LINKSYS®
A Division of Cisco Systems, Inc.

Linksys ATA Administrator Guide

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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
- RTP300—IP router with two FXS ports
- WRTP54G—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 SPA3102 PSTN Gateway"	This chapter describes how to configure the Linksys SPA3102 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"
Italic	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

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CHAPTER 1

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 1-4
- Technology Background, page 1-10
- Where to Go From Here, page 1-13

Overview

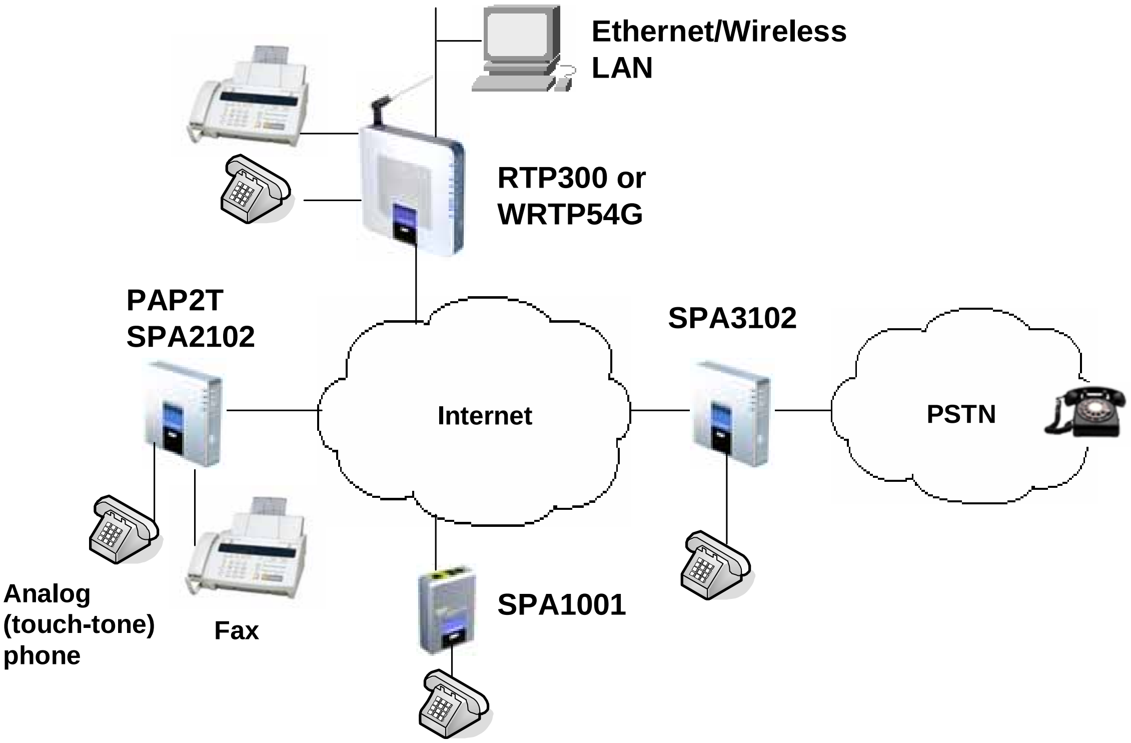
Table 1-1 summarizes the ports and features provided by the Linksys ATAs described in this document.

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
RTP300	Two (2)	—	One (1)	Four (4)	Two (2)	IP router with two FXS ports
WRTP54 G	Two (2)	—	One (1)	Four (4)	Two (2)	Wireless-G IP router with two FXS ports

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 RTP300 and WRTP54G provide QoS-enabled IP routers in addition to two ports for connecting analog telephone devices.

Figure 1-1 Linksys ATAs in a VoIP Network



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 1-5
- 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 mµ-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.

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	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: • Call Waiting Serv • Three Way Call Serv • Three Way Conf Serv 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.
Configurable Dial Plan with Interdigit Timers	The Linksys ATA has three configurable interdigit timers: • Initial timeout (T)—Handset off hook, no digit pressed yet. • Long timeout (L)—One or more digits pressed, more digits needed to reach a valid number (as per the dial plan). • Short timeout (S)—Current dialed number is valid, but more digits would also lead to a valid number.
Polarity Control	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.
Calling Party Control	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.
Report Generation and Event Logging	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.
Syslog and Debug Server Records	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.

Technology Background

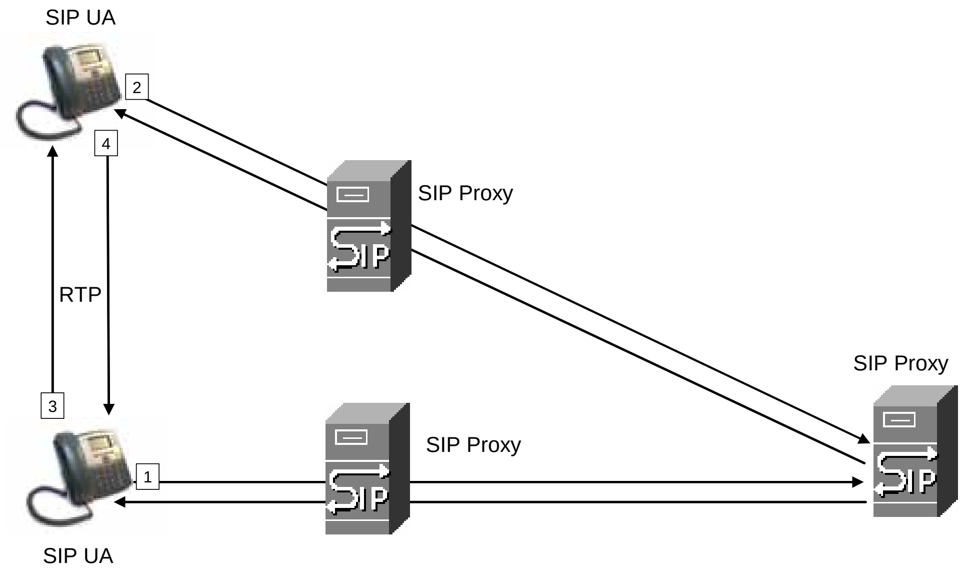
This section provides background information about the technology and protocols used by the ATA. It includes the following topics:

- Session Initiation Protocol, page 1-10
- Using the SPA9000 with a Firewall or Router, page 1-10
- Using the SPA9000 with a Firewall or Router, 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).

Using the SPA9000 with a Firewall or Router

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

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

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

Network Address Translation

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

- NAT Overview, page 1-11
- NAT Types, page 1-12
- Simple Traversal of UDP Through NAT, page 1-12
- SIP-NAT Interoperation, page 1-13

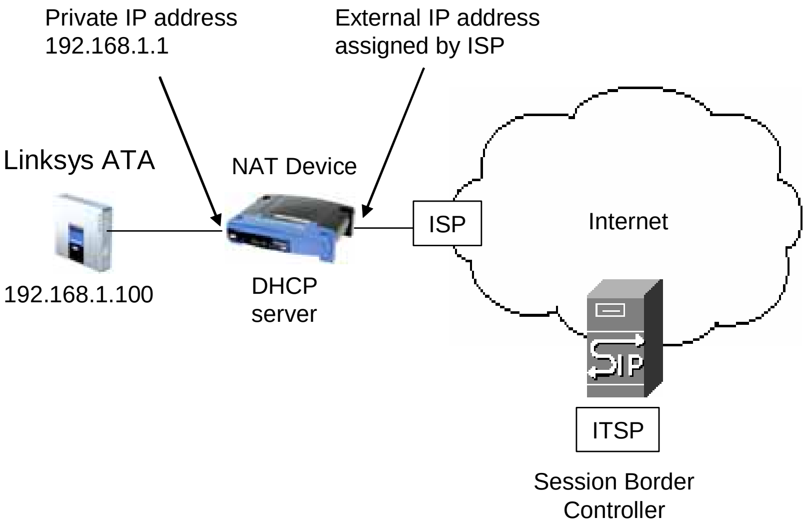
NAT Overview

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 for providing PSTN connectivity.	Chapter 4, "Configuring the SPA3102 PSTN Gateway"

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

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CHAPTER 2

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 2-8
- Using the Interactive Voice Response Interface, page 2-10
- Using the Administration Web Server, page 2-14

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 2-1
- PAP2T, page 2-2
- SPA1001, page 2-3
- SPA2102, page 2-4
- SPA3102, page 2-5
- RTP300, page 2-6
- WRTP54G, page 2-7

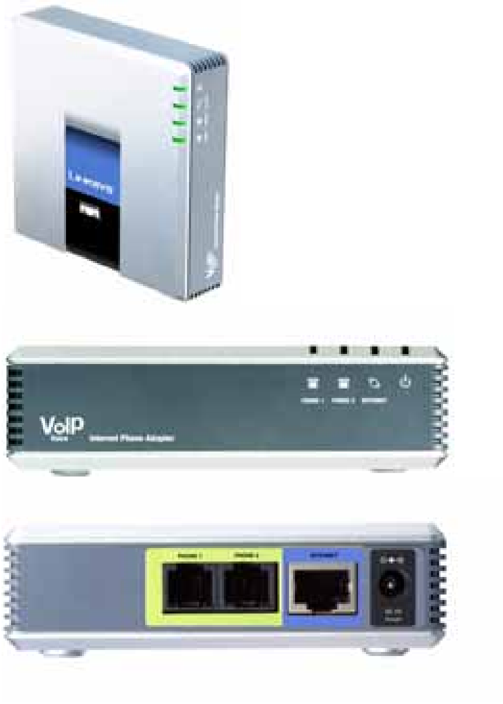
Caring for Your Hardware

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.

PAP2T

The PAP2T provides two FXS ports (see Figure 2-1).

Figure 2-1 PAP2T



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

Table 2-1 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.

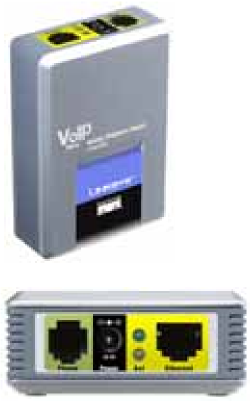
Table 2-2 PAP2T Back Panel

LED	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.

SPA1001

The SPA1001 provides one FXS port (see Figure 2-2).

Figure 2-2 SPA1001



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

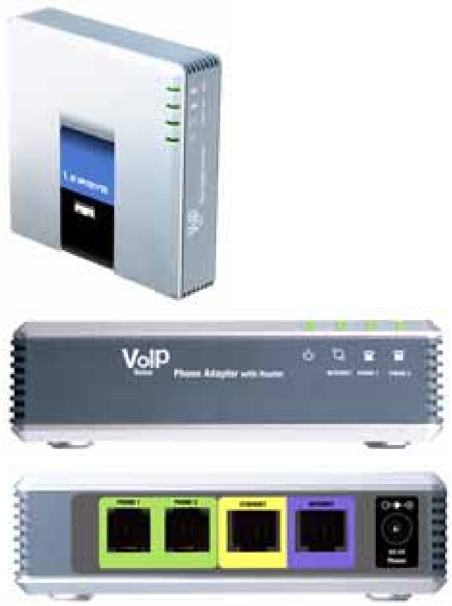
Table 2-3 SPA1001 Back Panel

LED	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 2-3 SPA2102



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

Table 2-4 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.

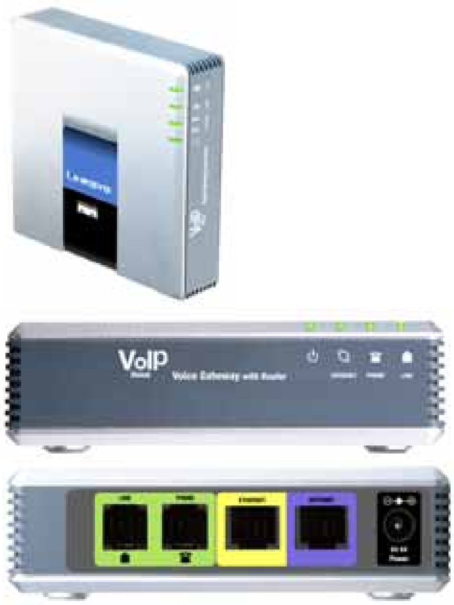
Table 2-5 SPA2102 Back Panel

LED	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.

Figure 2-4 SPA3102



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

Table 2-6 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.

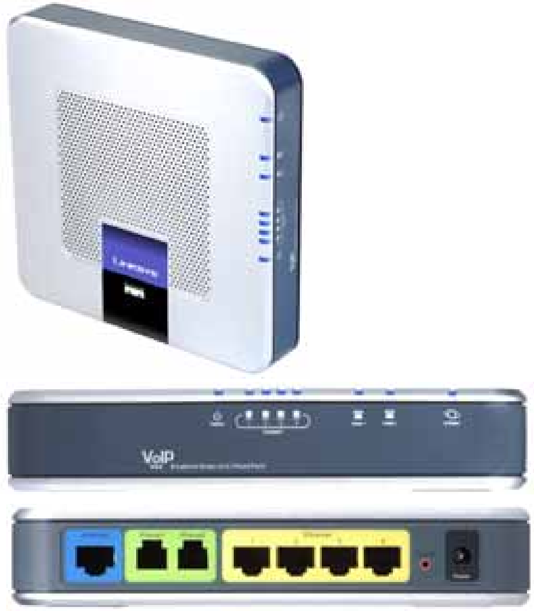
Table 2-7 SPA3102 Back Panel

LED	Function
Line	Connect to a standard telephone wall jack with an RJ-11cable.
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.

RTP300

The RTP300 provides an ATA with two FXS ports along with an IP router (see Figure 2-5). 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 2-5 RTP300



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

Table 2-8 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.

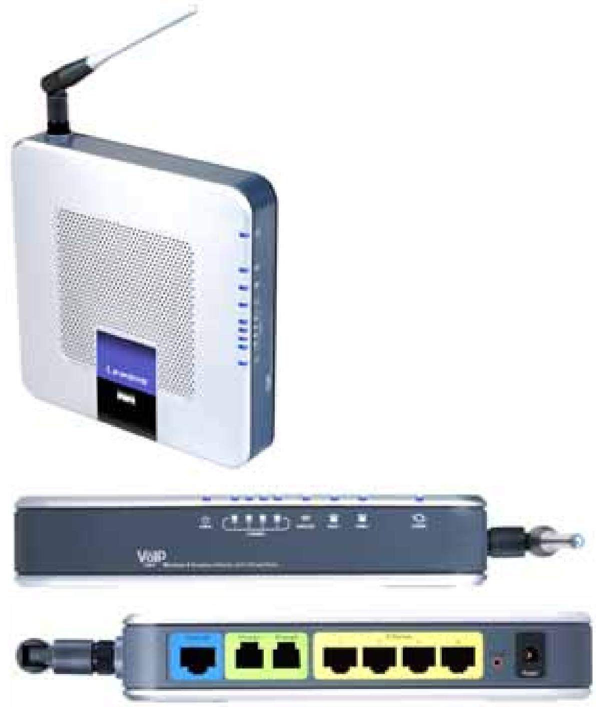
Table 2-9 RTP300 Back Panel

LED	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.

WRTP54G

The WRTP54G provides an ATA with two FXS ports and a Wireless-G multiport IP router (see Figure 2-6). 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.

Figure 2-6 WRTP54G



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

Table 2-10 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 2-11 RTP300 Back Panel

LED	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 2-11 RTP300 Back Panel

LED	Function
Ethernet 1-4	Connect to local IP devices, such as PCs, using an Ethernet cable.
Power	Connect to a 12v DC power supply.

Establishing Connectivity

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

- Bandwidth Requirements, page 2-8
- Making the Physical Connections, page 2-9

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 2-12 illustrates the bandwidth budget using different codecs.

Table 2-12 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 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 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 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.

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 2-10
- IVR Options, page 2-11
- Entering a Password through the IVR, page 2-13

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 2-13 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 2-13 summarizes the options provided by the IVR.

Table 2-13 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 in 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.
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.

Table 2-13 IVR Options (continued)

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.
User Factory Reset of Unit WARNING: ALL “User-Changeable” NON-DEFAULT SETTINGS WILL BE LOST! This might include network and service provider data.	877778	Enter 1 to confirm Enter *(star) to cancel operation	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.
Factory Reset of Unit WARNING: ALL NON-DEFAULT SETTINGS WILL BE LOST! This includes network and service provider data.	73738 (RESET)	Enter 1 to confirm Enter * (star) to cancel operation	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.

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 2-14
- Administrator Account Privileges, page 2-15

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.

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CHAPTER 3

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 a Linksys ATA, page 3-16
- 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.

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 workaround 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 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 3-5

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 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.
- 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 3-1 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, 1xxxxxxxxxx” sounds an “outside line” dial tone after the user presses 9, until the 1 is pressed.

Number Barring

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

Interdigit Timer Master Override

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

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

Thus, “L:8,(. . .)” 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.

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 xxxxxxx )
```

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

Dial Plan Timers

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

- Interdigit_Long_Timer
- 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 *n*
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 3-11
- Using a Mini-Certificate, page 3-11
- Generating a Mini-Certificate, page 3-12

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:

```
9CC9aYU1X5lJuU+EBZmi3AmcqE9U1LxEOGwopaGyGOh3VyhKgi6JaVtQZt87PiJINKW8XQj3B9Qqe3VgYx
WCQNa335YCnDsenASeBxuMIEaBCYd1l1fVEodJZOGwXwfAde0MhcbD0kj7LVlzcsTyk2TZYTccnZ75TuTj
j13qvYs=

5nEtOrkCa84/mEwl3D9tSvVLyliwQ+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>
Sm9lIFNtaXRoAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAxNDA4NTU1MTIzNAAAAAAAMDAwMDAwMDEwMTM0OvJakde2v
VMF3Rw4pPXL7lAgIagMpbLSAG2+++YlSqt198Cp9rP/xMGFfoPmDKGx6JFtkQ5sxLcuwgxpxpxkeXvpZKlYlpsb28L
4Rhg5qZA+Gqj1hDFCmG6dffZ9SJhxES767G0JIS+N8lQBLr0AuemotknSjjjOy8c+1lTCd2t44Mh0vmwNg4fDck2Yd
mTMBR516xJt4/uQ/LJQlni2kwqlm7scDvll5k232EvvvVtCK0AYa4eWd6fQOpiESCO9CC9aYU1X5lJuU+EBZmi3Amc
qE9U1LxEOGwopaGyGOh3VyhKgi6JaVtQZt87PiJINKW8XQj3B9Qqe3VgYxWCQNa335YCnDsenASeBxuMIEaBCYd1l1
fVEodJZOGwXwfAde0MhcbD0kj7LVlzcsTyk2TZYTccnZ75TuTjj13qvYs=
<SRTP Private Key>
b/DWc96X4YQraCnYzl5en1CIUhVQQqrvcr6Qd/8R52IEvJjOw/e+Klm4XiiFEPaKmU8UbooxKG36SEdKusp0AQ==
```

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 3-13
- Using a Streaming Audio Server, page 3-13
- Using the IVR with an SAS Line, page 3-14
- Example SAS with MOH, page 3-14
- SAS Line Not Registered with the Proxy Server, page 3-15

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.

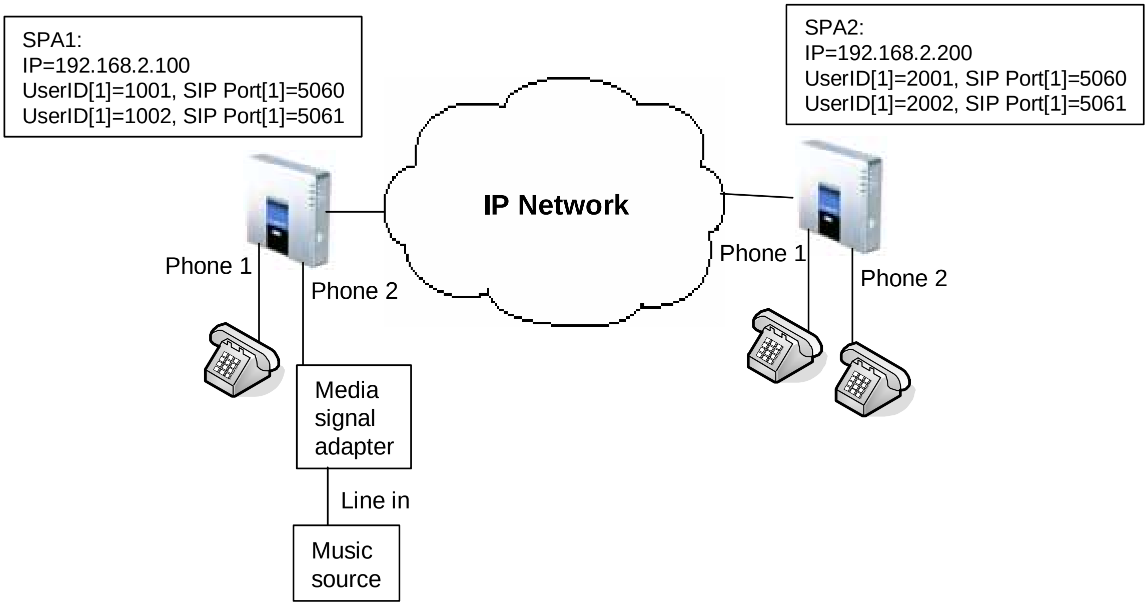
Step 3 Power on the Linksys ATA.

Step 4 Pick up handset and press * * * * to invoke IVR in the usual way.

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 3-1 MOH Application with a Linksys ATA Line Configured as an SAS


SPA1:
IP=192.168.2.100
UserID[1]=1001, SIP Port[1]=5060
UserID[1]=1002, SIP Port[1]=5061
SPA2:
IP=192.168.2.200
UserID[1]=2001, SIP Port[1]=5060
UserID[1]=2002, SIP Port[1]=5061
IP Network
Phone 1
Phone 2
Phone 1
Phone 2
Media signal adapter
Line in
Music source

The following configuration examples are based on the configuration shown in Figure 3-1.

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
```

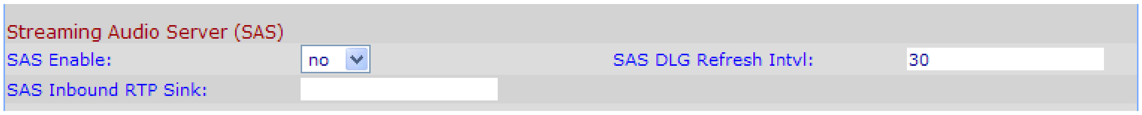
Configuring the Streaming Audio Server

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 3-2).

Figure 3-2 Enabling the Streaming Audio Server



Streaming Audio Server (SAS)
SAS Enable: no
SAS DLG Refresh Intvl: 30
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 a Linksys ATA

Note T.38 Fax is only supported on the SPA2102.

Procedure

To optimize fax completion rates, complete the following steps:

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 Canceller: 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.

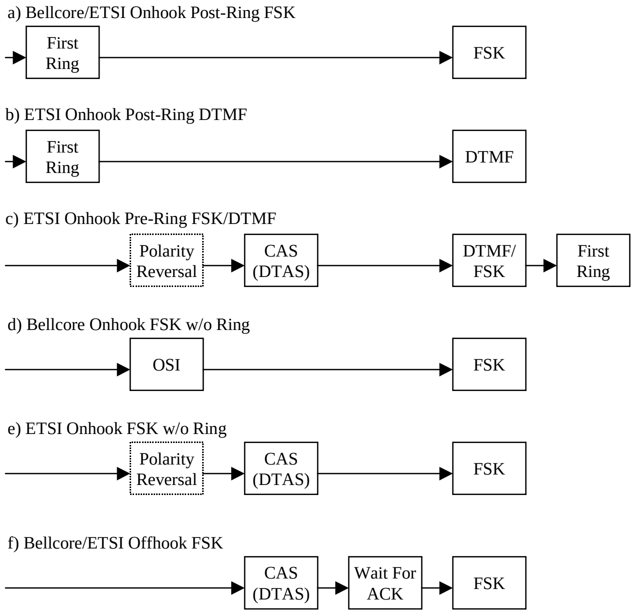
If you are an authorized Linksys Voice System partner, contact Linksys technical support:

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 3-3). 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 3-3 Linksys ATA Caller ID Delivery Architecture


a) Bellcore/ETSI Onhook Post-Ring FSK
First Ring
FSK
b) ETSI Onhook Post-Ring DTMF
First Ring
DTMF
c) ETSI Onhook Pre-Ring FSK/DTMF
Polarity Reversal
CAS (DTAS)
DTMF/ FSK
First Ring
d) Bellcore Onhook FSK w/o Ring
OSI
FSK
e) ETSI Onhook FSK w/o Ring
Polarity Reversal
CAS (DTAS)
FSK
f) Bellcore/ETSI Offhook FSK
CAS (DTAS)
Wait For ACK
FSK

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.

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.

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CHAPTER 4

Configuring the SPA3102 PSTN Gateway

This chapter describes how to configure the PSTN gateway provided by the 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 4-8
- Call Scenarios, page 4-9

Overview

The SPA3102 has the following ports:

- 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

The FXO port lets the SPA3102 act as a SIP-PSTN gateway, which bridges PSTN and VoIP service.

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 SPA3102 via VoIP to obtain PSTN service
- VoIP user—VoIP caller that has a user account (user-id and password) on the SPA3102
- PSTN caller—One who calls the SPA3102 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.

A second VoIP account can be configured in the SPA3102 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 4-1.

Table 4-1 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 4-2 lists the parameters used in two-stage dialing.

Table 4-2 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 4-3 lists parameters for terminating gateway call parameters.

Table 4-3 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 4-4 lists VoIP outbound call routing parameters.

Table 4-4 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 4-5 lists some examples.

Table 4-5 Specifying Gateway Addresses

Example	Description
<9,:>xx.<:@gw1	Dial 9 to start outside dial tone, followed by one or more digits, and route the call to Gateway 1.
[93]11<:@gw0>	Route 911 and 311 calls to the local PSTN gateway
<8,:1408>xxxxxxx<:@pstn.Linksys.com:5061;usr=joe;pwd=joe_pwd;nat>	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
<8,:1408>xxxxxxx<:@gw2:5061;usr=”Alex Bell”;pwd=”anything”;nat=no>	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 4-6.

Table 4-6 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
- 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 4-7 lists symmetric RTP parameters.

Table 4-7 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 4-8 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 4-11
- User Dialing 9 to Access PSTN-Gateway for Local Calls, page 4-12
- 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 than 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 4-12

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 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.

CHAPTER 5

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 5-20
- Line 1/2 Tab, page 5-35
- PSTN Line Tab (SPA3102), page 5-51
- User 1/2 Tab, page 5-68
- PSTN User Tab (SPA3102), page 5-73

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

Info Tab

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

- System Information (PAP2T), page 5-2
- Product Information, page 5-2
- System Status, page 5-3
- Line 1/2 Status, page 5-4
- PSTN Line Status (SPA3102), 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 ATA.
Host Name	Displays the current IP address assigned to the Linksys ATA.
Domain	Displays the network domain name of the Linksys ATA.
Current Netmask	Displays the network mask assigned to the Linksys ATA.
Current Gateway	Displays the default router assigned to the Linksys ATA.
Primary DNS	Displays the primary DNS server assigned to the Linksys ATA.
Secondary DNS	Displays the secondary DNS server assigned to the Linksys ATA.

Product Information

Field	Description
Product Name	Model number of the Linksys ATA.
Serial Number	Serial number of the Linksys ATA.
Software Version	Version number of the Linksys ATA software.
Hardware Version	Version number of the Linksys ATA hardware.
MAC Address	MAC address of the Linksys ATA.

Field	Description
Client Certificate	Status of the client certificate, which authenticates the Linksys ATA for use in the ITSP network. This field indicates if the client certificate is properly installed in the Linksys ATA.
Customization	For an RC unit, this field indicates whether the unit has been customized or not. Pending indicates a new RC unit that is ready for provisioning. If the unit has already retrieved its customized profile, this field displays the name of the company that provisioned the unit.

System Status

Field	Description
Current Time	Current date and time of the system; for example, 10/3/2003 16:43:00.
Elapsed Time	Total time elapsed since the last reboot of the system; for example, 25 days and 18:12:36.
Broadcast Pkts Sent	Total number of broadcast packets sent.
Broadcast Bytes Sent	Total number of broadcast packets received.
Broadcast Pkts Recv	Total number of broadcast bytes sent.
Broadcast Bytes Recv	Total number of broadcast bytes received and processed.
Broadcast Pkts Dropped	Total number of broadcast packets received but not processed.
Broadcast Bytes Dropped	Total number of broadcast bytes received but not processed.
RTP Packets Sent	Total number of RTP packets sent (including redundant packets).
RTP Bytes Sent	Total number of RTP packets received (including redundant packets).
RTP Packets Recv	Total number of RTP bytes sent.
RTP Bytes Recv	Total number of RTP bytes received.
SIP Messages Sent	Total number of SIP messages sent (including retransmissions).
SIP Bytes Sent	Total number of SIP messages received (including retransmissions).
SIP Messages Recv	Total number of bytes of SIP messages sent (including retransmissions).
SIP Bytes Recv	Total number of bytes of SIP messages received (including retransmissions).
External IP	External IP address used for NAT mapping.

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.
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.
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 (SPA3102)

Field	Description
Hook State	Hook state of the FXO port. Either On or Off.
Line Voltage	The voltage existing on the PSTN line.
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: • 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.
Call Type	May take one of the following values: • 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	May take one of the following values: • 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, page 5-8
- Optional Network Configuration, page 5-9
- Miscellaneous Settings, 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 ATA 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 ATA administration web server. The default is 80.
Enable Web Admin Access	Lets you enable or disable local access to the administration web server. Select yes or no from the drop-down menu. The default is yes.
Admin Passwd	Password for the administrator. The default is no password.
User Password	Password for the user. The default is no password.

Internet Connection Type

Field	Description
DHCP	Enable or disable DHCP. The default is yes.
Static IP	Static IP address of Linksys ATA, which takes effect if DHCP is disabled. The default is 0.0.0.0.

Field	Description
NetMask	The NetMask used by Linksys ATA when DHCP is disabled. The default is 255.255.255.0.
Gateway	The default gateway used by Linksys ATA when DHCP is disabled. The default is 0.0.0.0.

Optional Network Configuration

Field	Description
Host Name	The host name of the Linksys ATA.
Domain	The network domain of the Linksys ATA.
Primary DNS	DNS server used by Linksys ATA 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 ATA 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	Perform parallel or sequential DNS Query. With parallel DNS query mode, the Linksys ATA 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 ATA. The default is parallel.
Syslog Server	Specify the syslog server name and port. This feature specifies the server for logging Linksys ATA 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 ATA 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.

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

Miscellaneous Settings

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 5-13
- Response Status Code Handling, page 5-14
- RTP Parameters, page 5-15
- SDP Payload Types, page 5-16
- NAT Support Parameters, page 5-18

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 ATA 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.
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 ATA uses compact SIP headers in outbound SIP messages. If set to no, the Linksys ATA uses normal SIP headers. If inbound SIP requests contain compact headers, Linksys ATA 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 ATA 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 ATA 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.
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 ATA 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 ATA waits for the specified length of time before retrying. If this interval is 0, the Linksys ATA 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 ATA 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 ATA terminates the call. If value is set to 0, the Linksys ATA ignores the limit on ICMP errors. The default is 0.
RTCP Tx Interval	Interval for sending out RTCP sender reports on an active connection. It can range from 0 to 255 seconds. During an active connection, the Linksys ATA 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 ATA-1.0.31(b)). The NTP timestamp used in the SR is a snapshot of the Linksys ATA's local time, not the time reported by an NTP server. If the Linksys ATA 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 ATA web page. The default is 0.
No UDP Checksum	Select yes if you want the Linksys ATA to calculate the UDP header checksum for SIP messages. Otherwise, select no. The default is no.

Stats In BYE	Determines whether the Linksys ATA includes the P-RTP-Stat header or response to a BYE message. The header contains the RTP statistics of the current call. Select yes or no from the drop-down menu. The format of the P-RTP-Stat header is: P-RTP-State: PS=<packets sent>,OS=<octets sent>,PR=<packets received>,OR=<octets received>,PL=<packets lost>,JI=<jitter in ms>,LA=<delay in ms>,DU=<call duration in s>,EN=<encoder>,DE=<decoder>. The default is no.

SDP Payload Types

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

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

Field	Description
NSE Dynamic Payload	NSE dynamic payload type. The valid range is 96-127. The default is 100.
AVT Dynamic Payload	AVT dynamic payload type. The valid range is 96-127. The default is 101.
INFOREQ Dynamic Payload	INFOREQ dynamic payload type. There is no default.
G726r16 Dynamic Payload	G.726-16 dynamic payload type. The valid range is 96-127. The default is 98.
G726r24 Dynamic Payload	G.726-24 dynamic payload type. The valid range is 96-127. The default is 97.
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.
ENCAP RTP Dynamic Payload	The dynamic payload value (96 – 127) used for the encapsulating RTP packets when offering the SDP to loopback packets. This setting is used if the SPA is the offerer of the SDP. Otherwise, the value is decided by the peer. The default value is 112.
RTP-Start-Loopback Dynamic Payload	The dynamic payload value (96 – 127) used by the mirror in the self-generated RTP packets before receiving any RTP packets from the source. This setting is used only when the SPA is acting as the loopback source. Otherwise, the value is decided by the peer. This value must be different from any of the dynamic payload values that might be used by the source and the mirror (including the encaprtp payload type). This is necessary so that the source can easily tell when the mirror has switched from sending self-generated RTP packets to sending loopback packets. The default value is 113.

RTP-Start-Loopback Codec	The actual codec corresponding to RTP-Start-Loopback Dynamic Payload, whose codec name is used in the rtpmap attribute for the for the mirror self-generated RTP audio stream, prior to receiving any RTP packets from the source. The choices are {G711u, G711a, G726-16, G726-24, G726-32, G726-40, G729a, G723}. The default is G711u.
G723 Codec Name	G.723 codec name used in SDP. The default is G723.
EncapRTP Codec Name	EncapRTP codec name for the encapsulating RTP payload type to be used in the rtpmap attribute in the SDP. Both sides must use the same name to interoperate properly. The default is EncapRTP.

NAT Support Parameters

Field	**Description**
Handle VIA received	If you select yes, the Linksys ATA processes the received parameter in the VIA header (this is inserted by the server in a response to anyone of its requests). If you select no, the parameter is ignored. Select yes or no from the drop-down menu. The default is no.
Handle VIA rport	If you select yes, the Linksys ATA processes the rport parameter in the VIA header (this is inserted by the server in a response to anyone of its requests). If you select no, the parameter is ignored. Select yes or no from the drop-down menu. The default is no.
Insert VIA received	Inserts the received parameter into the VIA header of SIP responses if the received-from IP and VIA sent-by IP values differ. Select yes or no from the drop-down menu. The default is no.
Insert VIA rport	Inserts the rport parameter into the VIA header of SIP responses if the received-from IP and VIA sent-by IP values differ. Select yes or no from the drop-down menu. The default is no.

Substitute VIA Addr	Lets you use NAT-mapped IP:port values in the VIA header. Select yes or no from the drop-down menu. The default is no.
Send Resp To Src Port	Sends responses to the request source port instead of the VIA sent-by port. Select yes or no from the drop-down menu. The default is no.
STUN Enable	Enables the use of STUN to discover NAT mapping. Select yes or no from the drop-down menu. The default is no.
STUN Test Enable	If the STUN Enable feature is enabled and a valid STUN server is available, the Linksys ATA 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 ATA detects symmetric NAT or a symmetric firewall, NAT mapping is disabled. The default is no.
STUN Server	IP address or fully-qualified domain name of the STUN server to contact for NAT mapping discovery.
EXT IP	External IP address to substitute for the actual IP address of the Linksys ATA in all outgoing SIP messages. If 0.0.0.0 is specified, no IP address substitution is performed. If this parameter is specified, the Linksys ATA assumes this IP address when generating SIP messages and SDP (if NAT Mapping is enabled for that line). However, the results of STUN and VIA received parameter processing, if available, supersede this statically configured value. The default is 0.0.0.0.
EXT RTP Port Min	External port mapping number of the RTP Port Min. number. If this value is not zero, the RTP port number in all outgoing SIP messages is substituted for the corresponding port value in the external RTP port range. The default is 0.
NAT Keep Alive Intvl	Interval between NAT-mapping keep alive messages. The default is 15.

Regional Tab

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

- Call Progress Tones, page 5-20
- Distinctive Ring Patterns, page 5-22
- Distinctive Call Waiting Tone Patterns, page 5-23
- Distinctive Ring/CWT Pattern Names, page 5-23
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- Control Timer Values (sec), page 5-25
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- Vertical Service Announcement Codes, page 5-30
- Outbound Call Codec Selection Codes, page 5-31
- Miscellaneous, page 5-32

Call Progress Tones

Field	Description
Dial Tone	Prompts the user to enter a phone number. Reorder Tone is played automatically when <Dial Tone> or any of its alternatives times out. The default is 350@-19,440@-19;10(*/0/1+2).
Second Dial Tone	Alternative to the Dial Tone when the user dials a three-way call. The default is 420@-19,520@-19;10(*/0/1+2).
Outside Dial Tone	Alternative to the Dial Tone. It prompts the user to enter an external phone number, as opposed to an internal extension. It is triggered by a, (comma) character encountered in the dial plan. The default is 420@-19;10(*/0/1).
Prompt Tone	Prompts the user to enter a call forwarding phone number. The default is 520@-19,620@-19;10(*/0/1+2).
Busy Tone	Played when a 486 RSC is received for an outbound call. The default is 480@-19,620@-19;10(.5/.5/1+2).
Reorder Tone	Played when an outbound call has failed or after the far end hangs up during an established call. Reorder Tone is played automatically when <Dial Tone> or any of its alternatives times out. The default is 480@-19,620@-19;10(.25/.25/1+2).

Off Hook Warning Tone	Played when the caller has not properly placed the handset on the cradle. Off Hook Warning Tone is played when Reorder Tone times out. The default is 480@10,620@0;10(.125/.125/1+2).
Ring Back Tone	Played during an outbound call when the far end is ringing. The default is 440@-19,480@-19;*(2/4/1+2).
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).
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 ATA 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 ATA still considers the call as failed and keeps on retrying. The default is 0.5.
VMWI Refresh Intvl	Interval between VMWI refresh to the CPE. The default is 0.5.
Interdigit Long Timer	Long timeout between entering digits when dialing. The interdigit timer values are used as defaults when dialing. The Interdigit_Long_Timer is used after any one digit, if all valid matching sequences in the dial plan are incomplete as dialed. Range: 0–64 seconds. The default is 10.
Interdigit Short Timer	Short timeout between entering digits when dialing. The Interdigit_Short_Timer is used after any one digit, if at least one matching sequence is complete as dialed, but more dialed digits would match other as yet incomplete sequences. Range: 0–64 seconds. The default is 3.

CPC Delay	Delay in seconds after caller hangs up when the Linksys ATA starts removing the tip-and-ring voltage to the attached equipment of the called party. Range: 0–255 seconds. Linksys ATA 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.
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.
CW Deact Code	Disables call waiting on all calls. The default is *57.
CW Per Call Act Code	Enables call waiting for the next call. The default is *71.

CW Per Call Deact Code	Disables call waiting for the next call. The default is *70.
Block CID Act Code	Blocks caller ID on all outbound calls. The default is *67.
Block CID Deact Code	Removes caller ID blocking on all outbound calls. The default is *68.
Block CID Per Call Act Code	Blocks caller ID on the next outbound call. The default is *81.
Block CID Per Call Deact Code	Removes caller ID blocking on the next inbound call. The default is *82.
Block ANC Act Code	Blocks all anonymous calls. The default is *77.
Block ANC Deact Code	Removes blocking of all anonymous calls. The default is *87.
DND Act Code	Enables the do not disturb feature. The default is *78.
DND Deact Code	Disables the do not disturb feature. The default is *79.
CID Act Code	Enables caller ID generation. The default is *65.
CID Deact Code	Disables caller ID generation. The default is *85.
CWCID Act Code	Enables call waiting, caller ID generation. The default is *25.
CWCID Deact Code	Disables call waiting, caller ID generation. The default is *45.
Dist Ring Act Code	Enables the distinctive ringing feature. The default is *26
Dist Ring Deact Code	Disables the distinctive ringing feature. The default is *46.
Speed Dial Act Code	Assigns a speed dial number. The default is *74.

Secure All Call Act Code	Makes all outbound calls secure. The default is *16.
Secure No Call Act Code	Makes all outbound calls not secure. The default is *17.
Secure One Call Act Code	Makes the next outbound call secure. (It is redundant if all outbound calls are secure by default.) The default is *18.
Secure One Call Deact Code	Makes the next outbound call not secure. (It is redundant if all outbound calls are not secure by default.) The default is *19.
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	These codes tell the Linksys ATA what to do when the user places the current call on hold and is listening to the second dial tone. One or more *code can be configured into this parameter, such as *98, or *97|*98|*123, etc. Max total length is 79 chars. This parameter applies when the user places the current call on hold (by Hook Flash) and is listening to second dial tone. Each *code (and the following valid target number according to current dial plan) entered on the second dial-tone triggers the Linksys ATA to perform a blind transfer to a target number that is prepended by the service *code. For example, after the user dials *98, the Linksys ATA plays a special dial tone called the Prompt Tone while waiting for the user the enter a target number (which is checked according to dial plan as in normal dialing). When a complete number is entered, the Linksys ATA sends a blind REFER to the holding party with the Refer-To target equals to *98<target_number>. This feature allows the Linksys ATA to hand off a call to an application server to perform further processing, such as call park. The *codes should not conflict with any of the other vertical service codes internally processed by the Linksys ATA. You can empty the corresponding *code that you do not want to Linksys ATA to process.

Feature Dial Services Codes	These codes tell the Linksys ATA what to do when the user is listening to the first or second dial tone. One or more *code can be configured into this parameter, such as *72, or *72|*74|*67|*82, etc. Max total length is 79 chars. This parameter applies when the user has a dial tone (first or second dial tone). Enter *code (and the following target number according to current dial plan) entered at the dial tone triggers the Linksys ATA to call the target number prepended by the *code. For example, after user dials *72, the Linksys ATA plays a prompt tone awaiting the user to enter a valid target number. When a complete number is entered, the Linksys ATA sends a INVITE to *72<target_number> as in a normal call. This feature allows the proxy to process features like call forward (*72) or BLock Caller ID (*67). The *codes should not conflict with any of the other vertical service codes internally processed by the Linksys ATA. You can empty the corresponding *code that you do not want to Linksys ATA to process. You can add a parameter to each *code in Features Dial Services Codes to indicate what tone to play after the *code is entered, such as *72‘c‘|*67‘p‘. Below are a list of allowed tone parameters (note the use of back quotes surrounding the parameter w/o spaces) • ‘c‘ = <Cfwd Dial Tone> • ‘d‘ = <Dial Tone> • ‘m‘ = <MWI Dial Tone> • ‘o‘ = <Outside Dial Tone> • ‘p‘ = <Prompt Dial Tone> • ‘s‘ = <Second Dial Tone> • ‘x‘ = No tones are place, x is any digit not used above If no tone parameter is specified, the Linksys ATA plays Prompt tone by default. If the *code is not to be followed by a phone number, such as *73 to cancel call forwarding, do not include it in this parameter. In that case, simple add that *code in the dial plan and the Linksys ATA send INVITE *73@..... as usual when user dials *73.

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	Enter the rule for calculating daylight saving time; it should include the start, end, and save values. This rule is comprised of three fields. Each field is separated by ; (a semicolon) as shown below. Optional values inside [ ] (the brackets) are assumed to be 0 if they are not specified. Midnight is represented by 0:0:0 of the given date. This is the format of the rule: Start = <start-time>; end=<end-time>; save = <save-time>. • The <start-time> and <end-time> values specify the start and end dates and times of daylight saving time. Each value is in this format: <month> /<day> / <weekday>[/HH:[mm[:ss]]] • The <save-time> value is the number of hours, minutes, and/or seconds to add to the current time during daylight saving time. The <save-time> value can be preceded by a negative (-) sign if subtraction is desired instead of addition. The <save-time> value is in this format: [/[+|-]HH:[mm[:ss]]] • The <month> value equals any value in the range 1-12 (January-December). • The <day> value equals [+|-] any value in the range 1-31. • If <day> is 1, it means the <weekday> on or before the end of the month (in other words the last occurrence of < weekday> in that month). • The <weekday> value equals any value in the range 1-7 (Monday-Sunday). It can also equal 0. If the <weekday> value is 0, this means that the date to start or end daylight saving is exactly the date given. In that case, the <day> value must not be negative. If the <weekday> value is not 0 and the <day> value is positive, then daylight saving starts or ends on the <weekday> value on or after the date given. If the <weekday> value is not 0 and the <day> value is negative, then daylight saving starts or ends on the <weekday> value on or before the date given. • The abbreviation HH stands for hours (0-23). • The abbreviation mm stands for minutes (0-59). • The abbreviation ss stands for seconds (0-59). The default Daylight Saving Time Rule is start=4/1/7;end=10/-1/7;save=1.
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: • 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 ATA 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.
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 suppresion. The default is no.

Line 1/2 Tab

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

- Line Enable, page 5-35
- Streaming Audio Server (SAS), page 5-35
- NAT Settings, page 5-36
- Network Settings, page 5-37
- SIP Settings, page 5-37
- Call Feature Settings, page 5-40
- Proxy and Registration, page 5-40
- Subscriber Information, page 5-42
- Supplementary Service Subscription, page 5-42
- Audio Configuration, page 5-45
- VoIP Fallback to PSTN (SPA3102), page 5-48
- Gateway Accounts (SPA3102), page 5-47
- Dial Plan, page 5-48
- FXS Port Polarity Configuration, page 5-49

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

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
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 ATA 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 ATA'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	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. (Not in PAP2T) 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. (Not in PAP2T) The default is 6.
Network Jitter Level	Determines how jitter buffer size is adjusted by the Linksys ATA. 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 ATA 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 ATA 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	SIP messages are received at or sent from the proxy listen port. This feature controls which SIP messages to log. Choices are as follows: • none—No logging. • 1-line—Logs the start-line only for all messages. • 1-line excl. OPT—Logs the start-line only for all messages except OPTIONS requests/responses. • 1-line excl. NTFY—Logs the start-line only for all messages except NOTIFY requests/responses. • 1-line excl. REG—Logs the start-line only for all messages except REGISTER requests/responses. • 1-line excl. OPT|NTFY|REG—Logs the start-line only for all messages except OPTIONS, NOTIFY, and REGISTER requests/responses. • full—Logs all SIP messages in full text. • full excl. OPT—Logs all SIP messages in full text except OPTIONS requests/responses. • full excl. NTFY—Logs all SIP messages in full text except NOTIFY requests/responses. • full excl. REG—Logs all SIP messages in full text except REGISTER requests/responses. • full excl. OPT|NTFY|REG—Logs all SIP messages in full text except for OPTIONS, NOTIFY, and REGISTER requests/responses. The default is none.
RTP Log Intvl	The interval for the RTP log.
Restrict Source IP	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). The default is no.
Referor Bye Delay	Controls when the Linksys ATA 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. The default is 4.
Refer Target Bye Delay	For the Refer Target Bye Delay, enter the appropriate period of time in seconds. The default is 0.

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

Call Feature Settings

Field	Description
Blind Attn-Xfer Enable	Enables the Linksys ATA 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 ATA 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 ATA perform a transfer when a conference call has ended. Select yes or no from the drop-down menu. The default is yes.

Proxy and Registration

Field	Description
Proxy	SIP proxy server for all outbound requests.
Use Outbound Proxy	Enable the use of <Outbound Proxy>. If set to no, <Outbound Proxy> and <Use OB Proxy in Dialog) is ignored. The default is no.
Outbound Proxy	SIP Outbound Proxy Server where all outbound requests are sent as the first hop.

Field	Description
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 1st proxy's port on the list. The default is Normal.

Subscriber Information

Field	Description
Display Name	Display name for caller ID.
User ID	Extension number for this line.
Password	Password for this line.
Use Auth ID	To use the authentication ID and password for SIP authentication, select yes. Otherwise, select no to use the user ID and password. The default is no.
Auth ID	Authentication ID for SIP authentication.
Call Capacity	Maximum number of calls allowed on this line interface. Choices: {unlimited,1,2,3,…25 }. Default is 16. Note that the Linksys ATA 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 ATA 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 ATA.

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.

Field	Description
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)
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.

Field	Description
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.

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.
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 ATA 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: • 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.

Gateway Accounts (SPA3102)

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.

Field	Description
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)

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|xxxxxxxxxxxx.). The syntax for a dial plan expression is as follows:

Table 5-1 Dial Plan Script Syntax

Dial Plan Entry	Functionality
*xx	Allow arbitrary 2 digit star code
[3469]11	Allow x11 sequences
0	Operator
00	Int’l Operator
[2-9]xxxxxx	US local number
1xxx[2-9]xxxxxx	US 1 + 10-digit long distance number
xxxxxxxxxxxx.	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	Dial plan script for this line.
	The default(*xx|[3469]11|0|00|[2-9]xxxxxx|1xxx[2-9]xxxxxxS0|xxxxxxxxxxxx.)
	The dial plan syntax is expanded in the SPA3102 to allow the designation of three parameters to be used with a specific gateway:
	• uid – the authentication user-id
	• pwd – the authentication password
	• nat – if this parameter is present, use NAT mapping
	Each parameter is separated by a semi-colon (;).
	Furthermore, it recognizes gw0, gw1, …, gw4 as the locally configured gateways, where gw0 represents the local PSTN gateway in the same SPA3102 unit.
	Example 1:
	`*1xxxxxxxxxx<:@fwdnat.pulver.com:5082;uid=jsmith;pwd=xyz`
	Example 2:
	`*1xxxxxxxxxx<:@fwd.pulver.com;nat;uid=jsmith;pwd=xyz`
	Example 3:
	`[39]11<:@gw0>`

Enable IP Dialing	Enable or disable IP dialing. The default is no.
Emergency Number	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. The default is blank.

FXS Port Polarity Configuration

Field	Description
Idle Polarity	Polarity before a call is connected: Forward or Reverse. The default is Forward.
Caller Conn Polarity	Polarity after an outbound call is connected: Forward or Reverse. The default is Forward.

Callee Conn Polarity	Polarity after an inbound call is connected: Forward or Reverse. The default is Forward.

PSTN Line Tab (SPA3102)

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

- Line Enable, page 5-35
- NAT Settings, page 5-51
- Network Settings, page 5-52
- SIP Settings, page 5-53
- Proxy and Registration, page 5-55
- Subscriber Information, page 5-56
- Audio Configuration, page 5-57
- Dial Plans, page 5-60
- VoIP-To-PSTN Gateway Setup, page 5-60
- VoIP Users and Passwords (HTTP Authentication), page 5-61
- PSTN-to-VoIP Gateway Setup, page 5-62
- FXO Timer Values (sec), page 5-64
- PSTN Disconnect Detection, page 5-65
- International Control, page 5-66

Line Enable

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

NAT Settings

Field	Description
NAT Mapping Enable	To use externally mapped IP addresses and SIP/RTP ports in SIP messages, select yes. Otherwise, select no. The default is no.
NAT Keep Alive Enable	To send the configured NAT keep alive message periodically, select yes. Otherwise, select no. The default is no.

NAT Keep Alive Msg	Enter the keep alive message that should be sent periodically to maintain the current NAT mapping. If the value is $NOTIFY, a NOTIFY message is sent. If the value is $REGISTER, a REGISTER message without contact is sent. The default is $NOTIFY.
NAT Keep Alive Dest	Destination that should receive NAT keep alive messages. If the value is $PROXY, the messages are sent to the current or outbound proxy. The default is $PROXY.

Network Settings

Field	**Description**
SIP ToS/DiffServ Value	TOS/DiffServ field value in UDP IP packets carrying a SIP message. The default is 0x68.
SIP CoS Value [0-7]	CoS value for SIP messages. (Not in PAP2T) 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. (Not in PAP2T) The default is 6.
Network Jitter Level	Determines how jitter buffer size is adjusted by the Linksys ATA. 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 ATA 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 ATA 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	SIP messages are received at or sent from the proxy listen port. This feature controls which SIP messages to log. Choices are as follows: • none—No logging. • 1-line—Logs the start-line only for all messages. • 1-line excl. OPT—Logs the start-line only for all messages except OPTIONS requests/responses. • 1-line excl. NTFY—Logs the start-line only for all messages except NOTIFY requests/responses. • 1-line excl. REG—Logs the start-line only for all messages except REGISTER requests/responses. • 1-line excl. OPT|NTFY|REG—Logs the start-line only for all messages except OPTIONS, NOTIFY, and REGISTER requests/responses. • full—Logs all SIP messages in full text. • full excl. OPT—Logs all SIP messages in full text except OPTIONS requests/responses. • full excl. NTFY—Logs all SIP messages in full text except NOTIFY requests/responses. • full excl. REG—Logs all SIP messages in full text except REGISTER requests/responses. • full excl. OPT|NTFY|REG—Logs all SIP messages in full text except for OPTIONS, NOTIFY, and REGISTER requests/responses. The default is none.
RTP Log Intvl	The interval for the RTP log.
Restrict Source IP	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). The default is no.
Referor Bye Delay	Controls when the Linksys ATA 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. The default is 4.
Refer Target Bye Delay	For the Refer Target Bye Delay, enter the appropriate period of time in seconds. The default is 0.

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

Proxy and Registration

Field	Description
Proxy	SIP proxy server for all outbound requests.
Use Outbound Proxy	Enable the use of <Outbound Proxy>. If set to no, <Outbound Proxy> and <Use OB Proxy in Dialog) is ignored. The default is no.
Outbound Proxy	SIP Outbound Proxy Server where all outbound requests are sent as the first hop.
Use OB Proxy In Dialog	Whether to force SIP requests to be sent to the outbound proxy within a dialog. Ignored if <Use Outbound Proxy> is no or <Outbound Proxy> is empty. The default is yes.
Register	Enable periodic registration with the <Proxy>. This parameter is ignored if <Proxy> is not specified. The default is yes.
Make Call Without Reg	Allow making outbound calls without successful (dynamic) registration by the unit. If No, dial tone will not play unless registration is successful. The default is no.
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

Field	**Description**
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 1st proxy's port on the list. The default is Normal.

Subscriber Information

Field	**Description**
Display Name	Display name for caller ID.
User ID	Extension number for this line.
Password	Password for this line.
Use Auth ID	To use the authentication ID and password for SIP authentication, select yes. Otherwise, select no to use the user ID and password. The default is no.
Auth ID	Authentication ID for SIP authentication.
Call Capacity	Maximum number of calls allowed on this line interface. Choices: {unlimited,1,2,3,…25 }. Default is 16. Note that the Linksys ATA 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.

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 G729a 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 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.
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 ATA 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: • 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.

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.
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	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}. The default is 1.
VoIP Caller ID Pattern	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. For example: 1408*, 1512???1234. Note: ‘?’ matches any single digit; ‘*’ matches any number of digits. The default is blank.
VoIP Access List	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??. The default is blank.
VoIP Caller 1/2/3/4/5/6/7/8 PIN	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. The default is blank.
VoIP Caller 1/2/3/4/5/6/7/8 DP	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>. The default is 1.

VoIP Users and Passwords (HTTP Authentication)

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.

Field	Description
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.

PSTN-to-VoIP Gateway Setup

Field	Description
PSTN-To-VoIP Gateway Enable	Enable or disable PSTN-To-VoIP Gateway functionality. If set to no, gateway is disabled but PSTN calls still ring through Line 1 (if <PSTN Ring Thru Line 1> is enabled). The default is yes.
PSTN CID For VoIP CID	If set to yes, the outbound VoIP call will assume the caller-id of the PSTN caller, if PSTN caller ID is available. Otherwise, the PSTN Line’s VoIP account information is used. The PSTN Caller ID is after the application of <PSTN CID Name Prefix> and <PSTN CID Number Prefix>. The default is yes.
PSTN CID Number Prefix	A prefix to prepend to the PSTN caller ID number when ringing through Line 1 or used in outbound VoIP call. Note that most caller-id devices can only display 0-9. The default is blank.
PSTN CID Name Prefix	A prefix to prepend to the PSTN caller ID name when ringing through Line 1 or used in outbound VoIP call. The default is blank.
PSTN Caller Auth Method	Method to be used to authenticate a PSTN Caller for VoIP gateway services. Choose from {none, PIN}. The default is none.
PSTN PIN Max Retry	Number of trials to allow a PSTN Caller to enter a valid PIN number. The default is 3.

Field	Description
PSTN Ring Thru Line 1	If enabled, incoming calls will also ring Line 1 (after a delay as specified in <PSTN Ring Thru Delay>). Hence the Line 1 user can accept call waiting from the PSTN side. The caller-id from the PSTN service, if available, will be passed onto Line 1 also when ringing through (For this to work, the ring through delay must be set long enough such that PSTN caller-id is completely decoded before ring thru Line 1 starts). The default is yes.
PSTN Caller Default DP	Index of the dial plan in the dial plan pool to be used for the PSTN caller who does not require authentication (when authentication method is none, or when the caller's number is in the access list). Choose from {1, 2, 3, 4, 5, 6, 7, 8}. The default is 1.
PSTN Caller ID Pattern	A comma separated list of caller number templates such that PSTN callers with numbers not matching any of these templates will be rejected for VoIP 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 VoIP gateway service. The PSTN service must include Type I Caller-ID Delivery Service for this feature to work properly. If caller-id is blocked or not available, the caller-id is assumed to be Anonymous. For example: 1408*, 1512???1234, Anonymous The default is blank.
PSTN Access List	A comma separated list of caller number templates such that PSTN callers with numbers matching any of these templates will be accepted for VoIP gateway service without authentication. The default is blank.
PSTN Caller 1/2/3/4/5/6/7/8 PIN	One of 8 PIN numbers for authenticating PSTN callers to obtain VoIP gateway services. The PSTN Caller entering a PIN same as this PIN is referred as PSTN Caller 1/2/3/4/5/6/7/8. The default is blank.
PSTN Caller 1 DP	Index of the dial plan in the dial plan pool to be used with PSTN Caller 1. Choose from {1, 2, 3, 4, 5, 6, 7, 8}. The default is 1.

FXO 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.
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 1st 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 1st 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.

Field	Description
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 *on-time/off-time*, where *on-time* and *off-time* 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 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 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 2nd one triggers the disconnection. For VoIP gateway call, the 1st polarity reversal triggers the disconnection. The default is yes.
Detect 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.
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
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. The following restrictions apply: • Two 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 one segment set is allowed. • Total duration of the segment set is interpreted as the minimum duration of the tone to trigger detection. • Six 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

Field	Description
FXO Port Impedance	Desired impedance of the FXO Port. Choose from {600, 900, 270+. The default is 600.
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.
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.
Tip/Ring Voltage Adjust	Choices are {3.1, 3.2, 3.35, 3.5}. The default is 3.5.

Field	Description
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 5-68
- Selective Call Forward Settings, page 5-69
- Speed Dial Settings, page 5-69
- Supplementary Service Settings, page 5-69
- Distinctive Ring Settings, page 5-71
- Ring Settings, page 5-71

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

Call Forward Settings

Field	Description
Cfwd All Dest	Forward number for Call Forward All Service In addition to normal call forward destination as used in the other ATAs, on the SPA3102, you can specify the following additional parameters: • 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) The default is blank.
Cfwd Busy Dest	Forward number for Call Forward Busy Service. Same as Cfwd All Dest. The default is blank.
Cfwd No Ans Dest	Forward number for Call Forward No Answer Service. Same as Cfwd All Dest. In addition to normal call forward destination as used in the other ATAs, on the SPA3102, you can specify the following additional parameters: • 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) The default is blank.
Cfwd No Ans Delay	Delay in sec before Call Forward No Answer triggers. Same as Cfwd All Dest. The default is 20.

Selective Call Forward Settings

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

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 ATA 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 ATA.

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 ATA. 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.
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.
VMWI Ring Splash Len	Duration of ring splash when new messages arrive before the VMWI signal is applied (0 – 10.0s). The default is .5.

Field	Description
VMWI Ring Policy	The parameter controls when a ring splash is played when a the VM server sends a SIP NOTIFY message to the Linksys ATA indicating the status of the subscriber's mail box. 3 settings are available: New VM Available – ring as long as there is 1 or more unread voicemail New VM Becomes Available – ring when the number of unread voicemail changes from 0 to non-zero New VM Arrives – ring when the number of unread voicemail increases. The default is New VM Available.
Ring On No New VM	If enabled, the Linksys ATA will play a ring splash when the VM server sends SIP NOTIFY message to the Linksys ATA indicating that there are no more unread voicemails. Some equipment requires a short ring to precede the FSK signal to turn off VMWI lamp. The default is no.

PSTN User Tab (SPA3102)

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 5-73
- PSTN-To-VoIP Speed Dial Settings, page 5-73
- PSTN Ring Thru Line 1 Distinctive Ring Settings, page 5-73
- PSTN Ring Thru Line 1 Ring Settings, page 5-74
- PSTN/VoIP Caller Commands via DTMF, page 5-74

PSTN-To-VoIP Selective Call Forward Settings

Field	Description
Cfwd Sel1 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 Sel1 Dest	VoIP destination to forward a PSTN caller matching <Cfwd Sel1 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.

APPENDIX A

Acronyms

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

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

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

APPENDIX B

Glossary

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

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

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

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

Bundled service—Offering various services as a complete package.

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

SS7 (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.

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APPENDIX C

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	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	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 You must repeat this process at the start of each call.
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	If you choose to answer the second call, do one of the following: • 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) This puts your first call on hold and automatically connects you to your second call. 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.)
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	To temporarily disable Call Waiting (for the length of one call) do the following before placing a call: 1. Lift Receiver 2. Press *70 3. Listen for dial tone, then dial the number you want to call. Call Waiting is now disabled for the duration of this call only. To deactivate Call Waiting while on a call: 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. Call Waiting is now disabled for the duration of this call. To deactivate Call Waiting while on a permanent basis (until cancelled): 1. Lift the receiver 2. Listen for dial tone 3. Press *57 You will hear a confirmation tone signaling your request to cancel Call Waiting has been accepted.
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	If you have cancelled Call Waiting temporarily, no user action is required. If you deactivated call waiting and wish to reinstate the service, do the following: 1. Lift the receiver 2. Listen for dial tone 3. Press *57 You will hear a confirmation tone signaling your request to cancel Call Waiting has been accepted.

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	The SPA indicates that a message is waiting by playing stuttered dial tone when the user picks up the handset. To retrieve messages: 1. Lift the receiver 2. Listen for dial tone 3. Dial the phone number assigned to the SPA You will be connected to the voice mail server and prompted by a voice response system with instructions to listen to your messages.
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	While in a call with the party to be transferred: **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 **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.
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	While in a call with the party to be transferred: **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 The call is transferred when a complete number is entered. You will hear a short confirmation tone, followed by regular dial tone

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	**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. To hold a conference with the party on hold and the third party, simply press the switch hook or flash button.
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	If you are already on a call and wish to add a third party: 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 You now have the original caller and the third party together with you on the same call. If you want to initiate a new three-way call: 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	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). If you also have Call Transfer, you can also hang up at any time to transfer the original caller to the third party
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	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	1. Pick up the receiver 2. Listen for dial tone 3. Press *07
Expected call and network behavior	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. 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.
User action required to deactivate or end	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	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. Activation is confirmed with three short bursts of tone and your forwarding is activated. Alternatively, the user can activate this feature from a web browser interface.

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	1. Lift the receiver 2. Listen for dial tone 3. Press *73 You will hear a confirmation tone signaling your change has been accepted. Alternatively, the user can deactivate this feature from a web browser interface.

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	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. Activation is confirmed with three short bursts of tone and your forwarding is activated. Alternatively, the user can activate this feature from a web browser interface.
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	1. Lift the receiver 2. Listen for dial tone 3. Press *91 You will hear a confirmation tone signaling your change has been accepted. Alternatively, the user can deactivate this feature from a web browser interface.

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	1. Lift the receiver 2. Listen for dial tone 3. Press *92Listen for dial tone and enter the telephone number you are forwarding your call to. Activation is confirmed with three short bursts of tone and your forwarding is activated. Alternatively, the user can activate this feature from a web browser interface. **Note** The forward delay is entered from the web interface. Default is 20s.
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	1. Lift the receiver 2. Listen for dial tone 3. Press *93 You will hear a confirmation tone signaling your change has been accepted. Alternatively, the user can deactivate this feature from a web browser interface.

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	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	**1.** Pick up the receiver **2.** Listen for dial tone **3.** Press *26
Expected call and network behavior	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. 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.

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	**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 **Note** To enter IP addresses, a graphical user interface such as a web browser must be used.

Expected call and network behavior	1. Pick up the receiver 2. Listen for dial tone 3. 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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