



## Linksys 900 Series IP Phone Administrator Guide

Document Version 3.0



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*Linksys 900 Series IP Phone Administrator Guide*

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## Acronyms

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## Preface

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This guide describes administration and use of the Linksys SPA900 Series IP phones. It contains the following sections:

- [Document Audience, page xi](#)
- [Linksys 900 Series IP Telephones, page xi](#)
- [How This Document is Organized, page xii](#)
- [Document Conventions, page xii](#)
- [Related Documentation, page xiii](#)
- [Technical Support, page xiii](#)

## Document Audience

This document is written for the following audience:

- Service providers offering services using LVS products
- VARs and resellers who need LVS configuration references
- System administrators or anyone who performs LVS installation and administration



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**Note** This guide does not provide the configuration information required by specific service providers. Please consult with the service provider for specific service parameters.

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## Linksys 900 Series IP Telephones

The following summarizes the ports and features provided by the Linksys 900 Series IP phones described in this document.

- SPA901—One line, small, affordable, no display.
- SPA921—One-line business phone.
- SPA922—One-line business phone with Power over Ethernet (PoE) support and an extra Ethernet port for connecting another device to the LAN.

- SPA941—Default two lines, upgradeable to four lines.
- SPA942—Default is two lines, upgradeable to four lines. Power over Ethernet (PoE) support and an extra Ethernet port for connecting another device to the LAN.
- SPA962—Six lines, hi-res 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.

## How This Document is Organized

This document is divided into the following chapters and appendices.

Chapter	Contents
<a href="#">Chapter 1, “Introducing Linksys 900 Series IP Phones”</a>	This chapter introduces the Linksys 900 Series IP phones.
<a href="#">Chapter 2, “Getting Started”</a>	This chapter describes how to use the different administration and configuration tools provided for managing a Linksys 900 Series IP phone.
<a href="#">Chapter 3, “Managing Linksys 900 Series IP Phones”</a>	This chapter describes how to configure and monitor a Linksys 900 Series IP phone.
<a href="#">Chapter 5, “SPA900 Series Phone Field Reference”</a>	This chapter lists the function and usage for each field or parameter on the Linksys 900 Series IP phone administration web server pages.
<a href="#">Appendix A, “Acronyms”</a>	This appendix provides the expansion of acronyms used in this document.
<a href="#">Appendix B, “Glossary”</a>	This appendix defines the terms used in this document.

## Document Conventions

The following are the typographic conventions used in this document.

Typographic Element	Meaning
<b>Boldface</b>	Indicates an option on a menu or a literal value to be entered in a field.
<parameter>	Angle brackets (<>) are used to identify parameters that appear on the configuration pages of the 900 Series phone administration web server. The index at the end of this document contains an alphabetical listing of each parameter, hyperlinked to the appropriate table in <a href="#">Chapter 5, “SPA900 Series Phone Field Reference”</a>
<i>Italic</i>	Indicates a variable that should be replaced with a literal value.
Monospaced Font	Indicates code samples or system output.

## Related Documentation

The following documentation provides additional information about features and functionality of Linksys 900 Series IP phones:

- *AA Quick Guide*
- *IVR Quick Guide*
- *SPA Provisioning Guide*

The following documentation describes how to use other Linksys Voice System products:

- *SPA9000 Administrator Guide*
- *LVS CTI Integration Guide*
- *LVS Integration with ITSP Hosted Voicemail Guide*
- *SPA900 Series IP Phones Administrator 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 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



## Introducing Linksys 900 Series IP Phones

This guide describes the administration and use of Linksys analog telephone adapters (ATAs). This chapter introduces the functionality of the Linksys 900 Series IP phones and includes the following sections:

- [Overview, page 1-2](#)
- [Feature Descriptions, page 1-8](#)
- [Technology Background, page 1-12](#)
- [Where to Go From Here, page 1-16](#)

### 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 SPA900 Series IP Phones*

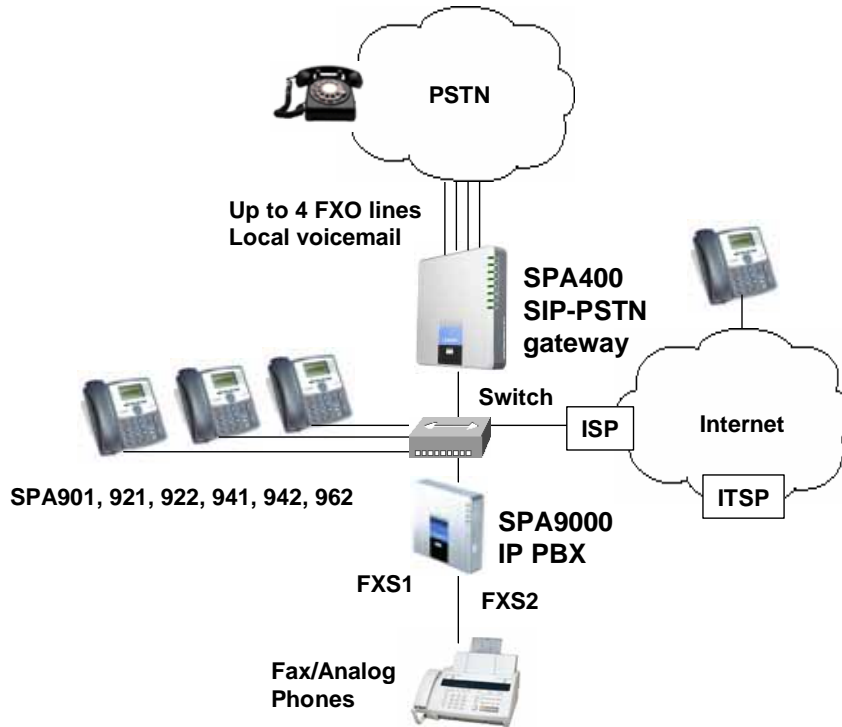
Product Name	RJ-45	Ethernet (LAN) Voice Lines	Additional Features/Notes
SPA901	One (1)	One (1)	Small, affordable, no display
SPA921	One (1)	One (1)	One-line business phone
SPA922	Two (2)	One (1)	Power over Ethernet (PoE) support
SPA941	One (1)	Four (4)	Default is 2-lines active, upgradeable
SPA942	Two (2)	Four (4)	Default is 2-lines active, upgradeable. Power over Ethernet (PoE) support
SPA962	Two (2)	Six (6)	Six lines, hi-res color display



**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, including the SPA3102, which acts as a SIP-PSTN gateway. As shown, the RTP400 and WRTP54G provide QoS-enabled IP routers in addition to two ports for connecting analog telephone devices.

Figure 1-1 Linksys SPA900 Series IP Phones in a VoIP Network



## SPA900 Series Features

The following telephony features are provided by the different models of the SPA900 Series IP phones:

- Shared Line Appearance \*\*
  - SPA901: Two Call Appearances Accessed Via Flash Key or Hook-Flash
  - 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](http://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

\*\* Feature requires support by SIP server

## SPA901 Features

The SPA901 provides the following features that are not needed with the SPA900 Series IP phones that provide an LCD display:

- Built-in Interactive Voice Response (IVR) system to check status and change configuration

SPA900 Series Features

- Ringer and Handset Volume Controls
- Handset Input Gain Adjustment

## SPA92x, SPA94x, and SPA962 Features

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.

*Table 1-2 Speech Quality Metrics*

Algorithm	Bandwidth	Complexity	MOS Score
G.711	64 kbps	Very low	4.5
G.726	16, 24, 32, 40 kbps	Low	4.1 (32 kbps)
G.729a	8 kbps	Low–medium	4
G.729	8 kbps	Medium	4
G.723.1	6.3, 5.3 kbps	High	3.8

**Note**

SPA900 Series IP phones support all the above voice coding algorithms.

The following factors contribute to voice quality:

- Audio compression algorithm—Speech signals are sampled, quantized, and compressed before they are packetized and transmitted to the other end. For IP Telephony, speech signals are usually sampled at 8000 samples per second with 12–16 bits per sample. The compression algorithm plays a large role in determining the voice quality of the reconstructed speech signal at the other end. SPA900 Series IP phones support the most popular audio compression algorithms for IP Telephony: G.711 a-law and  $\mu$ -law, G.726, G.729a, and G.723.1.

The encoder and decoder pair in a compression algorithm is known as a codec. The compression ratio of a codec is expressed in terms of the bit rate of the compressed speech. The lower the bit rate, the smaller the bandwidth required to transmit the audio packets. Although voice quality is usually lower with a lower bit rate, it is usually higher as the complexity of the codec gets higher at the same bit rate.

- Silence suppression—SPA900 Series IP phones apply silence suppression so that silence packets are not sent to the other end to conserve more transmission bandwidth. Instead, a noise level measurement can be sent periodically during silence suppressed intervals so that the other end can generate artificial comfort noise that mimics the noise at the other end (using a CNG or comfort noise generator).
- Packet loss—Audio packets are transported by UDP, which does not guarantee the delivery of the packets. Packets may be lost or contain errors that can lead to audio sample drop-outs and distortions and lower the perceived voice quality. SPA900 Series IP phones apply an error concealment algorithm to alleviate the effect of packet loss.
- Network jitter—The IP network can induce varying delay of received packets. The RTP receiver in SPA900 Series IP phones keeps a reserve of samples to absorb the network jitter, instead of playing out all the samples as soon as they arrive. This reserve is known as a jitter buffer. The bigger the jitter buffer, the more jitter it can absorb, but this also introduces bigger delay. Therefore, the jitter buffer size should be kept to a relatively small size whenever possible. If jitter buffer size is too small, many late packets may be considered as lost and thus lowers the voice quality. SPA900 Series IP phones dynamically adjust the size of the jitter buffer according to the network conditions that exist during a call.
- Echo—Impedance mismatch between the telephone and the IP Telephony gateway phone port can lead to near-end echo. SPA900 Series IP phones have a near-end echo canceller with at least 8 ms tail length to compensate for impedance match. SPA900 Series IP phones implement an echo suppressor with comfort noise generator (CNG) so that any residual echo is not noticeable.
- Hardware noise—Certain levels of noise can be coupled into the conversational audio signals because of the hardware design. The source can be ambient noise or 60 Hz noise from the power adaptor. The SPA900 Series hardware design minimizes noise coupling.
- End-to-end delay—End-to-end delay does not affect voice quality directly but is an important factor in determining whether subscribers can interact normally in a conversation taking place over an IP network. A reasonable delay figure should be about 50–100 ms. End-to-end delay larger than 300 ms is unacceptable to most callers. SPA900 Series IP phones support end-to-end delays well within acceptable thresholds.

## Feature Descriptions

SPA900 Series 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 SPA900 Series IP phones.

- [SIP Proxy Redundancy, page 1-8](#)
- [Supported Codecs, page 1-8](#)
- [Other Features, page 1-9](#)

### 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. SPA900 Series 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 static list of SIP proxy servers in the SPA900 Series IP phone configuration profile, where the list is arranged in order of priority. The SPA900 Series 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 SPA900 Series IP phone. One solution to this situation is through the use of DNS SRV records. SPA900 Series IP phones can be instructed to contact a SIP proxy server in a domain named in SIP messages. The SPA900 Series 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 SPA900 Series IP phone tries to contact the list of hosts in the order of their stated priority.

If the SPA900 Series 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 SPA900 Series IP phone to “match” a codec name with the far-end device/gateway codec name. SPA900 Series 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 SPA900 Series IP phones.

*Table 1-3 Codecs Supported by Linksys SPA900 Series IP Phones*

Codec (Voice Compression Algorithm)	Description
G.711 (A-law and $\mu$ -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	SPA900 Series 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, SPA900 Series IP phones can support dynamic payloads for G.726.

## Other Features

Table 1-4 summarizes the features provided by SPA900 Series IP Phones.

Table 1-4 Linksys ATA Features

Feature	Description
Music On Hold	On a connected call, SPA900 Series 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 the user 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 *) SPA900 Series 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	SPA900 Series 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 the user to hear the call progress tones (such as ringing) that are generated from the far-end network.
Jitter Buffer—Dynamic (Adaptive)	SPA900 Series 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.  SPA900 Series IP phones have a Network Jitter Level control setting for each line of service. The jitter level decides how aggressively SPA900 Series 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.

*Table 1-4 Linksys ATA 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	SPA900 Series IP phones have three configurable interdigit timers: <ul style="list-style-type: none"> <li>• Initial timeout (T)—Handset off hook; no digit pressed yet.</li> <li>• Long timeout (L)—One or more digits pressed, more digits needed to reach a valid number (as per the dial plan).</li> <li>• Short timeout (S)—Current dialed number is valid, but more digits would also lead to a valid number.</li> </ul>
Report Generation and Event Logging	SPA900 Series 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 digested authentication, for instance. The information may be organized as an XML page or HTML page.
Syslog and Debug Server Records	SPA900 Series IP phones support detailed logging of all activities for further debugging. The debug information may be sent to a configured Syslog server. SPA900 Series IP phones provide configuration settings that determine the type of activity/events that should be logged, as for instance, a debug level setting.
Dynamic Payload	When no static payload value is assigned per RFC 1890, SPA900 Series IP phones can support dynamic payloads for G.726.
Call Statistics and Reporting	The statistics collected by SPA900 Series 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

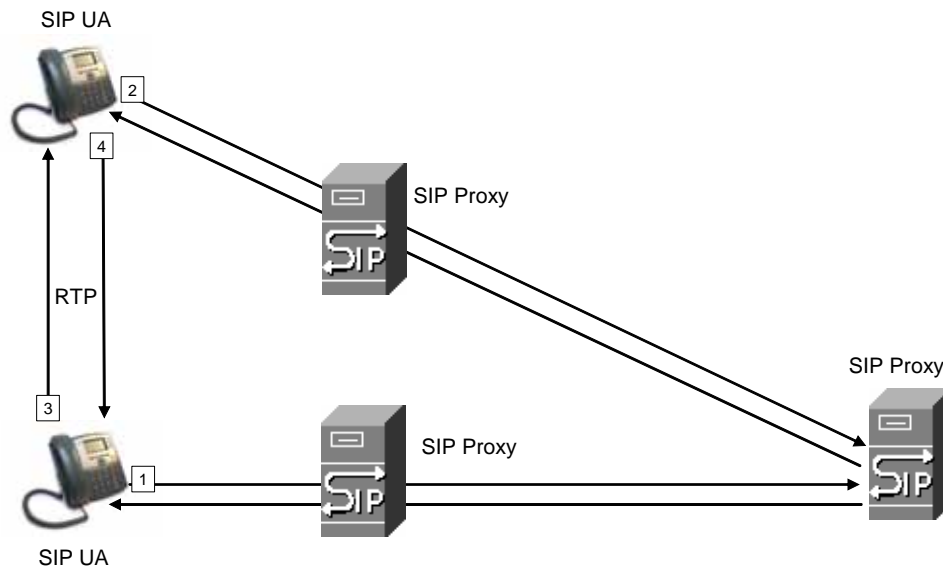
This section provides background information about the technology and protocols used by the ATA. It includes the following topics:

- [Session Initiation Protocol, page 1-12](#)
- [Using 900 Series Phones with a Firewall or Router, page 1-12](#)
- [Using 900 Series Phones with a Firewall or Router, page 1-12](#)

## 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 900 Series Phones with a Firewall or Router

When using a 900 Series 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 1-13](#)
- [NAT Types, page 1-14](#)
- [Simple Traversal of UDP Through NAT, page 1-14](#)
- [SIP-NAT Interoperation, page 1-15](#)

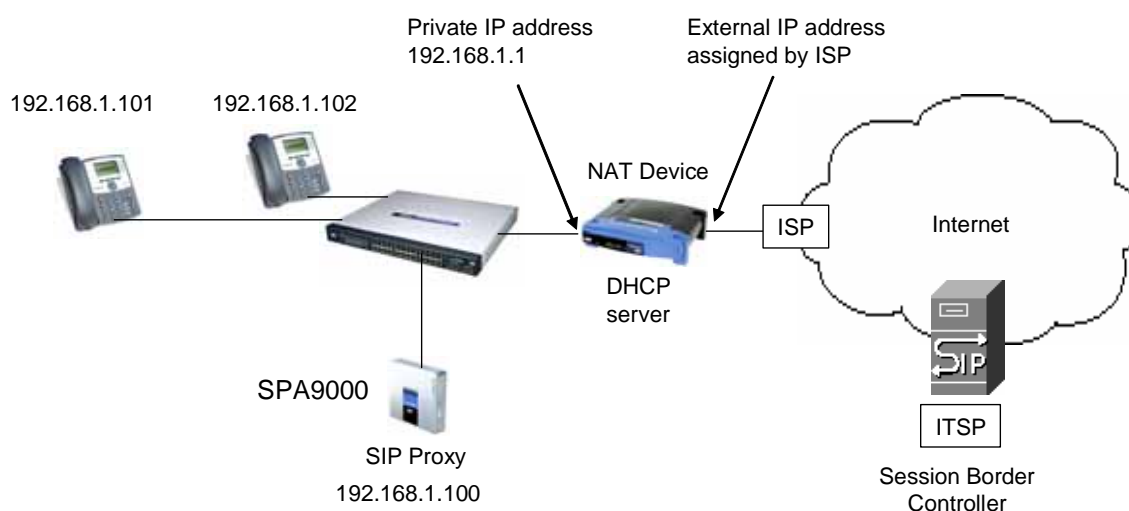
## NAT Overview

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 900 Series phone. If this is not available, you need to discuss with the ITSP how to use the NAT Support Parameters provided by the 900 Series phone, such as <Outbound Proxy> and <STUN Server Enable>.

A typical application of a NAT is to allow all the devices in a subscriber home 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

Simple Traversal of UDP through NATs (STUN) is a protocol defined by RFC 3489, that allows a client behind a NAT device to find out its public address, the type of NAT it is behind, 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:

---

**Step 1** Enable debugging on the 900 Series 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 SPA900 Series 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.

**Step 2** To determine the type of NAT your router is using set <STUN Test Enable> to **yes**.

**Step 3** View the syslog 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 **900 Series phone** system are embedded in the SIP messages sent by the device. If the **900 Series phone** system is sitting 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

If the ITSP offers an outbound NAT-Aware proxy, this discovers the public IP address from the remote endpoint and eliminates the need to modify the SIP message from the UAC.

---

The **900 Series 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 **900 Series 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 **900 Series 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 **900 Series phone** system continues to send periodic keep-alive packets through the mapping to extend its validity as necessary.

## Where to Go From Here

To do this ...	Refer to
Use the different administration and configuration tools provided for managing Linksys 900 Series IP phone.	<a href="#">Chapter 2, “Getting Started”</a>
Configure and monitor a Linksys 900 Series IP phone.	<a href="#">Chapter 3, “Managing Linksys 900 Series IP Phones”</a>
Refer to the function and usage for each field or parameter on the Linksys 900 Series IP phone administration web server pages.	<a href="#">Chapter 5, “SPA900 Series Phone Field Reference”</a>
Find the expansion of an acronym used in this document.	<a href="#">Appendix A, “Acronyms”</a>
Find the definition of a term used in this document.	<a href="#">Appendix B, “Glossary”</a>

The following documentation provides additional documentation for Linksys SPA900 Series IP phones:

- *IVR Quick Guide*
- *SPA Provisioning Guide*

The following documentation describes how to use other Linksys Voice System products:

- *SPA9000 Administrator Guide*
- *LVS CTI Integration Guide*
- *LVS Integration with ITSP Hosted Voicemail Guide*
- *Linksys Voice over IP Product Guide: SIP CPE for Massive Scale Deployment*
- *SPA 2.0 Analog Telephone Adapter Administrator Guide*

## Getting Started

This chapter describes the tools and utilities available for administering Linksys SPA900 Series phones. It includes the following sections:

- [Linksys 900 Series IP Phones, page 2-1](#)
- [Establishing Connectivity, page 2-9](#)
- [Using the Administration Web Server, page 2-11](#)
- [Web Interface URLs, page 2-13](#)
- [Provisioning, page 2-14](#)
- [Using the Interactive Voice Response Interface, page 2-16](#)



Note

If the SPA900 Series 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 SPA900 Series 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](#)
- [SPA941, page 2-5](#)
- [SPA942, page 2-7](#)
- [SPA962, page 2-7](#)

## 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 SPA921, SPA922, 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
- LED Indicates Line State – Active, Idle, On-Hold, Unregistered
- Line LED Configurable to 13 Different States (On/Off, Color, Flash)
- 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
- LED Test Function
- Two Ethernet LAN ports (100BaseT RJ-45)
- 802.3af Compliant Power over Ethernet (PoE)
- 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*



## SPA941

The SPA941 provides two lines by default and is upgradeable to four lines (see [Figure 2-4](#)).

Figure 2-4 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. The SPA942 has two ports. Use the other port to connect to a PC or other LAN device.
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 2-5](#)). The PA100 power supply must be ordered separately if you are not using a PoE switch. See the table for SPA941.

*Figure 2-5 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 2-6](#)). The PA100 power supply must be ordered separately if you are not using a PoE switch.

Figure 2-6 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	Reserved for future functionality and not for use as a telephone line port.

# Establishing Connectivity

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

- [Bandwidth Requirements, page 2-9](#)
- [Installing the SPA900 Series IP Phone, page 2-10](#)

## Bandwidth Requirements

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

Linksys recommends using the 900 Series 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 900 Series 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. 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 is reserved for future functionality of the SPA962. It is not for use as a telephone line port.



---

**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](http://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 3-32.

## Using the Administration Web Server

This section describes how to use the administration web server to configure the 900 Series phone. It includes the following topics:

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

## Connecting to the Administration Web Server

To access the 900 Series phone administration web server, perform the following steps.

**Step 1** Launch a web browser on a computer connected to the 900 Series 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 900 Series 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 900 Series phone may reboot.

## Administrator Account Privileges

The 900 Series 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 900 Series 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 900 Series 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 900 Series 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 is 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 900 Series 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 900 Series phone.



Note

---

The 900 Series 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 900 Series phone. This section includes the following topics:

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

For detailed information about provisioning the 900 Series phone, refer to the *Linksys SPA Provisioning Guide*.

## Provisioning Capabilities

The 900 Series phone provides for secure provisioning and remote upgrade. Provisioning is achieved through configuration profiles transferred to the device via TFTP, HTTP, or HTTPS.

The 900 Series 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 900 Series phone accepts profiles in XML format, or alternatively in a proprietary binary format, which is generated by a profile compiler tool available from Linksys. The 900 Series phone supports up to 256-bit symmetric key encryption of profiles. For the initial transfer of the profile encryption key (initial provisioning stage), the 900 Series 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 900 Series 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 900 Series phone via a remote profile resync.

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

## Configuration Profile

The 900 Series 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 900 Series 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 900 Series 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 LCD Display” section on page 3-1) or the administration web server (see the “Using the Administration Web Server” section on page 2-11).

This section includes the following topics:

- [Using the IVR Menu, 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/>

## Using the IVR Menu

To use the IVR menu, complete the following steps.

---

**Step 1** Connect an analog telephone to the Phone 1 or Phone 2 port of the 900 Series phone.



---

**Note** You can only access the IVR menu through an analog telephone, not an 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 900 Series 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 900 Series 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.
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

Table 2-2 IVR Options (continued)

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
Manual Reboot of Unit	732668	None	After you hear “Option Successful,” hang up. Unit reboots automatically.

**Table 2-2 IVR Options (continued)**

<p>User Factory Reset of Unit</p> <p><b>WARNING:</b></p> <p>ALL “User-Changeable” NON-DEFAULT SETTINGS WILL BE LOST!</p> <p>This might include network and service provider data.</p>	<p>877778</p>	<p>Enter 1 to confirm Enter *(star) to cancel operation</p>	<p>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.</p>
<p>Factory Reset of Unit</p> <p><b>WARNING:</b></p> <p>ALL NON-DEFAULT SETTINGS WILL BE LOST!</p> <p>This includes network and service provider data.</p>	<p>73738</p>	<p>Enter 1 to confirm Enter * (star) to cancel operation</p>	<p>SPA prompts for confirmation. After confirming, you hear “Option Successful.” Hang up. Unit reboots and all configuration parameters are reset to factory default values.</p>



**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 900 Series 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 \* \* \* \*.
-



## Managing Linksys 900 Series IP Phones

This chapter describes how to configure Linksys SPA900 Series IP phones and includes the following sections:

- [Using the LCD Display, page 3-1](#)
- [Localization, page 3-6](#)
- [Changing the Display Background \(SPA962\), page 3-8](#)
- [Call Appearances and Extensions, page 3-9](#)
- [Line Key LEDs, page 3-10](#)
- [Using Call Features, page 3-11](#)
- [Configuring a Dial Plan, page 3-22](#)
- [System Administration, page 3-27](#)
- [Troubleshooting FAQ, page 3-32](#)

### Using the LCD Display

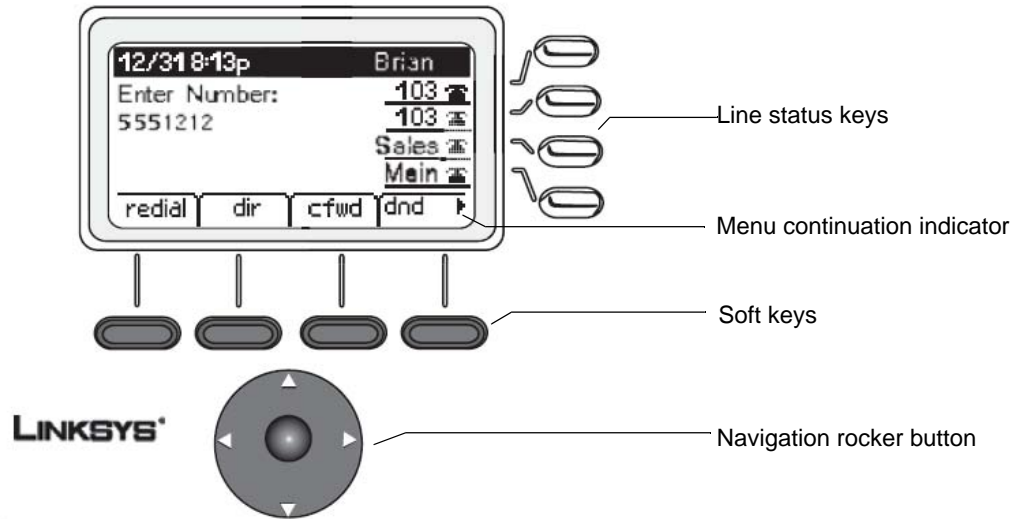
All SPA900 Series models provide an LCD display except for the SPA901, which you configure using the Interactive Voice Response (IVR) interface, described in the [“Using the Administration Web Server”](#) section on page 2-11. This section describes the basic functions of the LCD display and includes the following topics:

- [LCD Display Controls, page 3-2](#)
- [Entering and Saving Settings, page 3-5](#)
- [Call Appearances and Extensions, page 3-9](#)
- [Using Soft Keys, page 3-3](#)

## LCD Display Controls

The LCD display on the SPA900 Series IP phones provides 18 menu options for operating, configuring and monitoring the IP phone. For information about each of these options, refer to [Chapter 4, “LCD Command Reference Guide.”](#) [Figure 3-1](#) illustrates the main controls on the LCD display.

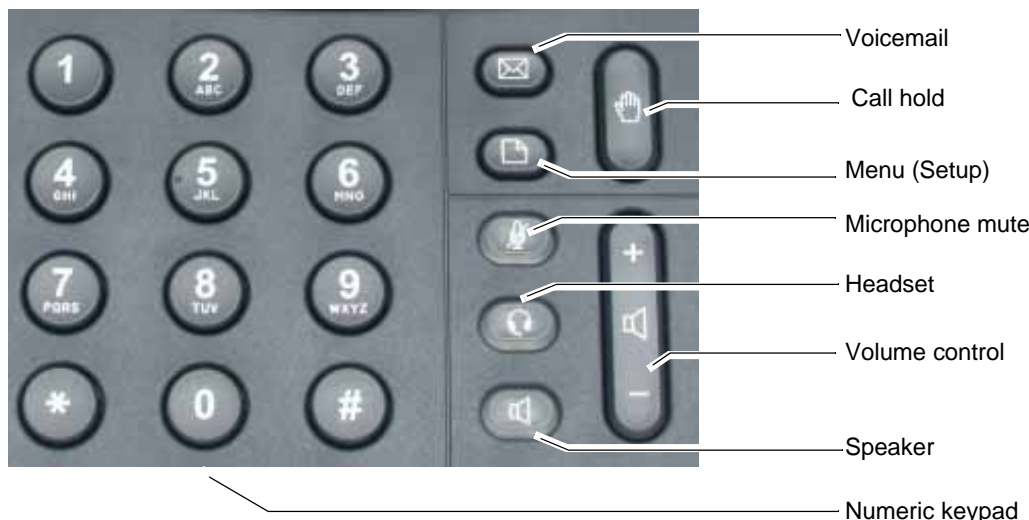
*Figure 3-1 SPA900 Series LCD Controls*



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.

To access an option, press the **Menu** button and press the **Navigation** button to select the option. Then push the left soft key beneath the **Select** option to select the option. [Figure 3-2](#) summarizes the function of the other buttons on the telephone keypad.

Figure 3-2 Telephone Keypad



To use the shortcut to an option, press the **Menu (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.

Table 3-1 General LCD Display Soft Key Labels

Label	Description
cfwd	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	Accesses the phone directory.
dnd	(Do not disturb) Sends incoming calls to voicemail, or gives a busy signal if voicemail is not available.
grPick	Picks up a call from an extension outside the group.
lcr	(Last call return) Dials the last call received.
park	Parks a call at a designated call park number. The call is active until it is unparked, or the caller disconnects (must be enabled before use).

*Table 3-1 General LCD Display Soft Key Labels*

Label	Description
pickup	Picks up a call from another phone in the group.
redial	Displays a list of recently called numbers.
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 Menu (Setup) key.

*Table 3-2 Menu (Setup) Soft Keys*

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	Adds an entry.
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	Cancels change before pressing <b>save</b> or <b>ok</b> .
change	Changes the ring tone feature.
clear	Clears the entire entry.
copy	In a directory, use the <b>Navigation</b> button to highlight an entry, then press to copy an entry.
delChr	Deletes the last character entered.
delete	Deletes an entry.
dial	Places a call to the number highlighted on the LCD screen.
edit	Edits part of an entry.
ok	Confirms an operation.
paste	Use the <b>Navigation</b> button to highlight an entry in the personal directory and press paste to place a copied entry
play	Plays ring tones.
save	Saves changes.
select	Chooses an item to change or view.
y/n	Toggles between yes and no settings.

## Entering and Saving Settings

The 900 Series phone GUI lets the user enter alphanumeric and numeric inputs. The following data types may be entered 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 two or more numeric string, alphanumeric string, and IP address types can apply to an entry, the phone shows a soft key for switching input modes among the applicable types. The soft keys changes when pressed to allow the user to scroll through the available modes, such as **num**, **alpha**, or **IP**. The cursor also changes according to [Table 3-3](#), to indicate the input mode being used:

*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 the user presses each digit key. The user scrolls through the choices by pressing the same key multiple times. To accept a symbol as input, the user does 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 900 Series phone automatically scrolls the input to the left so that the user can continue to enter input. On the other hand, the 900 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 900 Series phone displays (y/n); pressing the soft key toggles the current input between yes and no.

When the 900 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 the user press **save** or **ok** before exiting the edit screen for that setting.

However, there are a few exceptions where the user 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**



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>

The dictionaries are found in the FW and Tools section, LVS area. Currently dictionaries are available for the following languages:

- English
- Spanish
- German,
- Dutch
- Italian
- French
- Portuguese
- Danish

Changing the Display Background (SPA962)

- Swedish
- Czech
- Slovak

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

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

## Changing the Display Background (SPA962)

This section applies only to the SPA962, which has a color LCD display with a customized background. You use any 320 x 240 pixel .BMP image for the background image that appears on the LCD display.

You can include the definition of the required parameters in the configuration profile for the SPA962, or use the administration web server. The parameters are as follows:

- Picture download URL—The URL identifying the location of the .BMP file to be used as the background image, including the TFTP server name (or IP address), the directory, and the filename. For example:

```
tftp://myserver.mydomain.com/images/962back.bmp
```

The background image should be 320X240 pixels and must be in .BMP format.

- Current Background Picture—Select **Default** to use the default image; select **Downloaded** to use a customized background image.

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

- Step 1** Copy the 320X240 pixel .BMP image that 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 SPA962.
- Step 3** Click the **Phone** tab and scroll down to the **Background Picture** section at the bottom of the page (see [Figure 3-4](#)).

*Figure 3-4 Background Picture Parameters*



Background Picture  
Picture download URL:   
Current Background Picture:

- Step 4** Select **Downloaded** from the Current Background Picture pull-down selection list.



- Step 5** Type the URL for the .BMP image, including the IP address or hostname of the TFTP server, the directory, and filename in the Picture download URL field.
- Step 6** Click **Submit All Changes**.
- 

## Call Appearances and Extensions

A client station is a SPA900 Series phone with one or more extensions and call appearances provisioned. An extension in the SPA900 Series 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 SPA900 Series 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. SPA900 Series 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 SPA900 Series phones with multiple line key status indicators

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 an LedScript. The administrator can specify a different color and pattern for each of the following states of the call appearance.

- Idle: This call appearance is not in use. The user can use it 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

## 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 3-12](#)
- [Making Calls, page 3-12](#)
- [Answering and Ending Calls, page 3-13](#)
- [Hold and Resume, page 3-13](#)
- [Call Waiting, page 3-13](#)
- [Speed Dialing, page 3-14](#)
- [Three-Way Conferencing, page 3-14](#)
- [Attended Call Transfer, page 3-15](#)
- [Blind Call Transfer, page 3-15](#)
- [Call Back, page 3-16](#)
- [Message Waiting Indication \(MWI\), page 3-16](#)
- [Accessing Voicemail, page 3-16](#)
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- [Dialing Assistance, page 3-18](#)
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- [Call Logs, page 3-18](#)
- [Audio Volume Adjustment, page 3-20](#)
- [Managing Ring Tones, page 3-20](#)

## Selecting the Audio I/O Device and Line

The SPA900 Series 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 any one of the calls becomes active, the handset is selected automatically as the audio device if it is off-hook, else the speaker or the headset is selected according to configuration: the user can configure whether speaker or the headset device has higher preference.]

A line is selected by pressing the corresponding line key. To make or receive calls, the user must select a call appearance (line key) or an audio device. If the user selects a call appearance, then an audio device is selected automatically according to user preference setting. If the user selects an audio device, an idle call appearance is selected automatically in the order L1, L2, L3, and L4. The user may 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** or **xfer** 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. Selects an audio device
2. Dials the number

The 900 Series phone lets the user perform these two steps in either order. In other words, the user can select an audio device either before or after dialing. The 900 Series 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 descending 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. The user 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. The phones in the SPA900 Series 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 the user, 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 Menu 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. The user 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. The user can 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. The user 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 the user 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 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, the user is 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 the user 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 the one transferred.



## Call Back

This service mimics the call back service offered by the PSTN. The user can activate this service on a busy number such that he is 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, the user selects a line key and enters **\*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, the user picks up any line and enters **\*86**. When call back service is active, the corresponding Line key flashes green. The user 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, the user hears 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 the user 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

The user 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](http://ip_address/pdir.htm). The user can invoke the Directory menu on the phone in 2 ways:

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

The user 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 the user for confirmation before proceeding.

The user 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**. The user can edit the new entry and then press **save** to save the entry. The user 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 the user 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 the user dials 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, the user 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. The user 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 if 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. The user 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./callog.htm](http://ip_address./callog.htm)

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

```
[ext-id]name, phone, time-stamp
```

Where:

- ext-id is the extension on which the call is made from or received at (E1, E2, E3, or E4). If [En] is not present, it implies [E1].
- name is the call peer name. If this field is missing, the peer name is not available.
- phone is the call peer phone number (user-id). If this field is missing, the peer phone number is not available.
- t is the time-stamp at which the call log is entered. It has the format mm/dd hh/mm<a|p> (a = am and p = pm).

For example:

```
[E2]Joe Smith,14089991234,t=10/12 11:12a
```

Another example, when the name is not available:

```
14089991234,10/12 01:12p
```

Table 3-4 summarizes operations the user can perform with the call logs.

**Table 3-4 Call Log Operations**

Operation	Steps
To view a call log	<ol style="list-style-type: none"> <li>1. Press <b>Menu</b>.</li> <li>2. Select <b>Call History</b>.</li> <li>3. Select the log you want to view.</li> </ol>
To delete a log	<ol style="list-style-type: none"> <li>1. Highlight the entry.</li> <li>2. Press <b>del</b>.</li> </ol>
To edit a log	<ol style="list-style-type: none"> <li>1. Highlight the entry.</li> <li>2. Press <b>edit</b>.</li> <li>3. Press <b>save</b> to save the changes, or <b>cancel</b> to abort the change.</li> </ol>

*Table 3-4 Call Log Operations*

Operation	Steps
To call a log entry	<ol style="list-style-type: none"> <li>1. Highlight the entry.</li> <li>2. Press <b>dial</b>, or Off-Hook, or Turn on Speaker, or Turn on Headset, or press a line key.</li> </ol>
To save a log entry into Personal Directory (if the directory is not already full)	<ol style="list-style-type: none"> <li>1. Highlight the entry.</li> <li>2. Modify the entry as needed.</li> <li>3. Press <b>save</b> to save the entry in the personal directory.</li> </ol>

## Audio Volume Adjustment

There are four kinds of audio volume that can be adjusted:

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

To adjust the volume, press the VOL key to invoke the volume adjustment screen. Then use the UP/DOWN keys to increase or decrease the volume. 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, the user can assign a name for the ring tone, and specify two components:

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

The user can assign a default ring tone for each extension. The user 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, use the Upload Ringtone utility available from the Linksys website:

<http://www.sipura.com/Documents/ringtone.zip>

Figure 3-5 illustrates the user interface provided by this utility.

Figure 3-5 Upload Ringtone Utility



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

## Configuring a Dial Plan

The 900 Series 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 900 Series 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 900 Series 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.

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>xxxxxxx” would match “85551212” and transmit “16505551212”.

## Intersequence Tones

An “outside line” dial tone can be generated within a sequence by appending a comma (,) between digits. Thus, the sequence “9,1xxxxxxxxx” 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, “1900xxxxxxx!” automatically rejects all 900 area code numbers from being dialed.

## Interdigit Timer Master Override

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

- Long interdigit timer: L : delay-value ,
- Short interdigit timer: S : delay-value ,

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

## Local Timer Overrides

The long and short timeout values can be changed for a particular sequence starting at a particular point in the sequence. The syntax for long timer override is: “L delay-value<space>“. Note the terminating space character. The specified delay-value is measured in seconds. Similarly, to change the short timer override, use: S delay-value<space>.

## Pause

A sequence may require an explicit pause of some duration before continuing to dial digits, in order for the sequence to match. The syntax for this is similar to the timer override syntax: P delay-value <space>. The delay-value is measured in seconds.

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



## Implicit Sequences

The 900 Series 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 xxxxxx 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

<b>ParName</b>	Interdigit_Long_Timer
<b>Default</b>	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

<b>ParName</b>	Interdigit_Short_Timer
<b>Default</b>	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

<b>ParName</b>	Dial_Plan[n] for Each Line <i>n</i>
<b>Default</b>	(*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 SPA900 Series IP phone. It includes the following topics:

- [Reboot and Restart, page 3-28](#)
- [Factory Reset, page 3-28](#)
- [Password Protection, page 3-28](#)
- [Managing the Time/Date, page 3-28](#)
- [Daylight Saving Time, page 3-29](#)
- [Using Star Codes to Activate/Deactivate Services, page 3-30](#)
- [Disabling Services, page 3-31](#)
- [Error and Log Reporting, page 3-32](#)

## Reboot and Restart

Reboot cold boots the phone and restart DHCP. Restart warm boots the phone, without restarting DHCP. Both options require the user to confirm before proceeding.

## Factory Reset

Factory reset resets all parameters to default value. The personal directory and call logs are also cleared. This option require the user to confirm before proceeding.

## Password Protection

The user can configure a password to the phone to protect against access to the certain options.

## Managing the Time/Date

The 900 Series phone obtains the current time information in one of the following ways:

- **NTP Server**—One or two NTP servers can be configured to the phone. When it first boots up, the phone tries to contact the NTP server to get the current time. Then the phone periodically synchronizes the current time with the NTP server. The synchronization period is fixed at 1 hour. In between the update, the phone tracks the 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**—900 Series phone also lets the user 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 the user 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 900 Series 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]]]
- <month> = 1,2,12 (for Jan, Feb, Dec)
- <day> = [+|-]1,2,3,....,31
- <weekday> = 1, 2, 3,...., 7 (for Mon, Tue, ..., Sun), or 0 which has special meaning
- HH = hour (0-23)
- mm = minute (0-59)
- ss = second (0-59)

<start-time> and <end-time> specify the start and end time and date of daylight saving time and <save-time> is the amount of hour/min/sec to add to the current time during daylight saving period.

The <save-time> value can be preceded by a negative (-) sign if subtraction is desired instead of addition. 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. 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).

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:

```
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.

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

## Using Star Codes to Activate/Deactivate Services

The 900 Series phone accepts star codes to activate or deactivate certain services on the phone. Below is a list of services that are accessible with a star code, with the default star code shown in parenthesis:

- Call Return (\*69)—Calls the last caller, regardless which extension.
- Call Back(\*66)—Periodically redials the last busy number (every 30s by default) until it rings or until the trial expires (30 min by default), regardless which extension. Only 1 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 for all outbound calls. Applies to all extensions.
- 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)
- 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, the user 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

This is a list of star codes separated by a vertical bar (|), such as \*79|\*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

This is 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.

## 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 900 Series 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 status code values can be read from the IVR option XXX or from the SPA web-page.

### LED Blink Rate Definitions

- ON—LED remains solid on
- 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



Note

The Link LED flashes on transmit and receive (TX/RX) of packets. The LED displays solid off if no link is available. The LED displays solid on if link is up but no TX/RX activity is present.

# 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 SPA900 Series occasionally flash red?

A. A flashing red line status light on the SPA900 Series phone indicates a local LAN issue. If statistics on the switch show a high number of collisions on the ports connected to the 900 Series 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 to version 1.0.0.5.

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

A. Perform the following steps:

1. Check all the connections into the phone (5 Volt Power supply, RJ 45 Ethernet connection, and Handset cord.)
2. Check the wall socket and confirm that AC power is available.
3. 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, perform the following steps.

1. Press the **Menu** 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 **Menu** 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.

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)

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).



## LCD Command Reference Guide

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This chapter provides a systematic reference for the configuration and monitoring options available from the LCD display of 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 4-5
- 6 Call Forward, page 4-7
- 7 Time/Date, page 4-8
- 8 Voice Mail, page 4-8
- 9 Network, page 4-8
- 10 Product Info, page 4-12
- 11 Status, page 4-13
- 12 Reboot, page 4-14
- 13 Restart, page 4-14
- 14 Factory Reset, page 4-15
- 15 Set Password, page 4-15
- 16 Set LCD Contrast, page 4-15
- 17 CallPark Status, page 4-15
- 18 Language (SPA922, 942, and 962), page 4-16

# 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](#)

## Entering Names and Numbers into the Directory

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

To enter the **Directory** option, press the **Menu** button and then press the **1** button. Alternatively, you may press the Menu 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**, the user is 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.

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

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 Menu button and then press **2**. Alternatively, press the Menu 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 the user enters in 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.

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

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

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: Menu, 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 Menu option, press the **cancel** soft key.

## Answered Calls

**Shortcut: Menu, 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 Menu option, press the **cancel** soft key..

## Missed Calls

**Shortcut: Menu, 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 Menu option, press the **cancel** soft key..

## 4 Ring Tone

**Shortcut: Menu, 4, select extension, change, select ring tone**

This option lets the user select a different ring tone for each extension that the phone supports. It lets the user personalize his phone and can be useful in a shared office environment to provide each user with have a distinctive ring tone.

To set or change the ring tone for a specific extension, press Menu and then select ring tones by pressing **4**. Alternatively, press the Menu 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

To select the Preferences option, press the Menu 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: Menu, 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 Menu 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: Menu, 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 Menu 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: Menu, 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: Menu, 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 900 Series 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:** Menu, 5, 5, edit, y/n, ok

The Dial Assistance option allows the 900 Series 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:** Menu, 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.

# 6 Call Forward

To view or change the call forwarding setting, press the Menu 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:** Menu, 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:** Menu, 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 **Menu**, 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:** **Menu, 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:** **Menu, 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:** **Menu, 7, date, time, offset, save**

To set the date and time, press **Menu** 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:** **Menu, 8, voicemail\_number, save**

To set the number to call voicemail, press **Menu** 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 Menu button) to dial the mailbox.

## 9 Network

To view or modify network parameters, press the Menu 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

### 9.1 DHCP

**Shortcut:** Menu, **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 Menu 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:** Menu, **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:** Menu, 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:** Menu, 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:** Menu, 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:** Menu, 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 (the user cannot edit it)
- Manually set the address using the Non DHCP Default Route option (Option 9.10)

## 9.7 Enable Web Server

**Shortcut:** Menu, 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:** Menu, 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:** Menu, 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:** Menu, 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:** Menu, 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:** Menu, 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:** Menu, 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:** Menu, 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**.

# 10 Product Info

To view the parameters in the **Product Info**, press the **Menu** 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
- 10.5 Client Cert

## 10.1 Product Name

**Shortcut:** Menu, 10, select

To view the Product Name, press the Menu 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:** Menu, 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:** Menu, 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:** Menu, 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:** Menu, 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:** Menu, 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.

# 11 Status

To view the status of the phone and extension parameters, press the Menu 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:

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

## Phone

**Shortcut:** Menu, 11, select

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

## Ext 1/2/3/4

**Shortcut:** Menu, 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 date and time, and the time in seconds until the next registration. If the extension is not registered, the phone displays **Not Registered**.

## Line 1, 2,3,4

**Shortcut:** Menu, 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.

## 12 Reboot

**Shortcut:** Menu, **12**, **ok**

This option lets the user reboot the phone. To select Reboot, press the Menu button and press **12**, or scroll down until the selection is highlighted and then press **select** soft key. The LCD displays **Confirm System Reboot ?**. The display shows two options Reboot or Cancel. Select **Reboot** to reboot the phone. The option is only displayed for 5 seconds. After 5 seconds, the display returns to the Menu option.

## 13 Restart

**Shortcut:** Menu, **13**, **ok**

This option lets the user restart the phone. To select Restart, press the Menu 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?** This display shows the options ok or cancel. Select **ok** to restart the phone. This option is only displayed for 5 seconds. After 5 seconds, the display is returned to the Menu option.

## 14 Factory Reset

**Shortcut:** Menu, **14**, **ok**

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

The menu displays **All parameters are reset to factory default values, continue?** The display shows two options: ok or cancel. Select **ok** 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 Menu option.

## 15 Set Password

**Shortcut:** Menu, 15, *password*, ok

This option lets the user set a password to protect the phone from unauthorized use. To select this option, press the Menu 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.



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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.

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## 16 Set LCD Contrast

**Shortcut:** menu, 16, scroll knob left or right, save

This option lets the user adjust the LCD contrast. To use this option, press the Menu 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.

## 17 CallPark Status

**Shortcut:** menu, 17

This option lets the user 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**.

## 18 Language (SPA922, 942, and 962)

**Shortcut:** menu, 18, 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 the user change the language of the phone from the user menu. To change the language, the user performs the following steps:

1. Press the **Menu** 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 to the user through the LCD menu, refer to the [“Localization” section on page 3-6](#).



## SPA900 Series Phone Field Reference

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

- [Info Tab, page 5-1](#)
- [System Tab, page 5-5](#)
- [SIP Tab, page 5-8](#)
- [Regional Tab, page 5-17](#)
- [Ext 1/2/3/4/5/6 Tab, page 5-36](#)
- [Phone Tab, page 5-31](#)
- [Ext 1/2/3/4/5/6 Tab, page 5-36](#)
- [User Tab, page 5-48](#)

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

### Info Tab

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

- [System Information, page 5-1](#)
- [Product Information, page 5-2](#)
- [Phone Status, page 5-2](#)



Note

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

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## System Information

Field	Description
DHCP	Indicates if DHCP is enabled.
Current IP	Displays the current IP address assigned to the 900 Series phone.
Host Name	Displays the current IP address assigned to the 900 Series phone.
Domain	Displays the network domain name of the 900 Series phone.
Current Netmask	Displays the network mask assigned to the 900 Series phone.
Current Gateway	Displays the default router assigned to the 900 Series phone.
Primary DNS	Displays the primary DNS server assigned to the 900 Series phone.
Secondary DNS	Displays the secondary DNS server assigned to the 900 Series phone.

## Product Information

Field	Description
Product Name	Model number of the 900 Series phone.
Serial Number	Serial number of the 900 Series phone.
Software Version	Version number of the 900 Series phone software.
Hardware Version	Version number of the 900 Series phone hardware.
MAC Address	MAC address of the 900 Series phone.
Client Certificate	Status of the client certificate, which authenticates the 900 Series phone for use in the ITSP network. This field indicates if the client certificate is properly installed in the 900 Series phone.
Customization	For an RC unit, this field indicates whether the unit has been customized or not. Pending indicates a new RC unit that is ready for provisioning. If the unit has already retrieved its customized profile, this field displays the name of the company that provisioned the unit.
Licenses	Indicates any additional licenses that you have installed in the 900 Series 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	VLAN identifier used by the SPA900 Series IP phone.

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

Field	Description
Registration State	Displays whether the phone has registered with the SIP proxy or not.
Last Registration At	Last date and time the line was registered.
Next Registration In	Number of seconds before the next registration renewal.

Info Tab

Field	Description
Message Waiting	Indicates whether you have new voicemail waiting: Yes or No. This is updated when voicemail notification is received. You can also manually modify it to clear or set the flag. 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.
Mapped SIP Port	Port number of the SIP port mapped by NAT.

## Line 1/2/3/4/5/6 Status

Field	Description
Call 1/2 State	Status of the call.
Call 1/2 Tone	Type of tone used by the call.
Call 1/2 Encoder	Codec used for encoding.
Call 1/2 Decoder	Codec used for decoding.
Call 1/2 Type	Direction of the call.
Call 1/2 Remote Hold	Indicates whether the far end has placed the call on hold.
Call 1/2 Callback	Indicates whether the call was triggered by a call back request.
Call 1/2 Peer Name	Name of the internal phone.
Call 1/2 Peer Phone	Phone number of the internal phone.
Call 1/2 Duration	Duration of the call.
Call 1/2 Packets Sent	Number of packets sent.
Call 1/2 Packets Recv	Number of packets received.
Call 1/2 Bytes Sent	Number of bytes sent.
Call 1/2 Bytes Recv	Number of bytes received.
Call 1/2 Decode Latency	Number of milliseconds for decoder latency.
Call 1/2 Jitter	Number of milliseconds for receiver jitter.
Call 1/2 Round Trip Delay	Number of milliseconds for delay.
Call 1/2 Packets Lost	Number of packets lost.
Call 1/2 Packet Error	Number of invalid packets received.
Call 1/2 Mapped RTP Port	The port mapped for Real Time Protocol traffic for Call 1/2.
Call 1/2 Media Loopback	The media loopback status of the phone.

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

## System Configuration

Field	Description
Restricted Access Domains	This feature is used when implementing software customization.
Enable Web Server	Enable/disable web server of 900 Series phone This feature should only be used on firmware version 1.0.9 or later. The default is yes.
Web Server Port	Port number of the 900 Series phone administration web server. The default is 80.
Enable Web Admin Access	Lets you enable or disable local access to the administration web server. Select yes or no from the drop-down menu. The default is yes.
Admin Passwd	Password for the administrator. The default is no password.
User Password	Password for the user. The default is no password.

## Internet Connection Type

Field	Description
Connection Type	Select the type of connection used by the phone. The selections are DHCP, PPPoE, or Static IP. The default is DHCP.



## Static IP Settings

Static IP	Static IP address of 900 Series phone, which takes effect if DHCP is disabled. The default is 0.0.0.0.
NetMask	The NetMask used by 900 Series phone when DHCP is disabled. The default is 255.255.255.0.
Gateway	The default gateway used by 900 Series phone when DHCP is disabled. The default is 0.0.0.0.

## PPPoE Settings

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 900 Series phone.
Domain	The network domain of the 900 Series phone.
Primary DNS	DNS server used by 900 Series phone in addition to DHCP supplied DNS servers if DHCP is enabled; when DHCP is disabled, this is the primary DNS server. The default is 0.0.0.0.
Secondary DNS	DNS server used by 900 Series phone in addition to DHCP supplied DNS servers if DHCP is enabled; when DHCP is disabled, this is the secondary DNS server. The default is 0.0.0.0.
DNS Server Order	Specifies the method for selecting the DNS server. The options are Manual, Manual/DHCP, and DHCP/Manual.

System Tab

Field	Description
DNS Query Mode	Perform parallel or sequential DNS Query. With parallel DNS query mode, the 900 Series 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 900 Series phone. The default is parallel.
Syslog Server	Specify the syslog server name and port. This feature specifies the server for logging 900 Series 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 900 Series phone debug information. The level of detailed output depends on the debug level parameter setting.
Debug Level	The higher the debug level, the more debug information is generated. Zero (0) means no debug information is generated. To log SIP messages, Debug Level must be set to at least 2. The default is 0.
Primary NTP Server	IP address or name of primary NTP server.
Secondary NTP Server	IP address or name of secondary NTP server.

## VLAN Settings

Field	Description
Enable VLAN	Enable this parameter if the switch to which the 900 Series IP phone is connected uses VLAN tagging.
VLAN ID	Enter the VLAN tag for the VLAN to which the 900 Series 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). The default is Yes.

## SIP Tab

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

- [SIP Parameters, page 5-8](#)
- [SIP Timer Values \(sec\), page 5-10](#)
- [Response Status Code Handling, page 5-11](#)
- [RTP Parameters, page 5-12](#)
- [SDP Payload Types, page 5-13](#)
- [NAT Support Parameters, page 5-16](#)

## SIP Parameters

Field	Description
Max Forward	SIP Max Forward value, which can range from 1 to 255. The default is 70.
Max Redirection	Number of times an invite can be redirected to avoid an infinite loop. The default is 5.
Max Auth	Maximum number of times (from 0 to 255) a request may be challenged. The default is 2.
SIP User Agent Name	User-Agent header used in outbound requests. The default is \$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. The default is \$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. The default is blank.
SIP Accept Language	Accept-Language header used. There is no default (this indicates 900 Series 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. The default is application/dtmf-relay.
Remove Last Reg	Lets you remove the last registration before registering a new one if the value is different. Select yes or no from the drop-down menu. The default is no.

SIP Tab

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 900 Series phone uses compact SIP headers in outbound SIP messages. If set to no, the 900 Series phone uses normal SIP headers. If inbound SIP requests contain compact headers, 900 Series 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, 900 Series phone substitutes those headers with compact headers (if defined by RFC 261) if <Use Compact Header> parameter is set to yes. The default is no.
Escape Display Name	Lets you keep the Display Name private. Select yes if you want the 900 Series 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. The default is no.
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. The default is yes.
SIP-B Enable	Enables Sylanro 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 from an external application.
Hold Package	Enables support for the Broadsoft Hold Package, which enables a user to hold a call by clicking a button from an external application.
Conference Package	Enables support for the Broadsoft Conference Package, which enables a user to start a conference by clicking a button from an external application.
Notify Conference	If enabled, the unit will send out a NOTIFY with event=conference when starting a conference.
Random REG CID On Reboot	If set to yes, the 900 Series phone uses a different random call-ID for registration after the next software reboot. If set to no, the 900 Series phone tries to use the same call-ID for registration after the next software reboot. The 900 Series phone always uses a new random Call-ID for registration after a power-cycle, regardless of this setting. The default is no.
SIP TCP Port Min	Specifies the lowest TCP port number that can be used for SIP sessions.
SIP TCP Port Max	Specifies the highest TCP port number that can be used for SIP sessions.
CTI Enable	Enables or disables the Computer Telephone Interface feature provided by some servers.

## SIP Timer Values (sec)

Field	Description
SIP T1	RFC 3261 T1 value (RTT estimate), which can range from 0 to 64 seconds. The default is 5.
SIP T2	RFC 3261 T2 value (maximum retransmit interval for non-INVITE requests and INVITE responses), which can range from 0 to 64 seconds. The default is 4.
SIP T4	RFC 3261 T4 value (maximum duration a message remains in the network), which can range from 0 to 64 seconds. The default is 5.
SIP Timer B	INVITE time-out value, which can range from 0 to 64 seconds. The default is 32.
SIP Timer F	Non-INVITE time-out value, which can range from 0 to 64 seconds. The default is 32.
SIP Timer H	INVITE final response, time-out value, which can range from 0 to 64 seconds. The default is 32.
SIP Timer D	ACK hang-around time, which can range from 0 to 64 seconds. The default is 32.
SIP Timer J	Non-INVITE response hang-around time, which can range from 0 to 64 seconds. The default is 32.
INVITE Expires	INVITE request Expires header value. If you enter 0, the Expires header is not included in the request. The default is 240. Range: 0–(2 <sup>31</sup> –1).
ReINVITE Expires	ReINVITE request Expires header value. If you enter 0, the Expires header is not included in the request. The default is 30. Range: 0–(2 <sup>31</sup> –1).
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. The default is 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. The default is 7200.

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Reg Retry Intvl	Interval to wait before the 900 Series phone retries registration after failing during the last registration. The default is 30.
Reg Retry Long Intvl	When registration fails with a SIP response code that does not match<Retry Reg RSC>, the 900 Series phone waits for the specified length of time before retrying. If this interval is 0, the 900 Series phone stops trying. This value should be much larger than the Reg Retry Intvl value, which should not be 0. The default is 1200.
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 a Min-Expires header
Sub Retry Intvl	This value (in seconds) determines the retry interval when the last Subscribe request fails.
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. The default is 0, 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. The default is 0, 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. The default value is 0, which disables the exponential back-off feature.

## 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.
SIT2 RSC	SIP response status code to INVITE on which to play the SIT2 Tone.

SIT3 RSC	SIP response status code to INVITE on which to play the SIT3 Tone.
SIT4 RSC	SIP response status code to INVITE on which to play the SIT4 Tone.
Try Backup RSC	SIP response code that retries a backup server for the current request.
Retry Reg RSC	Interval to wait before the 900 Series phone retries registration after failing during the last registration. The default is 30.

## 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. The default is 16384.
RTP Port Max	Maximum port number for RTP transmission and reception. The default is 16482.
RTP Packet Size	Packet size in seconds, which can range from 0.01 to 0.16. Valid values must be a multiple of 0.01 seconds. The default is 0.030.
Max RTP ICMP Err	Number of successive ICMP errors allowed when transmitting RTP packets to the peer before the 900 Series phone terminates the call. If value is set to 0, the 900 Series phone ignores the limit on ICMP errors. The default is 0.

<p>RTCP Tx Interval</p>	<p>Interval for sending out RTCP sender reports on an active connection. It can range from 0 to 255 seconds. During an active connection, the 900 Series 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 &lt;User ID&gt;@&lt;Proxy&gt;, NAME is set to &lt;Display Name&gt; (or Anonymous if user blocks caller ID), and TOOL is set to the Vendor/Hardware-platform-software-version (such as Linksys/900 Series phone-1.0.31(b)). The NTP timestamp used in the SR is a snapshot of the 900 Series phone's local time, not the time reported by an NTP server. If the 900 Series phone receives a RR from the peer, it attempts to compute the round trip delay and show it as the &lt;Call Round Trip Delay&gt; value (ms) in the Info section of 900 Series phone web page.</p> <p>The default is 0.</p>
<p>No UDP Checksum</p>	<p>Select yes if you want the 900 Series phone to calculate the UDP header checksum for SIP messages. Otherwise, select no.</p> <p>The default is no.</p>
<p>Symmetric RTP</p>	<p>Enable symmetric RTP operation. If enabled, the SPA3102 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) the SPA3102 sends RTP to the destination as indicated in the inbound SDP.</p> <p>The default is yes.</p>
<p>Stats In BYE</p>	<p>Determines whether the 900 Series phone includes the P-RTP-Stat header or response to a BYE message. The header contains the RTP statistics of the current call. Select yes or no from the drop-down menu. The format of the P-RTP-Stat header is:</p> <p>P-RTP-State: PS=&lt;packets sent&gt;,OS=&lt;octets sent&gt;,PR=&lt;packets received&gt;,OR=&lt;octets received&gt;,PL=&lt;packets lost&gt;,JI=&lt;jitter in ms&gt;,LA=&lt;delay in ms&gt;,DU=&lt;call duration in s&gt;,EN=&lt;encoder&gt;,DE=&lt;decoder&gt;.</p> <p>The default is no.</p>



## SDP Payload Types

The configured dynamic payloads are used for outbound calls only where the 900 Series phone presents the SDP offer. For inbound calls with a SDP offer, 900 Series phone follow the caller dynamic payload type assignments.

900 Series phone uses the configured codec names in its outbound SDP. 900 Series phone ignores the codec names in incoming SDP for standard payload types (0 – 95). For dynamic payload types, 900 Series phone identifies the codec by the configured codec names. Comparison is case-insensitive.

Field	Description
NSE Dynamic Payload	NSE dynamic payload type. The valid range is 96-127. The default is 100.
AVT Dynamic Payload	AVT dynamic payload type. The valid range is 96-127. The default is 101.
INFOREQ Dynamic Payload	INFOREQ dynamic payload type. There is no default.
G726r16 Dynamic Payload	G.726-16 dynamic payload type. The valid range is 96-127. The default is 98.
G726r24 Dynamic Payload	G.726-24 dynamic payload type. The valid range is 96-127. The default is 97.
G726r32 Dynamic Payload	G.726-32 dynamic payload type. The default is 2.
G726r40 Dynamic Payload	G.726-40 dynamic payload type. The valid range is 96-127. The default is 96.
G729b Dynamic Payload	G.729b dynamic payload type. The valid range is 96-127. The default is 99.
NSE Codec Name	NSE codec name used in SDP. The default is NSE.
AVT Codec Name	AVT codec name used in SDP. The default is telephone-event.
G711u Codec Name	G.711u codec name used in SDP. The default is PCMU.
G711a Codec Name	G.711a codec name used in SDP. The default is PCMA.

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G726r16 Codec Name	G.726-16 codec name used in SDP. The default is G726-16.
G726r24 Codec Name	G.726-24 codec name used in SDP. The default is G726-24.
G726r32 Codec Name	G.726-32 codec name used in SDP. The default is G726-32.
G726r32 Codec Name	G.726-32 codec name used in SDP. The default is G726-32.
G726r40 Codec Name	G.726-40 codec name used in SDP. The default is G726-40.
G729a Codec Name	G.729a codec name used in SDP. The default is G729a.
G729b Codec Name	G.729b codec name used in SDP. The default is G729ab.
ENCAP RTP Dynamic Payload	The dynamic payload value (96 – 127) used for the encapsulating RTP packets when offering the SDP to loopback packets. This setting is used if the SPA is the offerer of the SDP. Otherwise, the value is decided by the peer. The default value is 112.
RTP-Start-Loopback Dynamic Payload	The dynamic payload value (96 – 127) used by the mirror in the self-generated RTP packets before receiving any RTP packets from the source. This setting is used only when the SPA is acting as the loopback source. Otherwise, the value is decided by the peer.  This value must be different from any of the dynamic payload values that might be used by the source and the mirror (including the encaprtp payload type). This is necessary so that the source can easily tell when the mirror has switched from sending self-generated RTP packets to sending loopback packets. The default value is 113.
RTP-Start-Loopback Codec	The actual codec corresponding to RTP-Start-Loopback Dynamic Payload, whose codec name is used in the rtpmap attribute for the for the mirror self-generated RTP audio stream, prior to receiving any RTP packets from the source. The choices are {G711u, G711a, G726-16, G726-24, G726-32, G726-40, G729a, G723}. The default is G711u.
G723 Codec Name	G.723 codec name used in SDP. The default is G723.

EncapRTP Codec Name	EncapRTP codec name for the encapsulating RTP payload type to be used in the rtpmap attribute in the SDP. Both sides must use the same name to interoperate properly.  The default is EncapRTP.
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## NAT Support Parameters

Field	Description
Handle VIA received	If you select yes, the 900 Series phone processes the received parameter in the VIA header (this is inserted by the server in a response to anyone of its requests). If you select no, the parameter is ignored. Select yes or no from the drop-down menu.  The default is no.
Handle VIA rport	If you select yes, the 900 Series phone processes the rport parameter in the VIA header (this is inserted by the server in a response to anyone of its requests). If you select no, the parameter is ignored. Select yes or no from the drop-down menu.  The default is 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.  The default is 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.  The default is 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.  The default is 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.  The default is no.
STUN Enable	Enables the use of STUN to discover NAT mapping. Select yes or no from the drop-down menu.  The default is no.

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STUN Test Enable	If the STUN Enable feature is enabled and a valid STUN server is available, the 900 Series 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 900 Series phone detects symmetric NAT or a symmetric firewall, NAT mapping is disabled.  The default is 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 900 Series 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 900 Series 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.  The default is 0.0.0.0.
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.  The default is 0.
NAT Keep Alive Intvl	Interval between NAT-mapping keep alive messages.  The default is 15.

## Linksys Key System Parameters

Field	Description
Linksys Key System	Enable or disable the Linksys Key System on the IP phone.  The default is yes.
Multicast Address	The multicast address is used by the SPA9000 to communicate with the SPA900 Series IP phones.  The default is 224.168.168.168:6061.
Force LAN Codec	The choices are: none, G.711u, or G.711a. The default is none.

## Regional Tab

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

- [Call Progress Tones, page 5-18](#)
- [Distinctive Ring Patterns, page 5-20](#)
- [Control Timer Values \(sec\), page 5-21](#)
- [Vertical Service Activation Codes, page 5-21](#)
- [Outbound Call Codec Selection Codes, page 5-26](#)
- [Miscellaneous, page 5-28](#)

## Call Progress Tones

Field	Description
Dial Tone	Prompts the user to enter a phone number. Reorder Tone is played automatically when <Dial Tone> or any of its alternatives times out. The default is 350@-19,440@-19;10(*0/1+2).
Second Dial Tone	Alternative to the Dial Tone when the user dials a three-way call. The default is 420@-19,520@-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. The default is 420@-19;10(*0/1).
Prompt Tone	Prompts the user to enter a call forwarding phone number. The default is 520@-19,620@-19;10(*0/1+2).
Busy Tone	Played when a 486 RSC is received for an outbound call. The default is 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. The default is 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. The default is 480@10,620@0;10(.125/.125/1+2).
Ring Back Tone	Played during an outbound call when the far end is ringing. The default is 440@-19,480@-19;*(2/4/1+2).

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Confirm Tone	Brief tone to notify the user that the last input value has been accepted. The default is 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 <i>SIP</i> screen. The default is 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 <i>SIP</i> screen. The default is 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 <i>SIP</i> screen. The default is 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 <i>SIP</i> screen. The default is 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. The default is 350@-19,440@-19;2(.1/.1/1+2);10(*0/1+2).
Cfwd Dial Tone	Played when all calls are forwarded. The default is 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. The default is 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. The default is 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. The default is 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. 600@-16;.3(.05/0.05/1)
Feature Invocation Tone	Played when a feature is implemented. (Not in PAP2T) The default is 350@-16;*(.1/.1/1).

## Distinctive Ring Patterns

Field	Description
Ring1 Cadence	Cadence script for distinctive ring 1. The default is 60(2/4).
Ring2 Cadence	Cadence script for distinctive ring 2. The default is 60(.3/.2, 1/.2,.3/4).
Ring3 Cadence	Cadence script for distinctive ring 3. The default is 60(.8/4,.8/4).
Ring4 Cadence	Cadence script for distinctive ring 4. The default is 60(.4/.2,.3/.2,.8/4).
Ring5 Cadence	Cadence script for distinctive ring 5. The default is 60(.4/.2,.3/.2,.8/4).
Ring6 Cadence	Cadence script for distinctive ring 6. The default is 60(.4/.2,.3/.2,.8/4).
Ring7 Cadence	Cadence script for distinctive ring 7. The default is 60(.4/.2,.3/.2,.8/4).
Ring8 Cadence	Cadence script for distinctive ring 8. The default is 60(.4/.2,.3/.2,.8/4).

## Control Timer Values (sec)

Field	Description
Reorder Delay	Delay after far end hangs up before reorder tone is played. 0 = plays immediately, inf = never plays. Range: 0–255 seconds. The default is 5.
Call Back Expires	Expiration time in seconds of a call back activation. Range: 0–65535 seconds. The default is 1800.
Call Back Retry Intvl	Call back retry interval in seconds. Range: 0–255 seconds. The default is 30.
Call Back Delay	Delay after receiving the first SIP 18x response before declaring the remote end is ringing. If a busy response is received during this time, the 900 Series phone still considers the call as failed and keeps on retrying. The default is 0.5.
VMWI Refresh Intvl	Interval between VMWI refresh to the CPE. The default is 0.5.
Interdigit Long Timer	Long timeout between entering digits when dialing. The interdigit timer values are used as defaults when dialing. The Interdigit_Long_Timer is used after any one digit, if all valid matching sequences in the dial plan are incomplete as dialed. Range: 0–64 seconds. The default is 10.
Interdigit Short Timer	Short timeout between entering digits when dialing. The Interdigit_Short_Timer is used after any one digit, if at least one matching sequence is complete as dialed, but more dialed digits would match other as yet incomplete sequences. Range: 0–64 seconds. The default is 3.

## Vertical Service Activation Codes

Vertical Service Activation Codes are automatically appended to the dial-plan. There is no need to include them in dial-plan, although no harm is done if they are included.

Field	Description
Call Return Code	This code calls the last caller. The default is *69.



Call Redial Code	Redials the last number called. (Not in PAP2T) The default is *07.
Blind Transfer Code	Begins a blind transfer of the current call to the extension specified after the activation code. The default is *98.
Call Back Act Code	Starts a callback when the last outbound call is not busy. The default is *66.
Call Back Deact Code	Cancel a callback. The default is *86.
Call Back Busy Act Code	Starts a callback when the last outbound call is busy. (Not in PAP2T) The default is *05
Cfwd All Act Code	Forwards all calls to the extension specified after the activation code. The default is *72.
Cfwd All Deact Code	Cancel call forwarding of all calls. The default is *73.
Cfwd Busy Act Code	Forwards busy calls to the extension specified after the activation code. The default is *90.
Cfwd Busy Deact Code	Cancel call forwarding of busy calls. The default is *91.
Cfwd No Ans Act Code	Forwards no-answer calls to the extension specified after the activation code. The default is *92.
Cfwd No Ans Deact Code	Cancel call forwarding of no-answer calls. The default is *93.
Cfwd Last Act Code	Forwards the last inbound or outbound calls to the extension specified after the activation code. The default is *63.
Cfwd Last Deact Code	Cancel call forwarding of the last inbound or outbound calls. The default is *83.
Block Last Act Code	Blocks the last inbound call. The default is *60.
Block Last Deact Code	Cancel blocking of the last inbound call. The default is *80.

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Accept Last Act Code	Accepts the last outbound call. It lets the call ring through when do not disturb or call forwarding of all calls are enabled. The default is *64.
Accept Last Deact Code	Cancels the code to accept the last outbound call. The default is *84.
CW Act Code	Enables call waiting on all calls. The default is *56.
CW Deact Code	Disables call waiting on all calls. The default is *57.
CW Per Call Act Code	Enables call waiting for the next call. The default is *71.
CW Per Call Deact Code	Disables call waiting for the next call. The default is *70.
Block CID Act Code	Blocks caller ID on all outbound calls. The default is *67.
Block CID Deact Code	Removes caller ID blocking on all outbound calls. The default is *68.
Block CID Per Call Act Code	Blocks caller ID on the next outbound call. The default is *81.
Block CID Per Call Deact Code	Removes caller ID blocking on the next inbound call. The default is *82.
Block ANC Act Code	Blocks all anonymous calls. The default is *77.
Block ANC Deact Code	Removes blocking of all anonymous calls. The default is *87.
DND Act Code	Enables the do not disturb feature. The default is *78.
DND Deact Code	Disables the do not disturb feature. The default is *79.
CID Act Code	Enables caller ID generation. The default is *65.

CID Deact Code	Disables caller ID generation. The default is *85.
CWCID Act Code	Enables call waiting, caller ID generation. The default is *25.
CWCID Deact Code	Disables call waiting, caller ID generation. The default is *45.
Dist Ring Act Code	Enables the distinctive ringing feature. The default is *26
Dist Ring Deact Code	Disables the distinctive ringing feature. The default is *46.
Speed Dial Act Code	Assigns a speed dial number. The default is *74.
Secure All Call Act Code	Makes all outbound calls secure. The default is *16.
Secure No Call Act Code	Makes all outbound calls not secure. The default is *17.
Secure One Call Act Code	Makes the next outbound call secure. (It is redundant if all outbound calls are secure by default.) The default is *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.) The default is *19.
Paging Code	The star code used for paging the other clients in the group. The default is *96.
Call Park Code	The star code used for parking the current call. The default is *38.
Call Pickup Code	The star code used for picking up a ringing call. The default is *36.
Call UnPark Code	The star code used for picking up a call from the call park. The default is *39.
Group Call Pickup Code	The star code used for picking up a group call. The default is *37.

Regional Tab

Media Loopback Code	<p>The star code used for enabling media loopback on the phone. The default is *03.</p>
Referral Services Codes	<p>These codes tell the 900 Series 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, and so forth. 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 900 Series 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 900 Series phone plays a special dial tone called the Prompt Tone while waiting for the user the enter a target number (which is checked according to dial plan as in normal dialing). When a complete number is entered, the 900 Series phone sends a blind REFER to the holding party with the Refer-To target equals to *98&lt;target_number&gt;. This feature allows the 900 Series 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 900 Series phone. You can empty the corresponding *code that you do not want to 900 Series phone to process.</p>

<p>Feature Dial Services Codes</p>	<p>These codes tell the 900 Series 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, and so forth. 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 900 Series phone to call the target number prepended by the *code. For example, after user dials *72, the 900 Series phone plays a prompt tone awaiting the user to enter a valid target number. When a complete number is entered, the 900 Series phone sends a INVITE to *72&lt;target_number&gt; 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 900 Series phone. You can empty the corresponding *code that you do not want to 900 Series 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"> <li>• 'c' = &lt;Cfwd Dial Tone&gt;</li> <li>• 'd' = &lt;Dial Tone&gt;</li> <li>• 'm' = &lt;MWI Dial Tone&gt;</li> <li>• 'o' = &lt;Outside Dial Tone&gt;</li> <li>• 'p' = &lt;Prompt Dial Tone&gt;</li> <li>• 's' = &lt;Second Dial Tone&gt;</li> <li>• 'x' = No tones are place, x is any digit not used above</li> </ul> <p>If no tone parameter is specified, the 900 Series 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 900 Series phone send INVITE *73@..... as usual when user dials *73.</p>
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## 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).

Field	Description
Prefer G711u Code	<p>Makes this codec the preferred codec for the associated call.</p> <p>The default is *017110.</p>

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Force G711u Code	Makes this codec the only codec that can be used for the associated call. The default is *027110.
Prefer G711a Code	Makes this codec the preferred codec for the associated call. The default is *017111
Force G711a Code	Makes this codec the only codec that can be used for the associated call. The default is *027111.
Prefer G723 Code	Makes this codec the preferred codec for the associated call. The default is *01723.
Force G723 Code	Makes this codec the only codec that can be used for the associated call. The default is *02723.
Prefer G726r16 Code	Makes this codec the preferred codec for the associated call. The default is *0172616.
Force G726r16 Code	Makes this codec the only codec that can be used for the associated call. The default is *0272616.
Prefer G726r24 Code	Makes this codec the preferred codec for the associated call. The default is *0172624.
Force G726r24 Code	Makes this codec the only codec that can be used for the associated call. The default is *0272624.
Prefer G726r32 Code	Makes this codec the preferred codec for the associated call. The default is *0172632.
Force G726r32 Code	Makes this codec the only codec that can be used for the associated call. The default is *0272632.
Prefer G726r40 Code	Makes this codec the preferred codec for the associated call. The default is *0172640.
Force G726r40 Code	Makes this codec the only codec that can be used for the associated call. The default is *0272640.
Prefer G729a Code	Makes this codec the preferred codec for the associated call. The default is *01729.
Force G729a Code	Makes this codec the only codec that can be used for the associated call. The default is *02729.

## Miscellaneous

Field	Description
Set Local Date (mm/dd)	Sets the local date (mm stands for months and dd stands for days). The year is optional and uses two or four digits.
Set Local Time (HH/mm)	Sets the local time (hh stands for hours and mm stands for 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. The default is GMT-08:00.
Time Offset (HH/mm)	This specifies the offset from GMT to use for the local system time.
FXS Port Impedance	Sets the electrical impedance of the FXS port. Choices are 600, 900, 600+2.16uF, 900+2.16uF, 270+750  150nF, 220+850  120nF, 220+820  115nF, or 200+600  100nF. The default is 600.

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 = &lt;start-time&gt;; end=&lt;end-time&gt;; save = &lt;save-time&gt;.</p> <ul style="list-style-type: none"> <li>• The &lt;start-time&gt; and &lt;end-time&gt; values specify the start and end dates and times of daylight saving time. Each value is in this format: &lt;month&gt; /&lt;day&gt; / &lt;weekday&gt;[/HH:[mm[:ss]]]</li> <li>• The &lt;save-time&gt; value is the number of hours, minutes, and/or seconds to add to the current time during daylight saving time. The &lt;save-time&gt; value can be preceded by a negative (-) sign if subtraction is desired instead of addition. The &lt;save-time&gt; value is in this format: [/[+/-]HH:[mm[:ss]]]</li> <li>• The &lt;month&gt; value equals any value in the range 1-12 (January-December).</li> <li>• The &lt;day&gt; value equals [+/-] any value in the range 1-31.</li> <li>• If &lt;day&gt; is 1, it means the &lt;weekday&gt; on or before the end of the month (in other words the last occurrence of &lt; weekday&gt; in that month).</li> <li>• The &lt;weekday&gt; value equals any value in the range 1-7 (Monday-Sunday). It can also equal 0. If the &lt;weekday&gt; value is 0, this means that the date to start or end daylight saving is exactly the date given. In that case, the &lt;day&gt; value must not be negative. If the &lt;weekday&gt; value is not 0 and the &lt;day&gt; value is positive, then daylight saving starts or ends on the &lt;weekday&gt; value on or after the date given. If the &lt;weekday&gt; value is not 0 and the &lt;day&gt; value is negative, then daylight saving starts or ends on the &lt;weekday&gt; value on or before the date given.</li> <li>• The abbreviation HH stands for hours (0-23).</li> <li>• The abbreviation mm stands for minutes (0-59).</li> <li>• The abbreviation ss stands for seconds (0-59).</li> </ul> <p>The default Daylight Saving Time Rule is start=4/1/7;end=10/-1/7;save=1.</p>
FXS Port Input Gain	<p>Input gain in dB, up to three decimal places. The range is 6.000 to -12.000. The default is -3.</p>
FXS Port Output Gain	<p>Output gain in dB, up to three decimal places. The range is 6.000 to -12.000. The Call Progress Tones and DTMF playback level are not affected by the &lt;FXS Port Output Gain&gt;.</p> <p>The default is -3.</p>
DTMF Playback Level	<p>Local DTMF playback level in dBm, up to one decimal place.</p> <p>The default is -16.0.</p>
DTMF Playback Length	<p>Local DTMF playback duration in milliseconds.</p> <p>The default is .1.</p>



<p>Inband DTMF Boost</p>	<p>Controls the amount of amplification applied DTMF signals. Choices are 3dB, 6dB, 9dB, 12dB, 15dB, or 18dB. The default is 12dB.</p>
<p>Dictionary Server Script</p>	<p>Defines the location of the dictionary server, the languages available and the associated dictionary. The syntax is as follows:</p> <pre>&lt;Dictionary_Server_Script ua="na"&gt; &lt;/Dictionary_Server_Script&gt;</pre> <p>The default is 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=&lt;language0&gt;;x0=&lt;dictionary0 filename&gt;; d1=&lt;language1&gt;;x1=&lt;dictionary1 filename&gt;; d2=&lt;language2&gt;;x2=&lt;dictionary2 filename&gt;; d3=&lt;language3&gt;;x3=&lt;dictionary3 filename&gt;; d4=&lt;language4&gt;;x4=&lt;dictionary4 filename&gt;; d5=&lt;language5&gt;;x5=&lt;dictionary5 filename&gt;; d6=&lt;language6&gt;;x6=&lt;dictionary6 filename&gt;; d7=&lt;language3&gt;;x7=&lt;dictionary7 filename&gt;; d8=&lt;language8&gt;;x8=&lt;dictionary8 filename&gt;; d9=&lt;language5&gt;;x9=&lt;dictionary9 filename&gt;;</pre> <p>The following is an example value:</p> <pre>&lt;Dictionary_Server_Script ua="na"&gt; serv=tftp://192.168.1.119/;d0=English;x0=enS_v101.xml;d1=Spanish;x1=esS_v101.xml &lt;/Dictionary_Server_Script&gt;</pre>
<p>Language Selection</p>	<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>&lt;Language_Selection ua="na"&gt; &lt;/Language_Selection&gt;</pre> <p>The default is blank and the maximum number of characters is 512. The following is an example:</p> <pre>&lt;Language_Selection ua="na"&gt; Spanish &lt;/Language_Selection&gt;</pre>

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<p>Caller ID Method</p>	<p>The following choices are available:</p> <ul style="list-style-type: none"> <li>• Bellcore (N.Amer,China)—CID, CIDCW, and VMWI. FSK sent after first ring (same as ETSI FSK sent after first ring) (no polarity reversal or DTAS).</li> <li>• DTMF (Finland, Sweden)—CID only. DTMF sent after polarity reversal (and no DTAS) and before first ring.</li> <li>• DTMF (Denmark)—CID only. DTMF sent after polarity reversal (and no DTAS) and before first ring.</li> <li>• ETSI DTMF—CID only. DTMF sent after DTAS (and no polarity reversal) and before first ring.</li> <li>• ETSI DTMF With PR—CID only. DTMF sent after polarity reversal and DTAS and before first ring.</li> <li>• ETSI DTMF After Ring—CID only. DTMF sent after first ring (no polarity reversal or DTAS).</li> <li>• ETSI FSK—CID, CIDCW, and VMWI. FSK sent after DTAS (but no polarity reversal) and before first ring. Waits for ACK from CPE after DTAS for CIDCW.</li> <li>• ETSI FSK With PR (UK)—CID, CIDCW, and VMWI. FSK is sent after polarity reversal and DTAS and before first ring. Waits for ACK from CPE after DTAS for CIDCW. Polarity reversal is applied only if equipment is on hook.</li> </ul> <p>The default is Bellcore(N.Amer, China).</p>
<p>Caller ID FSK Standard</p>	<p>The 900 Series phone supports bell 202 and v.23 standards for caller ID generation. Select the FSK standard you want to use, bell 202 or v.23.</p> <p>The default is bell 202.</p>
<p>Feature Invocation Method</p>	<p>Select the method you want to use, Default or Sweden default. (Not in PAP2T)</p> <p>The default is Default.</p>

# Phone Tab

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

[General, page 5-32](#)

[Line Key 1/2/3/4/5/6, page 5-32](#)

[Miscellaneous Line Key Settings, page 5-32](#)

[Line Key LED Pattern, page 5-33](#)

[Supplementary Services, page 5-34](#)

[Ring Tone, page 5-35](#)

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

[Background Picture \(SPA 962\), page 5-36](#)

## 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	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".

## 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).

Field	Description
Extension	Select an extension to be used for this line key. Choices are 1,2,3, or 4 for a four-line IP phone and 1 or 2 for a two-line IP phone. The default is 1.
Short Name	A short label shown on the LCD display for Line key 1/2/3/4.
Share Call Appearance	Yes indicates that Line Key 1/2/3/4 is a shared call appearance. Otherwise this call appearance is not shared (in other words it is private). The default is no.

## Miscellaneous Line Key Settings

Field	Description
SCA Line ID Mapping	<p>Specifies the shared call appearance line ID mapping. Choose Vertical First or Horizontal First. Each LED can hold two calls and the first call on an LED makes it light up. Horizontal first means the second call makes the same LED flash. Vertical first means the second call lights up the next LED.</p> <p>For example, if Extension 101 is assigned to two LEDs, and Vertical First is selected, the second call on Extension 101 lights up the second LED. The third call makes the first LED flash, and the fourth call makes the second LED flash.</p> <p>If Horizontal First is selected, the second call on Extension 101 makes the first LED flash. The third call lights up the second LED, and the fourth call makes the second LED flash.</p>

## Line Key LED Pattern

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.
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.

Field	Description
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.
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=0.
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.
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. The default value for these parameters is yes.

Field	Description
Conference Serv	Enable/disable Three way conference service
Attn Transfer Serv	Enable/disable attended-call-transfer service
Blind Transfer Serv	Enable/disable blind-call-transfer service
DND (Do Not Disturb) Serv	Enable/disable do-not-disturb service
Block ANC (Anonymous Call) Serv	Enable/disable block-anonymous-call service
Call Back Serv	Enable/disable call-back (a.k.a. repeating dialing) service

Field	Description
Block CID (Caller ID) Serv	Enable/disable blocking outbound Caller-ID service
Secure Call Serv	Enable/disable secure-call service
CFWD All Serv	Enable/disable call-forward-all service
CFWD Busy Serv	Enable/disable call-forward-on-busy service
CFWD On No Answer Serv	Enable/disable call-forward-on-no-answer service
Paging Serv	Enable/disable the paging service
Call Park Serv	Enable/disable the call park service
Call Pick Up Serv	Enable/disable the call pickup service
ACD Login Serv	Enable/disable the ACD Login Service, used for call centers. Typically enabled with the <SIP-B> parameter.
Group Call Pick Up Serv	Enable/disable the group call pickup service
ACD Ext	The extension used for handling ACD calls. Default 1 or 2

## 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. The default is n=Classic-1;w=3;c=1.
Ring 2	Ring tone script for Ring 2. The default is n=Classic-2;w=3;c=2.
Ring 3	Ring tone script for Ring 3. The default is n=Classic-3;w=3;c=3.
Ring 4	Ring tone script for Ring 4. The default is n=Classic-4;w=3;c=4.
Ring 5	Ring tone script for Ring 5. The default is n=Simple-1;w=2;c=1.
Ring 6	Ring tone script for Ring 6. The default is n=Simple-2;w=2;c=2.
Ring 7	Ring tone script for Ring 7. The default is n=Simple-3;w=2;c=3.
Ring 8	Ring tone script for Ring 8. The default is n=Simple-4;w=2;c=4.
Ring 9	Ring tone script for Ring 9. The default is n=Simple-5;w=2;c=5.
Ring 10	Ring tone script for Ring 10. The default is n=THx;w=4;c=7.

## Auto Input Gain (dB)

Field	Description
Handset Input Gain	The amount of amplification to apply to the audio input signal for the handset.
Headset Input Gain	The amount of amplification to apply to the audio input signal for the headset.
Speakerphone Input Gain	The amount of amplification to apply to the audio input signal for the speakerphone.

## Background Picture (SPA 962)

Field	Description
Picture download URL	<p>The URL identifying the location of the .BMP file to be used as the background image, including the TFTP server name (or IP address), the directory, and the filename. For example:</p> <pre>tftp://myserver.mydomain.com/images/962back.bmp</pre> <p>The background image should be 320X240 pixels and must be in .BMP format.</p>
Current Background Picture	Select Default to use the default image; select Downloaded to use a customized background image.

# Ext 1/2/3/4/5/6 Tab

This section describes the fields for the following headings on the Ext 1 and Ext 2 tabs:

- [General, page 5-36](#)
- [Share Line Appearance, page 5-36](#)
- [NAT Settings, page 5-37](#)
- [Network Settings, page 5-37](#)
- [SIP Settings, page 5-38](#)
- [Subscriber Information, page 5-43](#)
- [Dial Plan, page 5-45](#)



**Note**

In a configuration profile, the Line parameters must be appended with [1] or [2] to identify the line to which the setting applies.

## General

Field	Description
Line Enable	To enable this line for service, select yes. Otherwise, select no. The default is 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. The default is shared.
Shared User ID	The user identified assigned to the shared line appearance.
Subscription Expires	The subscription time with the SIP proxy (SPA9000). The default is 3600 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. The default is no.
NAT Keep Alive Enable	To send the configured NAT keep alive message periodically, select yes. Otherwise, select no. The default is no.
NAT Keep Alive Msg	Enter the keep alive message that should be sent periodically to maintain the current NAT mapping. If the value is \$NOTIFY, a NOTIFY message is sent. If the value is \$REGISTER, a REGISTER message without contact is sent. The default is \$NOTIFY.
NAT Keep Alive Dest	Destination that should receive NAT keep alive messages. If the value is \$PROXY, the messages are sent to the current or outbound proxy. The default is \$PROXY.

## Network Settings

Field	Description
SIP ToS/DiffServ Value	TOS/DiffServ field value in UDP IP packets carrying a SIP message. The default is 0x68.
SIP CoS Value [0-7]	CoS value for SIP messages. The default is 3.
RTP ToS/DiffServ Value	ToS/DiffServ field value in UDP IP packets carrying RTP data. The default is 0xb8.
RTP CoS Value [0-7]	CoS value for RTP data. The default is 6.
Network Jitter Level	Determines how jitter buffer size is adjusted by the 900 Series 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. The default is 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.  The default is up and down.
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## SIP Settings

Field	Description
SIP Transport	Transport type used for SIP messages. The options are UDP, TCP, or TLS. The default is UDP.
SIP Port	Port number of the SIP message listening and transmission port. The default is 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.  The default is no.
EXT SIP Port	The external SIP port number.
Auth Resync-Reboot	If this feature is enabled, the 900 Series phone authenticates the sender when it receives the NOTIFY resync reboot (RFC 2617) message. To use this feature, select yes. Otherwise, select no.  The default is yes.
SIP Proxy-Require	The SIP proxy can support a specific extension or behavior when it sees this header from the user agent. If this field is configured and the proxy does not support it, it responds with the message, unsupported. Enter the appropriate header in the field provided.
SIP Remote-Party-ID	To use the Remote-Party-ID header instead of the From header, select yes. Otherwise, select no.  The default is yes.
SIP GUID	The Global Unique ID is generated for each line for each device. When it is enabled, the 900 Series phone adds a GUID header in the SIP request. The GUID is generated the first time the unit boots up and stays with the unit through rebooting and even factory reset. This feature was requested by Bell Canada (Nortel) to limit the registration of SIP accounts.  The default is yes.

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"> <li>• none—No logging.</li> <li>• 1-line—Logs the start-line only for all messages.</li> <li>• 1-line excl. OPT—Logs the start-line only for all messages except OPTIONS requests/responses.</li> <li>• 1-line excl. NTFY—Logs the start-line only for all messages except NOTIFY requests/responses.</li> <li>• 1-line excl. REG—Logs the start-line only for all messages except REGISTER requests/responses.</li> <li>• 1-line excl. OPT NTFY REG—Logs the start-line only for all messages except OPTIONS, NOTIFY, and REGISTER requests/responses.</li> <li>• full—Logs all SIP messages in full text.</li> <li>• full excl. OPT—Logs all SIP messages in full text except OPTIONS requests/responses.</li> <li>• full excl. NTFY—Logs all SIP messages in full text except NOTIFY requests/responses.</li> <li>• full excl. REG—Logs all SIP messages in full text except REGISTER requests/responses.</li> <li>• full excl. OPT NTFY REG—Logs all SIP messages in full text except for OPTIONS, NOTIFY, and REGISTER requests/responses.</li> </ul> <p>The default is none.</p>
RTP Log Intvl	The interval for the RTP log.
Restrict Source IP	<p>If Lines 1 and 2 use the same SIP Port value and the Restrict Source IP feature is enabled, the proxy IP address for Lines 1 and 2 is treated as an acceptable IP address for both lines. To enable the Restrict Source IP feature, select yes. Otherwise, select no. If configured, the PAP2T will drop all packets sent to its SIP Ports originated from an untrusted IP address. A source IP address is untrusted if it does not match any of the IP addresses resolved from the configured &lt;Proxy&gt; (or &lt;Outbound Proxy&gt; if &lt;Use Outbound Proxy&gt; is yes).</p> <p>The default is no.</p>
Referor Bye Delay	<p>Controls when the 900 Series 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>The default is 4.</p>
Refer Target Bye Delay	<p>For the Refer Target Bye Delay, enter the appropriate period of time in seconds.</p> <p>The default is 0.</p>

Referee Bye Delay	For the Referee Bye Delay, enter the appropriate period of time in seconds. The default is 0.
Refer-To Target Contact	To contact the refer-to target, select yes. Otherwise, select no. The default is no.
Sticky 183	If this feature is enabled, the IP telephony ignores further 180 SIP responses after receiving the first 183 SIP response for an outbound INVITE. To enable this feature, select yes. Otherwise, select no. The default is no.
Auth INVITE	When enabled, authorization is required for initial incoming INVITE requests from the SIP proxy.

## Call Feature Settings

Field	Description
Blind Attn-Xfer Enable	Enables the 900 Series 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 900 Series 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. The default is no.
MOH Server	User ID or URL of the auto-answering streaming audio server. When only a user ID is specified, the current or outbound proxy is contacted. Music-on-hold is disabled if the MOH Server is not specified.
Message Waiting	Indicates whether the Message Waiting Indicator on the phone is lit. This parameter is toggled by a message from the SIP proxy to indicate if a message is waiting. The default is No.
Auth Page	Specifies whether to authenticate the invite before auto answering a page. The Auto Answer Page parameter, on the Supplementary Services section of the User tab, is set to Yes by default. The default is No.
Default Ring	Type of ring heard. This corresponds to the Ring Tone on the Phone tab.
Auth Page Realm	Identifies the Realm part of the Auth that is accepted when the Auth Page parameter is set to Yes. This parameter accepts alphanumeric characters.

Conference Bridge URL	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> ”
Auth Page Password	Identifies the password used when the Auth Page parameter is set to Yes. This parameter accepts alphanumeric characters.
Mailbox ID	Identifies the voicemail box number/ID for the phone.
Voice Mail Server	Identifies the SpecVM server for the phone, generally the IP address and port number of the VM server.
State Agent	Reserved feature.
CFWD Notify Serv	Specifies whether to enable a SIP-B feature regarding the sending of a Notify to the phone when a call is forwarded elsewhere. The default is No.
CFWD Notifier	Typically, this field is configured with the SIP proxy information.

## Proxy and Registration

Field	Description
Proxy	SIP proxy server for all outbound requests.
Use Outbound Proxy	Enable the use of <Outbound Proxy>. If set to no, <Outbound Proxy> and <Use OB Proxy in Dialog> is ignored. The default is 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. The default is yes.
Register	Enable periodic registration with the <Proxy>. This parameter is ignored if <Proxy> is not specified. The default is yes.
Make Call Without Reg	Allow making outbound calls without successful (dynamic) registration by the unit. If No, dial tone will not play unless registration is successful. The default is no.

Field	Description
Register Expires	Allow answering inbound calls without successful (dynamic) registration by the unit. If proxy responded to REGISTER with a smaller Expires value, the PAP2T will renew registration based on this smaller value instead of the configured value. If registration failed with an Expires too brief error response, the PAP2T will retry with the value given in the Min-Expires header in the error response. The default is 3600.
Ans Call Without Reg	Expires value in sec in a REGISTER request. PAP2T will periodically renew registration shortly before the current registration expired. This parameter is ignored if <Register> is no. Range: 0 – (231 – 1) sec
Use DNS SRV	Whether to use DNS SRV lookup for Proxy and Outbound Proxy. The default is no.
DNS SRV Auto Prefix	If enabled, the PAP2T will automatically prepend the Proxy or Outbound Proxy name with _sip._udp when performing a DNS SRV lookup on that name. The default is no.
Proxy Fallback Intvl	This parameter sets the delay (sec) after which the PAP2T 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 PAP2T 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 PAP2T will not attempt to fall back after a fail over). The default is 3600
Proxy Redundancy Method	PAP2T will make an internal list of proxies returned in DNS SRV records. In normal mode, this list will contain proxies ranked by weight and priority. if Based on SRV port is configured the PAP2T does normal first, and also inspect the port number based on 1 <sup>st</sup> proxy's port on the list. The default is Normal.

## Subscriber Information

Field	Description
Display Name	Display name for caller ID.
User ID	Extension number for this line.
Password	Password for this line.

Use Auth ID	To use the authentication ID and password for SIP authentication, select yes. Otherwise, select no to use the user ID and password. The default is no.
Auth ID	Authentication ID for SIP authentication.
Cfwd No Ans Delay	Delay, in seconds, before the call forwarding of no-answer calls feature is triggered. The default is 20.
Mini Certificate	Base64 encoded of Mini-Certificate concatenated with the 1024-bit public key of the CA signing the MC of all subscribers in the group. The default is empty.
SRTP Private Key	Base64 encoded of the 512-bit private key per subscriber for establishment of a secure call. The default is empty.

## 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 G.723/G.726 codec.

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. The default is G711u.
Silence Supp Enable	To enable silence suppression so that silent audio frames are not transmitted, select yes. Otherwise, select no. The default is 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. The default is no.

Silence Threshold	Select the appropriate setting for the threshold: high, medium, or low. The default is medium.
G729a Enable	To enable the use of the G.729a codec at 8 kbps, select yes. Otherwise, select no. The default is yes.
Echo Canc Enable	To enable the use of the echo canceller, select yes. Otherwise, select no. The default is yes.
G723 Enable	To enable the use of the G.723a codec at 6.3 kbps, select yes. Otherwise, select no. The default is yes.
Echo Canc Adapt Enable	To enable the echo canceller to adapt, select yes. Otherwise, select no. The default is yes.
G726-16 Enable	To enable the use of the G.726 codec at 16 kbps, select yes. Otherwise, select no. The default is yes.
Echo Supp Enable	To enable the use of the echo suppressor, select yes. Otherwise, select no. The default is yes.
G726-24 Enable	To enable the use of the G.726 codec at 24 kbps, select yes. Otherwise, select no. The default is yes.
FAX CED Detect Enable	To enable detection of the fax Caller-Entered Digits (CED) tone, select yes. Otherwise, select no. The default is yes.
G726-32 Enable	To enable the use of the G.726 codec at 32 kbps, select yes. Otherwise, select no. The default is yes.
G726-40 Enable	To enable the use of the G.726 codec at 40 kbps, select yes. Otherwise, select no. The default is yes.
DTMF Process INFO	(Not in PAP2T) To use the DTMF process info feature, select yes. Otherwise, select no. The default is yes.
DTMF Process AVT	(Not in PAP2T) To use the DTMF process AVT feature, select yes. Otherwise, select no. The default is yes.



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.  The default is Auto.
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.  The default is yes.

## Dial Plan

The default dial plan script for each line is as follows:  
(\*xx|[3469]11|0|00|[2-9]xxxxxx|1xxx[2-9]xxxxxx|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	Int'l 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 SPA3102 to allow the designation of three parameters to be used with a specific gateway:</p> <ul style="list-style-type: none"> <li>• uid – the authentication user-id</li> <li>• pwd – the authentication password</li> <li>• nat – if this parameter is present, use NAT mapping</li> </ul> <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 SPA3102 unit.</p> <p>Example 1:</p> <pre>*1xxxxxxxxxx&lt;:@fwdnat.pulver.com:5082;uid=jsmith;pwd=xyz</pre> <p>Example 2:</p> <pre>*1xxxxxxxxxx&lt;:@fwd.pulver.com;nat;uid=jsmith;pwd=xyz</pre> <p>Example 3:</p> <pre>[39]11&lt;:@gw0&gt;</pre>
Enable IP Dialing	<p>Enable or disable IP dialing.</p> <p>The default is no.</p>

## User Tab

This section describes the fields for the following headings on the User 1 and User 2 tabs:

- [Call Forward, page 5-47](#)
- [Speed Dial, page 5-48](#)
- [Supplementary Services, page 5-48](#)
- [Audio Volume, page 5-49](#)
- [Phone GUI Menu Color Settings \(SPA962\), page 5-49](#)

User 1/2 refers to the subscriber of Line 1 or Line 2. When a call is made from Line 1 or Line 2, 900 Series phone shall use the user and line settings for that Line; there is no user login support in 900 Series phone v1.0. Per user parameter tags must be appended with [1] or [2] (corresponding to line 1 or 2) in the configuration profile. It is omitted below for readability.

## Call Forward

Field	Description
Cfwd All Dest	<p>Forward number for Call Forward All Service</p> <p>In addition to normal call forward destination as used in the other ATAs, on the SPA3102, you can specify the following additional parameters:</p> <ul style="list-style-type: none"> <li>• gw0 – forward the caller to use the PSTN gateway</li> <li>• &lt;pstn-number&gt;@gw0 – forward to caller to the PSTN number (dialed automatically by the SPA through the PSTN gateway)</li> </ul> <p>The default is blank.</p>
Cfwd Busy Dest	<p>Forward number for Call Forward Busy Service. Same as Cfwd All Dest.</p> <p>The default is blank.</p>
Cfwd No Ans Dest	<p>Forward number for Call Forward No Answer Service. Same as Cfwd All Dest.</p> <p>In addition to normal call forward destination as used in the other ATAs, on the SPA3102, you can specify the following additional parameters:</p> <ul style="list-style-type: none"> <li>• gw0 – forward the caller to use the PSTN gateway</li> <li>• &lt;pstn-number&gt;@gw0 – forward to caller to the PSTN number (dialed automatically by the SPA through the PSTN gateway)</li> </ul> <p>The default is blank.</p>
Cfwd No Ans Delay	<p>Delay in sec before Call Forward No Answer triggers. Same as Cfwd All Dest.</p> <p>The default is 20.</p>

## Speed Dial

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. The default is blank.

## Supplementary Services

The 900 Series 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 900 Series phone.

Field	Description
CW Setting	Enables or disables call waiting. The default is Yes.
Block CID Setting	Block Caller ID on/off for all calls. The default is no.
Block ANC Setting	Block Anonymous Calls on or off. The default is no.
DND Setting	DND on or off. The default is no.
Secure Call Setting	If yes, all outbound calls are secure calls by default. The default is no.
Dial Assistance	If enabled, the phone LCD displays a list of up to 10 matching numbers as the user presses each key on the numeric keypad.
Auto Answer Page	If enabled, the phone automatically picks up a call and directs the audio signal to the speakerphone without user intervention.
Preferred Audio Device	Specifies the preferred audio device (Speaker or Headset). When set to Speaker, the user can enable the headset using the Preferences > Preferred Audio Device (Menu 5,6) option. When set to Headset, the same option is used to enable the speakerphone.
Send Audio to Speaker	If enabled, when the user is using the handset, the audio from the remote end will be played out on the speaker in addition to the handset (so that a supervisor and listen in).

Field	Description
Time Format	Toggles the LCD display between a 12-hour and 24-hour clock.
Date Format	Changes the order in which month and day are displayed.
Miss Call Shortcut	Enables or disables the display of missed calls on the LCD display.
Accept Media Loopback Request	Determines how the media loopback request is enabled. Choose automatic, never, or manual. The default is automatic.
Media Loopback Mode	Determines the media loopback mode. Choose source or mirror.
Media Loopback Type	Determines the media loopback type. Choose media or packet.

## Audio Volume

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.
Lcd Contrast	Sets the default contrast setting for the LCD display.

## Phone GUI Menu Color Settings (SPA962)

Field	Description
Color Scheme	Select the color scheme for the color LCD display on the SPA962. The choices are Silver, Light Blue, Light Green, or Pink. The default is Silver.



## Acronyms

A/D	Analog To Digital Converter
ANC	Anonymous Call
B2BUA	Back to Back User Agent
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
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



## Glossary

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**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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