defaults to 30.
Reg Min Expires	Minimum registration expiration time allowed from the proxy in the Expires header or as a Contact header parameter. If the proxy returns a value less than this setting, the minimum value is used. Defaults to 1.
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 Linksys IP phone 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 Linksys IP phone waits for the specified length of time before retrying. If this interval is 0, the Linksys IP phone stops trying. This value should be much larger than the Reg Retry Intvl value, which should not be 0. Defaults to 1200.

Field	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

Field	Description
SIT1 RSC	SIP response status code for the appropriate Special Information Tone (SIT). For example, if you set the SIT1 RSC to 404, when the user makes a call and a failure code of 404 is returned, the SIT1 tone is played. Reorder or Busy Tone is played by default for all unsuccessful response status code for SIT 1 RSC through SIT 4 RSC. Defaults to blank.
SIT2 RSC	SIP response status code to INVITE on which to play the SIT2 Tone. Defaults to blank.
SIT3 RSC	SIP response status code to INVITE on which to play the SIT3 Tone. Defaults to blank.
SIT4 RSC	SIP response status code to INVITE on which to play the SIT4 Tone. Defaults to blank.
Try Backup RSC (not applicable to SPA901)	SIP response code that retries a backup server for the current request. Defaults to blank.
Retry Reg RSC (not applicable to SPA901)	Interval to wait before the Linksys IP phone retries registration after failing during the last registration. Defaults to blank.

RTP Parameters

Field	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 (not applicable to SPA901)	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 (not applicable to SPA901)	Number of successive ICMP errors allowed when transmitting RTP packets to the peer before the Linksys IP phone terminates the call. If value is set to 0, the Linksys IP phone ignores the limit on ICMP errors. Defaults to 0.

Field	Description
RTCP Tx Interval (not applicable to SPA901)	Interval for sending out RTCP sender reports on an active connection. It can range from 0 to 255 seconds. During an active connection, the Linksys IP phone 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/Linksys IP phone-1.0.31(b)). The NTP timestamp used in the SR is a snapshot of the Linksys IP phone's local time, not the time reported by an NTP server. If the Linksys IP phone 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 Linksys IP phone web page. Defaults to 0.
No UDP Checksum (not applicable to SPA901)	Select yes if you want the Linksys IP phone to calculate the UDP header checksum for SIP messages. Otherwise, select no. Defaults to no.
Symmetric RTP (not applicable to SPA901)	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 (not applicable to SPA901)	Determines whether the Linksys IP phone includes the P-RTP-Stat header or response to a BYE message. The header contains the RTP statistics of the current call. Select yes or no from the drop-down menu. The format of the P-RTP-Stat header is: P-RTP-State: PS=<packets sent>,OS=<octets sent>,PR=<packets received>,OR=<octets received>,PL=<packets lost>,JI=<jitter in ms>,LA=<delay in ms>,DU=<call duration in s>,EN=<encoder>,DE=<decoder>. Defaults to no.

SDP Payload Types

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

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

Field	Description
AVT Dynamic Payload	AVT dynamic payload type. Ranges from 96-127. Defaults to 101.

Field	Description
INFOREQ Dynamic Payload (not applicable to SPA901)	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 (not applicable to SPA901)	G726r32 dynamic payload type. Defaults to 2.
G726r40 Dynamic Payload	G.726-40 dynamic payload type. Ranges from 96-127. Defaults to 96.
G729b Dynamic Payload	G.729b dynamic payload type. Ranges from 96-127. Defaults to 99.
EncapRTP Dynamic Payload (not applicable to SPA901)	EncapRTP Dynamic Payload type. Defaults to 112.
RTP-Start-Loopback Dynamic (not applicable to SPA901)	RTP-Start-Loopback Dynamic Payload. Defaults to 113.
RTP-Start-Loopback Codec (not applicable to SPA901)	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 (not applicable to SPA901)	AVT codec name used in SDP. Defaults to telephone-event.
G711u Codec Name (not applicable to SPA901)	G.711u codec name used in SDP. Defaults to PCMU.
G711a Codec Name (not applicable to SPA901)	G.711a codec name used in SDP. Defaults to PCMA.
G726r16 Codec Name (not applicable to SPA901)	G.726-16 codec name used in SDP. Defaults to G726-16.
G726r24 Codec Name (not applicable to SPA901)	G.726-24 codec name used in SDP. Defaults to G726-24.
G726r32 Codec Name (not applicable to SPA901)	G.726-32 codec name used in SDP. Defaults to G726-32.
G726r40 Codec Name (not applicable to SPA901)	G.726-40 codec name used in SDP. Defaults to G726-40.

Field	Description
G729a Codec Name (not applicable to SPA901)	G.729a codec name used in SDP. Defaults to G729a.
G729b Codec Name (not applicable to SPA901)	G.729b codec name used in SDP. Defaults to G729ab.
G723 Codec Name (not applicable to SPA901)	G.723 codec name used in SDP. Defaults to G723.
EncapRTP Codec Name (not applicable to SPA901)	EncapRTP codec name used in SDP. Defaults to encaprtsp.

NAT Support Parameters



Note

None of the following parameters apply to the SPA901.

Field	Description
Handle VIA received	If you select yes, the Linksys IP phone processes the received parameter in the VIA header (this is inserted by the server in a response to any of its requests). If you select no, the parameter is ignored. Select yes or no from the drop-down menu. Defaults to no.
Handle VIA rport	If you select yes, the Linksys IP phone 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.

Field	Description
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 Linksys IP phone 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 Linksys IP phone 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 Linksys IP phone in all outgoing SIP messages. If 0.0.0.0 is specified, no IP address substitution is performed. If this parameter is specified, the Linksys IP phone 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

Field	Description
Linksys Key System	Enable or disable the Linksys Key System on the IP phone. Defaults to yes.
Multicast Address	The multicast address is used by the SPA9000 to communicate with the 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, page 5-19](#)
- [Distinctive Ring Patterns, page 5-20](#)
- [Control Timer Values \(sec\), page 5-21](#)
- [Vertical Service Activation Codes, page 5-22](#)
- [Vertical Service Activation Codes, page 5-22](#)
- [Outbound Call Codec Selection Codes, page 5-25](#)
- [Miscellaneous, page 5-27](#)

Call Progress Tones



Note

None of the following parameters apply to the SPA901.

Field	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).
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)

Field	Description
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).
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



Note

None of the following parameters apply to the SPA901.

Field	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 to 60(.2/.4,.2/.4,.2/4).
Cadence 7	Cadence script for distinctive ring 7. Defaults to 60(4.5/4).
Cadence 8	Cadence script for distinctive ring 8. Defaults to 60(0.25/9.75)
Cadence 9	Cadence script for distinctive ring 9. Defaults to 60(.4/.2,.4/2).

Control Timer Values (sec)

Field	Description
Reorder Delay (Not applicable to SPA901)	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 (Not applicable to SPA901)	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 Linksys IP phone still considers the call as failed and keeps on retrying. Defaults to 0.5.

Field	Description
Interdigit Long Timer (Not applicable to SPA901)	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 (Not applicable to SPA901)	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.



Note

Some of the following parameters do not apply to the SPA901.

Field	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.

Field	Description
Cfwd No Ans Act Code	Forwards no-answer calls to the extension specified after the activation code. Defaults to *92.
Cfwd No Ans Deact Code	Cancels call forwarding of no-answer calls. Defaults to *93.
CW Act Code	Enables call waiting on all calls. Defaults to *56.
CW Deact Code	Disables call waiting on all calls. Defaults to *57.
CW Per Call Act Code	Enables call waiting for the next call. Defaults to *71.
CW Per Call Deact Code	Disables call waiting for the next call. Defaults to *70.
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.

Regional Tab

Field	Description
Secure One Call Act Code	Makes the next outbound call secure. (It is redundant if all outbound calls are secure by default.) Defaults to *18.
Secure One Call Deact Code	Makes the next outbound call not secure. (It is redundant if all outbound calls are not secure by default.) Defaults to *19.
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.
Referral Services Codes	<p>These codes tell the Linksys IP phone what to do when the user places the current call on hold and is listening to the second dial tone.</p> <p>One or more *code can be configured into this parameter, such as *98, or *97 *98 *123, etc. Max total length is 79 chars. This parameter applies when the user places the current call on hold (by Hook Flash) and is listening to second dial tone. Each *code (and the following valid target number according to current dial plan) entered on the second dial-tone triggers the Linksys IP phone to perform a blind transfer to a target number that is prepended by the service *code.</p> <p>For example, after the user dials *98, the Linksys IP phone plays a special dial tone called the Prompt Tone while waiting for the user to enter a target number (which is checked according to dial plan as in normal dialing). When a complete number is entered, the Linksys IP phone sends a blind REFER to the holding party with the Refer-To target equals to *98<target_number>. This feature allows the Linksys IP phone to hand off a call to an application server to perform further processing, such as call park.</p> <p>The *codes should not conflict with any of the other vertical service codes internally processed by the Linksys IP phone. You can empty the corresponding *code that you do not want to Linksys IP phone to process.</p>

Field	Description
Feature Dial Services Codes	<p>These codes tell the Linksys IP phone what to do when the user is listening to the first or second dial tone.</p> <p>One or more *code can be configured into this parameter, such as *72, or *72 *74 *67 *82, etc. Max total length is 79 chars. This parameter applies when the user has a dial tone (first or second dial tone). Enter *code (and the following target number according to current dial plan) entered at the dial tone triggers the Linksys IP phone to call the target number prepended by the *code. For example, after user dials *72, the Linksys IP phone plays a prompt tone awaiting the user to enter a valid target number. When a complete number is entered, the Linksys IP phone sends a INVITE to *72<target_number> as in a normal call. This feature allows the proxy to process features like call forward (*72) or BLock Caller ID (*67).</p> <p>The *codes should not conflict with any of the other vertical service codes internally processed by the Linksys IP phone. You can empty the corresponding *code that you do not want to Linksys IP phone to process.</p> <p>You can add a parameter to each *code in Features Dial Services Codes to indicate what tone to play after the *code is entered, such as *72‘c’ *67‘p’. Below are a list of allowed tone parameters (note the use of back quotes surrounding the parameter w/o spaces)</p> <ul style="list-style-type: none"> • ‘c’ = <Cfwd Dial Tone> • ‘d’ = <Dial Tone> • ‘m’ = <MWI Dial Tone> • ‘o’ = <Outside Dial Tone> • ‘p’ = <Prompt Dial Tone> • ‘s’ = <Second Dial Tone> • ‘x’ = No tones are place, x is any digit not used above <p>If no tone parameter is specified, the Linksys IP phone plays Prompt tone by default.</p> <p>If the *code is not to be followed by a phone number, such as *73 to cancel call forwarding, do not include it in this parameter. In that case, simple add that *code in the dial plan and the Linksys IP phone send INVITE *73@..... as usual when user dials *73.</p>

Vertical Service Announcement Codes



Note

None of the following parameters apply to the SPA901.

- 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. So no need to include them in dial-plan (although no harm to do so either).



Note

None of the following parameters apply to the SPA901.

Field	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.
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.

Field	Description
Force G729a Code	Makes this codec the only codec that can be used for the associated call. Defaults to *02729.

Miscellaneous

**Note**

Only the first four parameters apply to the SPA901:

- Set Local Date
- Set Local Time
- Time Zone
- Time Offset

Field	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.

Field	Description
Daylight Saving Time Rule	<p>Enter the rule for calculating daylight saving time; it should include the start, end, and save values. This rule is comprised of three fields. Each field is separated by ; (a semicolon) as shown below. Optional values inside [] (the brackets) are assumed to be 0 if they are not specified. Midnight is represented by 0:0:0 of the given date.</p> <p>This is the format of the rule: Start = <start-time>; end=<end-time>; save = <save-time>.</p> <ul style="list-style-type: none"> 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). <p>The default Daylight Saving Time Rule is start=4/1/7;end=10/-1/7;save=1.</p>
DTMF Playback Level	<p>Local DTMF playback level in dBm, up to one decimal place.</p> <p>Defaults to -16.</p>
DTMF Playback Length	<p>Local DTMF playback duration in milliseconds.</p> <p>Defaults to .1.</p>
Inband DTMF Boost	<p>Controls the amount of amplification applied DTMF signals.</p> <p>Choices are 0dB, 3dB, 6dB, 9dB, 12dB, 15dB, or 18dB.</p> <p>Defaults to 12dB.</p>

Field	Description
Dictionary Server Script	<p>Defines the location of the dictionary server, the languages available and the associated dictionary. The syntax is as follows:</p> <pre><Dictionary_Server_Script ua="na"> </Dictionary_Server_Script></pre> <p>Defaults to blank and the maximum number of characters is 512. The detailed format is as follows:</p> <pre>serv={server ip port and root path}; d0=<language0>;x0=<dictionary0 filename>; d1=<language1>;x1=<dictionary1 filename>; d2=<language2>;x2=<dictionary2 filename>; d3=<language3>;x3=<dictionary3 filename>; d4=<language4>;x4=<dictionary4 filename>; d5=<language5>;x5=<dictionary5 filename>; d6=<language6>;x6=<dictionary6 filename>; d7=<language3>;x7=<dictionary7 filename>; d8=<language8>;x8=<dictionary8 filename>; d9=<language5>;x9=<dictionary9 filename>;</pre> <p>The following is an example value:</p> <pre><Dictionary_Server_Script ua="na"> serv=tftp://192.168.1.119/;d0=English;x0=enS_v101.xml;d1=Spanish;x1=es_v101. xml </Dictionary_Server_Script></pre>
Language Selection	<p>Specifies the default language. The value needs to match one of the languages supported by the dictionary server. The script (dx value) is as follows:</p> <pre><Language_Selection ua="na"> </Language_Selection></pre> <p>Defaults to blank and the maximum number of characters is 512. The following is an example:</p> <pre><Language_Selection ua="na"> Spanish </Language_Selection></pre>

Phone Tab

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

[General, page 5-30](#)

[Line Key 1/2/3/4/5/6, page 5-30](#)

[Miscellaneous Line Key Settings, page 5-31](#)

[Line Key LED Pattern, page 5-31](#)

[Supplementary Services, page 5-33](#)

[Ring Tone, page 5-34](#)

[Auto Input Gain \(dB\), page 5-35](#)

General

Field	Description
Station Name	A name to identify this station (reserved for future use).
Voice Mail Number	A phone number or URL to check voicemail.
Text Logo (Not applicable to SPA901)	Sets the text logo to display when the phone boots up. Default is (blank) which displays Linksys. Up to 2 lines of text can be setup; each line should contain less than 32 characters. A newline character (\n) must be inserted between the 2 lines and escaped with %0a. For example, "Super%0aTelecom".
BMP Picture Download URL (Not applicable to SPA901)	<p>The URL locating the bitmap (.BMP) file to display on the LCD background. The URL must include the TFTP server name (or IP address), directory, and filename, for example: tftp://myserver.mydomain.com/images/downloadablebmppicture.bmp</p> <p>For the SPA962, you can use a 320 x 240 pixel .BMP image (1, 4, 8, 24, or 32-bit color) for the background picture. For the SPA942, the image should be 128 x 48 black and white (1-bit color).</p>
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.

Line Key 1/2/3/4/5/6

When used in the configuration profile, parameters in this section must be appended with [n], where n is 1, 2, 3, or 4).

This section does not apply to the WIP310 wireless phone.

Field	Description
Extension	Select an extension to be used for this line key. Choices are: One-line phone: 1 or disabled, Four-line phone: 1, 2, 3, 4, or disabled. Six-line phone: 1, 2, 3, 4, 5, 6, or disabled. Defaults to 1.
Short Name (Not applicable to SPA901)	A short label shown on the LCD display for Line key 1/2/3/4/5/6.
Share Call Appearance (Not applicable to SPA901)	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

This section does not apply to the WIP310 wireless phone.

Field	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

Line key LED colors are described in [“Line Key LEDs” section on page 3-22](#).



Note

None of the following parameters apply to the SPA901.

This section does not apply to the WIP310 wireless phone.

Field	Description
Idle LED	LED pattern during the Idle state, where the call appearance is not in use and is available to make a new call. Leaving this entry blank indicates the default value of c=g.

Field	Description
Remote Undefined LED	LED pattern during the Remote Undefined state, where the shared call state is undefined (the station is still waiting for the state information from the application server). Not applicable if the call appearance is not shared. Leaving this entry blank indicates the default value of c=r;p=d.
Local Seized LED	LED pattern during the Local Seized state, where this station has seized the call appearance to prepare for a new outbound call. Leaving this entry blank indicates the default value of c=r.
Remote Seized LED	LED pattern during the Remote Seized state, where the shared call appearance is seized by another station. Not applicable if the call appearance is not shared. Leaving this entry blank indicates the default value of c=r;p=d.
Local Progressing LED	LED pattern during the Local Progressing state, where this station is attempting on this call appearance an outgoing call that is in proceeding (i.e. the called number is ringing). Leaving this entry blank indicates the default value of c=r.
Remote Progressing LED	LED pattern during the Remote Progressing state, where another station is attempting on this shared call appearance an outbound call that is progressing. Not applicable if the call appearance is not shared. Leaving this entry blank indicates the default value of c=r;p=d.
Local Ringing LED	LED pattern during the Local Ringing state, when the call appearance is ringing. Leaving this entry blank indicates the default value of c=r;p=f.
Remote Ringing LED	LED pattern during the Remote Ringing state, where the shared call appearance is in ringing on another station. Not applicable if the call appearance is not shared. Leaving this entry blank indicates the default value of c=r;p=d.
Local Active LED	LED pattern during the Local Active state, where the call appearance is engaged in an active call. Leaving this entry blank indicates the default value of c=r.
Remote Active LED	LED pattern during the Remote Active state, where another station is engaged in an active call on this shared call appearance. Not applicable if this call appearance is not shared. Leaving this entry blank indicates the default value of c=r;p=d.
Local Held LED	LED pattern during the Local Held state, where the call appearance is held by this station. Leaving this entry blank indicates the default value of c=r;p=s.
Remote Held LED	LED pattern during the Remote Held state, where another station has placed this call appearance on hold. Not applicable if the call appearance is not shared. Leaving this entry blank indicates the default value of c=4;p=s.
Register Failed LED	LED pattern when the corresponding extension has failed to register with the proxy server. Leaving this entry blank indicates the default value of c=a.
Disabled LED	LED pattern when the Call Appearance is disabled (not available for any incoming or outgoing call). Leaving this entry blank indicates the default value of c=o.
Registering LED	LED Pattern when the corresponding extension is trying to register with the proxy server. Leaving this entry blank indicates the default value of c=r;p=s.
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.


Note

Some of the parameters in this section do not apply to the WIP310 wireless phone. None of the following parameters apply to the SPA901.

Field	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.
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.

Field	Description
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.
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.

Field	Description
Ring-1	Ring tone script for Ring 1. Defaults to n=Classic-1;w=3;c=1.
Ring-2	Ring tone script for Ring 2. Defaults to n=Classic-2;w=3;c=2.
Ring-3	Ring tone script for Ring 3. Defaults to n=Classic-3;w=3;c=3.
Ring-4	Ring tone script for Ring 4. Defaults to n=Classic-4;w=3;c=4.
Ring-5	Ring tone script for Ring 5. Defaults to n=Simple-1;w=2;c=1.
Ring-6	Ring tone script for Ring 6. Defaults to n=Simple-2;w=2;c=2.
Ring-7	Ring tone script for Ring 7. Defaults to n=Simple-3;w=2;c=3.
Ring-8	Ring tone script for Ring 8. Defaults to n=Simple-4;w=2;c=4.
Ring-9	Ring tone script for Ring 9. Defaults to n=Simple-5;w=2;c=5.
Ring-10	Ring tone script for Ring 10. Defaults to n=Office;w=4;c=1.

Auto Input Gain (dB)

This section does not apply to the WIP310 wireless phone.

Field	Description
Handset Input Gain	The amount of amplification to apply to the audio input signal for the handset. Defaults to zero.
Headset Input Gain (Not applicable to SPA901)	The amount of amplification to apply to the audio input signal for the headset. Defaults to zero.
Speakerphone Input Gain (Not applicable to SPA901)	The amount of amplification to apply to the audio input signal for the speakerphone. Defaults to zero.
Accoustic Echo Canceller Enable	Enable or disable the accoustic echo canceller. Defaults to yes (enable).

Extension Mobility

**Note**

Some of the parameters in this section do not apply to the WIP310 wireless phone. None of the following parameters apply to the SPA901.

This section does not apply to the WIP310 wireless phone.

Field	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:

- SPA901: 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:

- [General, page 5-36](#)
- [Share Line Appearance, page 5-37](#)
- [NAT Settings, page 5-37](#)
- [Network Settings, page 5-37](#)
- [SIP Settings, page 5-38](#)
- [Call Feature Settings, page 5-40](#)
- [Proxy and Registration, page 5-41](#)
- [Subscriber Information, page 5-43](#)
- [Audio Configuration, page 5-43](#)
- [, page 5-45](#)

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

Field	Description
Line Enable	To enable this line for service, select yes. Otherwise, select no. Defaults to yes.

Share Line Appearance

Field	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

Field	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 (Not applicable to SPA901)	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 (Not applicable to SPA901)	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



Note

Some of the parameters in this section do not apply to the WIP310 wireless phone. None of the following parameters apply to the SPA901.

Field	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.
Network Jitter Level	Determines how jitter buffer size is adjusted by the Linksys IP phone. 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



Note

Some of the parameters in this section do not apply to the WIP310 wireless phone. Only the SIP Port and SIP Debug Option parameters apply to the SPA901.

Field	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.

Field	Description
Auth Resync-Reboot	<p>If this feature is enabled, the Linksys IP phone authenticates the sender when it receives the NOTIFY resync reboot (RFC 2617) message. To use this feature, select yes. Otherwise, select no.</p> <p>Defaults to yes.</p>
SIP Proxy-Require	<p>The SIP proxy can support a specific extension or behavior when it sees this header from the user agent. If this field is configured and the proxy does not support it, it responds with the message, unsupported. Enter the appropriate header in the field provided.</p>
SIP Remote-Party-ID	<p>To use the Remote-Party-ID header instead of the From header, select yes. Otherwise, select no.</p> <p>Defaults to yes.</p>
Referor Bye Delay	<p>Controls when the Linksys IP phone sends BYE to terminate stale call legs upon completion of call transfers. Multiple delay settings (Referor, Refer Target, Referee, and Refer-To Target) are configured on this screen. For the Referor Bye Delay, enter the appropriate period of time in seconds.</p> <p>Defaults to 4.</p>
Refer Target Bye Delay	<p>For the Refer Target Bye Delay, enter the appropriate period of time in seconds.</p> <p>Defaults to 0.</p>
Sticky 183	<p>If this feature is enabled, the IP telephony ignores further 180 SIP responses after receiving the first 183 SIP response for an outbound INVITE. To enable this feature, select yes. Otherwise, select no.</p> <p>Defaults to no.</p>
Auth INVITE	<p>When enabled, authorization is required for initial incoming INVITE requests from the SIP proxy.</p>
Referee Bye Delay	<p>For the Referee Bye Delay, enter the appropriate period of time in seconds.</p> <p>Defaults to 0.</p>

Field	Description
SIP Debug Option	<p>SIP messages are received at or sent from the proxy listen port. This feature controls which SIP messages to log. Choices are as follows:</p> <ul style="list-style-type: none"> • none—No logging. • 1-line—Logs the start-line only for all messages. • 1-line excl. OPT—Logs the start-line only for all messages except OPTIONS requests/responses. • 1-line excl. NTFY—Logs the start-line only for all messages except NOTIFY requests/responses. • 1-line excl. REG—Logs the start-line only for all messages except REGISTER requests/responses. • 1-line excl. 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. <p>Defaults to none.</p>
Refer-To Target Contact	<p>To contact the refer-to target, select yes. Otherwise, select no.</p> <p>Defaults to no.</p>

Call Feature Settings



Note

Many of the following parameters do not apply to the SPA901.

Field	Description
Blind Attn-Xfer Enable	<p>Enables the Linksys IP phone to perform an attended transfer operation by ending the current call leg and performing a blind transfer of the other call leg. If this feature is disabled, the Linksys IP phone performs an attended transfer operation by referring the other call leg to the current call leg while maintaining both call legs. To use this feature, select yes. Otherwise, select no.</p> <p>Defaults to no.</p>

MOH Server	<p>User ID or URL of the auto-answering streaming audio server. When only a user ID is specified, the current or outbound proxy is contacted. Music-on-hold is disabled if the MOH Server is not specified.</p> <p>Defaults to imusic when used with a SPA9000 IP PBX.</p>
Message Waiting	<p>Indicates whether the Message Waiting Indicator on the phone is lit. This parameter is toggled by a message from the SIP proxy to indicate if a message is waiting. You can manually modify it to clear or set the flag in the Ext 1-6 tab.</p> <p>Setting this value to Yes can activate stutter tone and VMWI signal. This parameter is stored in long-term memory and survives after reboot or power cycle.</p> <p>Defaults to No.</p>
Auth Page	<p>Specifies whether to authenticate the invite before auto answering a page.</p> <p>Defaults to No.</p>
Default Ring	<p>Type of ring heard. This corresponds to the Ring Tone on the Phone tab. Choose from No Ring, 1 through 10, User 1, or User 2.</p> <p>Defaults to 1.</p>
Auth Page Realm	<p>Identifies the Realm part of the Auth that is accepted when the Auth Page parameter is set to Yes. This parameter accepts alphanumeric characters.</p> <p>Defaults to blank.</p>
Conference Bridge URL	<p>This is the URL used to join into a conference call, generally in the form of the word “conference” or <u>“user@IPaddress:port”</u>.</p> <p>Defaults to blank.</p>
Auth Page Password	<p>Identifies the password used when the Auth Page parameter is set to Yes. This parameter accepts alphanumeric characters.</p> <p>Defaults to blank.</p>
Mailbox ID	<p>Identifies the voicemail box number/ID for the phone.</p> <p>Defaults to blank.</p>
Voice Mail Server	<p>Identifies the SpecVM server for the phone, generally the IP address and port number of the VM server.</p>
State Agent	<p>Reserved feature.</p>
CFWD Notify Serv	<p>Specifies whether to enable a SIP-B feature regarding the sending of a Notify to the phone when a call is forwarded elsewhere.</p> <p>Defaults to No.</p>
CFWD Notifier	<p>Typically, this field is configured with the SIP proxy information.</p>

Proxy and Registration



Note

Many of the following parameters do not apply to the SPA901.

Field	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.
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

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

Subscriber Information

**Note**

Many of the following parameters do not apply to the SPA901.

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

Audio Configuration

A codec resource is considered as allocated if it has been included in the SDP codec list of an active call, even though it eventually may not be the one chosen for the connection. So, if the G.729a codec is enabled and included in the codec list, that resource is tied up until the end of the call whether or not the

call actually uses G.729a. If the G.729a resource is already allocated and since only one G.729a resource is allowed per device, no other low-bit-rate codec may be allocated for subsequent calls; the only choices are G711a and G711u. On the other hand, two G.723.1/G.726 resources are available per device.

Therefore it is important to disable the use of G.729a in order to guarantee the support of two simultaneous uses of the G.723/G.726 codecs.

Field	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.
Silence Supp Enable	To enable silence suppression so that silent audio frames are not transmitted, select yes. Otherwise, select no. Defaults to no.
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.
Echo Canc Enable	Enable the use of the echo canceller. Defaults to yes.
Echo Canc Adapt Enable	Enable the echo canceller to adapt. Defaults to yes.
Echo Supp Enable	Enable the use of echo suppressor. If <Echo Canc Enable> is “no,” this parameter is ignored. Defaults to yes.
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.
FAX CED Detect Enable	To enable detection of the fax Caller-Entered Digits (CED) tone, select yes. Otherwise, select no. Defaults to yes.

Field	Description
FAX CNG Detect Enable	Defaults to yes.
FAX Passthru Codec	Codec to use for FAX passthru (G711u or G711a). Defaults to G711u.
FAX Codec Symmetric	Force unit to use symmetric codec during FAX passthru. Defaults to yes.
FAX Passthru Method	Choose None, NSE, or ReINVITE. Defaults to NSE.
FAX Process NSE	Defaults to yes.
FAX Disable ECAN	Set to yes to automatically disable Echo Cancellor when FAX tone is detected. 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.
Hook Flash Tx Method	Select the method to signal Hook Flash events: <ul style="list-style-type: none"> • None: do not signal hook flash events • AVT: use RFC2833 AVT (event=16) • INFO: use SIP INFO method with the signal line "signal = hf" in the message body. The MIME type for this message body is taken from the <Hook Flash MIME Type> parameter. Defaults to None.
Symmetric RTP	Enable symmetric RTP operation. If enabled, the SPA sends RTP packets to the source address of the last received valid inbound RTP packet. If disabled, the SPA sends RTP to the destination as indicated in the inbound SDP. Defaults to yes.
Release Unused Codec	This feature 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. Otherwise, select no. Defaults to yes.

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

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

Dial Plan



Note

The following parameters do not apply to the SPA901.

The default dial plan script for each line is as follows:
(*xx|[3469]11|0|00|[2-9]xxxxxx|1xxx[2-9]xxxxxx|xxxxxxxxxxxxx.). The syntax for a dial plan expression is as follows:

Table 5-1 Dial Plan Script Syntax

Dial Plan Entry	Functionality
*xx	Allow arbitrary 2 digit star code
[3469]11	Allow x11 sequences
0	Operator
00	International Operator
[2-9]xxxxxx	US local number
1xxx[2-9]xxxxxx	US 1 + 10-digit long distance number
xxxxxxxxxxxxx.	Everything else (Int'l long distance, FWD, ...)

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.

Field	Description
Dial Plan	<p>Dial plan script for this line.</p> <p>The dial plan syntax is expanded in the SPA to allow the designation of three parameters to be used with a specific gateway:</p> <ul style="list-style-type: none">• uid – the authentication user-id• pwd – the authentication password• nat – if this parameter is present, use NAT mapping <p>Each parameter is separated by a semi-colon (;).</p> <p>Furthermore, it recognizes gw0, gw1, ..., gw4 as the locally configured gateways, where gw0 represents the local PSTN gateway in the same unit.</p> <p>Example 1:</p> <pre>*1xxxxxxxxxx<:@fwdnat.pulver.com:5082;uid=jsmith;pwd=xyz</pre> <p>Example 2:</p> <pre>*1xxxxxxxxxx<:@fwd.pulver.com;nat;uid=jsmith;pwd=xyz</pre> <p>Example 3:</p> <pre>[39]11<:@gw0></pre>
Enable IP Dialing	<p>Enable or disable IP dialing.</p> <p>Defaults to no.</p>

User

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

- [Call Forward, page 5-48](#)
- [Speed Dial, page 5-48](#)
- [Supplementary Services, page 5-48](#)
- [Web Information Service Settings \(SPA962\), page 5-49](#)
- [Traffic Service Information Settings \(SPA962\), page 5-49](#)
- [Audio Volume, page 5-50](#)
- [Phone GUI Menu Color Settings \(SPA962 only\), page 5-50](#)

Call Forward

See [Vertical Service Activation Codes, page 5-22](#) for information on call forwarding parameters.

Speed Dial



Note

For the SPA901, speed dial is discussed in [Call Forward, page 5-48](#)

This section does not apply to the WIP310 wireless phone.

Field	Description
Speed Dial 2/3/4/5/6/7/8/9	Target phone number (or URL) assigned to speed dial 2, 3, 4, 5, 6, 7, 8, or 9. Defaults to blank.

Supplementary Services

The Linksys IP phone provides native support of a large set of enhanced or supplementary services. All of these services are optional. The parameters listed in the following table are used to enable or disable a specific supplementary service. A supplementary service should be disabled if a) the user has not subscribed for it, or b) the Service Provider intends to support similar service using other means than relying on the Linksys IP phone.

Most supplementary service parameters are listed in [Supplementary Services, page 5-33](#).

Web Information Service Settings (SPA962)

**Note**

The following parameters apply only to the SPA962.

Field	Description
RSS Feed URL [1-5]	URLs for Local and World news, Finance, Sports, and Politics. For more information, including default URLs, see “RSS Newsfeeds (SPA962)” section on page 3-20 .
Feed Data Update Interval(m)	Interval at which the news feed is updated. Defaults to 15 seconds.
Scrolling Timer(s)	Scrolling timer. Defaults to 0.8 seconds.

Traffic Service Information Settings (SPA962)

**Note**

The following parameters apply only to the SPA962.

For more information on using the Web Server, see [“Configuring the Web Service” section on page 3-14](#),

Field	Description
State	Enter your local state. Defaults to CA.
City)	Enter your local city. Defaults to San Jose.
Street	Enter your local street. Defaults to Cisco.
Zip Code	Enter your local zip code. Defaults to 95134.

Audio Volume



Note

Many of the following parameters do not apply to the SPA901.

Field	Description
Ringer Volume	Sets the default volume for the ringer.
Speaker Volume	Sets the default volume for the 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:

- [General, page 5-51](#)
- [Unit 1, page 5-52](#)
- [Unit 2, page 5-53](#)

General

Field	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.
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.
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.
Call Pickup Code	The star code used for picking up a ringing call. Defaults to *98.

Unit 1

Field	Description
Unit 1 Key 1-32	<p>This field lets you enter a string that defines 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 composed of the following keywords, followed by an equal sign (=) and separated by semicolons (;):</p> <ul style="list-style-type: none"> fnc—defines which of the following functions are enabled for the specified key (separate more than one function with a plus sign (+)): <ul style="list-style-type: none"> blf—busy lamp field function used for monitoring line activity sd—speed dial function cp—call pickup (if supported by the SIP proxy server) sub—Use this keyword to identify the phones to be monitored. usr —Use this keyword to identify the specific users or extensions to be monitored. nme—(optional) Use this field with the SPA9000 to identify any alias that has been assigned to the extension in the Linksys IP phone configuration. vid—(optional) 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. <div>  <p>Note You only need to use the vid= keyword in the first entry assigned to each SPA962 extension. Subsequent keys will use the same extension.</p> </div> <p>The following is the syntax for a Broadsoft server:</p> <p>fnc=enabled_functions;sub=blf_list_uri@broadsoft;usr=dn@broadsoft</p> <p>Replace the variables (in italics), as follows:</p> <ul style="list-style-type: none"> <i>enabled_functions</i> with cp for call pickup, sd for speed dial, and blf for busy lamp field; separate multiple functions in any order with a plus sign (+) <i>blf_list_uri</i> with the URI assigned to the BLF monitoring list on the Broadsoft server. <i>broadsoft</i> with the IP address or hostname of the Broadsoft server. <i>dn</i> with the number associated with the user account on the Monitored Users list. <p>For example:</p> <p>fnc=sd+blf;sub=marketing@192.168.100.1;usr=reception@192.168.100.1</p> <div> <p>Note The nme keyword is not used because the Broadsoft server automatically uses the assigned user account name.</p> </div>

Unit 2

Field	Description
Unit 2 Key 1-32	See the description for Unit 1.

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:

Field	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.

A

Acronyms

A/D	Analog To Digital Converter
ANC	Anonymous Call
B2BUA	Back to Back User Agent
BLF	Busy Lamp Field
Bool	Boolean Values. Specified as “yes” and “no”, or “1” and “0” in the profile
CA	Certificate Authority
CAS	CPE Alert Signal
CDR	Call Detail Record
CID	Caller ID
CIDCW	Call Waiting Caller ID
CNG	Comfort Noise Generation
CPC	Calling Party Control
CPE	Customer Premises Equipment
CWCID	Call Waiting Caller ID
CWT	Call Waiting Tone
D/A	Digital to Analog Converter
dB	decibel
dBm	dB with respect to 1 milliwatt
DHCP	Dynamic Host Configuration Protocol
DNS	Domain Name Server
DRAM	Dynamic Random Access Memory
DSL	Digital Subscriber Loop
DSP	Digital Signal Processor
DTAS	Data Terminal Alert Signal (same as CAS)
DTMF	Dual Tone Multiple Frequency
FQDN	Fully Qualified Domain Name
FSK	Frequency Shift Keying
FXS	Foreign eXchange Station
GW	Gateway
ITU	International Telecommunication Union

HTML	Hypertext Markup Language
HTTP	Hypertext Transfer Protocol
HTTPS	HTTP over SSL
ICMP	Internet Control Message Protocol
IGMP	Internet Group Management Protocol
ILEC	Incumbent Local Exchange Carrier
IP	Internet Protocol
ISP	Internet Service Provider
ITSP	IP Telephony Service Provider
IVR	Interactive Voice Response
LAN	Local Area Network
LBR	Low Bit Rate
LBRC	Low Bit Rate Codec
MC	Mini-Certificate
MGCP	Media Gateway Control Protocol
MOH	Music On Hold
MOS	Mean Opinion Score (1-5, the higher the better)
ms	Millisecond
MSA	Music Source Adaptor
MWI	Message Waiting Indication
OSI	Open Switching Interval
PCB	Printed Circuit Board
PR	Polarity Reversal
PS	Provisioning Server
PSQM	Perceptual Speech Quality Measurement (1-5, the lower the better)
PSTN	Public Switched Telephone Network
NAT	Network Address Translation
OOB	Out-of-band
REQT	(SIP) Request Message
RESP	(SIP) Response Message
RSC	(SIP) Response Status Code, such as 404, 302, 600
RTP	Real Time Protocol
RTT	Round Trip Time
SAS	Streaming Audio Server
SDP	Session Description Protocol
SDRAM	Synchronous DRAM
sec	seconds

SIP	Session Initiation Protocol
SLA	Shared line appearance
SLIC	Subscriber Line Interface Circuit
SP	Service Provider
SPA	Linksys Phone Adaptor
SRTP	Secure Real-time Transport Protocol
SSL	Secure Socket Layer
TFTP	Trivial File Transfer Protocol
TCP	Transmission Control Protocol
UA	User Agent
uC	Micro-controller
UDP	User Datagram Protocol
URL	Uniform Resource Locator
VM	Voicemail
VMWI	Visual Message Waiting Indication/Indicator
VQ	Voice Quality
WAN	Wide Area Network
XML	Extensible Markup Language

B

Glossary

ACD (Automatic Call Distribution)—A switching system designed to allocate incoming calls to certain positions or agents in the order received and to hold calls not ready to be handled (often with a recorded announcement).

Area code—A 3-digit code used in North America to identify a specific geographic telephone location. The first digit can be any number between 2 and 9. The second and third digits can be any number.

Billing increment—The division by which the call is rounded. In the field it is common to see full-minute billing on the local invoice while 6-second rounding is the choice of most long-distance providers that bill their customers directly.

Blocked calls—Caused by an insufficient network facility that does not have enough lines to allow calls to reach a given destination. May also pertain to a call from an originating number that is blocked by the receiving telephone number.

Bundled service—Offering various services as a complete package.

Call completion—The point at which a dialed number is answered.

Call termination—The point at which a call is disconnected.

CDR (Call Detail Records)—A software program attached to a VoIP/telephone system that records information about the telephone number's activity.

Carrier's carrier—Companies that build fiber optic and microwave networks primarily selling to resellers and carriers. Their main focus is on the wholesale and not the retail market.

Casual access—When customers choose not to use their primary carriers to process the long-distance call being made. The customer dials the carrier's 101XXXX number.

CO (Central Office)—Switching center for the local exchange carrier.

Centrex—This service is offered by the LEC to the end user. The feature-rich Centrex line offers the same features and benefits as a PBX to a customer without the capital investment or maintenance charges. The LEC charges a monthly fee to the customer, who must agree to sign a term agreement.

Circuits—The communication path(s) that carry calls between two points on a network.

Customer Premise Equipment—The only part of the telecommunications system that the customer comes into direct contact with. Example of such pieces of equipment are telephones, key systems, PBXs, voice-mail systems, and call accounting systems as well as wiring telephone jacks. The standard for this equipment is set by the FCC, and the equipment is supplied by an interconnect company.

Dedicated access—Customers have direct access to the long-distance provider via a special circuit (T1 or private lines). The circuit is hardwired from the customer site to the POP and does not pass through the LEC switch. The dial tone is provided from the long-distance carrier.

Dedicated Access Line (DAL)—Provided by the local exchange carrier. An access line from the customer's telephone equipment directly to the long-distance company's switch or POP.

Demarcation point—This is where the LEC ownership and responsibility (wiring, equipment) ends and the customer's responsibilities begin.

Direct Inward Dialing (DID)—Allows an incoming call to bypass the attendant and ring directly to an extension. Available on most PBX systems and a feature of Centrex service.

Dual Tone Multifrequency (DTMF)—Better known as the push button keypad. DTMF replaces dial pulses with electronically produced tones for network signaling.

Enhanced service—Services that are provided in addition to basic long distance and accessed by way of a touchtone phone through a series of menus.

Exchange code (NXX)—The first three digits of a phone number.

Flat-rate pricing—The customer is charged one rate (sometimes two rates, one for peak and one for off-peak) rather than a mileage-sensitive program rate.

IXC (Interexchange Carrier)—A long-distance provider that maintains its own switching equipment.

IVR (Interactive Voice Response)—Provides a mechanism for information to be stored and retrieved using voice and a touchtone telephone.

Local loop—The local telephone company provides the transmission facility from the customer to the telephone company's office, which is engineered to carry voice and/or data.

North American Numbering Plan (NANP)—How telephone numbers are identified in North America. The telephone number can be identified based on their three separate components: (NPA), (NXX), and (XXXX).

PIN (Personal Identification Code)—A customer calling/billing code for prepaid and pay-as-you-go calling cards.

Private Branch Exchange—Advanced phone system commonly used by the medium to larger customer. It allows the customer to perform a variety of in-house routing (inside calling). The dial tone that is heard when the customer picks up the phone is an internal dial tone.

SS7 (900 Series phone Signaling Number 7)—Technology used by large carriers to increase the reliability and speed of transmission between switches.

Switch (switching)—Equipment that connects and routes calls and provides other interim functions such as least cost routing, IVR, and voicemail. It performs the "traffic cop" function of telecommunications via automated management decisions.

Touchtone (DTMF)—The tone recognized by a push button (touchtone) telephone.

Unified messaging—Platform that lets users send, receive, and manage all e-mail, voice, and fax messages from any telephone, PC, or information device.

Voicemail—A system that allows storage and retrieval of voice messages through voice-mail boxes.

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