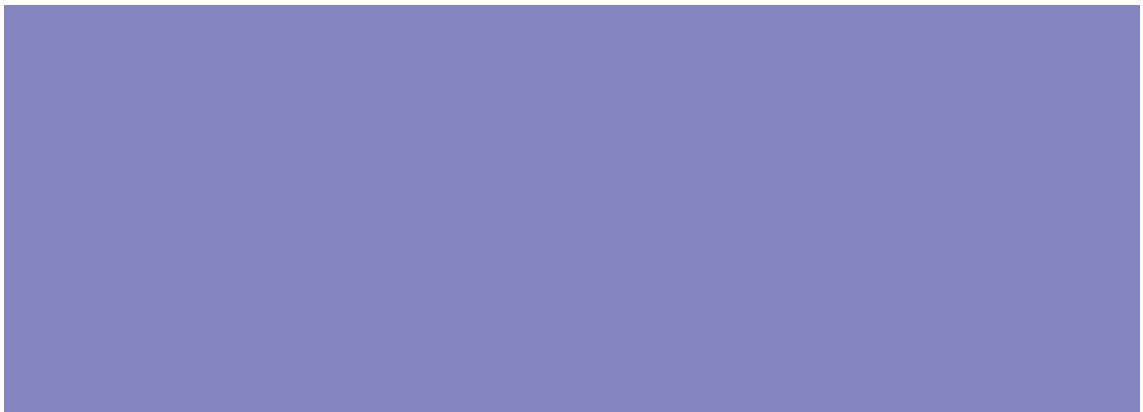




Linksys ATA Administrator Guide

Document Version 3.2



Corporate Headquarters

Linksys
121 Theory Drive
Irvine, CA 92617
USA
<http://www.linksys.com>
Tel: 949 823-1200
800 546-5797
Fax: 949 823-1100



Linksys ATA Administrator Guide

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Preface

This guide describes administration and use of the Linksys Analog Telephone Adapters (ATAs). It contains the following sections:

- [Document Audience, page xi](#)
- [Linksys Analog Telephone Adapters, 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 Linksys VoIP products
- VARs and resellers who need configuration information for Linksys VoIP products
- System administrators or anyone who performs Linksys VoIP product installation and administration



Note

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

Linksys Analog Telephone Adapters

The following summarizes the ports and features provided by the Linksys ATAs described in this document.

- PAP2T—Voice adapter with two FXS ports
- SPA1001—Small VoIP adapter
- SPA2102—Voice adapter with router

- SPA3102—Voice adapter with router and PSTN connectivity
- SPA8000—Voice adapter supporting up to eight FXS connections
- AG310—ADSL2+ gateway with VoIP and PSTN connectivity
- RTP300—IP router with two FXS ports
- WRP400—Wireless-G IP router with FXS ports
- WRT54G—Wireless-G IP router with two FXS ports
- WRT54GP2—Wireless-G IP router with two FXS ports

How This Document is Organized

This document is divided into the following chapters and appendices.

| Chapter | Contents |
|--|--|
| Chapter 1, “Introducing Linksys Analog Telephone Adapters” | This chapter introduces the Linksys Analog Telephone Adapters (ATAs). |
| Chapter 2, “Getting Started” | This chapter describes how to use the different administration and configuration tools provided for managing Linksys ATAs. |
| Chapter 3, “Configuring Linksys ATAs” | This chapter describes how to complete the basic configuration of a Linksys ATA. |
| Chapter 4, “Configuring the PSTN Gateway (FXO)” | This chapter describes how to configure the Linksys SPA3102 and AG310 for providing PSTN connectivity. |
| Chapter 5, “Linksys ATA Field Reference” | This chapter lists the function and usage for each field or parameter on the ATA administration web server pages. |
| Appendix A, “Acronyms” | This appendix provides the expansion of acronyms used in this document. |
| Appendix B, “Glossary” | This appendix defines the terms used in this document. |
| Appendix C, “User Guidelines” | This appendix summarizes the operations of ATA user features. |

Document Conventions

The following are the typographic conventions used in this document.

| Typographic Element | Meaning |
|---------------------|---|
| Boldface | 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 Linksys ATA administration web server. The index at the end of this document contains an alphabetical listing of each parameter, hyperlinked to the appropriate table in Chapter 5, “Linksys ATA Field Reference” |
| <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 ATAs:

- *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*

Technical Support

If you are an end user of Linksys VoIP 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:

- Linksys VoIP 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 Analog Telephone Adapters

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

- [Overview, page 1-1](#)
- [Ensuring Voice Quality, page 1-3](#)
- [Feature Descriptions, page 1-5](#)
- [Technology Background, page 1-9](#)
- [Where to Go From Here, page 1-13](#)

Overview

Table 1-1 summarizes the ports and features provided by the Linksys ATAs described in this document.

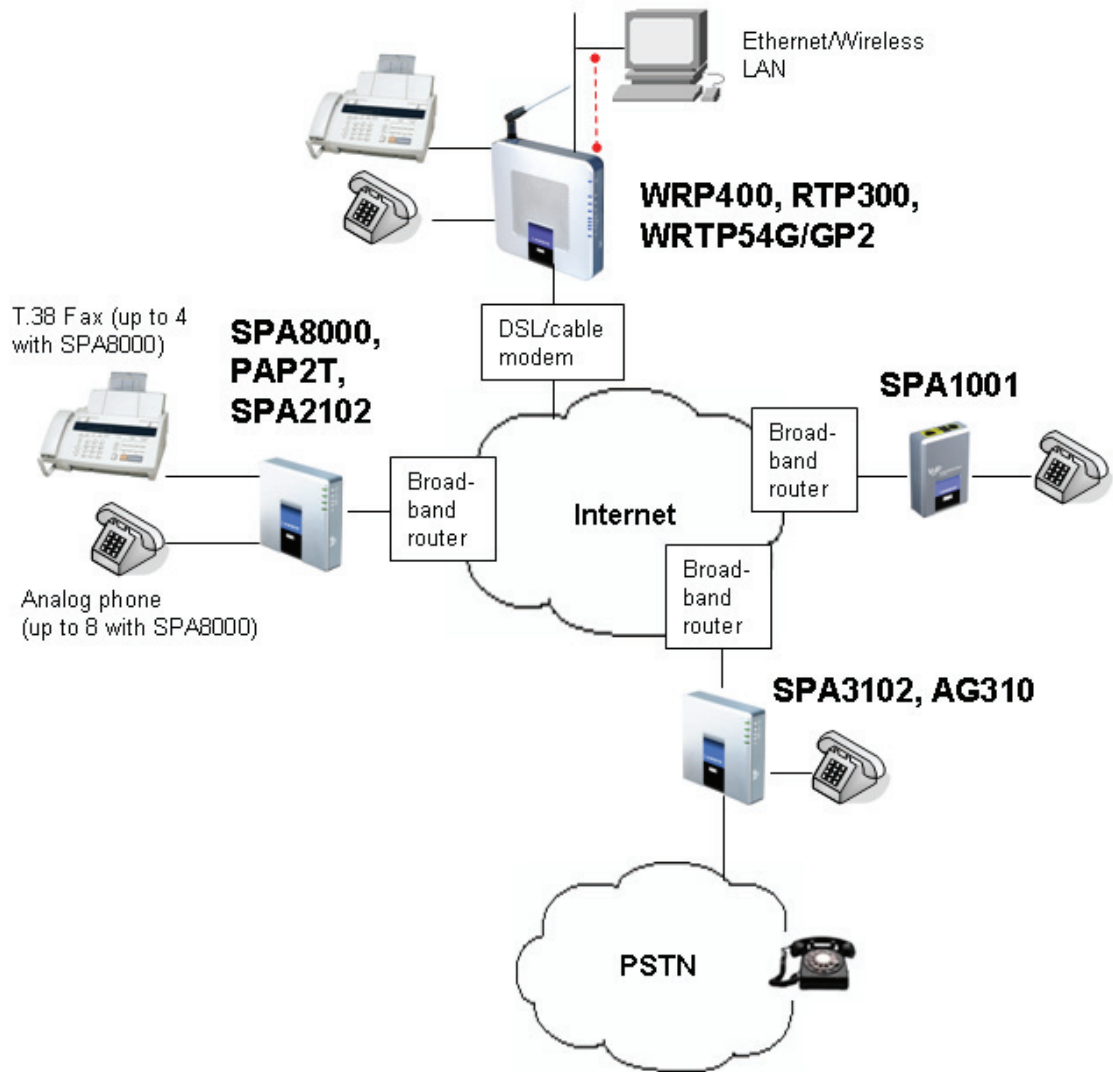
Table 1-1 Linksys ATAs

| Product Name | FXS (Analog Phone) | FXO PSTN Connection | RJ-45 Internet (WAN) | RJ-45 Ethernet (LAN) | Configurable Voice Lines | Description |
|-------------------------------|--------------------|---------------------|----------------------|----------------------|--------------------------|--|
| PAP2T | Two (2) | — | One (1) | — | Two (2) | Voice adapter with two FXS ports |
| SPA1001 | One (1) | — | One (1) | — | One (1) | Small VoIP adapter |
| SPA2102 | Two (2) | — | One (1) | — | One (1) | Voice adapter with router |
| SPA3102 | One (1) | One (1) | One (1) | One (1) | Two (2) | Voice adapter with router and PSTN connectivity |
| SPA8000 | Eight (8) | — | One (1) | Maintenance only | Four (4) | Voice adapter with support for up to eight FXS devices |
| RTP300 | Two (2) | — | One (1) | Four (4) | Two (2) | IP router with two FXS ports |
| WRP400 WRTP54G WRT54GP2 | Two (2) | — | One (1) | Four (4) | Two (2) | Wireless-G IP router with two FXS ports |
| AG310 | One (1) | One (1) | One (1) | One (1) | One (1) | ADSL2+ gateway with VoIP and PSTN connectivity |

Figure 1-1 illustrates how the different ATAs provide voice connectivity in a VoIP network, including the SPA3102, which acts as a SIP-PSTN gateway. As shown, the following devices also provide QoS-enabled IP routers in addition to ports for connecting analog telephone devices:

- WRP400
- RTP300
- WRTP54G
- WRTP54GP2

Figure 1-1 Linksys ATAs in the VoIP Network and PSTN



The AG310 and SPA3102 provide full PSTN connectivity in addition to a single FXS port. In addition, the AG310 provides an ADSL2+ gateway.

Each Linksys ATA is an intelligent low-density Voice over IP (VoIP) gateway that enables carrier-class residential and business IP Telephony services delivered over broadband or high-speed Internet connections. Linksys ATAs maintain the states of all the calls it terminates and makes the proper reaction to user input events (such as on/off hook or hook flash). Because the ATAs use the SIP standard, there is little or no involvement by a “middle-man” server or media gateway controller.

The response of a Linksys ATA includes playing a dial tone, collecting DTMF digits and comparing them against a dial plan, or terminating the call.

**Note**

The information contained in this guide is not a warranty from Linksys, a division of Cisco Systems, Inc. Customers planning to use Linksys ATAs in a VoIP service deployment are advised to test all functionality they plan to support before putting the ATA in service.

By implementing Linksys ATAs with the SIP protocol, intelligent endpoints at the edges of a network perform the bulk of the call processing. This allows the deployment of a large network with thousands of subscribers without complicated, expensive servers.

The ATA is a key element in the end-to-end IP Telephony solution. It provides one or more standard telephone RJ-11 phone ports (identical to the telephone phone wall jacks) to which the subscriber connects standard analog telephone equipment to access phone services. The ATA connects to a wide area IP network, such as the Internet, through a broadband (DSL or cable) modem or router.

Ensuring Voice Quality

Voice quality, as perceived by the subscribers of the IP Telephony service, should be equivalent (or better) compared to 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**

The Linksys ATA supports all the above voice coding algorithms.

The following sections describe the factors that 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. The Linksys ATA supports the most popular audio compression algorithms for IP Telephony: G.711 a-law and μ -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

The Linksys ATA applies 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 lowers the perceived voice quality. The Linksys ATA applies an error concealment algorithm to alleviate the effect of packet loss.

Network Jitter

The IP network can induce varying delay of the received packets. The RTP receiver in the Linksys ATA 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 the jitter buffer size is too small, late packets may be considered lost and this lowers the voice quality. The Linksys ATA can 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. The Linksys ATA has a near-end echo canceller with at least 8 ms tail length to compensate for impedance match. The Linksys ATA also implements an echo suppressor with comfort noise generator (CNG) so that any residual echo is not noticeable.

Hardware Noise

Certain levels of noise can be coupled into the conversational audio signals because of the hardware design. The source can be ambient noise or 60 Hz noise from the power adaptor. The Linksys ATA 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. Reasonable delay figure should be about 50–100 ms. End-to-end delay larger than 300 ms is unacceptable to most callers. The Linksys ATA supports end-to-end delays well within acceptable thresholds.

Feature Descriptions

The Linksys ATA is a full featured, fully programmable phone adapter 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 the Linksys ATA.

- [SIP Proxy Redundancy, page 1-5](#)
- [Supported Codecs, page 1-6](#)
- [Streaming Audio Server and Music on Hold, page 1-6](#)
- [Silence Suppression and Comfort Noise Generation, page 1-7](#)
- [Modem and Fax Pass-Through, page 1-7](#)
- [Adaptive Jitter Buffer, page 1-7](#)
- [Other Features, page 1-8](#)

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. The Linksys ATA supports the use of backup SIP proxy servers so that service disruption should be nearly eliminated.

A relatively simple way to support proxy redundancy is to configure a static list of SIP proxy servers to the Linksys ATA in its configuration profile where the list is arranged in some order of priority. The Linksys ATA attempts to contact the highest priority proxy server whenever possible. When the currently selected proxy server is not responding, the Linksys ATA automatically retries the next proxy server in the list.

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 it is not feasible to configure a static list of proxy servers for each domain.

One solution in this situation is through the use of DNS SRV records. The Linksys ATA can be instructed to contact a SIP proxy server in a domain named in the SIP message. The Linksys ATA consults the DNS server to get a list of hosts in the given domain that provides SIP services. If an entry

exists, the DNS server returns an SRV record that contains a list of SIP proxy servers for the domain, with their host names, priority, listening ports, and so on. The Linksys ATA tries to contact the list of hosts in the order of their stated priority.

If the Linksys ATA is currently using a lower priority proxy server, it periodically probes the higher priority proxy to see whether it is back on line, and switches back to the higher priority proxy when possible.

Supported Codecs

Negotiation of the optimal voice codec sometimes depends on the ability of the Linksys ATA device to match a codec name with the codec used by the far-end device. The Linksys ATA allows the network administrator to individually name the various codecs that are supported so that the Linksys ATA can successfully negotiate the codec with the far-end equipment. The administrator can select which low-bit-rate codec is to be used for each line. G.711a and G.711u are always enabled.

Table 1-3 Codecs Supported by Linksys ATAs

| Codec (Voice Compression Algorithm) | Description |
|-------------------------------------|--|
| G.711 (A-law and μ -law) | This very low complexity codec supports uncompressed 64 kbps digitized voice transmission at one through ten 5 ms voice frames per packet. This codec provides the highest voice quality and uses the most bandwidth of any of the available codecs. |
| G.726 | This low complexity codec supports compressed 16, 24, 32, and 40 kbps digitized voice transmission at one through ten 10 ms voice frames per packet. This codec provides high voice quality. |
| G.729A | The ITU G.729 voice coding algorithm is used to compress digitized speech. Linksys supports G.729. G.729A is a reduced complexity version of G.729. It requires about half the processing power to code G.729. The G.729 and G.729A bit streams are compatible and interoperable, but not identical. |
| G.723.1 | The Linksys ATA supports the use of ITU G.723.1 audio codec at 6.4 kbps. Up to two channels of G.723.1 can be used simultaneously. For example, Line 1 and Line 2 can be using G.723.1 simultaneously, or Line 1 or Line 2 can initiate a three-way conference with both call legs using G.723.1. |



Note

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

Streaming Audio Server and Music on Hold

This feature allows you to attach an audio source to one of the Linksys ATA FXS ports and use it as a streaming audio source device. The corresponding Line (1 or 2) can be configured as a streaming audio server (SAS) such that when the Line is called, the Linksys ATA answers the call automatically and

starts streaming audio to the calling party provided the FXS port is off-hook. If the FXS port is on-hook when the incoming call arrives, the Linksys ATA replies with a SIP 503 response code to indicate “Service Not Available.”

If an incoming call is auto-answered, but later the FXS port becomes on-hook, the Linksys ATA does not terminate the call but continues to stream silence packets to the caller. If an incoming call arrives when the SAS line has reached full capacity, the Linksys ATA replies with a SIP 486 response code to indicate “Busy Here”. The SAS line can be set up to refresh each streaming audio session periodically (via SIP re-INVITE) to detect whether the connection to the caller is down. If the caller does not respond to the refresh message, the SAS line terminates the call so that the streaming resource can be used for other callers.

On a connected call, the Linksys ATA may place the remote party on call. The only way to do this on the SPA2102 is to perform a hook-flash to initiate a three-way call, or to swap two calls during call-waiting. If the remote party indicates that they can still receive audio while the call is holding, the SPA2102 can be set up to contact an auto-answering SAS as described in and have it stream audio to the holding party. When used this way, the SAS is referred to as a MOH Server.

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.

Modem and Fax Pass-Through

Modem Pass-through Mode can be triggered only by predialing the <Modem Line Toggle Code>. FAX Pass-through Mode is triggered by CED/CNG tone or NSE events. Echo canceller is automatically disabled for Modem Pass-through Mode only. Echo canceller is automatically disabled only if <FAX Disable ECAN> (Line 1/2) is set to “yes” for that line (in that case FAX pass-through is the same as Modem pass-through). Call waiting and silence suppression is automatically disabled for both FAX and Modem pass-through as before. In addition, out-of-band DTMF Tx is disabled during modem or fax pass-through.

Adaptive Jitter Buffer

The Linksys ATA 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.

Feature Descriptions

The Linksys ATA has a Network Jitter Level control setting for each line of service. The jitter level decides how aggressively the Linksys ATA tries 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.

Other Features

The following table summarizes other features provided by Linksys ATAs.

Table 1-4 Linksys ATA Features

| Feature | Description |
|--|---|
| International Caller ID Delivery | In addition to support of the Bellcore (FSK) and Swedish/Danish (DTMF) methods of Caller ID (CID) delivery, release 2.0 adds a large subset of ETSI-compliant methods to support international CID equipment. Different types of CID delivery method can be obtained by mixing-and-matching some of the steps as shown. |
| 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 *) The Linksys ATA 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 | The Linksys ATA has 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. |
| Full Duplex Audio | Full-duplex is the ability to communicate in two directions simultaneously so that more than one person can speak at a time. Half-duplex means that only one person can talk at a time, like a CB radio or walkie-talkie, which is unnatural in normal free-flowing two-way communications. The Linksys ATA supports full-duplex audio. |
| Echo Cancellation—Up to 8 ms Echo Tail | The SPA3102 supports hybrid line echo cancellation. This feature uses the G.165 echo canceller to eliminate up to 8 ms of line echo. This feature does not provide acoustic echo cancellation on endpoint devices; that is, an end user speakerphone. |

Table 1-4 Linksys ATA Features

| Feature | Description |
|---|---|
| Signaling Hook Flash Event | <p>The Linksys ATA can signal hook flash events to the remote party on a connected call. This feature can be used to provide advanced mid-call services with third-party-call-control. Depending on the features that the service provider offers using third-party-call-control, the following three Linksys ATA features may be disabled to correctly signal a hook-flash event to the softswitch:</p> <ul style="list-style-type: none"> • Call Waiting Serv • Three Way Call Serv • Three Way Conf Serv <p>You can configure the length of time allowed for detection of a hook flash using the Hook Flash Timer parameter on the Regional tab of the administration web server.</p> |
| Configurable Dial Plan with Interdigit Timers | <p>The Linksys ATA has three configurable interdigit timers:</p> <ul style="list-style-type: none"> • Initial timeout (T)—Handset off hook, no digit pressed yet. • Long timeout (L)—One or more digits pressed, more digits needed to reach a valid number (as per the dial plan). • Short timeout (S)—Current dialed number is valid, but more digits would also lead to a valid number. |
| Polarity Control | <p>The Linksys ATA allows the polarity to be set when a call is connected and when a call is disconnected. This feature is required to support some pay phone system and answering machines.</p> |
| Calling Party Control | <p>Calling Party Control (CPC) signals to the called party equipment that the calling party has hung up during a connected call by removing the voltage between the tip and ring momentarily. This feature is useful for auto-answer equipment, which then knows when to disengage.</p> |
| Report Generation and Event Logging | <p>The Linksys ATA reports a variety of status and error reports to assist service providers to diagnose problems and evaluate 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.</p> |
| Syslog and Debug Server Records | <p>The Linksys ATA supports detailed logging of all activities for further debugging. The debug information may be sent to a configured Syslog server. Via the configuration parameters, the Linksys ATA allows some settings to select which type of activity/events should be logged, as for instance, a debug level setting.</p> |

Technology Background

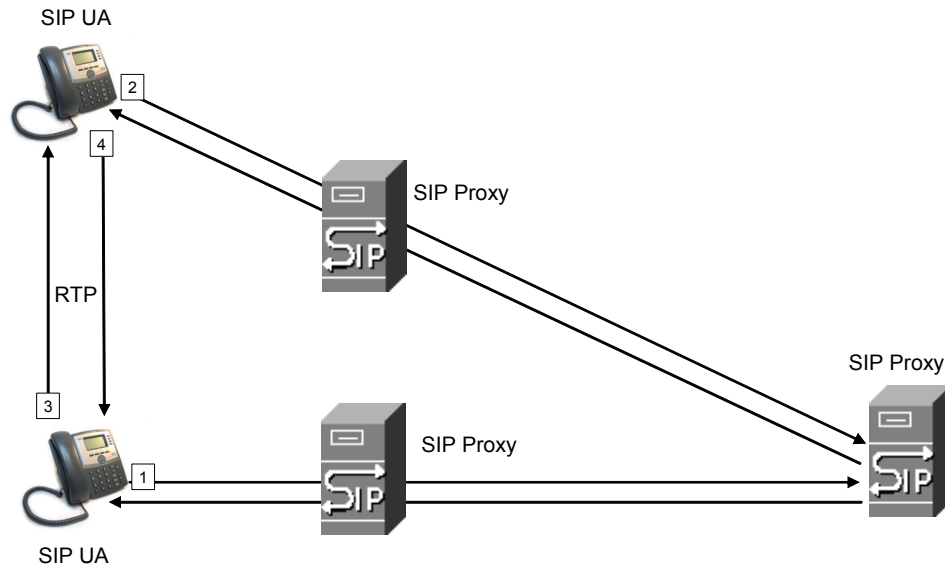
This section provides background information about the technology and protocols used by the ATA. It includes the following topics:

- [Session Initiation Protocol, page 1-10](#)
- [Network Address Translation, page 1-10](#)

Session Initiation Protocol

Linksys ATAs 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).

Network Address Translation

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

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

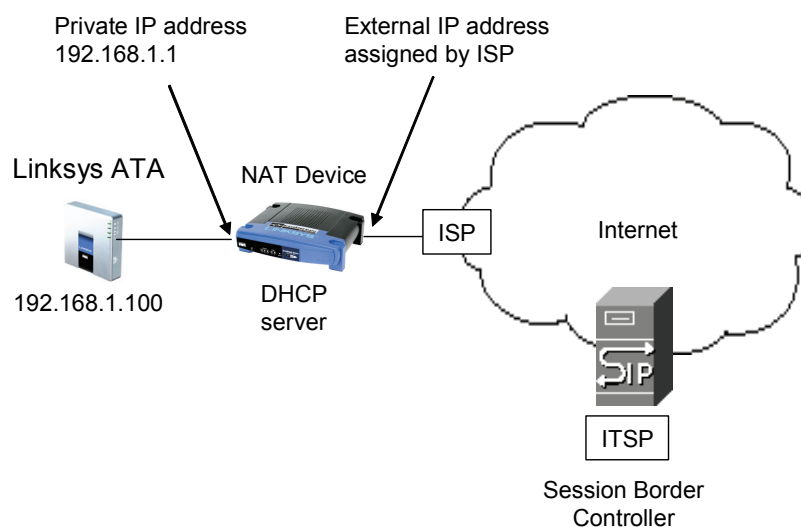
NAT Overview

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

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.

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.

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



The ITSP may support NAT mapping using a Session Border Controller (see [Figure 1-3](#)). This is the preferred option because it eliminates the need for managing NAT on the Linksys ATA. If this is not available, you will need to discuss with the ITSP how to use the NAT Support Parameters provided by the Linksys ATA, such as <Outbound Proxy> and <STUN Server Enable>.

NAT Types

The different ways that NAT is implemented is sometimes 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 Linksys ATA:
1. Make sure you do not have firewall running on your PC that could block the syslog port (by default this is 514).
 2. On the administration web server, System tab, set <Debug Server> to the IP address and port number of your syslog server.
Note that this address and port number has to be reachable from the Linksys ATA.
 3. Set <Debug level> to **3** but you do not need to 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 or not.
-

SIP-NAT Interoperation

In the case of SIP, the addresses where messages/data should be sent to a Linksys ATA system are embedded in the SIP messages sent by the device. If the Linksys ATA 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 Linksys ATA system must substitute the private IP address information with the proper external IP address/port in the mapping chosen by the underlying NAT to communicate with a particular public peer address/port. For this, the Linksys ATA system needs to perform the following tasks:

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

This can be done with the help of an external device, such as a STUN server. A STUN server

responds to a special NAT-Mapping-Discovery request by sending back a message to the source IP address/port of the request, where the message contains the source IP address/port of the original request. The Linksys ATA system can send this request when it first attempts to communicate with a SIP entity over the Internet. It then stores the mapping discovery results returned by the server.

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

If the entity is a SIP Registrar, the information should be carried in the Contact header that overwrites the private address/port information. If the entity is another SIP UA when establishing a call, the information should be carried in the Contact header as well as in the SDP embedded in SIP message bodies. The VIA header in outbound SIP requests might also need to be substituted with the public address if the UAS relies on it to route back responses.

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

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

Where to Go From Here

| To | Refer to |
|--|---|
| Use the different administration and configuration tools provided for managing Linksys ATAs. | Chapter 2, “Getting Started” |
| Complete the basic configuration of a Linksys ATA. | Chapter 3, “Configuring Linksys ATAs” |
| Configure the Linksys SPA3102 or AG310 for providing PSTN connectivity. | Chapter 4, “Configuring the PSTN Gateway (FXO)” |
| Look up the function and usage for each field or parameter on the ATA administration web server pages. | Chapter 5, “Linksys ATA Field Reference” |
| Look up the expansion of acronyms used in this document. | Appendix A, “Acronyms” |
| Define the terms used in this document. | Appendix B, “Glossary” |
| Understand the operations of ATA user features. | Appendix C, “User Guidelines” |

The following documentation provides additional information about features and functionality of Linksys ATAs:

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

The following documentation describes how to use and configure other Linksys VoIP products:

- *SPA9000 Administrator Guide*
- *LVS CTI Integration Guide*
- *LVS Integration with ITSP Hosted Voicemail Guide*

Where to Go From Here

- *SPA900 Series IP Phones Administrator Guide*
- *Linksys Voice over IP Product Guide: SIP CPE for Massive Scale Deployment*

Getting Started

This chapter provides a brief description of each Linksys ATA and describes the tools and utilities available for administration. It includes the following sections:

- [Linksys Analog Telephone Adapters \(ATAs\), page 2-1](#)
- [Establishing Connectivity, page 2-14](#)
- [Connecting the SPA8000, page 2-16](#)
- [Using the Interactive Voice Response Interface, page 2-18](#)
- [Using the Administration Web Server, page 2-22](#)

The following ATAs provide Layer 2 (data link) and Layer 3 (IP) configuration options:

- AG310
- WRP400
- WRTP54/54G
- RTP300

For information about configuring these options, including private virtual circuits (PVCs), PPPoE, and IP routing, refer to the user guide for the specific device.

For information about configuring the Public Switch Telephone Network (PSTN) connectivity options for the SPA3102 and AG310 refer to [Chapter 3, “Configuring Linksys ATAs.”](#)

Linksys Analog Telephone Adapters (ATAs)

Linksys ATAs convert voice traffic into data packets for transmission over an IP network. This section illustrates and summarizes the ports and LEDs provided by each model. It includes the following topics:

- [Caring for Your Hardware, page 2-2](#)
- [PAP2T, page 2-3](#)
- [SPA2102, page 2-6](#)
- [SPA3102, page 2-7](#)
- [SPA8000, page 2-9](#)
- [AG310, page 2-2](#)
- [WRP400, page 2-10](#)

Linksys Analog Telephone Adapters (ATAs)

- [WRTP54G](#), page 2-11
- [WRT54GP2](#), page 2-13
- [RTP300](#), page 2-4

Caring for Your Hardware

The Linksys ATA is an electronic device 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.

AG310

The AG310 provides a high-speed ADSL2/+ modem along with a four-port Ethernet switch and a PSTN gateway (see [Figure 1-4](#)).



Note

Throughout this document, when references are made to the software configuration of the SPA3102, the information also applies to the AG310, which provides the same functionality as the SPA3102, as well as an ADSL modem and a four-port Ethernet switch.

Figure 1-4 AG310



The following tables describe the LEDs on the front panel and the ports on the back panel of the device.

Table 1-5 AG310 Front Panel

| LED | Function |
|--------------|---|
| Power | Steady green indicates power on and Internet connection. Flashing indicates not connected to the Internet, booting or firmware upgrade. |
| Ethernet 1-4 | Steady green indicates an active connection to the network. Flashing indicates traffic. |
| Voice Status | Steady green indicates that a voice call is currently in progress. |
| Phone | Steady green indicates active or registered connection to the ITSP through the Phone port. Flashing indicates device is in use or off hook. |
| Line | Steady green when a telephone or fax machine has an active connection to traditional phone service through the Line port. |
| DSL | Steady green indicates that the DSL connection is active. |
| Internet | Steady green indicates active connection. Flashing indicates traffic. |

Table 1-6 AG310 Back Panel

| Port | Function |
|--------------|--|
| DSL | Connect to the telephone wall jack providing DSL service using a RJ-11 cable. |
| Line | Connect to a standard telephone wall jack with an RJ-11 cable. |
| Phone | Connect to an analog telephone or fax machine with an RJ-11 cable. |
| Ethernet 1-4 | Connect to network devices, such as a PC or a switch, using an Ethernet cable. |
| Power | Connect to the power supply. |

PAP2T

The PAP2T provides two FXS ports (see [Figure 1-5](#)).

Linksys Analog Telephone Adapters (ATAs)

Figure 1-5 PAP2T



The following tables describe the LEDs on the front panel and the ports on the back panel of the device.

Table 1-7 PAP2T Front Panel

| LED | Function |
|-----------|---|
| Phone 1/2 | Steady green indicates active or registered connection to the ITSP through the Phone port. Flashing indicates device is in use or off hook. |
| Ethernet | Steady green indicates active connection. Flashing indicates traffic. |
| Power | Steady green indicates power on and Internet connection. Flashing indicates not connected to the Internet, booting or firmware upgrade. |

Table 1-8 PAP2T Back Panel

| Port | Function |
|-----------|--|
| Phone 1/2 | Connect to an analog telephone or fax machine with an RJ-11 cable. |
| Ethernet | Connect to a router or broadband (cable/DSL) modem. |
| Power | Connect to the 5-volt DC power adapter. |

RTP300

The RTP300 provides an ATA with two FXS ports along with an IP router (see [Figure 1-6](#)). The RTP300 provides connectivity to an analog telephone as well as Internet connectivity to a LAN with a built-in four-port switch. The IP router provides QoS support, an integrated firewall and VPN pass-through, and full routing with DHCP and NAT.

Figure 1-6 RTP300



The following tables describe the LEDs on the front panel and the ports on the back panel of the device.

Table 1-9 RTP300 Front Panel

| LED | Function |
|--------------|---|
| Ethernet 1-4 | Steady green indicates an active connection to the network. Flashing indicates traffic. |
| Phone 1/2 | Steady green when telephone or fax machine has an active or registered connection to the ITSP through the Phone port. Flashing indicates phone is in use or off hook. |
| Internet | Steady green indicates active connection. Flashing indicates traffic. |

Table 1-10 RTP300 Back Panel

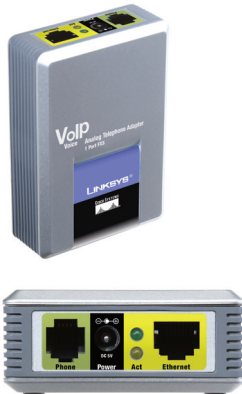
| Port | Function |
|--------------|---|
| Internet | Connect to the broadband (cable/DSL) modem using an Ethernet cable. |
| Phone 1/2 | Connect to an analog telephone or fax machine using an RJ-11 cable. |
| Ethernet 1-4 | Connect to local IP devices, such as PCs, using an Ethernet cable. |
| Power | Connect to a 12v DC power supply. |

SPA1001

The SPA1001 provides one FXS port (see [Figure 1-7](#)).

Linksys Analog Telephone Adapters (ATAs)

Figure 1-7 SPA1001



The following table describes the LEDs and ports on the back panel of the device.

Table 1-11 SPA1001 Back Panel

| LED/Port | Function |
|----------|--|
| Phone | Connect to an analog telephone or fax machine with an RJ-11 cable. |
| Power | Connect to the 5-volt DC power adapter. |
| Act | Steady green indicates active or registered connection to the ITSP through the Phone port. Flashing indicates device is in use or off hook. Steady yellow indicates active Ethernet connection. Flashing indicates traffic. |
| Ethernet | Connect to a router or broadband (cable/DSL) modem. |

SPA2102

The SPA2102 provides two FXS ports and an Ethernet port for connecting to a local PC or other Ethernet device.

Figure 1-8 SPA2102



The following tables describe the LEDs on the front panel and the ports on the back panel of the device.

Table 1-12 SPA2102 Front Panel

| LED | Function |
|-----------|---|
| Power | Steady green indicates power on and Internet connection. Flashing indicates not connected to the Internet, booting or firmware upgrade. |
| Internet | Steady green indicates active connection. Flashing indicates traffic. |
| Phone 1/2 | Steady green indicates active or registered connection to the ITSP through the Phone port. Flashing indicates device is in use or off hook. |

Table 1-13 SPA2102 Back Panel

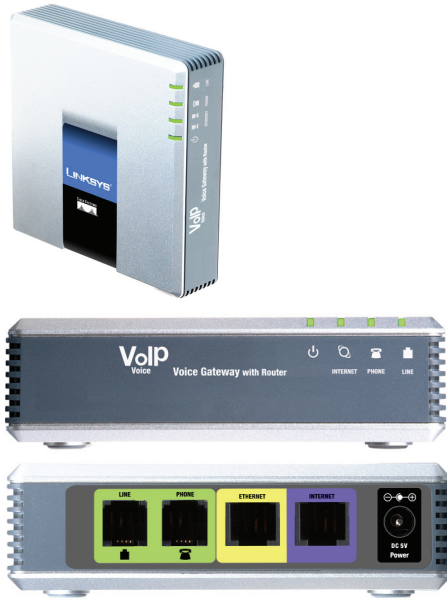
| Port | Function |
|-----------|---|
| Phone 1/2 | Connect to an analog telephone or fax machine with an RJ-11 cable. |
| Ethernet | Connect to a network device, such as a PC or a switch with an Ethernet cable. |
| Internet | Connect to a router or broadband (cable/DSL) modem. |
| Power | Connect to the 5 volt DC power adapter. |

SPA3102

The SPA3102 acts as a PSTN gateway providing connectivity to the PSTN.

Linksys Analog Telephone Adapters (ATAs)

Figure 1-9 SPA3102



The following tables describe the LEDs on the front panel and the ports on the back panel of the device.

Table 1-14 SPA3102 Front Panel

| LED | Function |
|----------|---|
| Power | Steady green indicates power on and Internet connection. Flashing indicates not connected to the Internet, booting or firmware upgrade. |
| Internet | Steady green indicates active connection. Flashing indicates traffic. |
| Phone | Steady green indicates active or registered connection to the ITSP through the Phone port. Flashing indicates device is in use or off hook. |
| Line | Steady green when a telephone or fax machine has an active connection to traditional phone service through the Line port. |

Table 1-15 SPA3102 Back Panel

| Port | Function |
|----------|---|
| Line | Connect to a standard telephone wall jack with an RJ-11 cable. |
| Phone | Connect to an analog telephone or fax machine with an RJ-11 cable. |
| Ethernet | Connect to a network device, such as a PC or a switch with an Ethernet cable. |
| Internet | Connect to a router or broadband (cable/DSL) modem. |
| Power | Connect to the power supply. |

SPA8000

The SPA8000 is an analog telephone adapter that supports connecting up to eight analog telephone devices, including up to four fax devices. The SPA8000 consists of four hardware modules, each of which is similar in functionality to the SPA2102. However, IP routing functionality is not provided by the SPA8000.

Figure 1-10 SPA8000



The following tables describe the LEDs on the front panel and the ports on the back panel of the device.


Table 1-16 SPA8000 Front Panel

| LED | Function |
|----------------------|--|
| Power | Steady green indicates power on and Internet connection. Flashing indicates not connected to the Internet, booting or firmware upgrade. |
| Ethernet | Link activity status. Steady green indicates active connection. Flashing indicates traffic. |
| Status 1/2/3/4 | Indicates the status of each of the four hardware modules in the SPA8000. Steady green indicates active connection. Flashing indicates that the associated module is restarting. |
| Line 1/2/3/4/5/6/7/8 | Steady green indicates active or registered connection to the ITSP. Flashing indicates device is in use or off hook. |

Table 1-17 SPA8000 Back Panel

| Port | Function |
|---|--|
| RJ21 port | Provides a single connector that supports up to eight FXS connections. |
| RJ11 ports: Line 1/2 Line 3/4 Line 5/6 Line 7/8 | Connect each of the eight FXS ports to an analog telephone or fax machine using an RJ-11 cable. These ports are distributed among four hardware modules of two ports each. You can connect a single fax machine to each pair of ports (for a maximum of four). |

Table 1-17 SPA8000 Back Panel

| Port | Function |
|----------|--|
| AUX | Ethernet maintenance port. Connect to a network device, such as a PC or a switch with an Ethernet cable for accessing the administration web server on the SPA8000. The attached device is assigned an IP address dynamically by the internal DHCP server provided by the SPA8000. |
| Ethernet | Connect to a switch, router or broadband (cable/DSL) modem for access to the Internet. Note When connected directly to the Internet, connectivity between FXS devices on the SPA8000 depends on the Outbound Proxy service provided by the ITSP. |
| Power | Connect to the 12-volt DC power adapter. Caution  This 12-volt power adapter is to be used only with the SPA8000. Using it with any other Linksys product may damage the unit. |

WRP400

The WRP400 provides an ATA with two FXS ports and a Wireless-G multiport IP router (see [Figure 1-11](#)). The WRP400 provides connectivity to an analog telephone as well as Internet connectivity to a LAN with a built-in four-port switch. The IP router provides QoS support, an integrated firewall and VPN pass-through, and full routing with DHCP and NAT.

Figure 1-11 WRP400



The following tables describe the LEDs on the front panel and the ports on the back panel of the device

Table 1-18 WRP400 *Front Panel*

| LED | Function |
|--------------|---|
| Ethernet 1-4 | Steady green indicates an active connection to the network. Flashing indicates traffic. |
| Wireless | Steady green indicates an active connection to the network. Flashing indicates traffic. |
| Phone 1/2 | Steady green when telephone or fax machine has an active or registered connection to the ITSP through the Phone port. Flashing indicates phone is in use or off hook. |
| Internet | Steady green indicates active connection. Flashing indicates traffic. |

Table 1-19 WRTP54G *Back Panel*

| Port | Function |
|--------------|---|
| Internet | Connect to the broadband (cable/DSL) modem using an Ethernet cable. |
| Phone 1/2 | Connect to an analog telephone or fax machine using an RJ-11 cable. |
| Ethernet 1-4 | Connect to local IP devices, such as PCs, using an Ethernet cable. |
| Power | Connect to the 12-volt power supply. |

WRTP54G

The WRTP54G provides an ATA with two FXS ports and a Wireless-G multiport IP router (see [Figure 1-12](#)). The WRTP54G provides connectivity to an analog telephone as well as Internet connectivity to a LAN with a built-in four-port switch. The IP router provides QoS support, an integrated firewall and VPN pass-through, and full routing with DHCP and NAT.

Linksys Analog Telephone Adapters (ATAs)

Figure 1-12 WRTP54G



The following tables describe the LEDs on the front panel and the ports on the back panel of the device

Table 1-20 WRTP54G Front Panel

| LED | Function |
|--------------|---|
| Ethernet 1-4 | Steady green indicates an active connection to the network. Flashing indicates traffic. |
| Wireless | Steady green indicates an active connection to the network. Flashing indicates traffic. |
| Phone 1/2 | Steady green when telephone or fax machine has an active or registered connection to the ITSP through the Phone port. Flashing indicates phone is in use or off hook. |
| Internet | Steady green indicates active connection. Flashing indicates traffic. |

Table 1-21 WRTP54G Back Panel

| Port | Function |
|--------------|---|
| Internet | Connect to the broadband (cable/DSL) modem using an Ethernet cable. |
| Phone 1/2 | Connect to an analog telephone or fax machine using an RJ-11 cable. |
| Ethernet 1-4 | Connect to local IP devices, such as PCs, using an Ethernet cable. |
| Power | Connect to the 12-volt power supply. |

WRT54GP2

The WRT54GP2 provides an ATA with two FXS ports and a Wireless-G multiport IP router (see [Figure 1-13](#)). The WRT54GP2 provides connectivity to an analog telephone as well as Internet connectivity to a LAN with a built-in four-port switch. The IP router provides QoS support, an integrated firewall and VPN pass-through, and full routing with DHCP and NAT.

Figure 1-13 WRT54GP2



The following tables describe the LEDs on the front panel and the ports on the back panel of the device.

Table 1-22 WRT54G P2 Front Panel

| LED | Function |
|--------------|---|
| Power | Steady green indicates that the power supply is connected and the device is functioning. |
| WLAN | Steady green indicates an active connection to the network. Flashing indicates traffic. |
| Ethernet 1-3 | Steady green indicates an active connection to the network. Flashing indicates traffic. |
| Phone 1/2 | Steady green when telephone or fax machine has an active or registered connection to the ITSP through the Phone port. Flashing indicates phone is in use or off hook. |
| Internet | Steady green indicates active connection. Flashing indicates traffic. |

Table 1-23 WRT54G Back Panel

| Port | Function |
|-----------|---|
| Internet | Connect to the broadband (cable/DSL) modem using an Ethernet cable. |
| Phone 1/2 | Connect to an analog telephone or fax machine using an RJ-11 cable. |

Table 1-23 WRTP54G Back Panel

| Port | Function |
|--------------|--|
| Ethernet 1-3 | Connect to local IP devices, such as PCs, using an Ethernet cable. |
| Power | Connect to the power supply. |

Establishing Connectivity

This section describes how to connect the Linksys ATA hardware. It includes the following topics:

- [Bandwidth Requirements, page 2-14](#)
- [Making the Physical Connections, page 2-15](#)

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 Linksys ATA 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 1-24](#) illustrates the bandwidth budget using different codecs.

Table 1-24 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 milliseconds 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>

Making the Physical Connections

Make sure that you have the following package contents:

- Linksys phone adapter unit
- Ethernet cable
- RJ-11 phone cable (SPA3102/AG310 Only)
- SPA Quickstart Guide
- Volt power adapter

You also need the following:

- One or two analog touchtone telephones (or fax Machine)
- Access to an IP network via an Ethernet connection for WAN connectivity
- Access to a PSTN network connection (SPA3102/AG310 only).

Perform the following steps to connect the Linksys ATA.

-
- Step 1** Insert a standard RJ-45 Ethernet cable (included) into the LAN port and connect the other end to the broadband router or modem.
- Step 2** (Optional) For a Linksys ATA with more than one Ethernet port, you can connect the extra port to a PC or other Ethernet device.
- Step 3** Insert a standard RJ-11 telephone cable into the Phone 1 port.
- Step 4** (Optional) For a Linksys ATA with more than one FXS port, you can connect the extra port to a second analog telephone or a fax machine.



Note Do not connect an RJ-11 telephone cable from the Linksys ATA to the wall jack to prevent any chance of connection to the circuit switched Telco network.

- Step 5** (SPA3102/AG310 only) Connect the RJ-11 phone cable (included) to the SPA3102 LINE port and to your telephone wall jack.
- Step 6** Connect the included power adapter to the Linksys ATA power port, and then plug the power adapter into an electrical outlet.

The power LED on the front panel will light up as soon as the device powers on.

- Step 7** Power on the broadband modem or router.
-

Connecting the SPA8000

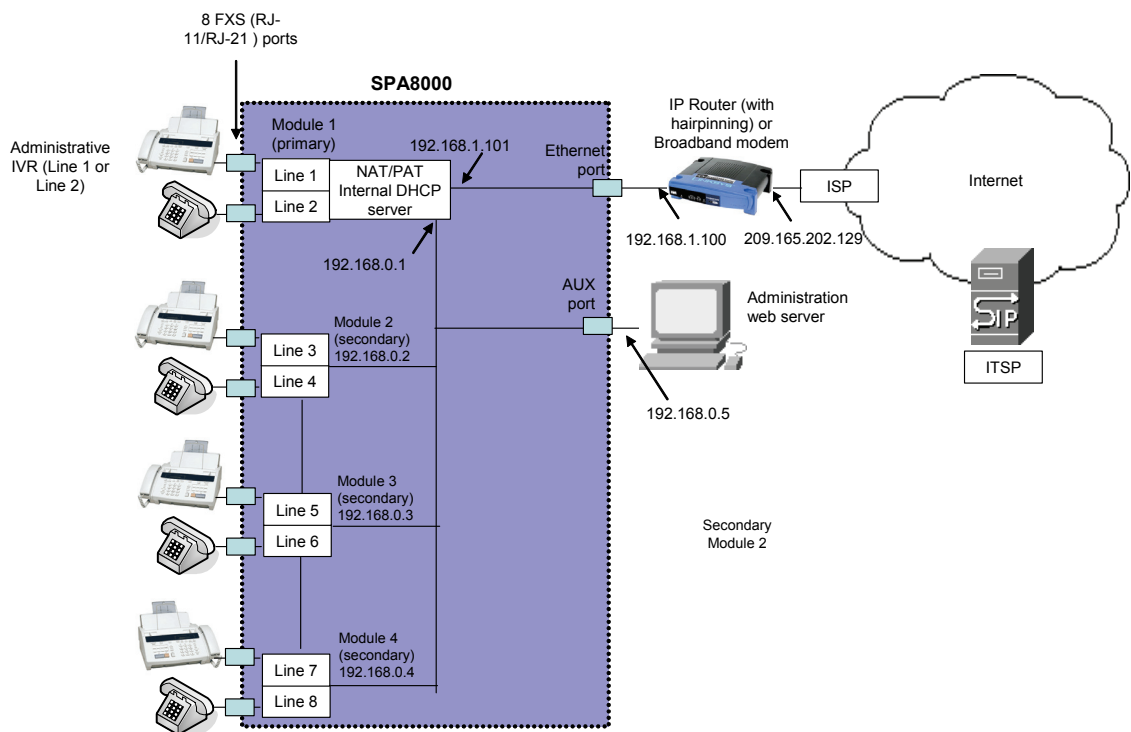
The SPA8000 provides up to eight analog telephone connections, and is designed to function as a network endpoint. This section describes the architecture and connectivity requirements of the SPA8000 and includes the following topics:

- [SPA8000 Architecture](#), page 2-16
- [Connectivity Requirements](#), page 2-17

SPA8000 Architecture

As shown in [Figure 1-14](#), the SPA8000 consists of four hardware modules: one primary (Module 1) and three secondary modules (Modules 2, 3, and 4). Each module provides two FXS ports. The primary module supports Line 1 and Line 2, while the three secondary modules support Line 3-4, 5-6, and 7-8.

Figure 1-14 SPA8000 Internal Configuration



The four modules are interconnected through an internal IP network, configured automatically as a single subnet, which by default is assigned the network address 192.168.0.0 with a subnet mask of 255.255.255.0. If a network conflict is detected from the device connected on the AUX port, the internal network is automatically changed to 192.168.1.0.

The primary module is responsible for the management functions of the entire SPA8000, in addition to handling calls on Line 1 and Line 2. These management functions include remote provisioning, firmware upgrades, the administration web server, IVR functions, and SIP NOTIFY control messages (over Line 1 or 2). The IVR functions are accessed by connecting an analog telephone to Line 1 or Line 2.

The secondary modules (Module 2, 3, and 4) obtain configuration and firmware upgrades from the primary module and are not managed directly. The primary module routes all the SIP and RTP traffic to and from the secondary modules over the SPA8000 internal LAN, which also includes the AUX maintenance port. The AUX maintenance port cannot be used to forward IP traffic. It is designed to connect to the management functions on the SPA8000, such as the administration web server.

The secondary modules are assigned IP addresses by the DHCP server that runs on the primary module. The primary module also performs Network Address Translation (NAT) and Port Address Translation (PAT) on the IP addresses and ports used by the secondary modules, translating them into addresses and port numbers that can be transmitted over the network connected to the Ethernet port.

For proper operation, the service provider should use an Outbound Proxy to forward all voice traffic when the SPA8000 is located behind a router. If necessary, explicit port ranges can be specified for SIP and RTP for each of the four modules.

When performing a firmware upgrade, either through remote provisioning or through the administration web server, the SPA8000 downloads and upgrades the primary module first. After successful upgrade of the primary module, the three secondary modules update their firmware image directly from the primary module.

The primary module is also connected through the Ethernet port to the external local area network, or directly to the Internet. This module receives its IP address through DHCP, or its address can be statically assigned using the IVR or the administration web server. To access the IVR, connect an analog telephone to the Phone 1 or Phone 2 port. To access the administration web server, connect a PC to the AUX port.

**Note**

In the illustration, one fax machine is connected to each pair of ports to illustrate that only one T.38 connection is supported by each of the four pairs of RJ-11 ports. Up to four fax machines can be connected to the SPA8000, but they must be distributed as shown.

The secondary modules (Line 3/4, 5/6, and 7/8) are assigned IP addresses by an internal DHCP server provided by the SPA8000, and these addresses are not configurable.

**Note**

The SPA8000 is not designed to forward IP packets to devices connected to its AUX port and that configuration is not supported.

Connectivity Requirements

The SPA8000 can be connected to a local router through its LAN port, or directly to the Internet. Each phone connected to an RJ-11 (analog) port on the SPA8000 connects to other devices through SIP, which is transmitted over the IP network.

In order to ensure connectivity between the devices connected to its FXS ports, the SPA8000 requires the following functionality to be supplied on the network connected to its Ethernet port:

- Connection to an IP router with hairpinning support
- Connection to an outbound Proxy server

When a phone connected to a port on the SPA8000 communicates with another phone, it sends a SIP packet onto the internal LAN, which is then forwarded to the external local area network or directly to the Internet. The source address and source port on the original packet is assigned by the SPA8000

DHCP server, and the address and port is translated by the SPA8000 using Network Address Translation (NAT) and Port Address Translation (PAT). The packet must then be routed back to the internal network on the SPA8000 by the local router or the ISP router.

Problems can occur with calls between phones connected to the SPA8000 when an outbound proxy or a router with hairpinning support is not available. The SPA8000 cannot directly connect the two telephone devices, but requires a local or remote router to route the packet back to its destination on the local network from which it originated.

The necessary routing can be provided by a router with hairpinning support, or by an outbound SIP proxy, which is typically provided by the Internet Telephony Service Provider (ITSP). When relying on the ITSP for interconnecting phones on the SPA8000, local phones connected to the SPA8000 are unable to communicate with each other if the Internet connection is not available for any reason. It is recommended you connect the SPA8000 to a local router that provides hairpinning support to prevent this problem.

Using the Interactive Voice Response Interface

This section describes how to read or write basic network configuration settings using a touchtone telephone connected to one of the FXS (RJ-11) phone ports of the Linksys ATA. It includes the following topics:

- [Using the IVR Menu, page 2-18](#)
- [IVR Options, page 2-19](#)
- [Entering a Password through the IVR, page 2-21](#)

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 Linksys ATA.



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 1-25](#) to identify the option required.

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

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

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

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

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

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

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

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

IVR Options

Table 1-25 summarizes the options provided by the IVR.

Table 1-25 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. Please enter option followed by the pound key or hang-up to exit." |
| Exit IVR Menu | 3948 | None | |
| Check DHCP | 100 | None | IVR announces if 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 |
| Check Network Mask | 120 | None | IVR announces the current network mask of Linksys ATA. |

Table 1-25 IVR Options (continued)

| | | | |
|--|--------|---|---|
| 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 the Linksys ATA. |
| 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 “Disable,” 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 Linksys ATA in hex string format. |
| Check Firmware Version | 150 | None | IVR announces the version of the firmware running on the Linksys ATA. |
| 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 1-25 IVR Options (continued)

| | | | |
|--|----------------------|--|--|
| <p>User Factory Reset of Unit</p> <p>WARNING:</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>Linksys ATA 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>WARNING:</p> <p>ALL NON-DEFAULT SETTINGS WILL BE LOST!</p> <p>This includes network and service provider data.</p> | <p>73738 (RESET)</p> | <p>Enter 1 to confirm Enter * (star) to cancel operation</p> | <p>Linksys ATA 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" only require a password 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 only applies to the password input.

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

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

Notes:

- The final # key is not included in the password value.
 - Saved settings take effect when the telephone is hung-up, and if necessary, the Linksys ATA 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 * * * *.

Using the Administration Web Server

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

- [Connecting to the Administration Web Server, page 2-22](#)
- [Administrator Account Privileges, page 2-23](#)

Connecting to the Administration Web Server

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

-
- Step 1** Launch a web browser on a computer connected to the Linksys ATA.
You access the administration web server through the Internet (WAN) interface on the Linksys ATA.
 - Step 2** Determine the address of the administration web server.
 - a. Connect an analog telephone to the Phone 1 or Phone 2 port on the Linksys ATA.
 - b. Press **** on the keypad to access the IVR menu.
 - c. Press **110#** to determine the Internet (WAN) IP address.
 - Step 3** Direct the browser to the IP address of the Linksys ATA.

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 phones, click **PBX Status**.

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

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

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

Administrator Account Privileges

The Linksys ATA supports 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.

Configuring Linksys ATAs

This chapter describes how to perform site-specific configuration required to use a Linksys ATA or to enable specific features. It includes the following sections:

- [Initial Configuration, page 3-1](#)
- [Web Interface URLs, page 3-3](#)
- [Provisioning, page 3-4](#)
- [Configuring a Dial Plan, page 3-5](#)
- [Secure Call Implementation, page 3-10](#)
- [Configuring a Streaming Audio Server, page 3-12](#)
- [Using a FAX Machine with the SPA2102 or SPA8000, page 3-15](#)
- [Managing Caller ID Service, page 3-17](#)
- [Troubleshooting and Configuration FAQ, page 3-18](#)

Initial Configuration

Step 1 After connecting the Linksys ATA to your LAN, launch the web browser on your PC and direct it to the following URL:

192.168.0.1/advanced

192.168.0.1 is the default local IP address of a Linksys ATA that provides a built-in IP router.

If the ITSP supplied a password for administrative access, you will see a login screen. In the *User Name* field, enter **user**, the default user name for user access, or **admin** for administrator access. Then enter the user password supplied by your ITSP.

If your ATA includes a built-in IP router, the *Router - Status* screen appears.

Step 2 Click the **WAN Setup** tab.

Step 3 Complete the WAN configuration for DHCP, static IP addressing, or PPPoE.

If you are using DHCP for dynamically assigning the IP address to the Linksys ATA, complete the following steps:

- a. Select **DHCP** from the *Connection Type* drop-down menu.

Initial Configuration

If you use a cable modem, you may need to configure the MAC Clone Settings. (Contact your ISP for more information.)

- b. If your service uses a specific PC MAC address, then select **yes** from the *Enable MAC Clone Service* setting.
- c. Then enter the PC's MAC address in the *Cloned MAC Address* field.

If you are using static IP addressing, complete the following steps:

- a. Select **Static IP** from the *Connection Type* drop-down menu.
- b. In the Static IP Settings section, enter the IP address in the *Static IP* field, the subnet mask in the *NetMask* field, and the default gateway IP address in the *Gateway* field.
- c. In the Optional Settings section, enter the DNS server address(es) in the *Primary DNS* and optional *Secondary DNS* fields.

If you are using PPPoE, complete the following steps:

- a. Select **PPPoE** from the *Connection Type* drop-down menu.
This is the correct setting for most DSL users.
- b. Enter the values provided by the ITSP in the following fields:
 - PPPoE Login Name
 - PPPoE Login Password
 - PPPoE Service Name

Step 4 Configure the Linksys ATA parameters on the Line tab with the appropriate values for the specific ITSP.

If the Linksys ATA has more than one Line tab, each line tab must be configured separately. Each line tab can be configured for a different ITSP.

The minimum required configuration to connect the Linksys ATA to an ITSP includes configuring the following parameters on the appropriate Line tab with the values provided for a specific ITSP:

- User ID (Subscriber Information section)
- Password (Subscriber Information section)
- Proxy (Proxy and Registration section)

Other parameters may need to be changed from the defaults, depending on the requirements of a specific ITSP. Some of the commonly changed sections include the following:

- Streaming Audio Server—to enable an external music source for enabling music on hold (see the [“Configuring a Streaming Audio Server”](#) section on page 3-12 for further information).
- NAT Settings—to workaroud NAT issues (see the [“Troubleshooting and Configuration FAQ”](#) section on page 3-18 for further information)
- Subscriber Information—to configure security parameters (see the [“Secure Call Implementation”](#) section on page 3-10 for further information)
- Dial Plan—to configure a dial plan for a specific line (see the [“Configuring a Dial Plan”](#) section on page 3-5 for further information)

Step 5 After making any necessary changes to the Linksys ATA parameters, click the **Submit All Changes** button.

The changes are saved, the Linksys ATA reboots, and the configuration changes are applied to the running configuration.

Web Interface URLs

The Linksys ATA web interface supports several functions through special URLs:

- Upgrade
- Reboot
- Resync

Administrator account privilege is needed for these functions.

Upgrade URL

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

**Note**

If the value of the <Upgrade Enable> parameter in the Provisioning page is **No**, you cannot upgrade the Linksys ATA 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 Linksys ATA to do a resync to a profile specified in the URL, which can identify either a TFTP, HTTP, or HTTPS server.

**Note**

The SPA resyncs only when it is idle.

The syntax of the Resync URL is as follows:

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

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

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

If no port is specified, the default port is used (69 for TFTP, 80 for HTTP, and 443 for HTTPS).

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

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

Reboot URL

The Reboot URL lets you reboot the Linksys ATA.



Note

The Linksys ATA reboots only when it is idle.

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

Provisioning

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

- [Provisioning Capabilities, page 3-4](#)
- [Configuration Profile, page 3-4](#)

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

Provisioning Capabilities

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

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

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

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

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

Configuration Profile

The Linksys ATA 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. Here is an example of a valid XML profile:

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

```
<Upgrade_Enable>Yes</Upgrade_Enable>  
</flat-profile>
```

Binary format profiles contain Linksys ATA parameter values and user access permissions for the parameters. By convention, the profile uses the extension .cfg (for example, spa2102.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. Here 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 *Linksys SPA Provisioning Guide* for further details.

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

Configuring a Dial Plan

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

The Linksys ATA 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.



Note

When using the SPA3102 or AG310 as a PSTN gateway, gateway calls can be restricted on a per-caller basis using dial plans. Up to eight dial plans can be configured to restrict gateway calls in either direction.

Dial Plan Digit Sequences

The plans contain a series of digit sequences, separated by a vertical bar (|). The collection of sequences is enclosed in parentheses.

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

Any one of a set of terminating events triggers the Linksys ATA 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.

Configuring a Dial Plan

- 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, then the <Interdigit_Short_Timeout> applies.

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

Table 1-26 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 |



Note

Early production versions of the SPA2102 supported dual-line telephones on a single FXS port. A subsequent hardware change revised the FXS ports on the SPA2102 and this function is to be reserved for future development.

Dial Plan Rules

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



Note

White space is ignored, but 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”.

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

Local Timer Overrides

The long and short time-out 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.

Configuring a Dial Plan

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

Implicit Sequences

The Linksys ATA 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, then IP dialing is also accepted on the associated line.

Dial Plan Examples

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

```
( 1 xxx xxxxxxxx )
```

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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


Dial Plan Timers

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

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

Interdigit Long Timer

| | |
|----------------|-----------------------|
| ParName | Interdigit_Long_Timer |
| Default | 10 |

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

Interdigit Short Timer

| | |
|----------------|------------------------|
| ParName | Interdigit_Short_Timer |
| Default | 3 |

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

Dial Plans

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

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

Secure Call Implementation

This section describes secure call implementation with a Linksys ATA. It includes the following topics:

- [Enabling Secure Calls, page 3-10](#)
- [Secure Call Details, page 3-10](#)
- [Using a Mini-Certificate, page 3-11](#)
- [Generating a Mini-Certificate, page 3-11](#)

Enabling Secure Calls

A secure call is established in two stages. The first stage is no different from normal call setup. The second stage starts after the call is established in the normal way with both sides ready to stream RTP packets. I

In the second stage, the two parties exchange information to determine if the current call can switch over to the secure mode. The information is transported by base64 encoding embedded in the message body of SIP INFO requests, and responses using a proprietary format. If the second stage is successful, the Linksys ATA plays a special Secure Call Indication Tone for a short time to indicate to both parties that the call is secured and that RTP traffic in both directions is being encrypted.

If the user has a phone that supports call waiting caller ID (CIDCW) and that service is enabled, the CID will be updated with the information extracted from the Mini-Certificate received from the remote party. The Name field of the CID will be prepended with a '\$' symbol. Both parties can verify the name and number to ensure the identity of the remote party.

The signing agent is implicit and must be the same for all Linksys ATAs that communicate securely with each other. The public key of the signing agent is pre-configured into the Linksys ATA by the administrator and is used by the Linksys ATA to verify the Mini-Certificate of its peer. The Mini-Certificate is valid if it has not expired, and it has a valid signature.

The Linksys ATA can be configured so that, by default, all outbound calls are either secure or not secure. If secure by default, the user has the option to disable security when making a call by dialing *19 before dialing the target number. If not secure by default, the user can make a secure outbound call by dialing *18 before dialing the target number. However, the user cannot force inbound calls to be secure or not secure; that depends on whether the caller has security enabled or not.

The Linksys ATA will not switch to secure mode if the CID of the called party from its Mini-Certificate does not agree with the user-id used in making the outbound call. The Linksys ATA performs this check after receiving the Mini-Certificate of the called party

Secure Call Details

Looking at the second stage of setting up a secure call in greater detail, this stage can be further divided into two steps.

1. The caller sends a "Caller Hello" message (base64 encoded and embedded in the message body of a SIP INFO request) to the called party with the following information:
 - Message ID (4B)
 - Version and flags (4B)
 - SSRC of the encrypted stream (4B)

- Mini-Certificate (252B)

Upon receiving the Caller Hello, the called party responds with a Callee Hello message (base64 encoded and embedded in the message body of a SIP response to the caller's INFO request) with similar information, if the Caller Hello message is valid. The caller then examines the Callee Hello and proceeds to the next step if the message is valid.

2. The caller sends the "Caller Final" message to the called party with the following information:
 - Message ID (4B)
 - Encrypted Master Key (16B or 128b)
 - Encrypted Master Salt (16B or 128b)

The Master Key and Master Salt are encrypted with the public key from the called party mini-certificate. The Master Key and Master Salt are used by both ends for deriving session keys to encrypt subsequent RTP packets. The called party then responds with a Callee Final message (which is an empty message).

Using a Mini-Certificate

The Linksys ATA Mini-Certificate (MC) contains the following information:

- User Name (32B)
- User ID or Phone Number (16B)
- Expiration Date (12B)
- Public Key (512b or 64B)
- Signature (1024b or 512B)

The MC has a 512-bit public key used for establishing secure calls. The administrator must provision each subscriber of the secure call service with an MC and the corresponding 512-bit private key. The MC is signed with a 1024-bit private key of the service provider, which acts as the CA of the MC. The 1024-bit public key of the CA signing the MC must also be provisioned for each subscriber.

The CA public key is used by the Linksys ATA to verify the MC received from the other end. If the MC is invalid, the Linksys ATA will not switch to secure mode. The MC and the 1024-bit CA public key are concatenated and base64 encoded into the single parameter <Mini Certificate>. The 512-bit private key is base64 encoded into the <SRTP Private Key> parameter, which should be kept secret, like a password.

Because the secure call establishment relies on exchange of information embedded in message bodies of SIP INFO requests/responses, the service provider must ensure that the network infrastructure allows the SIP INFO messages to pass through with the message body unmodified.

Generating a Mini-Certificate

Linksys provides a configuration tool called `gen_mc` for the generation of MC and private keys with the following syntax:

```
gen_mc ca-key user-name user-id expire-date
```

Where:

- *ca-key* is a text file with the base64 encoded 1024-bit CA private/public key pairs for signing/verifying the MC, such as the following:

Configuring a Streaming Audio Server

```
9CC9aYU1X51JuU+EBZmi3AmcqE9U1LxE0GwopaGyG0h3VyhKgi6JaVtQZt87PiJINKW8XQj3B9Qqe3VgYx
WCQNa335YcNdsenASeBxumIEaBCYd111fVEodJZOGwXfAde0MhcbD0kj7LV1zcsTyk2TZYTccnZ75TuTj
j13qvYs=
```

```
5nEtOrkCa84/mEw13D9tSvVLy1iwQ+u/Hd+C8u5SNk7hsAUZaA9TqH8Iw0J/IqSrsf6scsmundY5j7Z5mK
5J9uBxSB8t8vamFGD0pF4zhNtbrVvIXKI9kmp4vph1C5jzO9gDfs3MF+zjyYrVUFdM+pXtDBxmM+fGUfrp
AuXb7/k=
```

- *user-name* is the name of the subscriber, such as “Joe Smith”. Maximum length is 32 characters
- *user-id* is the User ID of the subscriber, which must match exactly the user-id used in the INVITE when making the call, such as “14083331234”. The maximum length is 16 characters.
- *expire-date* is the expiration date of the MC, such as “00:00:00 1/1/34” (34=2034). Internally the date is encoded as a fixed 12B string: 000000010134

The tool generates the <Mini Certificate> and <SRTP Private Key> parameters that can be provisioned to the Linksys ATA.

For Example:

```
gen_mc ca_key "Joe Smith" 14085551234 "00:00:00 1/1/34"
```

Produces the following Mini Certificate and SRTP Private Key:

```
<Mini Certificate>
Sm91IFNtaXR0AAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAxNDA4NTU1MTIzNAAAAAAAAAMDawMDawMDEwMTM00vJakde2v
VMF3Rw4pPXL71AgIagMpbLSAG2+++Y1Sqt198Cp9rP/xMGFfoPmDKGx6JFtkQ5sxLcuwgxpPxkeXvpZK1Y1psb28L
4Rhg5qZA+Gqj1hDFCmG6dffZ9SjhxES767G0JIS+N81QBLr0AuemotknSjjjOy8c+11TCd2t44Mh0vmwNg4fDck2Yd
mTMBR516xJt4/uQ/LJQlni2kwqlm7scDv115k232EvvvVtCK0AYa4eWd6fQOpiESCO9CC9aYU1X51JuU+EBZmi3Amc
qE9U1LxE0GwopaGyG0h3VyhKgi6JaVtQZt87PiJINKW8XQj3B9Qqe3VgYxWCQNa335YcNdsenASeBxumIEaBCYd111
fVEodJZOGwXfAde0MhcbD0kj7LV1zcsTyk2TZYTccnZ75TuTjj13qvYs=
<SRTP Private Key>
b/DWc96X4YQraCnYz15en1CIUhVQqrvcr6Qd/8R52IEvJjOw/e+Klm4Xi iFEpaKmU8UbooxKG36SEdKusp0AQ==
```

Configuring a Streaming Audio Server

This section describes how to use and configure a streaming audio server (SAS). It includes the following topics:

- [Music On Hold, page 3-12](#)
- [Using a Streaming Audio Server, page 3-13](#)
- [Using the IVR with an SAS Line, page 3-13](#)
- [Example SAS with MOH, page 3-14](#)
- [SAS Line Not Registered with the Proxy Server, page 3-14](#)

Music On Hold

On a connected call, the Linksys ATA may place the remote party on hold by performing a hook-flash to initiate a three-way call or by swapping two calls during call-waiting. If the remote client indicates that it can still receive audio while the call is holding, the Linksys ATA can be configured to contact an auto-answering streaming audio server (SAS) to stream audio to the holding party. When used this way, the SAS is referred to as an MOH Server.

Using a Streaming Audio Server

The SAS feature lets you use attach an audio source to one of the Linksys ATA FXS ports (Phone 1 or Phone 2 on the PAP2T) and use it as a streaming audio source device. If the Linksys ATA has multiple FXS ports, either or both of the associated lines (Line 1 and Line 2 on the PAP2T) can be configured as an SAS server.

To connect an external music source to an FXS port, use a media signal adapter, which provides a line in from a media source and a RJ-11 port for connecting to the FXS port on the Linksys ATA. The following is a URL for a device that has been tested with Linksys ATAs:

http://www.neogadgets.com/cart/cart.php?target=product&product_id=17&substring=music+coupler

After installing the music source using the media signal adapter and completing the required configuration on the Linksys ATA, when the line is called and the FXS port is off hook, the Linksys ATA answers the call automatically and streams audio to the caller.

If the FXS port is on-hook when the incoming call arrives, the Linksys ATA replies with a SIP 503 response code (Service Not Available). The SAS line will not ring for incoming calls even if the attached equipment is on-hook.

If an incoming call is auto-answered, but later the FXS port changes to on-hook, the SPA does not terminate the call but continues to stream silence packets to the caller. If an incoming call arrives when the SAS line has reached full capacity, the SPA replies with a SIP 486 response (Busy Here).

The SAS line can be set up to refresh each streaming audio session periodically using a SIP re-INVITE message, which detects if the connection to the caller is down. If the caller does not respond to the refresh message, the SAS line terminates the call so that the streaming resource can be used for other callers.

Each SAS server can maintain up to five simultaneous calls. If the second line on the Linksys ATA is disabled, then the SAS line can maintain up to 10 simultaneous calls. Further incoming calls will receive a busy signal (SIP 486 Response).

If no calls are in session, battery is removed from tip-and-ring of the FXS port. Some audio source devices have an LED to indicate the battery status. This can be used as a visual indication as to whether audio streaming is in progress.

Set up the Proxy and Subscriber Information for the SAS Line as you normally would with a regular user account.

Call Forwarding, Call Screening, Call Blocking, DND, and Caller-ID Delivery features are not available on an SAS line.

Using the IVR with an SAS Line

The IVR can still be used on an SAS line, but the user needs to follow the following steps:

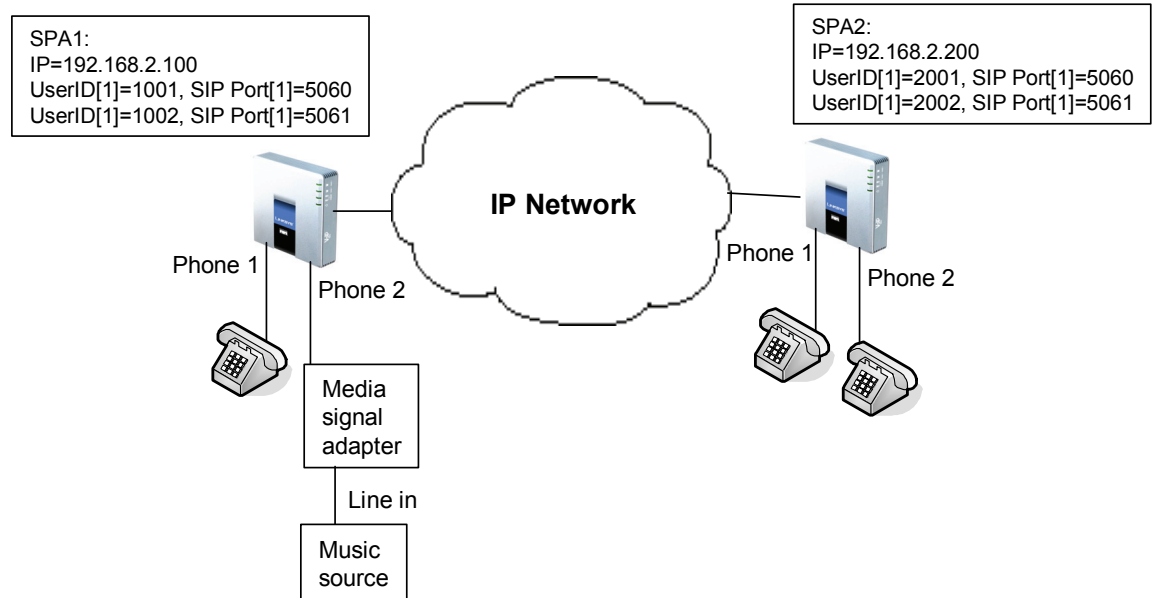
-
- Step 1** Power off the Linksys ATA.
 - Step 2** Connect a phone to the port and make sure the phone is on-hook.
 - Step 3** Power on the Linksys ATA.
 - Step 4** Pick up handset and press * * * * to invoke IVR in the usual way.
-

Configuring a Streaming Audio Server

If the Linksys ATA boots and finds that the SAS line is on-hook, it will not remove battery from the line so that IVR may be used. But if the Linksys ATA boots up and finds that the SAS line is off-hook, it will remove battery from the line because no audio session is in progress.

Example SAS with MOH

Figure 1-15 MOH Application with a Linksys ATA Line Configured as an SAS



The following configuration examples are based on the configuration shown in [Figure 1-15](#).

SAS Line Registered with the Proxy Server

In this example, the SAS Line is registered with the Proxy Server as the other subscribers.

On Linksys ATA 1:

```
SAS Enable[1] = no
MOH Server [1] = 1002
SAS Enable[2] = yes
```

On Linksys ATA 2:

```
SAS Enable[1] = no
MOH Server [1] = 1002
SAS Enable[2] = no
MOH Server [2] = 1002
```

SAS Line Not Registered with the Proxy Server

In this example, the SAS Line is not registered with the Proxy Server for the other subscribers.

On Linksys ATA 1:

```
SAS Enable[1] = no
MOH Server [1] = 1002@192.168.2.100:5061 or 1002@127.0.0.1:5061
SAS Enable[2] = yes
```

On Linksys ATA 2:

```
SAS Enable[1] = no
MOH Server [1] = 1002@192.168.2.100:5061
SAS Enable[2] = no
MOH Server [2] = 1002@192.168.2.100:5061
```

Configuring the Streaming Audio Server

The following provides step-by-step procedures for implementing an SAS with an external music source.

- Step 1** Connect an RJ-11 adapter between the music source and an FXS port on the Linksys ATA (Phone 1 or Phone 2).
- Step 2** On the administration web server, click the SIP tab and scroll down to the Streaming Audio Server section (see [Figure 1-16](#)).

Figure 1-16 Enabling the Streaming Audio Server

Streaming Audio Server (SAS)

SAS Enable: SAS DLG Refresh Intvl:

SAS Inbound RTP Sink:

- Step 3** On the SAS Enable pull-down selection list, select **yes**.
- Step 4** In the MOH Server field of the Call Feature Settings section, enter the User ID configured for the line attached to the audio source.

If the line is not registered with the SIP proxy, enter the IP address and SIP port number configured for the line.
- Step 5** Click **Submit All Changes**.

Using a FAX Machine with the SPA2102 or SPA8000



Note

T.38 Fax is only supported on the SPA2102 and the SPA8000. The SPA2102 supports a single connection, while the SPA8000 supports one connection for each pair of ports (1/2, 3/4, 5/6, and 7/8) for a maximum of four connections.

Procedure

To optimize fax completion rates, complete the following steps:

Using a FAX Machine with the SPA2102 or SPA8000

-
- Step 1** Upgrade the ATA firmware to the latest version
- Step 2** Ensure that you have enough bandwidth for uplink and downlink.
- For G.711 fallback, it is recommend to have ~100Kbps.
 - For T.38, allocate at least 50 kbps.
- Step 3** To optimize G.711 fallback fax completion rates, use the following settings on the Line tab of the Linksys ATA:
- Network Jitter Buffer: very high
 - Jitter buffer adjustment: disable
 - Call Waiting: no
 - 3 Way Calling: no
 - Echo Cancellor: no
 - Silence suppression: no
 - Preferred Codec: G.711
 - Use pref. codec only: yes

- Step 4** If you are using a Cisco media gateway for PSTN termination, disable T.38 (fax relay) and enable fax using modem passthrough.

For example:

```
modem passthrough nse payload-type 110 codec g711ulaw
fax rate disable
fax protocol pass-through g711ulaw
```

- Step 5** Enable T.38 fax on the SPA 2102 by configuring the following parameter on the Line tab for the FXS port to which the FAX machine is connected:

FAX_Passthru_Method: ReINVITE



Note If a T.38 call cannot be set-up, then the call should automatically revert to G.711 fallback.

- Step 6** If you are using a Cisco media gateway use the following settings:

Make sure the Cisco gateway is correctly configured for T.38 with the SPA dial peer. For example:

```
fax protocol T38
fax rate voice
fax-relay ecm disable
fax nsf 000000
no vad
```

Troubleshooting

If have problems sending or receiving faxes, complete the following steps:

-
- Step 1** Send a test Fax in a controlled environment between two Linksys ATAs.
- Step 2** Determine the success rate.
- Step 3** Monitor the network and record the following statistics:
- Jitter

- Loss
- Delay

Step 4 If faxes fail consistently, capture a copy of the web interface settings by selecting **Save As > Web page, complete** from the administration web server page.

Step 5 Enable and capture the debug log. For instructions, refer to Item 4. in “[Troubleshooting and Configuration FAQ](#)” at the end of this chapter.

Step 6 Identify the type of Fax machine connected to the ATA.

Step 7 Contact technical support.

If you are an end user of Linksys VoIP products, contact the reseller or Internet telephony service provider (ITSP) that supplied the equipment.

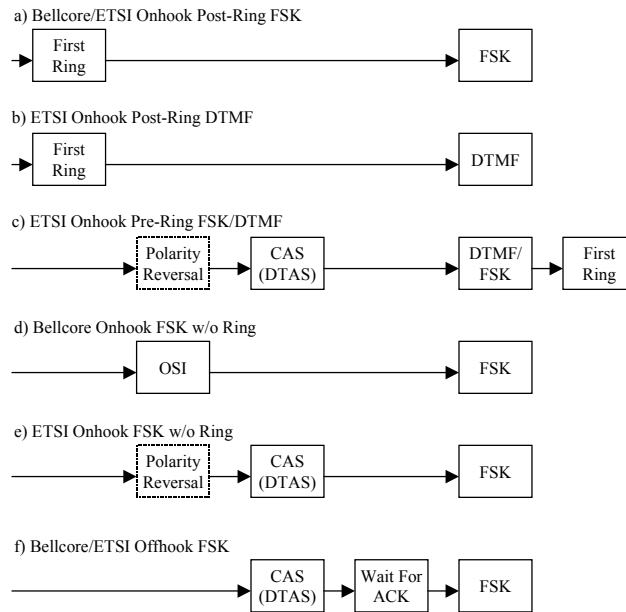
If you are an authorized Linksys Voice System partner, contact Linksys technical support.

Managing Caller ID Service

The choice of caller ID (CID) method affects the following features:

- On Hook Caller ID Associated with Ringing – This type of Caller ID is used for incoming calls when the attached phone is on hook. See figure below (a) – (c). All CID methods can be applied for this type of CID.
- On Hook Caller ID Not Associated with Ringing – This feature is used for send VMWI signal to the phone to turn the message waiting light on and off (see Figure 1 (d) and (e)). This is available only for FSK-based CID methods: (Bellcore, ETSI FSK, and ETSI FSK With PR).
- Off Hook Caller ID – This is used to delivery caller-id on incoming calls when the attached phone is off hook (see [Figure 1-17](#)). This can be call waiting caller ID (CIDCW) or to notify the user that the far end party identity has changed or updated (such as due to a call transfer). This is available only for FSK-based CID methods: (Bellcore, ETSI FSK, and ETSI FSK With PR).

Figure 1-17 Linksys ATA Caller ID Delivery Architecture



Troubleshooting and Configuration FAQ

This section provides solutions to problems that may occur during the installation and operation of the Linksys ATA. If you can't find an answer here, check the Linksys website at www.linksys.com.

1. ***I want to use a different computer to access the administration web server. I entered `http://ipaddress`, but this address did not work.***
Use the Interactive Voice Response Menu to find out the Linksys ATA Internet IP address. Follow these steps:
 - A. Use a telephone connected to the Phone 1 port of the Linksys ATA.
 - B. Press **** (in other words, press the star key four times).
 - C. Wait until you hear "Linksys configuration menu. Please enter the option followed by the # (pound) key or hang up to exit."
 - D. Press **110#**.
 - E. You hear the IP address assigned to the Linksys ATA Internet (external) interface. Write it down.
 - F. Press **7932#**.
 - G. Press **1** to enable WAN access to the administration web server.
 - H. Open the web browser on a networked computer.
 - I. Enter `http://(Internet IP address of the Linksys ATA)`.

2. ***I'm trying to access the Linksys ATA administration web server, but I do not see the login screen. Instead, I see a screen saying, "404 Forbidden."***
If you are using Windows Explorer, perform the following steps until you see the administration web server login screen (Netscape Navigator requires similar steps):
 - A. Click **File**. Make sure *Work Offline* is NOT checked.

- B. Press **CTRL + F5**. This is a hard refresh, which forces Windows Explorer to load new webpages, not cached ones.
- C. Click **Tools**. Click **Internet Options**. Click the **Security** tab. Click the **Default level** button. Make sure the security level is Medium or lower. Then click the **OK** button.

3. *How do I save my current SPA configuration?*

Currently, the only way is to do HTTPGET from an HTTP client, from which you get the entire HTML page. Alternatively, from your browser you can select **File > Save as > HTML** from any of the administration web server pages. Do this in Admin, Advanced mode. This saves all the tabs into one HTML file. This HTML file is helpful to provide to our support team when you have a problem or technical question.

4. *How do I debug my SPA? Is there a syslog?*

SPA sends out debug information via syslog to a syslog server. The ports can be configured (by default the port is 514).

- A. Make sure you do not have firewall running on your PC that could block port 514.
- B. On the administration web server System tab, set <Debug Server> as the IP address and port number of your syslog server. Note that this address has to be reachable from the Linksys ATA).
- C. Also, set <Debug level> to **3**.
You do not need to change the value of the <syslog server> parameter.
- D. To capture SIP signaling messages, under the Line tab, set <SIP Debug Option> to **Full**.
The file output is syslog.<portnum>.log (for the default port setting, syslog.514.log)

5. *How do I access the Linksys ATA if I forget my password?*

By default, the User and Admin accounts have no password. If the ITSP set the password for either account and you do not know what it is, you need to contact the ITSP. If the password for the user account was configured after you received the Linksys ATA, you can reset the device to the user factory default, which preserves any provisioning completed by the ITSP. If the Admin account needs to be reset, you have to perform a full factory reset, which also erases any provisioning.

To reset the Linksys ATA to the factory defaults, perform the following steps:

- A. Connect an analog phone to the Linksys ATA and access the IVR by pressing ****.
 - B. Press the appropriate code to reset the unit:
 - Press 877778# to reset the unit to the defaults as it shipped from the ITSP. This will reset the User account password to the default of blank.
 - Press 73738# to perform a full reset of unit to the defaults as it shipped from Linksys. This will reset the Admin account password to the default of blank.
 - C. Press 1 to confirm the operation.
Press * to cancel the operation.
 - D. Login to the unit using the User or Admin account without a password and reconfigure the unit as necessary.
- ### 6. *My Linksys ATA is behind a NAT device or firewall and I'm unable to make a call or I'm only receiving a one-way connection. What should I do?*
- A. Configure your router to port forward "TCP port 80" to the ip address currently being used by SPA. If you do this often, we suggest that you use static IP address for the SPA, instead of DHCP. (For help with port forwarding, consult your router documentation)
 - B. On the Line tab of the administration web server, change the value of <Nat Mapping Enable> to **yes**. On the SIP tab; change <Substitute VIA Addr> to **yes**, and <EXT IP> to the IP address of your router.

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- C. Make sure you are not blocking the UDP PORT 5060,5061 and port for UDP packets in the range of 16384-16482. Also, disable “SPI” if this feature is provided by your firewall. Identify the SIP server to which the Linksys ATA is registering, if it supports NAT, using the <Outbound Proxy> parameter.
- D. Add a STUN server to allow traversal of UDP packets through the NAT device. On the SIP tab of the administration web server, set <STUN Enable> to **yes**, and enter the IP address of the STUN server in <STUN Server>.

STUN (Simple Traversal of UDP through NATs) 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>



Note

STUN does not work with a symmetric NAT router. Enable debug through syslog (see FAQ#10), and set <STUN Test Enable> to **yes**. The messages indicate whether you have symmetric NAT or not.

Configuring the PSTN Gateway (FXO)

This chapter describes how to configure the PSTN gateway provided by Analog Telephone Adapters (ATAs) with one or more FXO ports, which includes the AG310 and SPA3102. It includes the following sections:

- [Overview, page 4-1](#)
- [How VoIP-To-PSTN Calls Work, page 4-2](#)
- [How PSTN-To-VoIP Calls Work, page 4-4](#)
- [Configuring VoIP Failover to PSTN, page 4-7](#)
- [Sharing One VoIP Account Between the FXS and PSTN Lines, page 4-7](#)
- [Other Options, page 4-7](#)
- [Call Scenarios, page 4-8](#)



Note

Throughout this chapter, when references are made to the software configuration of the SPA3102, the information also applies to the AG310. The AG310 provides the same functionality as the SPA3102, as well as an ADSL modem and a four-port Ethernet switch.

Overview

The SPA3102 and AG310 have the following ports for connection to telephony devices:

- FXS port (Phone)—Connected to a standard analog telephone or fax machine, configured using the Line tab
- FXO port (Line)—Connected to a standard telephone wall jack for connectivity to the PSTN, configured using the PSTN Line tab

Line 1 does not provide a gateway because it provides only VoIP service. The VoIP-To-PSTN calling function is referred to as a *PSTN gateway*, and PSTN-To-VoIP calling function as a *VoIP gateway*. Note the following definitions:

- VoIP caller—One who calls the AG310 via VoIP to obtain PSTN service
- VoIP user—VoIP caller that has a user account (user-id and password) on the AG310
- PSTN caller—One who calls the AG310 from the PSTN to obtain VoIP service

Line 1 can be configured with a regular VoIP account and can be used in the same way as the Line 1 of any Linksys ATA.

With the SPA3102 and AG310, a second VoIP account can be configured to support PSTN gateway calls exclusively. A different SIP port should be assigned to Line 1 and the PSTN Line. The same VoIP account may be used for both Line 1 and the PSTN Line if a different SIP port is assigned to each.

VoIP callers can be authenticated by one of the following methods:

- No Authentication—All callers are accepted for service
- PIN—Caller is prompted to enter a PIN right after the call is answered
- HTTP digest—SIP INVITE must contain a valid authorization header

PSTN callers can be authenticated by one of the following methods:

- No authentication—All callers are accepted for service
- PIN—Caller is prompted to enter a PIN right after the call is answered

How VoIP-To-PSTN Calls Work

To obtain PSTN services through the SPA3102, the VoIP caller establishes a connection with the PSTN Line by way of a standard SIP INVITE request addressed to the PSTN Line. The PSTN Line can be configured to support one-stage and two-stage dialing as described in the following sections.

One-Stage Dialing

The Request-URI of the INVITE to the PSTN Line should have the form <Dialed-Number>@<SPA-Address>, where <Dialed-Number> is the number dialed by the VoIP caller, and <SPA-Address> is a valid address of the SPA3102, such as 10.0.0.100:5061.

If the FXO port is currently in use (off-hook) or the PSTN line is being used by another extension, the SPA3102 replies to the INVITE with a 503 response. Otherwise, it compares the <Dialed-Number> with the <User ID> of the PSTN Line. If they are the same, the SPA3102 interprets this as a request for two-stage dialing (see the “[Two-Stage Dialing](#)” section on page 4-3). If they are different, the SPA3102 processes the <Dialed-Number> using the corresponding <Dial Plan>.

If dial plan processing fails, the SPA3102 replies with a 403 response. Otherwise, it replies with a 200 and at the same time takes the FXO port off hook and dials the target number returned after processing the dial plan.



Note

If <User ID> on the PSTN Line is blank, <Registration> should be disabled for the PSTN Line.

If HTTP Digest Authentication is enabled, the SPA3102 challenges the INVITE with a 401 response if it does not have a valid Authorization header. The Authorization header should include a <User ID *n*> parameter, where *n* refers to one of eight VoIP user accounts that can be configured on the SPA3102. The credentials are computed based on the corresponding password using Message Digest 5 (MD5). The <User ID *n*> must match one of the VoIP accounts stored on the SPA3102. Each VoIP user account contains the information listed in [Table 1-27](#).

Table 1-27 VoIP User Account Information

| Parameter | Group | Description | Values |
|---------------------------------|--------------|---|---------------------|
| User ID 1/2/3/4/5/6/7/ 8 | PSTN Line | The username value. | 31-character string |
| Password 1/2/3/4/5/6/7/ 8 | PSTN Line | The password value. | 31-character string |
| User 1/2/3/4/5/6/7/ 8 DP | PSTN Line | Specifies the dial plan to be used for this VoIP user. If 0, dial plan processing is disabled; the given target number is dialed to the PSTN as is. | Choice of 0-8 |



Note If Authentication is disabled, a default dial plan is used for all unknown VoIP users.

Two-Stage Dialing

In two-stage dialing, the SPA3102 takes the FXO port off-hook but does not automatically dial any digits after accepting the call. To invoke two-stage dialing, the VoIP caller should INVITE the PSTN Line without the user-id in the Request-URI or with a user-id that matches exactly the <User ID *n*> of the PSTN Line. A different user-id in the Request-URI is treated as a request for one-stage dialing if one-stage dialing is enabled, or dropped by the SPA3102 (as if no user-id is given) if one-stage dialing is disabled.



Note If Authentication is disabled, a default dial plan is assigned to all VoIP callers.

HTTP Digest Authentication can be also used for two-stage dialing, as in one-stage dialing. If using HTTP Digest Authentication or Authentication is disabled, the VoIP caller should hear the PSTN dial tone right after the call is answered (by a SIP 200 response).

If PIN Authentication is enabled, the VoIP caller is prompted to enter a PIN number after the SPA3102 answers the call. The PIN number must end with a # key. The inter-PIN-digit timeout is 10 seconds (not configurable). Up to eight VoIP caller PIN numbers can be configured on the SPA3102. A dial plan can be selected for each PIN number. If the caller enters a wrong PIN or the SPA3102 times out waiting for more PIN digits, the SPA3102 tears down the call immediately with a BYE request.



Note When the source address of the INVITE is 127.0.0.1, authentication is automatically disabled because this is a call by the local user. This applies to both one-stage and two-stage dialing.

Table 1-28 lists the parameters used in two-stage dialing.

Table 1-28 Two-Stage Dialing

| Parameter | Group | Description | Values |
|---------------------------------------|--------------|---|---------------------|
| VoIP Caller 1/2/3/4/5/6/7/8 PIN | PSTN Line | The PIN for VoIP Caller 1, 2, 3, 4, 5, 6, 7, or 8. | 31-character string |
| VoIP Caller 1/2/3/4/5/6/7/8 DP | PSTN Line | Specifies which dial plan to be used for this VoIP caller. If 0, dial plan processing is disabled; the given target number is dialed to the PSTN as is. | Choice of 1 to 8 |

How PSTN-To-VoIP Calls Work

PSTN-To-VoIP calls can be made with two-stage dialing only. The only authentication method available is the PIN method.

The SPA3102 takes the FXO port off hook after a configurable number of rings. If PIN Authentication is enabled, it prompts the caller to enter the PIN number followed by a # key. The Inter-PIN-digit timeout is set at 10 seconds. Up to eight PSTN PIN numbers can be configured in the SPA3102. If the given PIN does not match any of the PSTN PIN values, the SPA3102 plays the reorder tone to the FXO port for up to 10 seconds, and then takes the FXO port on-hook. If the given PIN matches one of PSTN PIN values, the SPA3102 plays dial tone to the FXO port and is ready to accept digits for the target VoIP number from the PSTN caller. The collected digits are processed by the dial plan associated with the PIN number.



Note

If Authentication is disabled, a default dial plan is used for all PSTN callers.

Terminating Gateway Calls

There are two call legs in a PSTN gateway call: the PSTN call leg and the VoIP call leg. A gateway call is terminated when either call leg is ended. The SPA3102 takes the FXO port on-hook when the call terminates so the PSTN line can be used again. The SPA3102 detects that the PSTN call leg is ended when one of the following conditions occurs during a call:

- The PSTN Line voltage drops to a very low value (this occurs if the line is disconnected from the PSTN service or if the PSTN switch provides a CPC signal)
- A polarity reversal or disconnect tone is detected at the FXO port
- When there is no voice activity for a configurable period of time in either direction at the FXO port

When any of the above occurs, the SPA3102 takes the FXO port on hook and sends a BYE request to end the VoIP call leg. On the other hand, when the SPA3102 receives a SIP BYE from the VoIP during a call, it takes the FXO port on hook to end the PSTN call leg.

In addition, the SPA3102 can also send a refresh signal periodically to the VoIP call leg to determine whether the call leg is still up. If a refresh operation fails, the SPA3102 ends both call legs.

Table 1-29 lists parameters for terminating gateway call parameters.

Table 1-29 Terminating Gateway Call Parameters

| Parameter | Group | Description | Values |
|---------------------------|-----------|---|--|
| Detect CPC: | PSTN Line | If yes, SPA3102 detects CPC as a disconnect signal. Default = Yes | Yes or No |
| Detect Long Silence: | PSTN Line | If yes, SPA3102 detects prolonged silence period as a disconnect signal. Default = Yes | Yes or No |
| Long Silence Duration: | PSTN Line | The minimum duration of continuous silence before the SPA3102 disconnects the call, if <Detect Long Silence> is enabled. Default = 30 (s) | 10-255 |
| Disconnect Tone: | PSTN Line | Tone Script of the disconnect tone to detect. Default = “480@-30,620@-30;4(.25/.25/1+2)”. Note: The SPA3102 supports two frequency components. If the tone has only one frequency, use the same value for both frequencies. Each cadence segment must have the same frequency. The level value is the threshold to detect each tone. The total duration is the minimum duration of the tone to be recognized as the disconnect tone | ToneScript |
| Detect Polarity Reversal: | PSTN Line | If yes, SPA3102 interprets polarity reversal as a disconnect signal. On an inbound PSTN call, SPA3102 disconnects on the first polarity reversal. On an outbound PSTN call, SPA3102 disconnects on the second polarity reversal (because the first polarity reversal indicates the outbound call is connected). Default = Yes | Yes or No |
| Detect Disconnect Tone: | PSTN Line | If yes, SPA3102 interprets the disconnect tone as specified in <Disconnect Tone> as the disconnect signal. Default = Yes | Yes or No |
| Silence Threshold: | PSTN Line | This is the signal energy threshold. Below this threshold is considered silence. Default = medium | very low, low, medium, high, very high |

VoIP Outbound Call Routing

Calls made from Line 1 are routed through the configured Line 1 service provider, by default. You can override this behavior by IP dialing, through which the calls can be routed to any IP address entered by the user. SPA3102 allows flexible call routing with four sets of gateway parameters and configurable dial plans. [Table 1-30](#) lists VoIP outbound call routing parameters.

Table 1-30 VoIP Outbound Call Routing Parameters

| Parameters | Group | Description | Values |
|----------------------------|--------|--|---------------------------|
| Gateway 1 | Line 1 | Fully qualified domain name (or IP address) of a gateway. If the port number is not specified, 5060 is assumed. Default value is [blank] | Domain name or IP address |
| GW1 Nat Mapping Enable | Line 1 | Whether to enable NAT mapping when using Gateway 1. Default is “no”. | Yes or No |
| GW1 User ID | Line 1 | The authentication user name when using Gateway 1. Default is [blank] | 31-character string |
| GW2 Password | Line 1 | The authentication password when using Gateway 1. Default is [blank] | 31-character string |
| Similar for GW 2, 3, and 4 | Line 1 | Similar for GW 2, 3, and 4 | |

Gateways 1 to 4 can be specified in a dial plan with the special identifier gw1, gw2, gw3, or gw4. Also, gw0 represents the internal PSTN gateway via the FXO port. You can specify in the dial plan to use gw x ($x = 0,1,2,3,4$) when making certain calls. In general, you can specify any gateway address in the dial plan. In addition, three parameters are added that can be used with call routing:

- `usr`—User-id used for authentication with the given gateway
- `pwd`—Password used for authentication with the given gateway
- `nat`—Enable or disable NAT mapping when calling the gateway

Table 1-31 lists some examples.

Table 1-31 Specifying Gateway Addresses

| Example | Description |
|--|--|
| <code><9,:>xx.<:@gw1</code> | Dial 9 to start outside dial tone, followed by one or more digits, and route the call to Gateway 1. |
| <code>[93]11<:@gw0></code> | Route 911 and 311 calls to the local PSTN gateway |
| <code><8,:1408>xxxxxx<:@pstn.Linksys.com:5061;usr=joe;pwd=joe_pwd;nat></code> | Dial 8 to start outside dial tone, prepend 1408 followed by seven digits, and route the call to pstn.Linksys.com:5061, with user-id = joe, and pwd = bell_pwd, and enable NAT mapping |
| <code><8,:1408>xxxxxx<:@gw2:5061;usr="Alex Bell";pwd="anything";nat=no></code> | Dial 8 to start outside dial tone, prepend 1408 followed by seven digits, and route the call to Gateway 2, but use the given port, user-id, and password, and no pstn.Linksys.com:5061, and with user-id = “Alex Bell” and pwd = bell_pwd, and disable NAT mapping |

You can set up multiple PSTN gateways at different locations and configure Line 1 to use a different gateway when dialing specific numbers.

Configuring VoIP Failover to PSTN

When power is disconnected from the SPA3102, the FXS port is connected to the FXO port. In this case, the telephone attached to the FXS port is electrically connected to the PSTN service via the FXO port. When power is applied to the SPA3102, the FXS port is disconnected from the FXO port. However, if the PSTN line is in use when the power is applied to the SPA3102, the relay is not flipped until the PSTN line is released. This is done so that the SPA3102 does not interrupt any call in progress on the PSTN line.

When Line 1 VoIP service is down (because of registration failure or loss of network link), SPA3102 can be configured to automatically route all outbound calls to the internal gateway using the parameters listed in [Table 1-32](#).

Table 1-32 Automatically Routing Outbound Calls to Internal Gateway

| Parameter | Group | Description | Range |
|--------------------|--------|---|-------|
| Auto PSTN Fallback | Line 1 | If enabled, SPA3102 automatically routes outbound calls to Gateway 0 when registration fails or network link is down. | Bool |

Sharing One VoIP Account Between the FXS and PSTN Lines

The FXS (Line 1) and FXO (PSTN Line) can share a single VoIP account if they use different SIP ports. If the service provider allows multiple registration contacts and simultaneous ringing, both lines can register periodically with the service provider. In this case, both lines receive inbound calls to this VoIP account. The PSTN Line should be configured with a sufficiently long answer delay before the call is automatically answered to allow for the function of the PSTN gateway.

If the service provider does not allow more than one register contact, the PSTN Line should not register. In this case, only Line 1 rings on the inbound call to this VoIP account because it is the only line registered with the service provider.

Line 1 can have the call forwarded to the PSTN Line after a few seconds using the Call-Forward-On-No-Answer feature with gw0 as the forward destination. Similarly, Line 1 can apply Call-Forward-All, Call-Forward-On-Busy, and Call-Forward-Selective feature, and direct the caller to the PSTN-Gateway.

Only PIN authentication is allowed when a VoIP caller is forwarded to the PSTN-gateway from Line 1. If HTTP Authentication is used, the caller is not authenticated.

Another option when using the Forward-To-GW0 feature is to forward the caller to a specific PSTN number, using the syntax <PSTN-number>@gw0 in the forward destination. When using this with Call-Forward-Selective, you can develop some interesting applications. For example, you can forward all callers with 408 area code to 14081234567, or all callers with 800 area code to 18005558355 (This is the number for Tell Me). When this syntax is used, authentication is not used and the target PSTN number is automatically dialed by the SPA3102 after the caller is forwarded to gw0.

Other Options

This section describes other options provided by the SPA3102. It includes the following topics:

- [PSTN Call to Ring Line 1, page 4-8](#)

Call Scenarios

- Symmetric RTP, page 4-8
- Call Progress Tones, page 4-8

PSTN Call to Ring Line 1

This feature allows a PSTN caller to ring Line 1. When the PSTN line rings, the PSTN Line makes a local VoIP call to Line 1. If Line 1 is busy, it stops. After a given number of rings, the VoIP gateway picks up the call.

Symmetric RTP

Symmetric RTP is used to send audio RTP to the source IP and port of the inbound RTP packets. This facilitates NAT traversal.

Table 1-33 lists symmetric RTP parameters.

Table 1-33 Symmetric RTP Parameters

| Parameter | Group | Description | Range |
|---------------|-----------|---|-----------|
| Symmetric RTP | Line 1 | Enable symmetric RTP operation. If enabled, SPA3102 sends RTP packets to the source address of the last received valid inbound RTP packet. If disabled, SPA3102 sends RTP to the destination as indicated in the inbound SDP. Default is yes. | Yes or No |
| Symmetric RTP | PSTN Line | Same as above. Default is yes. | Yes or No |

Call Progress Tones

Table 1-34 Call Progress Tones

| Call Progress Tone | Description |
|--------------------|---|
| VoIP PIN Tone | This tone is played to prompt a VoIP caller to enter a PIN number. |
| PSTN PIN Tone | This tone is played to prompt a PSTN caller to enter a PIN number. |
| Outside Dial Tone | During two-stage PSTN-gateway dialing and with a dial plan assigned, the SPA3102 collects digits from the VoIP caller and processes the number using the dial plan. The SPA3102 plays the <Outside Dial Tone> to prompt the VoIP caller to enter the PSTN number. This tone should be specified to sound different from the PSTN dial tone. |

Call Scenarios

This section describes some typical scenarios where the SPA3102 can be applied. Some terms are introduced in the first few sections and reused in later sections. This section includes the following topics:

- [PSTN to VoIP Call with and Without Ring-Thru, page 4-9](#)
- [VoIP to PSTN Call with and Without Authentication, page 4-9](#)
- [Call Forwarding to PSTN Gateway, page 4-10](#)
- [User Dialing 9 to Access PSTN-Gateway for Local Calls, page 4-11](#)
- [Using the PSTN-Gateway for 311 and 911 Calls, page 4-12](#)
- [Auto-Fallback to the PSTN-Gateway, page 4-12](#)

PSTN to VoIP Call with and Without Ring-Thru

The PSTN caller calls the PSTN line connected to the FXO port. Ring-Thru is disabled. After it rings for a delay equals to the value in <PSTN Answer Delay>, the VoIP gateway answers the call and prompts the PSTN caller to enter a PIN number (assuming PIN authentication is enabled). After a valid PIN is entered, a regular dial plan is played to prompt the PSTN caller to dial the VoIP number. A dial plan is selected according to the PIN number entered by the caller. If authentication is disabled, the default PSTN dial plan is used. Note that the dial plan choice cannot be 0 for a PSTN caller.



Note

A <PSTN Access List> in terms of Caller ID (ANI) patterns can be configured into the SPA3102 to automatically grant access to the PSTN caller without entering the PIN. In this case, the default PSTN dial plan is also used.

The same scenario can be implemented using Ring-Thru. When the PSTN line rings, Line 1 rings also. This feature is called *Ring-Thru*. If Line 1 is picked up before the VoIP gateway auto-answers, it is connected to the PSTN call. Line 1 hears a call waiting tone if it is already connected to another call.

VoIP to PSTN Call with and Without Authentication

This section describes three scenarios with and without authentication and includes the following topics:

- [Using PIN Authentication, page 4-9](#)
- [Using HTTP Digest Authentication, page 4-10](#)
- [Without Authentication, page 4-10](#)

Using PIN Authentication

This scenario assumes that the PSTN Line has a different VoIP account than the Line 1 account. The VoIP caller calls the FXO number, which auto-answers after <VoIP Answer Delay>. The SPA3102 then prompts the VoIP caller for a PIN. When a valid PIN is entered, the SPA3102 plays the <Outside Dial Tone> and prompts the caller to dial the PSTN number.

The number dialed is processed by the dial plan corresponding to the VoIP caller. If the dial plan choice is 0, no dial plan is needed and the user hears the PSTN dial tone right after the PIN is entered. If the dial plan choice is not 0, the final number returned from the dial plan after the complete number is dialed by the caller is dialed to the PSTN. The caller does not hear the PSTN dial tone (except for a little leakage before the first digit of the final number is auto-dialed by the SPA3102).

If the PSTN Line is busy (off-hook, ringing, or PSTN line not connected) when the VoIP caller calls, the SPA3102 replies with 503. If the PIN number is invalid or entered after the VoIP call leg is connected, the SPA3102 plays the reorder tone to the VoIP caller and eventually ends the call when the reorder tone times out.



Note

If <VoIP Caller ID Pattern> is specified and the VoIP caller ID does not match any of the given patterns, the SPA3102 rejects the call with a 403. This rule applies regardless of the authentication method, even when the source IP address of the INVITE request is in the <VoIP Access List>.

Using HTTP Digest Authentication

The same scenario can be implemented with HTTP digest authentication when the calling device supports the configuration of a auth-ID and password to access the SPA3102 PSTN gateway. When the VoIP caller calls the PSTN Line, the SPA3102 challenges the INVITE request with a 401 response. The calling device should then provide the correct credentials in a subsequent retry of the INVITE, computed with the auth-ID and password using MD5.

If the credentials are correct, the target number specified in the user-id field of the INVITE Request-URI is processed by the dial plan corresponding to the VoIP user (assuming the dial plan choice is not 0). The final number is then auto-dialed by the SPA3102.

If the credentials are incorrect, the SPA3102 challenges the INVITE again. If the auth-ID does not exist in the SPA3102 configuration, the SPA3102 replies 403 to the INVITE. If the target number is invalid according to the corresponding dial plan, the SPA3102 also replies 403 to the INVITE. Again, if the PSTN Line is busy at the time of the call, the SPA3102 replies 503.



Note

HTTP Digest Authentication is one way to perform one-stage dialing of a VoIP-To-PSTN call. The other way is with no authentication require. However, if the target number is not specified in the Request-URI or the number matches the account user-id of the PSTN Line, the call reverts to two-stage dialing.

Without Authentication

This scenario can also be implemented without authentication, using one-stage or two-stage dialing, as in the HTTP Authentication case. The default VoIP caller dial plan is used in this scenario.

Authentication is performed when the method is none or when the source IP address of the inbound INVITE matches one of the <VoIP Access List> patterns.

Call Forwarding to PSTN Gateway

This section describes a number of scenarios that forward calls to the PSTN gateway. It includes the following topics:

- [Forward-On-No-Answer to the PSTN Gateway, page 4-11](#)
- [Forward-All to the PSTN gateway, page 4-11](#)
- [Forward to a Particular PSTN Number, page 4-11](#)
- [Forward-On-Busy to PSTN Gateway or Number, page 4-11](#)
- [Forward-Selective to PSTN Gateway or Number, page 4-11](#)

Forward-On-No-Answer to the PSTN Gateway

In this scenario, Line 1 is configured to <Forward-On-No-Answer> to the PSTN Gateway. The scenario is implemented by setting User 1 to forward to gw0 on no answer, with <No Answer Delay> set to six seconds.

The caller calls Line 1 and if Line 1 is not picked up after six seconds, the PSTN Line picks up the call and the call reverts to a PSTN-Gateway call, as described above. In this case, HTTP authentication is not allowed because Line 1 does not authenticate inbound INVITE requests. If you need to authenticate the VoIP caller in this case, you must select the PIN authentication method, or else the caller is *not* authenticated.

**Note**

If the PSTN Line is busy at the moment of the forward, it does not answer the VoIP call. The call forward rule is ignored and Line 1 continues to ring.

Forward-All to the PSTN gateway

In this scenario, Line 1 is configured with <Forward-All> to the PSTN gateway. This scenario is the same as the previous case, except the FXO picks up the Line 1 call immediately.

If the PSTN Line is busy at the moment of the call, the PSTN Line does not pick up the call, the call forward rule is ignored, and Line 1 continues to ring.

Forward to a Particular PSTN Number

In this scenario, the forward destination is set to <target-number>@gw0>. This is the same as in the previous examples, except that the SPA3102 automatically dials the given target number on the PSTN line right after it answers the VoIP call leg. This is a special case of one-stage dialing where the target number is specified in the configuration. The caller is not authenticated in this case regardless of the authentication method. However, the caller is still limited by the <VoIP Caller ID Pattern> parameter

Forward-On-Busy to PSTN Gateway or Number

This scenario is similar to the previous cases of call forwarding to gw0, but this applies when Line 1 is active.

Forward-Selective to PSTN Gateway or Number

This scenario is similar to the previous cases of call forwarding to gw0, but this applies when the caller matches the specific caller-id pattern.

User Dialing 9 to Access PSTN-Gateway for Local Calls

To implement this scenario, add the rule “<9,:1408>xxxxxxx<:@gw0>” to the Line 1 dial plan. When user dials 9, SPA3102 plays outside dial tone. The user then dials seven digits and the SPA3102 prepends 1408 before dialing the final number on the PSTN line.

Using the PSTN-Gateway for 311 and 911 Calls

To implement this scenario, add the rule “[39]11<:@gw0>” to Line 1. When the user dials 311 or 911, the call is routed to the PSTN gateway.



Note

If the PSTN Line is busy after the user dials 311 or 911, the call still fails. For true life-line supports, therefore, the PSTN Line cannot be shared.

Auto-Fallback to the PSTN-Gateway

To implement this scenario, enable <Auto PSTN Fallback>. When registration fails or link is down, the SPA3102 automatically calls “fallback@gw0” when user picks up Line 1. The SPA3102 does *not* reboot when the link is down. However, the SPA3102 reboots when the link is back up and Line 1 and PSTN Line are not in use.

Linksys ATA Field Reference

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

- Info Tab, page 5-2
- System Tab, page 5-8
- SIP Tab, page 5-11
- Regional Tab, page 5-19
- Line Tab, page 5-33
- PSTN Line Tab, page 5-49
- User 1/2 Tab, page 5-65
- PSTN User Tab (SPA3102/AG310), page 5-70



Note

Throughout this chapter, references to the SPA3102 also apply to the AG310. The AG310 provides the same functionality as the SPA3102, as well as an ADSL modem and a four-port Ethernet switch.

For information about the tabs on the Routing page, see the documentation for any Linksys router. For information about the Provisioning page, see the *Linksys SPA Provisioning Guide*.

Info Tab

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

- [System Information \(PAP2T\)](#), page 5-2
- [System Status \(VoIP\)](#), page 5-2
- [Line 1/2 Status](#), page 5-3
- [PSTN Line Status](#), page 5-5



Note

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

System Information (PAP2T)

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

System Status (VoIP)

| 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. |

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

Line 1/2 Status

| Field | Description |
|----------------------|---|
| Display Name | The value that is shown through Caller ID. |
| User ID | The user ID assigned by the ITSP (often the phone number) used for registering the SIP user agent with the SIP proxy at the ITSP. |
| Hazardous Potential | The amount of hazardous potential is displayed. |
| Foreign Voltage | The amount of foreign voltage is displayed. |
| Long Resistive Fault | The amount of long resistive fault is displayed. |
| REN | The Ringer Equivalence Number (REN) is displayed. |
| Hook State | Status of phone readiness. On indicates that the phone is ready for use; Off indicates that the phone is in use. |
| Registration State | Indicates if the line has registered with the SIP proxy. |
| Last Registration At | Last date and time the line was registered. |
| Next Registration In | Number of seconds before the next registration renewal. |

Info Tab

| | |
|---------------------------|---|
| 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. |
| Call Back Active | Indicates whether a call back request is in progress: Yes or No. |
| Last Called Number | Last number called. |
| Last Caller Number | Number of the last caller. |
| Mapped SIP Port | Port number of the SIP port mapped by NAT. |
| 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 FAX | Status of the fax pass-through mode. |
| 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. |

PSTN Line Status



Note

References to the SPA3102 also apply to the AG310.

| Field | Description |
|-----------------------------|--|
| (PSTN) Hook State | Hook state of the FXO port. Either On or Off. |
| (PSTN) Line Voltage | The voltage existing on the PSTN line. |
| (PSTN) Loop Current | The current (milliamperes) existing on the local loop. |
| Registration State | Indicates if the line has registered with the SIP proxy. |
| Last Registration At | Last date and time the line was registered. |
| Next Registration In | Number of seconds before the next registration renewal. |
| Last Called VoIP Number | The last VoIP number called from the FXO Line. |
| Last Called PSTN Number | The PSTN number dialed by the SPA (logged only if a non-trivial dial plan is used). |
| Last VoIP Caller | The last VoIP caller to the FXO Line. |
| Last PSTN Caller | Name and number of the last PSTN caller. |
| Last PSTN Disconnect Reason | Reason for SPA hanging up the FXO port. Can be one of the following: <ul style="list-style-type: none"> • PSTN Disconnect Tone • PSTN Activity Timeout • CPC Signal • Polarity Reversal • VoIP Call Failed • VoIP Call Ended • Invalid VoIP Destination • Invalid PIN • PIN Digit Timeout • VoIP Dialing Timeout • PSTN Gateway Call Timeout • VoIP Gateway Call Timeout |
| PSTN Activity Timer | Shows the time (ms) before the SPA disconnects the current gateway unless the PSTN side has some audio activity. |
| Mapped SIP Port | Same as Line 1. |

Info Tab

| | |
|----------------------------|---|
| Call Type | <p>May take one of the following values:</p> <ul style="list-style-type: none"> • PSTN Gateway Call = VoIP-To-PSTN Call • VoIP Gateway Call = PSTN-To-VoIP Call • PSTN To Line 1 = PSTN call ring through and answered by Line 1 • Line 1 Forward to PSTN Gateway = VoIP calls Line 1 then forwarded to PSTN GW • Line 1 Forward to PSTN Number = VoIP calls Line 1 then forwarded to PSTN number • Line 1 To PSTN Gateway • Line 1 Fallback To PSTN Gateway |
| VoIP State | Same as Line 1 Call 1. |
| PSTN State | <p>May take one of the following values:</p> <ul style="list-style-type: none"> • Idle • Collecting PSTN Pin • Invalid PSTN PIN • PSTN Caller Accepted • Connected to PSTN |
| VoIP Tone | Same as Line 1 Call 1 (Indicate what tone is being played to the VoIP call leg). |
| PSTN Tone | Indicate what tone is being played to the PSTN call leg. |
| VoIP Peer Name | Same as Line 1 Call 1 (Name of the party at the VoIP call leg). |
| PSTN Peer Name | Name of the party at the PSTN call leg. |
| VoIP Peer Number | Same as Line 1 Call 1 (Phone number of the party at the VoIP call leg). |
| PSTN Peer Number | Phone number of the party at the PSTN call leg. |
| VoIP Call Encoder | Same as Line 1 Call 1 (Audio encoder being used for the VoIP call leg). |
| VoIP Call Decoder | Same as Line 1 Call 1 (Audio decoder being used for the VoIP call leg). |
| VoIP Call FAX | Same as Line 1 Call 1. |
| VoIP Call Remote Hold | Same as Line 1 Call 1. |
| VoIP Call Duration | Same as Line 1 Call 1. |
| VoIP Call Packets Sent | Same as Line 1 Call 1. |
| VoIP Call Packets Recv | Same as Line 1 Call 1. |
| VoIP Call Bytes Sent | Same as Line 1 Call 1. |
| VoIP Call Bytes Recv | Same as Line 1 Call 1. |
| VoIP Call Decode Latency | Same as Line 1 Call 1. |
| VoIP Call Jitter | Same as Line 1 Call 1. |
| VoIP Call Round Trip Delay | Same as Line 1 Call 1. |

| | |
|---------------------------|------------------------|
| VoIP Call Packets Lost | Same as Line 1 Call 1. |
| VoIP Call Packet Error | Same as Line 1 Call 1. |
| VoIP Call Mapped RTP Port | Same as Line 1 Call 1. |

System Tab

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

- [System Configuration](#), page 5-8
- [Internet Connection Type \(PAP2T\)](#), page 5-8
- [Optional Network Configuration \(PAP2T\)](#), page 5-9
- [Miscellaneous Settings \(Not in PAP2T\)](#), page 5-10

System Configuration

| Field | Description |
|---------------------------|--|
| Restricted Access Domains | This feature is used when implementing software customization. |
| Enable Web Server | Enable/disable web server of Linksys IP 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 Linksys IP 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 (PAP2T)

| Field | Description |
|-----------|---|
| DHCP | Enable or disable DHCP. The default is yes. |
| Static IP | Static IP address of Linksys IP phone, which takes effect if DHCP is disabled. The default is 0.0.0.0. |
| NetMask | The NetMask used by Linksys IP phone when DHCP is disabled. The default is 255.255.255.0. |

| Field | Description |
|---------|--|
| Gateway | The default gateway used by Linksys IP phone when DHCP is disabled. The default is 0.0.0.0. |

Optional Network Configuration (PAP2T)

| Field | Description |
|----------------------|--|
| Host Name | The host name of the Linksys IP phone. |
| Domain | The network domain of the Linksys IP phone. |
| Primary DNS | DNS server used by Linksys IP phone in addition to DHCP supplied DNS servers if DHCP is enabled; when DHCP is disabled, this is the primary DNS server. The default is 0.0.0.0. |
| Secondary DNS | DNS server used by Linksys IP phone in addition to DHCP supplied DNS servers if DHCP is enabled; when DHCP is disabled, this is the secondary DNS server. 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. |
| DNS Query Mode | Do parallel or sequential DNS Query. With parallel DNS query mode, the Linksys IP phone sends the same request to all the DNS servers at the same time when doing a DNS lookup, the first incoming reply is accepted by the Linksys IP phone. The default is parallel. |
| Syslog Server | Specify the syslog server name and port. This feature specifies the server for logging Linksys IP phone system information and critical events. If both Debug Server and Syslog Server are specified, Syslog messages are also logged to the Debug Server. |
| Debug Server | The debug server name and port. This feature specifies the server for logging Linksys IP phone debug information. The level of detailed output depends on the debug level parameter setting. |
| Debug Level | The 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. |

Miscellaneous Settings (Not in PAP2T)

| Field | Description |
|---------------|---|
| Syslog Server | Specifies the IP address of the syslog server. |
| Debug Server | Specifies the IP address of the debug server, which logs debug information. The level of detailed output depends on the debug level parameter setting. |
| Debug Level | Determines the level of debug information that is generated. Select 0, 1, 2, or 3 from the drop-down menu. The higher the debug level, the more debug information is generated. The default is 0, which indicates that no debug information is generated. |

SIP Tab

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

- [SIP Parameters, page 5-11](#)
- [SIP Timer Values \(sec\), page 5-12](#)
- [Response Status Code Handling, page 5-14](#)
- [RTP Parameters, page 5-14](#)
- [SDP Payload Types, page 5-15](#)
- [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 Linksys IP phone does not include this header). If empty, the header is not included. |
| DTMF Relay MIME Type | MIME Type used in a SIP INFO message to signal a DTMF event. The default is application/dtmf-relay. |
| Hook Flash MIME Type | MIME Type used in a SIP INFO message to signal a hook flash event. The default is application/hook-flash. |
| 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 Linksys IP phone uses compact SIP headers in outbound SIP messages. If set to no, the Linksys IP phone uses normal SIP headers. If inbound SIP requests contain compact headers, Linksys IP phone reuses the same compact headers when generating the response regardless the settings of the <Use Compact Header> parameter. If inbound SIP requests contain normal headers, Linksys IP phone substitutes those headers with compact headers (if defined by RFC 261) if <Use Compact Header> parameter is set to yes. The default is no. |
| Escape Display Name | Lets you keep the Display Name private. Select yes if you want the Linksys IP phone to enclose the string (configured in the Display Name) in a pair of double quotes for outbound SIP messages. Any occurrences of or \ in the string is escaped with \ and \\ inside the pair of double quotes. Otherwise, select no. The default is no. |
| RFC 2543 Call Hold | The default is no. |
| Mark All AVT Packets | |
| SIP TCP Port Min | |
| 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: $0-(2^{31}-1)$. |
| 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: $0-(2^{31}-1)$. |
| 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. |
| Reg Retry Intvl | Interval to wait before the Linksys IP 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 Linksys IP phone waits for the specified length of time before retrying. If this interval is 0, the Linksys IP phone stops trying. This value should be much larger than the Reg Retry Intvl value, which should not be 0. The default is 1200. |
| 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 Linksys IP 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 Linksys IP phone terminates the call. If value is set to 0, the Linksys IP phone ignores the limit on ICMP errors. The default is 0. |

| | |
|------------------|---|
| RTCP Tx Interval | <p>Interval for sending out RTCP sender reports on an active connection. It can range from 0 to 255 seconds. During an active connection, the Linksys IP phone can be programmed to send out compound RTCP packet on the connection. Each compound RTP packet except the last one contains a SR (Sender Report) and a SDES.(Source Description). The last RTCP packet contains an additional BYE packet. Each SR except the last one contains exactly 1 RR (Receiver Report); the last SR carries no RR. The SDES contains CNAME, NAME, and TOOL identifiers. The CNAME is set to <User ID>@<Proxy>, NAME is set to <Display Name> (or Anonymous if user blocks caller ID), and TOOL is set to the Vendor/Hardware-platform-software-version (such as Linksys/Linksys IP phone-1.0.31(b)). The NTP timestamp used in the SR is a snapshot of the Linksys IP phone's local time, not the time reported by an NTP server. If the Linksys IP phone receives a RR from the peer, it attempts to compute the round trip delay and show it as the <Call Round Trip Delay> value (ms) in the Info section of Linksys IP phone web page.</p> <p>The default is 0.</p> |
| No UDP Checksum | <p>Select yes if you want the Linksys IP phone to calculate the UDP header checksum for SIP messages. Otherwise, select no.</p> <p>The default is no.</p> |
| Stats In BYE | <p>Determines whether the Linksys IP phone includes the P-RTP-Stat header or response to a BYE message. The header contains the RTP statistics of the current call. Select yes or no from the drop-down menu. The format of the P-RTP-Stat header is:</p> <p>P-RTP-State: PS=<packets sent>,OS=<octets sent>,PR=<packets received>,OR=<octets received>,PL=<packets lost>,JI=<jitter in ms>,LA=<delay in ms>,DU=<call duration in s>,EN=<encoder>,DE=<decoder>.</p> <p>The default is no.</p> |

SDP Payload Types

| Field | Description |
|-------------------------|--|
| NSE Dynamic Payload | <p>NSE dynamic payload type. The valid range is 96-127.</p> <p>The default is 100.</p> |
| AVT Dynamic Payload | <p>AVT dynamic payload type. The valid range is 96-127.</p> <p>The default is 101.</p> |
| INFOREQ Dynamic Payload | <p>INFOREQ dynamic payload type.</p> <p>There is no default.</p> |
| G726r16 Dynamic Payload | <p>G.726-16 dynamic payload type. The valid range is 96-127.</p> <p>The default is 98.</p> |
| G726r24 Dynamic Payload | <p>G.726-24 dynamic payload type. The valid range is 96-127.</p> <p>The default is 97.</p> |

SIP Tab

| | |
|-------------------------|---|
| 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. |
| 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. |
| 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. |
| G723 Codec Name | G.723 codec name used in SDP. The default is G723. |
| EncapRTP Codec Name | EncapRTP codec name used in SDP. The default is EncapRTP. |

NAT Support Parameters

| Field | Description |
|-------|-------------|
|-------|-------------|

| | |
|-----------------------|---|
| Handle VIA received | <p>If you select yes, the Linksys IP phone processes the received parameter in the VIA header (this is inserted by the server in a response to anyone of its requests). If you select no, the parameter is ignored. Select yes or no from the drop-down menu.</p> <p>The default is no.</p> |
| Handle VIA rport | <p>If you select yes, the Linksys IP phone processes the rport parameter in the VIA header (this is inserted by the server in a response to anyone of its requests). If you select no, the parameter is ignored. Select yes or no from the drop-down menu.</p> <p>The default is no.</p> |
| Insert VIA received | <p>Inserts the received parameter into the VIA header of SIP responses if the received-from IP and VIA sent-by IP values differ. Select yes or no from the drop-down menu.</p> <p>The default is no.</p> |
| Insert VIA rport | <p>Inserts the rport parameter into the VIA header of SIP responses if the received-from IP and VIA sent-by IP values differ. Select yes or no from the drop-down menu.</p> <p>The default is no.</p> |
| Substitute VIA Addr | <p>Lets you use NAT-mapped IP:port values in the VIA header. Select yes or no from the drop-down menu.</p> <p>The default is no.</p> |
| Send Resp To Src Port | <p>Sends responses to the request source port instead of the VIA sent-by port. Select yes or no from the drop-down menu.</p> <p>The default is no.</p> |
| STUN Enable | <p>Enables the use of STUN to discover NAT mapping. Select yes or no from the drop-down menu.</p> <p>The default is no.</p> |
| STUN Test Enable | <p>If the STUN Enable feature is enabled and a valid STUN server is available, the Linksys IP phone can perform a NAT-type discovery operation when it powers on. It contacts the configured STUN server, and the result of the discovery is reported in a Warning header in all subsequent REGISTER requests. If the Linksys IP phone detects symmetric NAT or a symmetric firewall, NAT mapping is disabled.</p> <p>The default is no.</p> |
| STUN Server | <p>IP address or fully-qualified domain name of the STUN server to contact for NAT mapping discovery.</p> |
| EXT IP | <p>External IP address to substitute for the actual IP address of the Linksys IP phone in all outgoing SIP messages. If 0.0.0.0 is specified, no IP address substitution is performed.</p> <p>If this parameter is specified, the Linksys IP phone assumes this IP address when generating SIP messages and SDP (if NAT Mapping is enabled for that line). However, the results of STUN and VIA received parameter processing, if available, supersede this statically configured value.</p> <p>The default is 0.0.0.0.</p> |

SIP Tab

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| 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. |

Regional Tab

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

- [Call Progress Tones, page 5-19](#)
- [Distinctive Ring Patterns, page 5-21](#)
- [Distinctive Call Waiting Tone Patterns, page 5-21](#)
- [Distinctive Ring/CWT Pattern Names, page 5-22](#)
- [Ring and Call Waiting Tone Spec, page 5-23](#)
- [Control Timer Values \(sec\), page 5-23](#)
- [Vertical Service Activation Codes, page 5-24](#)
- [Vertical Service Announcement Codes, page 5-28](#)
- [Outbound Call Codec Selection Codes, page 5-28](#)
- [Miscellaneous, page 5-30](#)

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). |

Regional Tab

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| 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). |
| 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 SIP 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 SIP 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 SIP 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 SIP 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). |
| VoIP PIN Tone | Specification of the tone played to prompt a VoIP caller for a PIN number (if PIN authentication is selected and the caller requires authentication to use the PSTN gateway). The default is 600@-10;*(0/1/1,.1/.1/1,.1/.1/1,.1/.5/1) |
| PSTN PIN Tone | Specification of the tone played to prompt a PSTN caller for a PIN number (if PIN authentication is selected and the caller requires authentication to use the VoIP gateway). The default is 600@-10;*(0/.7/1,.2/.1/1,.2/.1/1,.2/.5/1). |
| 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). |

Distinctive Call Waiting Tone Patterns

| Field | Description |
|--------------|---|
| CWT1 Cadence | Cadence script for distinctive CWT 1. The default is 30(.3/9.7). |
| CWT2 Cadence | Cadence script for distinctive CWT 2. The default is 30(.1/.1, .1/9.7). |
| CWT3 Cadence | Cadence script for distinctive CWT 3. The default is 30(.1/.1, .1/.1, .1/9.5). |
| CWT4 Cadence | Cadence script for distinctive CWT 4. The default is 30(.1/.1, .3/.1, .1/9.3). |
| CWT5 Cadence | Cadence script for distinctive CWT 5. The default is 30(.3/.1,.1/.1,.3/9.1). |

Regional Tab

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|--------------|---|
| CWT6 Cadence | Cadence script for distinctive CWT 6. The default is 30(.1/.1, .3/.1, .1/9.3). |
| CWT7 Cadence | Cadence script for distinctive CWT 7. The default is 30(.1/.1, .3/.1, .1/9.3). |
| CWT8 Cadence | Cadence script for distinctive CWT 8. The default is 2.3(..3/2). |

Distinctive Ring/CWT Pattern Names

| Field | Description |
|------------|---|
| Ring1 Name | Name in an INVITE's Alert-Info Header to pick distinctive ring/CWT 1 for the inbound call. The default is Bellcore-r1. |
| Ring2 Name | Name in an INVITE's Alert-Info Header to pick distinctive ring/CWT 2 for the inbound call. The default is Bellcore-r2. |
| Ring3 Name | Name in an INVITE's Alert-Info Header to pick distinctive ring/CWT 3 for the inbound call. The default is Bellcore-r3. |
| Ring4 Name | Name in an INVITE's Alert-Info Header to pick distinctive ring/CWT 4 for the inbound call. The default is Bellcore-r4. |
| Ring5 Name | Name in an INVITE's Alert-Info Header to pick distinctive ring/CWT 5 for the inbound call. The default is Bellcore-r5. |
| Ring6 Name | Name in an INVITE's Alert-Info Header to pick distinctive ring/CWT 6 for the inbound call. The default is Bellcore-r6. |
| Ring7 Name | Name in an INVITE's Alert-Info Header to pick distinctive ring/CWT 7 for the inbound call. The default is Bellcore-r7. |
| Ring8 Name | Name in an INVITE's Alert-Info Header to pick distinctive ring/CWT 8 for the inbound call. The default is Bellcore-r8. |

Ring and Call Waiting Tone Spec

| Field | Description |
|----------------|--|
| Ring Waveform | Waveform for the ringing signal. The default is Sinusoid. |
| Ring Frequency | Frequency of the ringing signal. Valid values are 10–100 (Hz). The default is 25. |
| Ring Voltage | Ringing voltage. 60–90 (V). The default is 70. |
| CWT Frequency | Frequency script of the call waiting tone. All distinctive CWTs are based on this tone. The default is 440@-10. |

Control Timer Values (sec)

| Field | Description |
|-----------------------|---|
| Hook Flash Timer Min | Minimum on-hook time before off-hook qualifies as hook-flash. Less than this the on-hook event is ignored. Range: 0.1–0.4 seconds. The default is 0.1. |
| Hook Flash Timer Max | Maximum on-hook time before off-hook qualifies as hook-flash. More than this the on-hook event is treated as on-hook (no hook-flash event). Range: 0.4–1.6 seconds. The default is 0.9. |
| Callee On Hook Delay | Phone must be on-hook for at this time in sec before the Linksys IP phone will tear down the current inbound call. It does not apply to outbound calls. Range: 0–255 seconds. The default is 0. |
| 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 Linksys IP phone still considers the call as failed and keeps on retrying. The default is 0.5. |

Regional Tab

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|------------------------|---|
| 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. |
| CPC Delay | Delay in seconds after caller hangs up when the Linksys IP phone starts removing the tip-and-ring voltage to the attached equipment of the called party. Range: 0–255 seconds. Linksys IP phone has had polarity reversal feature since release 1.0 which can be applied to both the caller and the callee end. This feature is generally used for answer supervision on the caller side to signal to the attached equipment when the call has been connected (remote end has answered) or disconnected (remote end has hung up). This feature should be disabled for the called party (in other words, by using the same polarity for connected and idle state) and the CPC feature should be used instead. Without CPC enabled, reorder tone will is played after a configurable delay. If CPC is enabled, dial tone will be played when tip-to-ring voltage is restored Resolution is 1 second. The default is 2. |
| CPC Duration | Duration in seconds for which the tip-to-ring voltage is removed after the caller hangs up. After that, tip-to-ring voltage is restored and dial tone applies if the attached equipment is still off-hook. CPC is disabled if this value is set to 0. Range: 0 to 1.000 second. Resolution is 0.001 second. The default is 0 (CPC disabled). |

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. |

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|-------------------------|---|
| Call Back Act Code | Starts a callback when the last outbound call is not busy. The default is *66. |
| Call Back Deact Code | Cancels 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 | Cancels 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 | Cancels 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 | Cancels 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 | Cancels 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 | Cancels blocking of the last inbound call. The default is *80. |
| 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. |

Regional Tab

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|-------------------------------|---|
| 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. |

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|----------------------------|---|
| 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. |
| Conference Act Code | If this code is specified, the user must enter it before dialing the third party for a conference call. Enter the code for a conference call. |
| Attn-Xfer Act Code | If the code is specified, the user must enter it before dialing the third party for a call transfer. Enter the code for a call transfer. |
| Modem Line Toggle Code | Toggles the line to a modem. The default is *99. Modem pass-through mode can be triggered only by pre-dialing this code. |
| FAX Line Toggle Code | Toggles the line to a fax machine. (Not in PAP2T) The default is #99. |
| Referral Services Codes | <p>These codes tell the Linksys IP phone what to do when the user places the current call on hold and is listening to the second dial tone.</p> <p>One or more *code can be configured into this parameter, such as *98, or *97 *98 *123, etc. Max total length is 79 chars. This parameter applies when the user places the current call on hold (by Hook Flash) and is listening to second dial tone. Each *code (and the following valid target number according to current dial plan) entered on the second dial-tone triggers the Linksys IP phone to perform a blind transfer to a target number that is prepended by the service *code.</p> <p>For example, after the user dials *98, the Linksys IP phone plays a special dial tone called the Prompt Tone while waiting for the user to enter a target number (which is checked according to dial plan as in normal dialing). When a complete number is entered, the Linksys IP phone sends a blind REFER to the holding party with the Refer-To target equals to *98<target_number>. This feature allows the Linksys IP phone to hand off a call to an application server to perform further processing, such as call park.</p> <p>The *codes should not conflict with any of the other vertical service codes internally processed by the Linksys IP phone. You can empty the corresponding *code that you do not want to Linksys IP phone to process.</p> |

Regional Tab

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

Vertical Service Announcement Codes

| Field | Description |
|------------------------------|--|
| Service Annc Base Number | Base number for service announcements. |
| Service Annc Extension Codes | Extension codes for service announcements. |

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 | Makes this codec the preferred codec for the associated call. The default is *017110. |
| 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. |
| 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 = <start-time>; end=<end-time>; save = <save-time>.</p> <ul style="list-style-type: none"> The <start-time> and <end-time> values specify the start and end dates and times of daylight saving time. Each value is in this format: <month> /<day> / <weekday>[/HH:[mm[:ss]]] The <save-time> value is the number of hours, minutes, and/or seconds to add to the current time during daylight saving time. The <save-time> value can be preceded by a negative (-) sign if subtraction is desired instead of addition. The <save-time> value is in this format: [/[+/-]HH:[mm[:ss]]] The <month> value equals any value in the range 1-12 (January-December). The <day> value equals [+/-] any value in the range 1-31. If <day> is 1, it means the <weekday> on or before the end of the month (in other words the last occurrence of < weekday> in that month). The <weekday> value equals any value in the range 1-7 (Monday-Sunday). It can also equal 0. If the <weekday> value is 0, this means that the date to start or end daylight saving is exactly the date given. In that case, the <day> value must not be negative. If the <weekday> value is not 0 and the <day> value is positive, then daylight saving starts or ends on the <weekday> value on or after the date given. If the <weekday> value is not 0 and the <day> value is negative, then daylight saving starts or ends on the <weekday> value on or before the date given. The abbreviation HH stands for hours (0-23). The abbreviation mm stands for minutes (0-59). The abbreviation ss stands for seconds (0-59). <p>The default Daylight Saving Time Rule is start=4/1/7;end=10/-1/7;save=1.</p> |

| | |
|------------------------|---|
| FXS Port Input Gain | Input gain in dB, up to three decimal places. The range is 6.000 to -12.000. The default is -3. |
| FXS Port Output Gain | 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 <FXS Port Output Gain>. The default is -3. |
| DTMF Playback Level | Local DTMF playback level in dBm, up to one decimal place. The default is -16.0. |
| DTMF Playback Length | Local DTMF playback duration in milliseconds. The default is .1. |
| Detect ABCD | To enable local detection of DTMF ABCD, select yes. Otherwise, select no. The default is yes. Setting has no effect if DTMF Tx Method is INFO; ABCD is always sent OOB regardless in this setting. |
| Playback ABCD | To enable local playback of OOB DTMF ABCD, select yes. Otherwise, select no. The default is yes. |
| Caller ID Method | The following choices are available: <ul style="list-style-type: none"> • 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). • DTMF (Finland, Sweden)—CID only. DTMF sent after polarity reversal (and no DTAS) and before first ring. • DTMF (Denmark)—CID only. DTMF sent after polarity reversal (and no DTAS) and before first ring. • ETSI DTMF—CID only. DTMF sent after DTAS (and no polarity reversal) and before first ring. • ETSI DTMF With PR—CID only. DTMF sent after polarity reversal and DTAS and before first ring. • ETSI DTMF After Ring—CID only. DTMF sent after first ring (no polarity reversal or DTAS). • 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. • 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. The default is Bellcore(N.Amer, China). |
| Caller ID FSK Standard | The Linksys IP 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. The default is bell 202. |
| FXS Port Power Limit | The choices are from 1 to 8. (PAP2T only) The default is 3. |

Regional Tab

| | |
|---------------------------|---|
| Feature Invocation Method | Select the method you want to use, Default or Sweden default. (Not in PAP2T) The default is Default. |
| More Echo Suppression | Enable or disable more echo suppression. The default is no. |
| GR909 Test | To use this test, select yes . Otherwise, keep the default, no . |

Line Tab

This section describes the fields for the following headings on the Line tabs:

- [Line Enable](#), page 5-33
- [Streaming Audio Server \(SAS\)](#), page 5-33
- [NAT Settings](#), page 5-34
- [Network Settings](#), page 5-35
- [SIP Settings](#), page 5-35
- [Call Feature Settings](#), page 5-38
- [Proxy and Registration](#), page 5-38
- [Subscriber Information](#), page 5-40
- [Supplementary Service Subscription](#), page 5-40
- [Audio Configuration](#), page 5-42
- [VoIP Fallback to PSTN \(SPA3102/AG310\)](#), page 5-46
- [Gateway Accounts \(SPA3102/AG310\)](#), page 5-45
- [Dial Plan](#), page 5-46
- [FXS Port Polarity Configuration](#), page 5-47

In a configuration profile, the Line parameters must be appended with the appropriate numeral (for example, [1] or [2]) to identify the line to which the setting applies. The number of lines varies with the model of the ATA. For example, the SPA2102 provides two Line tabs (Line 1 and Line 2), while the SPA8000 provides eight tabs (Line1 through Line 8).

The SPA2102 provides one User tab for each Line tab (User 1 and User 2), where many of the line-specific configuration parameters are contained. The SPA8000 does not provide User tabs, but consolidates all the line-specific parameters on the Line tab.

Line Enable

| Field | Description |
|-------------|---|
| Line Enable | To enable this line for service, select yes. Otherwise, select no. The default is yes. |

Streaming Audio Server (SAS)

| Field | Description |
|-------|-------------|
|-------|-------------|

Line Tab

| | |
|-----------------------|--|
| SAS Enable | To enable the use of the line as a streaming audio source, select yes. Otherwise, select no. If enabled, the line cannot be used for outgoing calls. Instead, it auto-answers incoming calls and streams audio RTP packets to the caller. The default is no. |
| SAS DLG Refresh Intvl | If this is not zero, it is the interval at which the streaming audio server sends out session refresh (SIP re-INVITE) messages to determine whether the connection to the caller is still active. If the caller does not respond to the refresh message, the Linksys IP phone ends this call with a SIP BYE message. The range is 0 to 255 seconds (0 means that the session refresh is disabled). The default is 30. |
| SAS Inbound RTP Sink | This setting works around devices that do not play inbound RTP if the streaming audio server line declares itself as a send-only device and tells the client not to stream out audio. Enter a Fully Qualified Domain Name (FQDN) or IP address of an RTP sink; this is used by the Linksys IP phone's streaming audio server line in the SDP of its 200 response to an inbound INVITE message from a client. The purpose of this parameter is to work around devices that do not play inbound RTP if the SAS line declares itself as a send-only device and tells the client not to stream out audio. This parameter is a FQDN or IP address of a RTP sink to be used by the SPA SAS line in the SDP of its 200 response to inbound INVITE from a client. It will appear in the c = line and the port number and, if specified, in the m = line of the SDP. If this value is not specified or equal to 0, then c = 0.0.0.0 and a=sendonly will be used in the SDP to tell the SAS client to not to send any RTP to this SAS line. If a non-zero value is specified, then a=sendrecv and the SAS client will stream audio to the given address. Special case: If the value is \$IP, then the SAS line's own IP address is used in the c = line and a=sendrecv. In that case the SAS client will stream RTP packets to the SAS line. The default value is empty. |

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 | <p>Destination that should receive NAT keep alive messages. If the value is \$PROXY, the messages are sent to the current or outbound proxy.</p> <p>The default is \$PROXY.</p> |
|---------------------|---|

Network Settings

| Field | Description |
|--------------------------|---|
| SIP ToS/DiffServ Value | <p>TOS/DiffServ field value in UDP IP packets carrying a SIP message.</p> <p>The default is 0x68.</p> |
| SIP CoS Value [0-7] | <p>CoS value for SIP messages. (Not in PAP2T)</p> <p>The default is 3.</p> |
| RTP ToS/DiffServ Value | <p>ToS/DiffServ field value in UDP IP packets carrying RTP data.</p> <p>The default is 0xb8.</p> |
| RTP CoS Value [0-7] | <p>CoS value for RTP data. (Not in PAP2T)</p> <p>The default is 6.</p> |
| Auto PSTN Failback | <p>If enabled, the SPA will automatically route all calls to the PSTN gateway when the Line 1 proxy is down (registration failure or network link down).</p> <p>The default is yes.</p> |
| Network Jitter Level | <p>Determines how jitter buffer size is adjusted by the Linksys IP phone. Jitter buffer size is adjusted dynamically. The minimum jitter buffer size is 30 milliseconds or (10 milliseconds + current RTP frame size), whichever is larger, for all jitter level settings. However, the starting jitter buffer size value is larger for higher jitter levels. This setting controls the rate at which the jitter buffer size is adjusted to reach the minimum. Select the appropriate setting: low, medium, high, very high, or extremely high.</p> <p>The default is high.</p> |
| Jitter Buffer Adjustment | <p>Controls how the jitter buffer should be adjusted. Select the appropriate setting: up and down, up only, down only, or disable.</p> <p>The default is up and down.</p> |

SIP Settings

| Field | Description |
|----------|--|
| SIP Port | <p>Port number of the SIP message listening and transmission port.</p> <p>The default is 5060.</p> |

Line Tab

| | |
|---------------------|---|
| 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 Linksys IP phone authenticates the sender when it receives the NOTIFY resync reboot (RFC 2617) message. To use this feature, select yes. Otherwise, select no. 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 | (Not in PAP2T) The Global Unique ID is generated for each line for each device. When it is enabled, the Linksys IP 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"> • none—No logging. • 1-line—Logs the start-line only for all messages. • 1-line excl. OPT—Logs the start-line only for all messages except OPTIONS requests/responses. • 1-line excl. NTFY—Logs the start-line only for all messages except NOTIFY requests/responses. • 1-line excl. REG—Logs the start-line only for all messages except REGISTER requests/responses. • 1-line excl. OPT NTFY REG—Logs the start-line only for all messages except OPTIONS, NOTIFY, and REGISTER requests/responses. • full—Logs all SIP messages in full text. • full excl. OPT—Logs all SIP messages in full text except OPTIONS requests/responses. • full excl. NTFY—Logs all SIP messages in full text except NOTIFY requests/responses. • full excl. REG—Logs all SIP messages in full text except REGISTER requests/responses. • full excl. OPT NTFY REG—Logs all SIP messages in full text except for OPTIONS, NOTIFY, and REGISTER requests/responses. <p>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 <Proxy> (or <Outbound Proxy> if <Use Outbound Proxy> is yes).</p> <p>The default is no.</p> |
| Referor Bye Delay | <p>Controls when the Linksys IP phone sends BYE to terminate stale call legs upon completion of call transfers. Multiple delay settings (Referor, Refer Target, Referee, and Refer-To Target) are configured on this screen. For the Referor Bye Delay, enter the appropriate period of time in seconds.</p> <p>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 | <p>For the Referee Bye Delay, enter the appropriate period of time in seconds.</p> <p>The default is 0.</p> |

Line Tab

| | |
|---------------------------|---|
| 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. |
| Reply 182 On Call Waiting | To use this feature, select yes . Otherwise, keep the default, no . |

Call Feature Settings

| Field | Description |
|-------------------------|---|
| Blind Attn-Xfer Enable | Enables the Linksys IP phone to perform an attended transfer operation by ending the current call leg and performing a blind transfer of the other call leg. If this feature is disabled, the Linksys IP phone performs an attended transfer operation by referring the other call leg to the current call leg while maintaining both call legs. To use this feature, select yes. Otherwise, select no. 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. |
| Xfer When Hangup Conf | Makes the Linksys IP phone perform a transfer when a conference call has ended. Select yes or no from the drop-down menu. The default is yes. |
| Conference Bridge URL | This feature supports external conference bridging for n-way conference calls ($n > 2$), instead of mixing audio locally. To use this feature, set this parameter to that of the server's name, for example, conf@myserver.com:12345 or conf (which uses the Proxy value as the domain). |
| Conference Bridge Ports | Select the maximum number of conference call participants. The range is 3 to 10. The default is 3. |

Proxy and Registration

| Field | Description |
|-------|---|
| Proxy | SIP proxy server for all outbound requests. |

| Field | Description |
|-------------------------|---|
| 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. |
| 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 st proxy's port on the list. The default is Normal. |

Line Tab

| Field | Description |
|---------------------------|--|
| Voice Mail Server | Enter the URL or IP address of the server. |
| Mailbox Subscribe Expires | <<help here>> |

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. |
| Directory Number | Enter the number for this line. |
| Call Capacity | Maximum number of calls allowed on this line interface. Choices: {unlimited,1,2,3,...25}. Default is 16. Note that the Linksys IP phone does not distinguish between incoming and outgoing calls when talking about call capacity. Note: unlimited = 16 |
| 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. |

Supplementary Service Subscription

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

| Field | Description |
|-------------------|---|
| Call Waiting Serv | Enable Call Waiting Service. The default is yes. |
| Block CID Serv | Enable Block Caller ID Service. The default is yes. |
| Block ANC Serv | Enable Block Anonymous Calls Service The default is yes. |
| Dist Ring Serv | Enable Distinctive Ringing Service The default is yes. |
| Cfwd All Serv | Enable Call Forward All Service The default is yes. |
| Cfwd Busy Serv | Enable Call Forward Busy Service The default is yes. |
| Cfwd No Ans Serv | Enable Call Forward No Answer Service The default is yes. |
| Cfwd Sel Serv | Enable Call Forward Selective Service The default is yes. |
| Cfwd Last Serv | Enable Forward Last Call Service The default is yes. |
| Block Last Serv | Enable Block Last Call Service The default is yes. |
| Accept Last Serv | Enable Accept Last Call Service The default is yes. The default is yes. |
| DND Serv | Enable Do Not Disturb Service The default is yes. |
| CID_Serv | Enable Caller ID Service The default is yes. |
| CWCID Serv | Enable Call Waiting Caller ID Service The default is yes. |
| Call Return Serv | Enable Call Return Service The default is yes. |
| Call Redial Serv | Enable Call Redial Service (Not in PAP2T) |

Line Tab

| Field | Description |
|---------------------------|--|
| Call Back Serv | Enable Call Back Service |
| Three Way Call Serv | Enable Three Way Calling Service. Three Way Calling is required for Three Way Conference and Attended Transfer. The default is yes. |
| Three Way Conf Serv | Enable Three Way Conference Service. Three Way Conference is required for Attended Transfer. The default is yes. |
| Attn Transfer Serv | Enable Attended Call Transfer Service. Three Way Conference is required for Attended Transfer. The default is yes. |
| Unattn Transfer Serv | Enable Unattended (Blind) Call Transfer Service. The default is yes. |
| MWI Serv | Enable MWI Service. MWI is available only if a Voice Mail Service is set-up in the deployment. The default is yes. |
| VMWI Serv | Enable VMWI Service (FSK). The default is yes. |
| Speed Dial Serv | Enable Speed Dial Service. The default is yes. |
| Secure Call Serv | Enable Secure Call Service. The default is yes. |
| Referral Serv | Enable Referral Service. See <Referral Services Codes> for more details. The default is yes. |
| Feature Dial Serv | Enable Feature Dial Service. See <Feature Dial Services Codes> for more details. The default is yes. |
| Service Announcement Serv | Enable Service Announcement Service. The default is yes. (Not in PAP2T) |

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. |
| FAX CNG Detect Enable | To enable detection of the fax Calling Tone (CNG), select yes. Otherwise, select no. The default is yes. |

Line Tab

| | |
|----------------------|---|
| G726-40 Enable | To enable the use of the G.726 codec at 40 kbps, select yes. Otherwise, select no. The default is yes. |
| FAX Passthru Codec | Select the codec for fax passthrough, G711u or G711a. The default is G711u. |
| DTMF Process INFO | (Not in PAP2T) To use the DTMF process info feature, select yes. Otherwise, select no. The default is yes. |
| FAX Codec Symmetric | To force the Linksys IP phone to use a symmetric codec during fax passthrough, 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. |
| FAX Passthru Method | Select the fax passthrough method: None, NSE, or ReINVITE. The default is NSE. |
| 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. |
| FAX Process NSE | To use the fax process NSE feature, select yes. Otherwise, select no. The default is yes. |
| Hook Flash Tx Method | Select the method for signaling hook flash events: None , AVT , or INFO . None does not signal hook flash events. AVT uses RFC2833 AVT (event = 16). INFO uses SIP INFO with the single line signal=hf in the message body. The MIME type for this message body is taken from the Hook Flash MIME Type setting. The default is None. |
| FAX Disable ECAN | If enabled, this feature automatically disables the echo canceller when a fax tone is detected. To use this feature, select yes. Otherwise, select no. The default is no. |
| 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. |
| FAX Enable T38 | To enable the use of the ITU-T T.38 standard for faxing, select yes. Otherwise, select no. T.38 is supported by the SPA2102 and the SPA8000. The SPA2102 supports a single T.38 connection. The SPA8000 supports one T.38 connection for each of its four modules (Line 1-2, 3-4, 5-6, and 7-8). The SPA8000 supports a maximum of four connections, but it does not support two fax devices connected to the same module. The default is yes. |

| | |
|----------------------|---|
| FAX T38 Redundancy | Select the appropriate number. The default is 1 . |
| Fax Tone Detect Mode | If you want the Gateway to detect the fax tone whether the Gateway is a caller or callee, then select caller or callee . If you want the Gateway to detect the fax tone only if the Gateway is the caller, then select caller only . If you want the Gateway to detect the fax tone only if the Gateway is the callee, then select callee only . |
| FAX Tone Detect Mode | This parameter has three possible values: <ul style="list-style-type: none"> • caller or callee - SPA will detect FAX tone whether it is callee or caller • caller only - SPA will detect FAX tone only if it is the caller • callee only - SPA will detect FAX tone only if it is the callee The default is caller or callee. |
| Symmetric RTP | (SPA3102 only) 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. The default is yes. |



Note

References in this section to the SPA3102 also apply to the AG310.

Gateway Accounts (SPA3102/AG310)

| Field | Description |
|------------------------------|---|
| Gateway1/2/3/4 | The first of 4 gateways that can be specified to be used in the <Dial Plan> to facilitate call routing specification (that overrides the given proxy information). This gateway is represented by gw1 in the <Dial Plan>. For example, the rule 1408xxxxxxx<:@gw1> can be added to the dial plan such that when the user dials 1408+7digits, the call will be routed to Gateway 1. Without the <:@gw1> syntax, all calls are routed to the given proxy by default (except IP dialing). The default is blank. |
| GW1/2/3/4 NAT Mapping Enable | If enabled, the SPA will use NAT mapping when contacting Gateway 1. The default is no. |
| GW1/2/3/4 Auth ID | This is the authentication user-id to be used by the SPA to authenticate itself to Gateway 1. The default is blank. |
| GW1/2/3/4 Password | This is the password to be used by the SPA to authenticate itself to Gateway 1. The default is blank. |

VoIP Fallback to PSTN (SPA3102/AG310)

| Field | Description |
|--------------------|--|
| Auto PSTN Fallback | If enabled, the SPA will automatically route all calls to the PSTN gateway when the Line 1 proxy is down (registration failure or network link down). 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 1-35 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 default(*xx [3469]11 0 00 [2-9]xxxxxx 1xxx[2-9]xxxxxxS0 xxxxxxxxxxxxx.)</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"> • uid – the authentication user-id • pwd – the authentication password • nat – if this parameter is present, use NAT mapping <p>Each parameter is separated by a semi-colon (;).</p> <p>Furthermore, it recognizes gw0, gw1, ..., gw4 as the locally configured gateways, where gw0 represents the local PSTN gateway in the same SPA3102 unit.</p> <p>Example 1:</p> <pre>*1xxxxxxxxxxx<:@fwdnat.pulver.com:5082;uid=jsmith;pwd=xyz</pre> <p>Example 2:</p> <pre>*1xxxxxxxxxxx<:@fwd.pulver.com;nat;uid=jsmith;pwd=xyz</pre> <p>Example 3:</p> <pre>[39]11<:@gw0></pre> |
| PSTN Fallback Dial Plan | <<help here>> |
| Enable IP Dialing | <p>Enable or disable IP dialing.</p> <p>The default is no.</p> |
| Emergency Number | <p>Comma separated list of emergency number patterns. If outbound call matches one of the pattern, SPA will disable hook flash event handling. The condition is restored to normal after the phone is on-hook. Blank signifies no emergency number. Maximum number length is 63 characters.</p> <p>The default is blank.</p> |



Note

References in this section to the SPA3102 also apply to the AG310.

FXS Port Polarity Configuration

| Field | Description |
|----------------------|---|
| Idle Polarity | <p>Polarity before a call is connected: Forward or Reverse.</p> <p>The default is Forward.</p> |
| Caller Conn Polarity | <p>Polarity after an outbound call is connected: Forward or Reverse.</p> <p>The default is Forward.</p> |
| Callee Conn Polarity | <p>Polarity after an inbound call is connected: Forward or Reverse.</p> <p>The default is Forward.</p> |

PSTN Line Tab

This section describes the fields for the following headings on the PSTN Line tab on the SPA3102 and AG310:

- [Line Enable](#), page 5-33
- [NAT Settings](#), page 5-49
- [Network Settings](#), page 5-50
- [SIP Settings](#), page 5-50
- [Proxy and Registration \(SPA3102/AG310\)](#), page 5-53
- [Subscriber Information \(SPA3102/AG310\)](#), page 5-54
- [Audio Configuration \(SPA3102/AG310\)](#), page 5-55
- [Dial Plans](#), page 5-57
- [VoIP-To-PSTN Gateway Setup](#), page 5-57
- [VoIP Users and Passwords \(HTTP Authentication\)](#), page 5-59
- [FXO \(PSTN\) Timer Values \(sec\)](#), page 5-60
- [PSTN Disconnect Detection](#), page 5-62
- [International Control \(Settings\)](#), page 5-63

Line Enable

| Field | Description |
|-------------------|--|
| Line Enable | To enable this line for service, select yes. Otherwise, select no. The default is yes. |
| PSTN Contact List | Select the appropriate list: None , Phone 1+2 , Phone 1 , or Phone 2 . The default is Phone 1+2 . |

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. |

PSTN Line Tab

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

SIP Settings

| Field | Description |
|----------|---|
| 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 Linksys IP phone authenticates the sender when it receives the NOTIFY resync reboot (RFC 2617) message. To use this feature, select yes. Otherwise, select no. 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 | (Not in PAP2T) The Global Unique ID is generated for each line for each device. When it is enabled, the Linksys IP 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"> • none—No logging. • 1-line—Logs the start-line only for all messages. • 1-line excl. OPT—Logs the start-line only for all messages except OPTIONS requests/responses. • 1-line excl. NTFY—Logs the start-line only for all messages except NOTIFY requests/responses. • 1-line excl. REG—Logs the start-line only for all messages except REGISTER requests/responses. • 1-line excl. OPT NTFY REG—Logs the start-line only for all messages except OPTIONS, NOTIFY, and REGISTER requests/responses. • full—Logs all SIP messages in full text. • full excl. OPT—Logs all SIP messages in full text except OPTIONS requests/responses. • full excl. NTFY—Logs all SIP messages in full text except NOTIFY requests/responses. • full excl. REG—Logs all SIP messages in full text except REGISTER requests/responses. • full excl. OPT NTFY REG—Logs all SIP messages in full text except for OPTIONS, NOTIFY, and REGISTER requests/responses. <p>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 <Proxy> (or <Outbound Proxy> if <Use Outbound Proxy> is yes).</p> <p>The default is no.</p> |
| Referor Bye Delay | <p>Controls when the Linksys IP phone sends BYE to terminate stale call legs upon completion of call transfers. Multiple delay settings (Referor, Refer Target, Referee, and Refer-To Target) are configured on this screen. For the Referor Bye Delay, enter the appropriate period of time in seconds.</p> <p>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 | <p>For the Referee Bye Delay, enter the appropriate period of time in seconds.</p> <p>The default is 0.</p> |

| | |
|-------------------------|---|
| 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. |

Proxy and Registration (SPA3102/AG310)

| 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. |
| 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. |

PSTN Line Tab

| Field | Description |
|-------------------------|--|
| Proxy Fallback Intvl | <p>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).</p> <p>The default is 3600</p> |
| Proxy Redundancy Method | <p>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.</p> <p>if Based on SRV port is configured the PAP2T does normal first, and also inspect the port number based on 1st proxy's port on the list.</p> <p>The default is Normal.</p> |

Subscriber Information (SPA3102/AG310)

| Field | Description |
|------------------|--|
| Display Name | Display name for caller ID. |
| User ID | Extension number for this line. |
| Password | Password for this line. |
| Use Auth ID | <p>To use the authentication ID and password for SIP authentication, select yes. Otherwise, select no to use the user ID and password.</p> <p>The default is no.</p> |
| Auth ID | Authentication ID for SIP authentication. |
| Call Capacity | <p>Maximum number of calls allowed on this line interface. Choices: {unlimited,1,2,3,...25}. Default is 16. Note that the Linksys IP phone does not distinguish between incoming and outgoing calls when talking about call capacity. Note: unlimited = 16</p> |
| Mini Certificate | <p>Base64 encoded of Mini-Certificate concatenated with the 1024-bit public key of the CA signing the MC of all subscribers in the group.</p> <p>The default is empty.</p> |
| SRTP Private Key | <p>Base64 encoded of the 512-bit private key per subscriber for establishment of a secure call.</p> <p>The default is empty.</p> |

Audio Configuration (SPA3102/AG310)

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 G.711a and G.711u. 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 G729a 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 G723a 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 G726 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 G726 codec at 24 kbps, select yes. Otherwise, select no. The default is yes. |

PSTN Line Tab

| | |
|-----------------------|--|
| 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 G726 codec at 32 kbps, select yes. Otherwise, select no. The default is yes. |
| FAX CNG Detect Enable | To enable detection of the fax Calling Tone (CNG), select yes. Otherwise, select no. The default is yes. |
| G726-40 Enable | To enable the use of the G726 codec at 40 kbps, select yes. Otherwise, select no. The default is yes. |
| FAX Passthru Codec | Select the codec for fax passthrough, G711u or G711a. The default is G711u. |
| DTMF Process INFO | (Not in PAP2T) To use the DTMF process info feature, select yes. Otherwise, select no. The default is yes. |
| FAX Codec Symmetric | To force the Linksys IP phone to use a symmetric codec during fax passthrough, 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. |
| FAX Passthru Method | Select the fax passthrough method: None, NSE, or ReINVITE. The default is NSE. |
| 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. |
| FAX Process NSE | To use the fax process NSE feature, select yes. Otherwise, select no. The default is yes. |
| Hook Flash Tx Method | Select the method for signaling hook flash events: None , AVT , or INFO . None does not signal hook flash events. AVT uses RFC2833 AVT (event = 16). INFO uses SIP INFO with the single line signal=hf in the message body. The MIME type for this message body is taken from the Hook Flash MIME Type setting. The default is None. |
| FAX Disable ECAN | If enabled, this feature automatically disables the echo canceller when a fax tone is detected. To use this feature, select yes. Otherwise, select no. The default is no. |

| | |
|----------------------|---|
| 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. |
| FAX Enable T38 | To enable the use of the ITU-T T.38 standard for faxing, select yes. Otherwise, select no. The default is yes. |
| FAX Tone Detect Mode | This parameter has three possible values: <ul style="list-style-type: none"> • caller or callee - SPA will detect FAX tone whether it is callee or caller • caller only - SPA will detect FAX tone only if it is the caller • callee only - SPA will detect FAX tone only if it is the callee The default is caller or callee. |
| Symmetric RTP | (SPA3102 only) 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. The default is yes. |



Note

References in this section to the SPA3102 also apply to the AG310.

Dial Plans

| Field | Description |
|---------------------------|--|
| Dial Plan 1/2/3/4/5/6/7/8 | Dial plan script for this line. The default is (xx.) Dial plans in the dial plan pool to be associated with a VoIP Caller or a PSTN Caller. Each dial plan in the pool is referenced by a index 1 to 8 corresponding to Dial Plan 1 to 8. The dial plan syntax is the same as that used for Line 1. |

VoIP-To-PSTN Gateway Setup

| Field | Description |
|-----------------------------------|---|
| VoIP-To-PSTN Gateway Enable | Enable or disable VoIP-To-PSTN Gateway functionality. The default is yes. |
| VoIP Caller Authentication Method | Method to be used to authenticate a VoIP Caller to access the PSTN gateway. Choose from {none, PIN, HTTP Digest}. The default is none. |

PSTN Line Tab

| Field | Description |
|------------------------|---|
| VoIP PIN Max Retry | Number of trials to allow VoIP caller to enter a PIN number (used only if authentication method is set to PIN). The default is 3. |
| One Stage Dialing | Enable one-stage dialing (applicable if authentication method is none, or HTTP Digest, or caller is in the Access List). The default is yes. |
| Line 1 VoIP Caller DP | Index of the dial plan in the dial plan pool to be used when the VoIP Caller is calling from Line 1 of the same SPA3102 unit during normal operation (in other words, not due to fallback to PSTN service when Line 1 VoIP service is down). Choose from {none, 1, 2, 3, 4, 5, 6, 7, 8} Note Authentication is skipped for Line 1 VoIP caller. The default is 1. |
| Default VoIP Caller DP | Index of the dial plan in the dial plan pool to be used when the VoIP Caller is not authenticated. Choose from {none, 1, 2, 3, 4, 5, 6, 7, 8}. The default is 1. |

| Field | Description |
|---------------------------------|---|
| Line 1 Fallback DP | <p>Index of the dial plan in the dial plan pool to be used when the VoIP Caller is calling from Line 1 of the same SPA3102 unit due to fallback to PSTN service when Line 1 VoIP service is down. Choose from {none, 1, 2, 3, 4, 5, 6, 7, 8}.</p> <p>The default is 1.</p> |
| VoIP Caller ID Pattern | <p>A comma separated list of caller number templates such that callers with numbers not matching any of these templates will be rejected for PSTN gateway service regardless of the setting of the authentication method. The comparison is applied before access list is applied. If this parameter is blank (not specified), all callers will be considered for PSTN gateway service.</p> <p>For example: 1408*, 1512???1234.</p> <p>Note: '?' matches any single digit; '*' matches any number of digits.</p> <p>The default is blank.</p> |
| VoIP Access List | <p>A comma separated list of IP address templates, such that callers with source IP address matching any of the templates will be accepted for PSTN gateway service without further authentication. For example: 192.168.*.*, 66.43.12.1??.</p> <p>The default is blank.</p> |
| VoIP Caller 1/2/3/4/5/6/7/8 PIN | <p>One of 8 PIN numbers that can be specified to control access to the PSTN gateway by a VoIP Caller, when the <VoIP Caller Authentication Method> is set to PIN.</p> <p>The default is blank.</p> |
| VoIP Caller 1/2/3/4/5/6/7/8 DP | <p>Index of the dial plan in the dial plan pool to be associated with the VoIP caller who enters the PIN that matches <VoIP Caller 1/2/3/4/5/6/7/8 PIN>.</p> <p>The default is 1.</p> |

VoIP Users and Passwords (HTTP Authentication)

| Field | Description |
|-----------------------------------|---|
| VoIP User 1/2/3/4/5/6/7/8 Auth ID | <p>The first of 8 user-id's that a VoIP Caller can use to authenticate itself to the SPA using the HTTP Digest method (in other words, by embedding an Authorization header in the SIP INVITE message sent to the SPA. If the credentials are missing or incorrect, the SPA will challenge the caller with a 401 response). The VoIP caller whose authentication user-id equals to this ID is referred to VoIP User 1 of this SPA.</p> <p>Note: If the caller specifies an authentication user-id that does not match any of the VoIP User Auth ID's, the INVITE will be rejected with a 403 response.</p> <p>The default is blank.</p> |
| VoIP User 1/2/3/4/5/6/7/8 DP | <p>Index of the dial plan in the dial plan pool to be used with VoIP User 1.</p> <p>The default is 1.</p> |

PSTN Line Tab

| Field | Description |
|------------------------------------|--|
| VoIP User 1/2/3/4/5/6/7/8 Password | The password to be used with VoIP User 1. The user assumes the identity of VoIP User 1 must therefore compute the credentials using this password, or the INVITE will be challenged with a 401 response The default is blank. |

| Field | Description |
|------------------------------------|--|
| VoIP User 1/2/3/4/5/6/7/8 Auth ID | The first of 8 user-id's that a VoIP Caller can use to authenticate itself to the SPA using the HTTP Digest method (in other words, by embedding an Authorization header in the SIP INVITE message sent to the SPA. If the credentials are missing or incorrect, the SPA will challenge the caller with a 401 response). The VoIP caller whose authentication user-id equals to this ID is referred to VoIP User 1 of this SPA. Note: If the caller specifies an authentication user-id that does not match any of the VoIP User Auth ID's, the INVITE will be rejected with a 403 response. The default is blank. |
| VoIP User 1/2/3/4/5/6/7/8 DP | Index of the dial plan in the dial plan pool to be used with VoIP User 1. The default is 1. |
| VoIP User 1/2/3/4/5/6/7/8 Password | The password to be used with VoIP User 1. The user assumes the identity of VoIP User 1 must therefore compute the credentials using this password, or the INVITE will be challenged with a 401 response The default is blank. |

Ring Settings

| Field | Description |
|--------------|----------------------|
| Default Ring | 1-8, Follow Line Cfg |

FXO (PSTN) Timer Values (sec)

| Field | Description |
|-------------------|---|
| VoIP Answer Delay | Delay in seconds before auto-answering inbound VoIP calls for the FXO account. The range is 0-255. The default is 3. |

| Field | Description |
|---------------------------|---|
| PSTN Answer Delay | Delay in seconds before auto-answering inbound PSTN calls after the PSTN starts ringing. The range is 0-255. The default is 16. |
| VoIP PIN Digit Timeout | Timeout to wait for the 1 st or subsequent PIN digits from a VoIP caller. The range is 0-255. The default is 10. |
| PSTN PIN Digit Timeout | Timeout to wait for the 1 st or subsequent PIN digits from a PSTN caller. The range is 0-255. The default is 10. |
| VoIP DLG Refresh Intvl | Interval between (SIP) Dialog refresh messages sent by the SPA to detect if the VoIP call-leg is still up. If value is set to 0, SPA will not send refresh messages and VoIP call-leg status is not checked by the SPA. The refresh message is a SIP ReINVITE and the VoIP peer must response with a 2xx response. If VoIP peer does not reply or response is not greater than 2xx, the SPA will disconnect both PSTN and VoIP call legs automatically. The range is 0-255. The default is 30. |
| PSTN Ring Thru Delay | Delay in seconds before starting to ring thru Line 1 after the PSTN starts ringing. In order for Line 1 to have the caller-id information, the delay should be set to larger than the delay required to complete the PSTN caller-id delivery (such as 5s). The range is 0-255. The default is 5. |
| PSTN-To-VoIP Call Max Dur | Limit on the duration of a PSTN-To-VoIP Gateway Call. Unit is in seconds. 0 means unlimited. The range is 0-2147483647. The default is 0. |
| VoIP-To-PSTN Call Max Dur | Limit on the duration of a VoIP-To-PSTN Gateway Call. Unit is in seconds. 0 means unlimited. The range is 0-2147483647. The default is 0. |
| PSTN Dialing Delay | Delay after hook before the SPA dials a PSTN number. The range is 0-255. The default is 1. |
| PSTN Ring Timeout | Delay after a ring burst before the SPA decides that PSTN ring has ceased. The range is 0-255. The default is 5. |
| PSTN Dial Digit Len | Determines the on/off time when transmitting digits through the FXO port. The syntax is <i>on-time/off-time</i> , where <i>on-time</i> and <i>off-time</i> are expressed in seconds with up to two decimal places, within the permitted range, which is from .05 to 3.00. The default is .1/.1. If this value is blank, the default is used. |

PSTN Line Tab

| Field | Description |
|--------------------------|---|
| PSTN Hook Flash Len | The length of the hook flash in seconds. During a PSTN-to-VoIP gateway call, the Linksys ATA processes the out-of-band hook flash signal sent from the VoIP peer through a hook-flash (momentary on-hook signal) on the FXO port. This allows the VoIP peer to initiate a three-way conference call and subsequent call transfer. The duration of the on-hook signal can be configured using this parameter. The default is 0.25. The permitted range is limited to 0.02 to 1.6 seconds. |
| PSTN Ring Thru CWT Delay | Specify the delay before incoming PSTN calls will ring Line 1 using a Call Waiting Tone. The default is 3 . |
| PSTN Ring Timeout | Specify the delay after a ring burst before the Gateway decides that the PSTN ring has ended. The default is 5 . |
| PSTN Dialing Delay | Specify the delay after the PSTN phone line is on-hook before the Gateway dials a PSTN number. The default is 1 . |
| PSTN Dial Digit Len | Specify the on/off time when the Gateway transmits digits through the Line (FXO) port. The syntax is on-time/off-time, expressed in seconds with up to two decimal places. The permitted range is 0.05 to 3.00. The default is .1/.1 . |

PSTN Disconnect Detection

| Field | Description |
|----------------------------|---|
| Detect CPC | CPC is a brief removal of tip-and-ring voltage. If enabled, the SPA will disconnect both call legs when this signal is detected during a gateway call. The default is yes. |
| Detect Polarity Reversal | If enabled, SPA will disconnect both call legs when this signal is detected during a gateway call. If it is a PSTN gateway call, the 1st polarity reversal is ignored and the 2 nd one triggers the disconnection. For VoIP gateway call, the 1 st polarity reversal triggers the disconnection. The default is yes. |
| Detect (PSTN) Long Silence | If enabled, SPA will disconnect both call legs when the PSTN side has no voice activity for a duration longer than the length specified in <Long Silence Duration> during a gateway call The default is yes. |
| Min CPC Duration | Specify the minimum duration of a low tip-and-ring voltage (below 1V) for the Gateway to recognize it as a CPC signal or PSTN line removal. The default is 0.2 . |
| Detect Disconnect Tone | If enabled, SPA will disconnect both call legs when it detects the disconnect tone from the PSTN side during a gateway call. Disconnect tone is specified in the <Disconnect Tone> parameter, which depends on the region of the PSTN service. The default is yes. |

| Field | Description |
|------------------------------|--|
| (PSTN) Long Silence Duration | This is minimum length of PSTN silence (or inactivity) in seconds to trigger a gateway call disconnection if <Detect Long Silence> is yes. The default is 30. |
| Silence Threshold | This parameter adjusts the sensitivity of PSTN silence detection. Choose from {very low, low, medium, high, very high}. The higher the setting, the easier to detect silence and hence easier to trigger a disconnection. The default is medium. |
| Disconnect Tone | This is the tone script which describes to the SPA the tone to detect as a disconnect tone. The syntax follows a standard Tone Script with some restrictions. Default value is standard US reorder (fast busy) tone, for 4 seconds. Restrictions: 2 frequency components must be given. If single frequency is desired, the same frequency is used for both The tone level value is not used. -30 (dBm) should be used for now. Only 1 segment set is allowed Total duration of the segment set is interpreted as the minimum duration of the tone to trigger detection 6 segments of on/off time (seconds) can be specified. A 10% margin is used to validated cadence characteristics of the tone. The default is 480@-30,620@-30;4(.25/.25/1+2). |

International Control (Settings)

| Field | Description |
|--------------------|--|
| FXO Port Impedance | Desired impedance of the FXO Port. Choose from {600, 900, 270+}. The default is 600. |
| Ring Frequency Min | Specify the lower limit of the ring frequency used to detect the ring signal. The default is 10 . |
| SPA To PSTN Gain | dB of digital gain (or attenuation if negative) to be applied to the signal sent from the SPA to the PSTN side. The range is -15 to 12. The default is 0. |
| Ring Frequency Max | Specify the higher limit of the ring frequency used to detect the ring signal. The default is 100 . |
| PSTN To SPA Gain | dB of digital gain (or attenuation if negative) to be applied to the signal sent from the PSTN side to the SPA. The range is -15 to 12. The default is 0. |

PSTN Line Tab

| Field | Description |
|------------------------------|---|
| Ring Validation Time | Specify the minimum signal duration required by the Gateway for recognition as a ring signal. The default is 256 ms . |
| Tip/Ring Voltage Adjust | Choices are {3.1, 3.2, 3.35, 3.5}. The default is 3.5. |
| Operational Loop Current Min | Choices for mA are: {10, 12, 14, 16}. The default is 10. |
| On-Hook Speed | Choose from {Less than 0.5ms, 3ms (ETSI), 26ms (Australia)}. The default is Less than 0.5ms. |
| Current Limiting Enable | Enable or disable current limiting. The default is no. |
| Ring Frequency Min | Minimum ring frequency to detect. The range is 5-100. The default is 10. |
| Ring Frequency Max | Maximum ring frequency to detect. The range is 5-100. The default is 100. |
| Ring Validation Time | Choose from {100, 150, 200, 256, 384, 512, 640, 1024} (ms). The default is 256ms. |
| Ring Indication Delay | Choose from {0, 512, 768, 1024, 1280, 1536, 1792} (ms). The default is 512ms. |
| Ring Timeout | Choose from {0, 128, 256, 384, 512, 640, 768, 896, 1024, 1152, 1280, 1408, 1536, 1664, 1792, 1920} (ms). The default is 640 ms. |
| Ring Threshold | Choose from {13.5–16.5, 19.35–2.65, 40.5–49.5} (Vrms). The default is 13.5-16.5 Vrms. |
| Ringer Impedance | Choose from {High, Synthesized(Poland, S.Africa, Slovenia)}. The default is high. |
| Line-In-Use Voltage | Determines the voltage threshold at which the SPA-3000 assumes the PSTN is in use by another handset sharing the same line (and will declare PSTN gateway service not available to incoming VoIP callers). The default value is 40v. |

User 1/2 Tab

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

- [Call Forward Settings, page 5-65](#)
- [Selective Call Forward Settings, page 5-66](#)
- [Speed Dial Settings, page 5-67](#)
- [Supplementary Service Settings, page 5-67](#)

- [Distinctive Ring Settings, page 5-68](#)
- [Ring Settings, page 5-68](#)

User 1/2 refers to the subscriber of Line 1 or Line 2.



Note

For the SPA8000, the settings on this page occur on each Line tab ([1] to [8]).

When a call is made from Line 1 or Line 2, Linksys IP phone shall use the user and line settings for that Line; there is no user login support in Linksys IP 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.



Note

References in this section to the SPA3102 also apply to the AG310.

Call Forward Settings

| 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"> • gw0 – forward the caller to use the PSTN gateway • <pstn-number>@gw0 – forward to caller to the PSTN number (dialed automatically by the SPA through the PSTN gateway) <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> |

| Field | Description |
|-------------------|---|
| 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"> • gw0 – forward the caller to use the PSTN gateway • <pstn-number>@gw0 – forward to caller to the PSTN number (dialed automatically by the SPA through the PSTN gateway) <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> |

Selective Call Forward Settings

| Field | Description |
|--------------------------------|---|
| Cfwd Sel1/2/3/4/5/6/7/8 Caller | <p>Caller number pattern to trigger Call Forward Selective 1, 2, 3, 4, 5, 6, 7, or 8.</p> <p>The default is blank.</p> |
| Cfwd Sel1/2/3/4/5/6/7/8 Dest | <p>Forward number for Call Forward Selective 1, 2, 3, 4, 5, 6, 7, or 8.</p> <p>Same as Cfwd All Dest.</p> <p>The default is blank.</p> |
| Block Last Caller | <p>ID of caller blocked via the Block Last Caller service.</p> <p>The default is blank.</p> |
| Accept Last Caller | <p>ID of caller accepted via the Accept Last Caller service.</p> <p>The default is blank.</p> |
| Cfwd Last Caller | <p>The Caller number that is actively forwarded to <Cfwd Last Dest> by using the Call Forward Last activation code</p> <p>The default is blank.</p> |
| Cfwd Last Dest | <p>Forward number for the <Cfwd Last Caller></p> <p>Same as Cfwd All Dest.</p> <p>The default is blank.</p> |

Speed Dial Settings

| 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 Service Settings

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

| Field | Description |
|---------------------|---|
| CW Setting | Call Waiting on/off for all calls. 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. |
| CID Setting | Caller ID Generation on or off. The default is yes. |
| CWCID Setting | Call Waiting Caller ID Generation on or off. The default is yes. |
| Dist Ring Setting | Distinctive Ring on or off. The default is yes. |
| Secure Call Setting | If yes, all outbound calls are secure calls by default. The default is no. |
| Miss Call Shortcut | Enables or disables the display of missed calls on the LCD display. |
| Message Waiting | This is updated when there is voicemail notification received by the Linksys IP phone. The user 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 will survive after reboot or power cycle. The default is no. |

| Field | Description |
|-------------------------------|--|
| 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. |

Distinctive Ring Settings

Caller number patterns are matched from Ring 1 to Ring 8. The first match (not the closest match) will be used for alerting the subscriber.

| Field | Description |
|----------------------------|--|
| Ring1/2/3/4/5/6/7/8 Caller | Caller number pattern to play Distinctive Ring/CWT 1, 2, 3, 4, 5, 6, 7, or 8. The default is blank. |

Ring Settings

| Field | Description |
|----------------------|--|
| Default Ring | Default ringing pattern, 1 – 8, for all callers. The default is 1. |
| Default CWT | Default CWT pattern, 1 – 8, for all callers. The default is 2. |
| Hold Reminder Ring | Ring pattern for reminder of a holding call when the phone is on-hook. The default is None. |
| Call Back Ring | Ring pattern for call back notification. The default is None. |
| Cfwd Ring Splash Len | Duration of ring splash when a call is forwarded (0 – 10.0s). The default is 0. |
| Cblk Ring Splash Len | Duration of ring splash when a call is blocked (0 – 10.0s). The default is 0. |

| Field | Description |
|----------------------|--|
| VMWI Ring Splash Len | <p>Duration of ring splash when new messages arrive before the VMWI signal is applied (0 – 10.0s).</p> <p>The default is .5.</p> |
| VMWI Ring Policy | <p>The parameter controls when a ring splash is played when a the VM server sends a SIP NOTIFY message to the Linksys IP phone indicating the status of the subscriber’s mail box. 3 settings are available:</p> <p>New VM Available – ring as long as there is 1 or more unread voicemail</p> <p>New VM Becomes Available – ring when the number of unread voicemail changes from 0 to non-zero</p> <p>New VM Arrives – ring when the number of unread voicemail increases.</p> <p>The default is New VM Available.</p> |
| Ring On No New VM | <p>If enabled, the Linksys IP phone will play a ring splash when the VM server sends SIP NOTIFY message to the Linksys IP phone indicating that there are no more unread voicemails. Some equipment requires a short ring to precede the FSK signal to turn off VMWI lamp.</p> <p>The default is no.</p> |

PSTN User Tab (SPA3102/AG310)

This section describes the fields for the following headings on the PSTN User tab on the SPA3102:

- [PSTN-To-VoIP Selective Call Forward Settings, page 5-70](#)
- [PSTN-To-VoIP Speed Dial Settings, page 5-70](#)
- [PSTN Ring Thru Line 1 Distinctive Ring Settings, page 5-70](#)
- [PSTN Ring Thru Line 1 Ring Settings, page 5-71](#)
- [PSTN/VoIP Caller Commands via DTMF, page 5-71](#)



Note

References in this section to the SPA3102 also apply to the AG310.

PSTN-To-VoIP Selective Call Forward Settings

| Field | Description |
|--------------------|---|
| Cfwd Sell Caller | First of eight PSTN Caller Number Patterns to be blocked for VoIP gateway services or forwarded to a certain VoIP number. If the caller is blocked, the SPA will not auto-answers the call. |
| Cfwd Sell Dest | VoIP destination to forward a PSTN caller matching <Cfwd Sell Caller>. If this entry is blank, the PSTN caller is blocked for VoIP service. |
| Block Last Caller | ID of caller blocked via the Block Last Caller service |
| Accept Last Caller | ID of caller accepted via the Accept Last Caller service |

PSTN-To-VoIP Speed Dial Settings

| Field | Description |
|----------------------------|---|
| Speed Dial 2/3/4/5/6/7/8/9 | The VoIP number to call when the PSTN caller dials a single digit '2' |

PSTN Ring Thru Line 1 Distinctive Ring Settings

| Field | Description |
|----------------------------|--|
| Ring1/2/3/4/5/6/7/8 Caller | First of 8 PSTN Caller Number Patterns such that the corresponding ring will be used to ring through Line 1 if the PSTN caller matches this pattern. |

PSTN Ring Thru Line 1 Ring Settings

| Field | Description |
|--------------|--|
| Default Ring | The default ring to be used to ring through Line 1. Choose from {1,2,3,4,5,6,7,8,Follow Line 1}. If Follow Line 1 is selected, the ring to be used is determined by Line 1's distinctive ring settings. The default is 1. |

PSTN/VoIP Caller Commands via DTMF

| Field | Description |
|-------|--|
| **# | Disconnect the PSTN line (SPA will take the FXO port on-hook). |
| **1 | End the current call and restarts dial tone. |

Acronyms

| | |
|-------|--|
| ACS | Auto-Configuration Server |
| 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

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.

Auto-Configuration Server—A server used for automatic, remote provisioning of CPEs, as defined by DSL Forum specifications TR-069, TR-098, and TR-104.

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 (Linksys ATA 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.

User Guidelines

This appendix provides documentation for the use of the SPA products. It includes the following sections:

- [Basic Services, page C-1](#)
- [Enhanced Services, page C-2](#)

The SPA can be configured to the custom requirements of the service provider, so that from the subscriber point of view, the service behaves exactly as the service provider wishes, with varying degrees of control left with the end user. This means that a service provider can leverage the programmability of the SPA to offer sometimes subtle yet continually valuable and differentiated services optimized for the network environment or target markets.

This chapter describes how some of the supported basic, enhanced, and supplementary services can be implemented. The implementations described below are not the only way to achieve the desired service behavior.

Basic Services

Originating a Phone Call

| | |
|---|---|
| Service description | Placing telephone a call to another telephone or telephony system (IVR, conference bridge, and so on). This is the most basic service. |
| User action required to activate or use | When the user picks up the handset, the SPA provides dial tone and is ready to collect dialing information via DTMF digits from the telephone touchtone key pad. |
| Expected call and network behavior | Although it is possible to support overlapped dialing within the context of SIP, the SPA collects a complete phone number and sends the full number in a SIP INVITE message to the proxy server for further call processing. To minimize dialing delay, the SPA maintains a dial plan and matches it against the cumulative number entered by the user. The SPA also detects invalid phone numbers not compatible with the dial plan and alerts the user via a configurable tone (Reorder) or announcement. |
| User action required to deactivate or end | Hang up the telephone. |

Receiving a Phone Call

| | |
|---|---|
| Service description | The SPA can receive calls from the PSTN or other IP Telephony subscribers |
| User action required to activate or use | When the telephone rings, pick up the handset and begin talking. |
| Expected call and network behavior | Each subscriber is assigned an E.164 ID (phone number) so that they may be reached from wired or wireless callers on the PSTN or IP network. The SPA supplies ring voltage to the attached telephone set to alert the user of incoming calls. |
| User action required to deactivate or end | Hang up the telephone. |

Enhanced Services

Caller ID

| | |
|---|--|
| Service description | If available, the SPA supports the generation and pass through of Caller ID information. |
| User action required to activate or use | No user action required. The user telephone equipment must support Caller ID to display the caller name and/or number. |
| Expected call and network behavior | In between ringing bursts, the SPA can generate a Caller-ID signal to the attached phone when the phone is on-hook. As part of the INVITE message, the SPA sends the caller name and number as it is configured in the profile. |
| User action required to deactivate or end | No user action required. See CLIP and CLIR. |

Calling Line Identification Presentation (CLIP)

| | |
|---|--|
| Service description | Some users choose to block their Caller ID information for all outgoing calls. However, there may be circumstances where sending Caller ID information for a call is desired; that is, trying to reach a party that does not accept Caller ID blocked calls. |
| User action required to activate or use | <ol style="list-style-type: none"> 1. Lift the receiver 2. Listen for dial tone 3. Press *82 4. Listen for dial tone 5. Dial the telephone number you are calling |

| | |
|---|--|
| Expected call and network behavior | Caller ID is sent to the distant party for this call only. Users must repeat this process at the start of each call. |
| User action required to deactivate or end | No action required. This service is only in effect for the duration of the current call. |

Calling Line Identification Restriction (CLIR)—Caller ID Blocking

| | |
|---|---|
| Service description | This feature allows the user to block the delivery of their Caller ID to the number they are calling. This feature must be activated before dialing each call and is only in effect for the duration of each call. |
| User action required to activate or use | <ol style="list-style-type: none"> 1. Lift the receiver 2. Listen for dial tone 3. Press *81 4. Listen for dial tone 5. Dial the telephone number you are calling <p>You must repeat this process at the start of each call.</p> |
| Expected call and network behavior | The user activates this service to hide their Caller ID when making an outgoing call. |
| User action required to deactivate or end | No action required. This service is only in effect for the duration of the current call. |

Call Waiting

| | |
|---|---|
| Service description | The user can accept a call from a third party while engaging in an active call. The SPA alerts the subscriber of the second incoming call by playing a call waiting tone. |
| User action required to activate or use | <p>If you choose to answer the second call, do one of the following:</p> <ul style="list-style-type: none"> • Press and release your phone switch hook (the button you release when you take your phone off the hook) • Press the flash button (if your phone has one) <p>This puts your first call on hold and automatically connects you to your second call.</p> <p>To put your second caller back on hold and return to your first caller, press the switch hook or flash button again. (You can alternate between calls as often as you like.)</p> |
| Expected call and network behavior | If the user is on a call when another call comes in, they hear a series of beeps/tones alerting them to the second call. The person calling hears normal ringing. |
| User action required to deactivate or end | See Cancel Call Waiting. |

Disable or Cancel Call Waiting

| | |
|---|---|
| Service description | The SPA supports disabling of call waiting permanently or on a per-call basis. |
| User action required to activate or use | <p>To temporarily disable Call Waiting (for the length of one call) do the following before placing a call:</p> <ol style="list-style-type: none"> 1. Lift Receiver 2. Press *70 3. Listen for dial tone, then dial the number you want to call. <p>Call Waiting is now disabled for the duration of this call only.</p> <p>To deactivate Call Waiting while on a call:</p> <ol style="list-style-type: none"> 1. Press the switch hook or flash button briefly. This puts the first call on hold. 2. Listen for three short tones and then a dial tone. 3. Press *70 4. Listen for dial tone then return to your call by pressing the switch hook or flash button. <p>Call Waiting is now disabled for the duration of this call.</p> <p>To deactivate Call Waiting while on a permanent basis (until cancelled):</p> <ol style="list-style-type: none"> 1. Lift the receiver 2. Listen for dial tone 3. Press *57 <p>You will hear a confirmation tone signaling your request to cancel Call Waiting has been accepted.</p> |
| Expected call and network behavior | Callers who dial your number receive a busy signal, or the caller is forwarded to voice mail or another predetermined forwarding number, if available. |
| User action required to deactivate or end | <p>If you have cancelled Call Waiting temporarily, no user action is required.</p> <p>If you deactivated call waiting and wish to reinstate the service, do the following:</p> <ol style="list-style-type: none"> 1. Lift the receiver 2. Listen for dial tone 3. Press *57 <p>You will hear a confirmation tone signaling your request to cancel Call Waiting has been accepted.</p> |


Call-Waiting with Caller ID

| | |
|---|--|
| Service description | When the user is on the phone and has Call Waiting active, the new caller Caller ID information is displayed on the user phone display screen at the same time the user is hearing the Call Waiting beeps/tones. |
| User action required to activate or use | The telephone equipment connected to the SPA must support Call-Waiting with Caller ID. |
| Expected call and network behavior | In between call waiting tone bursts, the SPA can generate a Caller-ID signal to the attached phone when it is off hook. |
| User action required to deactivate or end | Not applicable. |

Voice Mail

| | |
|---|--|
| Service description | Service providers may provide voice mail service to their subscribers. Users have the ability to retrieve voice mail via the telephone connected to the SPA. |
| User action required to activate or use | <p>The SPA indicates that a message is waiting by playing stuttered dial tone when the user picks up the handset.</p> <p>To retrieve messages:</p> <ol style="list-style-type: none"> 1. Lift the receiver 2. Listen for dial tone 3. Dial the phone number assigned to the SPA <p>You will be connected to the voice mail server and prompted by a voice response system with instructions to listen to your messages.</p> |
| Expected call and network behavior | When voice mail is available for a subscriber, a notification message is sent from the voice mail server to the SPA. When the user dials their own phone number, the SPA connects the subscriber to their voice mail system, which can then connect them to their individual voice mail box. |
| User action required to deactivate or end | Follow instructions of the voice mail system or simply hang up the telephone. |

Attendant Call Transfer

| | |
|---|--|
| Service description | Attendant Call Transfer lets a customer use their touchtone phone to send a call to any other phone, inside or outside their business, including a wireless phones. |
| User action required to activate or use | <p>While in a call with the party to be transferred:</p> <ol style="list-style-type: none"> 1. Press the switch hook or flash button on the phone to place the party on hold 2. Listen for three short tones followed by dial tone 3. Dial the number to which you will transfer the caller 4. Stay on the line until the called number answers 5. Announce the call 6. Press the switch hook or flash button adding the held party to the call 7. Hang up to connect the two parties and transfer the call <p> Note You can hook flash while the third party is ringing to start an early conference, and then hang up to complete the transfer without waiting for the third party to answer first.</p> |
| Expected call and network behavior | When the user presses the switch hook or flash button, the transferee is placed on hold. When the user successfully dials the transfer number and the party answers, the transferee can be added to the call by pressing the switch hook or flash button, creating a three-way conference. When the user hangs up the phone, the transferee and the called party remain in a call. |
| User action required to deactivate or end | Not applicable. |

Unattended or “Blind” Call Transfer

| | |
|---|--|
| Service description | Unattended or “Blind” Call Transfer lets a customer use their touchtone phone to send a call to any other phone, inside or outside their business, including a wireless phones. |
| User action required to activate or use | <p>While in a call with the party to be transferred:</p> <ol style="list-style-type: none"> 1. Press the switch hook or flash button on the phone to place the party on hold 2. Enter *98 3. Dial the number to which you will transfer the caller <p>The call is transferred when a complete number is entered. You will hear a short confirmation tone, followed by regular dial tone</p> |

| | |
|---|--|
| Expected call and network behavior | When the user presses the switch hook or flash button, the transferee is placed on hold. When the user successfully dials the transfer number, the transferee automatically calls the dialed number. |
| User action required to deactivate or end | Not applicable. |

Call Hold

| | |
|---|---|
| Service description | Call Hold lets you put a caller on hold for an unlimited period of time. It is especially useful on phones without the hold button. Unlike a hold button, this feature provides access to a dial tone while the call is being held. |
| User action required to activate or use | Press the switch hook or flash button on the phone to place the first party on hold. You will hear a dial tone. To make another call, enter the new number. To return to call on hold, hang up, and the phone set rings with the first call on the line (or hook flash again) |
| Expected call and network behavior | |
| User action required to deactivate or end | Hang up the telephone. |

Three-Way Calling

| | |
|---|--|
| Service description | The user can originate a call to a third party while engaging in an active call. |
| User action required to activate or use | <ol style="list-style-type: none"> 1. Press the switch hook or flash button on the phone to place the first party on hold 2. Listen for three short tones followed by dial tone 3. Dial the number of the third party. 4. When the third party answers you may have a conversation with them while the other party is on hold. <p>To hold a conference with the party on hold and the third party, simply press the switch hook or flash button.</p> |
| Expected call and network behavior | The SPA supports up to two calls per line. The SPA can conference two calls by bridging the second and third parties. |
| User action required to deactivate or end | Hang up the telephone. |

Three-Way Ad-Hoc Conference Calling

| | |
|---|---|
| Service description | This feature allows the user to conference up to two other numbers on the same line to create a three-way call. |
| User action required to activate or use | <p>If you are already on a call and wish to add a third party:</p> <ol style="list-style-type: none"> 1. Press the switch hook or flash button 2. Listen for dial tone 3. Dial the third party normally 4. When the third party number starts to ring press the switch hook or flash button again <p>You now have the original caller and the third party together with you on the same call.</p> <p>If you want to initiate a new three-way call:</p> <ol style="list-style-type: none"> 1. Call the first party in the normal manner 2. Follow the directions for adding a third party (see instructions above) |
| Expected call and network behavior | <p>The SPA can host a three-way conference and perform three-way audio mixing (without the need of an external conference bridge device or service).</p> <p>If you also have Call Transfer, you can also hang up at any time to transfer the original caller to the third party</p> |
| User action required to deactivate or end | |

Call Return

| | |
|---|--|
| Service description | The SPA supports a service that allows the SPA to automatically dial the last caller's number. |
| User action required to activate or use | <ol style="list-style-type: none"> 1. Pick up the receiver 2. Listen for dial tone 3. Press *69 to dial back the last caller that tried to reach you. |
| Expected call and network behavior | This service gives the user the convenience of recalling the last incoming call to their number automatically. |
| User action required to deactivate or end | No user action required. |

Automatic Call Back

| | |
|---|--|
| Service description | This feature allows the user to place a call to the last number they tried to reach whether the call was answered, unanswered, or busy, by dialing an activation code. |
| User action required to activate or use | <ol style="list-style-type: none"> 1. Pick up the receiver 2. Listen for dial tone 3. Press *07 |
| Expected call and network behavior | <p>If the number called is idle, the call rings through and completes normally. If the called number is busy, the user hears a special announcement and the feature monitors the called number for up to 30 minutes. When both lines are idle, the user hears a special ring.</p> <p>During the monitoring process, the user can continue to originate and receive calls without affecting the Call Return on Busy request. Call Return on Busy requests can be canceled by dialing the deactivation code.</p> |
| User action required to deactivate or end | <ol style="list-style-type: none"> 1. Lift the receiver 2. Listen for dial tone 3. Press *__ |

Call FWD—Unconditional


| | |
|---|--|
| Service description | All calls are immediately forwarded to the designated forwarding number. The SPA does not ring or provide call waiting when Call FWD—Unconditional is activated. |
| User action required to activate or use | <ol style="list-style-type: none"> 1. Lift the receiver 2. Listen for dial tone 3. Press *72 4. Listen for dial tone and enter the telephone number you are forwarding your call to. <p>Activation is confirmed with three short bursts of tone and your forwarding is activated.</p> <p>Alternatively, the user can activate this feature from a web browser interface.</p> |

| | |
|---|--|
| Expected call and network behavior | This feature allows a user the option to divert (forward) all calls to their telephone number to any number using the touchtone keypad of their telephone or web browser interface. This service is activated or deactivated from the phone being forwarded or the web browser interface. |
| User action required to deactivate or end | <ol style="list-style-type: none"> 1. Lift the receiver 2. Listen for dial tone 3. Press *73 <p>You will hear a confirmation tone signaling your change has been accepted.</p> <p>Alternatively, the user can deactivate this feature from a web browser interface.</p> |

Call FWD – Busy

| | |
|---|--|
| Service description | Calls are forwarded to the designated forwarding number if the subscriber line is busy because of the following: primary line already in a call, primary and secondary line in a call or conference. |
| User action required to activate or use | <ol style="list-style-type: none"> 1. Lift the receiver 2. Listen for dial tone 3. Press *90 4. Listen for dial tone and enter the telephone number you are forwarding your call to. <p>Activation is confirmed with three short bursts of tone and your forwarding is activated.</p> <p>Alternatively, the user can activate this feature from a web browser interface.</p> |
| Expected call and network behavior | This feature allows a user the option to divert (forward) calls to their telephone number to any number when their phone is busy or in conference by using the touchtone keypad of their telephone or web browser interface. This service is activated or deactivated from the phone being forwarded or the web browser interface. |
| User action required to deactivate or end | <ol style="list-style-type: none"> 1. Lift the receiver 2. Listen for dial tone 3. Press *91 <p>You will hear a confirmation tone signaling your change has been accepted.</p> <p>Alternatively, the user can deactivate this feature from a web browser interface.</p> |

Call FWD—No Answer

| | |
|---|--|
| Service description | Calls are forwarded to the designated forwarding number after a configurable time period elapses while the SPA is ringing and does not answer. |
| User action required to activate or use | <ol style="list-style-type: none"> 1. Lift the receiver 2. Listen for dial tone 3. Press *92 Listen for dial tone and enter the telephone number you are forwarding your call to. <p>Activation is confirmed with three short bursts of tone and your forwarding is activated.</p> <p>Alternatively, the user can activate this feature from a web browser interface.</p> <p> Note The forward delay is entered from the web interface. Default is 20s.</p> |
| Expected call and network behavior | This feature allows a user the option to divert (forward) calls to their telephone number to any other dialable number when their phone is not answered by using the touchtone keypad of their telephone or web browser interface. This service is activated or deactivated from the phone being forwarded or the web browser interface. |
| User action required to deactivate or end | <ol style="list-style-type: none"> 1. Lift the receiver 2. Listen for dial tone 3. Press *93 <p>You will hear a confirmation tone signaling your change has been accepted.</p> <p>Alternatively, the user can deactivate this feature from a web browser interface.</p> |


Anonymous Call Blocking

| | |
|---|--|
| Service description | By setting the corresponding configuration parameter on the SPA, the subscriber has the option to block incoming calls that do not reveal the Caller ID of the caller. |
| User action required to activate or use | <ol style="list-style-type: none"> 1. Pick up the receiver 2. Listen for dial tone 3. To activate, press *77 |
| Expected call and network behavior | When activated by the user, callers hear a busy tone. |
| User action required to deactivate or end | To de-activate, press *87 |

Distinctive/Priority Ringing and Call Waiting Tone

| | |
|---|--|
| Service description | The SPA supports a number of ringing and call waiting tone patterns to be played when incoming calls arrive. The choice of alerting pattern to use is carried in the incoming SIP INVITE message inserted by the SIP proxy server (or other intermediate application server in the service provider domain). |
| User action required to activate or use | <ol style="list-style-type: none"> 1. Pick up the receiver 2. Listen for dial tone 3. Press *26 |
| Expected call and network behavior | <p>With this service, incoming calls from multiple telephone numbers can be automatically identified by distinctive ringing. A distinctive ringing pattern (that is, short-long-short) accompanies incoming calls from the designated telephone numbers.</p> <p>If the user is engaged in conversation and a call from one of the designated numbers arrives, a distinctive call waiting tone (that is, short-long-short) accompanies the incoming call. Calls from other telephone numbers ring normally.</p> |

Speed Calling—Up to Eight Numbers or IP Addresses

| | |
|---|--|
| Service description | The SPA supports user programming of up to eight long distance, local, international, or emergency numbers and/or IP addresses for fast and easy access. |
| User action required to activate or use | <ol style="list-style-type: none"> 1. Pick up the receiver 2. Listen for dial tone 3. Press *74 4. Dial the single digit code under which the number is to be stored (2-9) 5. Dial the complete number to be stored just as if you were going to dial it yourself 6. Listen for Confirmation tone (two short beeps) 7. Hang up or repeat the sequence |
| |  <p>Note To enter IP addresses, a graphical user interface such as a web browser must be used.</p> |

| | |
|---|---|
| Expected call and network behavior | <ol style="list-style-type: none">1. Pick up the receiver2. Listen for dial tone3. Press single digit code assigned to the stored number (2-9)4. Press # to signal dialing complete The number is automatically dialed normally. |
| User action required to deactivate or end | None |

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