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Getting Started

Thank you for choosing the Cisco SPA100 Series Phone Adapters. This chapter provides more information about the features of the product and the web-based configuration utility.

Feature Overview

Cisco SPA100 Series Analog Telephone Adapters (ATAs) provide your standard analog phones with access to Internet phone services through two standard telephone RJ-11 phone ports. The ATA connects to the Internet through a broadband (DSL or cable) modem or router. The ATA can be used with an on-site call-control system or an Internet-based call-control system.

The 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. An ATA maintains the state of each call it terminates and reacts appropriately to user input events (such as on/off hook or hook flash). The ATAs use the Session Initiation Protocol (SIP) open standard so there is little or no involvement by a “middle-man” server or media gateway controller. SIP allows inter-operation with all ITSPs that support SIP.

The system supports four simultaneous calls, including “active” calls and “on-hold” calls. A phone can handle one on-hold call and one active call simultaneously.

Understanding Voice Service Operations

The ATA allows calls to be made by using SIP-based Voice-over-IP (VoIP) services and traditional telephone Public Switched Telephone Network (PSTN) services. Calls can be placed and received by using an analog phone or fax machine.

The ATA maintains the state of each call and makes the proper reaction to user input events (such as on/off hook or hook flash). Because the ATA uses the Session Initiation Protocol (SIP), it is compatible with most Internet Telephony Service Provider (ITSP) offerings.

ATA Voice Features

The ATA can be custom provisioned within a wide range of configuration parameters. The following sections describe the factors that contribute to voice quality:

- **Supported Codecs**
- **SIP Proxy Redundancy**
- **Other ATA Voice Features**

Supported Codecs

The ATA supports the codecs listed below. You can use the default settings or configure the codec settings in the *Audio Configuration* section of the **Line 1 and Line 2 Settings (PHONE Port1 and PHONE2)** page.

Codec	Description
G.711 (A-law and mu-law)	Very low complexity codecs that support uncompressed 64 kbps digitized voice transmissions at one through ten 5 ms voice frames per packet. These codecs provide the highest narrow-band voice quality and uses the most bandwidth of any of the available codecs.
G.726-32	Low complexity codec that supports compressed 32 kbps digitized voice transmission at one through ten 10 ms voice frames per packet. This codec provides high voice quality.
G.729a	ITU G.729 voice coding algorithm used to compress digitized speech. G.729a is a reduced complexity version of G.729 requiring about half the processing power of G.729. The G.729 and G.729a bit streams are compatible and interoperable, but not identical.

SIP Proxy Redundancy

In typical commercial IP Telephony deployments, all calls are established through a SIP proxy server. A typical SIP proxy server can handle 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 ATA supports the use of backup SIP proxy servers (through DNS SRV) so that service disruption is minimized.

An easy way to support proxy redundancy is to configure your DNS server with a list of SIP proxy addresses. The ATA can be instructed to contact a SIP proxy server in a domain named in the SIP message. The ATA consults the DNS server to get a list of hosts in the given domain that provide 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 ATA tries to contact the list of hosts in the order of their stated priority.

If the ATA is currently using a lower priority proxy server, it periodically probes the higher priority proxy to see whether it is online, and switches back to the higher priority proxy when possible. You can use the default settings or configure the Proxy Redundancy Method in the *Proxy and Registration* section of the **Line 1 Settings (PHONE Port)** page

Other ATA Voice Features

- **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 average bandwidth required for a single call. VAD distinguishes between speech and non-speech signals, and Silence Suppression removes the natural silences that occur in a conversation. Therefore the IP bandwidth is used only to transmit speech. Comfort Noise Generation provides artificially-generated background white noise (sounds) to reassure callers that their calls are still connected during the silent periods. You can enable this feature in the *Audio Configuration* section of the **Line 1 and Line 2 Settings (PHONE Port1 and PHONE2)** page.

- **Modem and Fax Pass-Through**

  - Modem pass-through mode can be triggered by predialing the Vertical Service Activation Code for the Modem Line Toggle Code. You can configure this setting in the *Vertical Service Activation Codes* section of the **Regional** page.

  - FAX pass-through mode is triggered by the detection of a CED/CNG tone or an NSE event.

- Echo canceller is automatically disabled for Modem passthrough mode.
- Echo canceller is disabled for FAX pass-through if the parameter FAX Disable ECAN (Line 1 or 2 tab) 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 are automatically disabled for both FAX and Modem pass-through. In addition, out-of-band DTMF transmission is disabled during modem or fax passthrough.

- **Adaptive Jitter Buffer**
The ATA can buffer incoming voice packets to minimize the impact of variable network delays. This process is known as jitter buffering. The size of the jitter buffer adjusts to changing network conditions. The ATA has a Network Jitter Level control setting for each line of service. The jitter level determines how aggressively the 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. You can use the default settings or configure this feature in the *Network Settings* section of the **Line 1 and Line 2 Settings (PHONE Port1 and PHONE2)** page.

- **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. You can configure this setting in the *RTP Parameters* section of the **SIP** page.

- **DTMF Relay**
The 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. You can configure this setting in the *RTP Parameters* section of the **SIP** page.

- **Call Progress Tones**
The ATA has configurable call progress tones. Call progress tones are generated locally on the ATA so that an end user is advised of status (such as ringback) Parameters for each type of tone (for instance a dial tone played back to an end user) may include frequency and amplitude of each component, and cadence information. You can keep the default settings or configure these tones in the *Call Progress Tones* section of the **Regional** page.

- **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.

- **Echo Cancellation**
Impedance mismatch between the telephone and the IP Telephony gateway phone port can lead to near-end echo. The ATA has a near-end echo canceller that compensates for impedance mismatch. The ATA also implements an echo suppressor with Comfort Noise Generator (CNG) so that any residual echo is not noticeable. This feature is enabled by default. You can configure this setting in the *Audio Configuration* of the **Line 1 and Line 2 Settings (PHONE Port1 and PHONE2)** page.

- **Hook Flash Events**
The ATA can signal hook flash events to the proxy during 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, you may need to disable Call Waiting Service, Three Way Conference Service, or Three Way Call Service to correctly signal a hook flash event to the softswitch. You can configure these settings in the *Supplementary Service Subscription* section of the **Line 1 and Line 2 Settings (PHONE Port1 and PHONE2)** page.

  - You can configure the length of time allowed for detection of a hook flash by adjusting the Hook Flash Timer parameter in the *Control Timer Values* section of the **SIP** page.

- **Configurable Dial Plan with Interdigit Timers**
The ATA has three configurable interdigit timers: an initial timeout signaling that a phone is taken off hook, a long timeout signaling the end of a dialed string, and a short timeout, signaling that more digits are expected. For more information, see **Configuring Dial Plans, page 138**.

- **Polarity Control**
The 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. You can configure these settings in the *FXS Port Polarity Configuration* section of the **Line 1 and Line 2 Settings (PHONE Port1 and PHONE2)** page.

- **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 momentarily removing the voltage between the tip and the ring. This feature is useful for

auto-answer equipment. You can configure these settings in the *Control Timer Values* section of the **Regional** page.

- **Event Logging**
  You can enable logging and select the relative priority of events to be logged. The information can be sent to a Syslog Server. You can configure the syslog and debug settings in the *Miscellaneous Settings* section of the **System** page.

- **Encryption of SIP messages using SIP over TLS**
  You can enable SIP over Transport Layer Security (TLS) to encrypt the SIP messages between the service provider and the your business. SIP over TLS relies on the widely-deployed and standardized TLS protocol to encrypt the signaling messages. You can configure the SIP Transport parameter in the *SIP Settings* section of the **Line 1 and Line 2 Settings (PHONE Port1 and PHONE2)** page.

- **Secure Calling using SRTP**

  Voice packets are encrypted by using Secure Real-Time Transport Protocol (SRTP). This function is implemented on a standards basis (RFC4568). Secure call service (Secure Call Serv) is enabled by default in the *Supplementary Service Subscription* section of the **Line 1 and Line 2 Settings (PHONE Port1 and PHONE2)** page. When this service is enabled, users can activate secure calling by pressing the star (*) key before dialing a phone number. Alternatively, you can enable the Secure Call Setting to encrypt all calls from a user's phone. See the *Supplementary Service Settings* section of the **User 1 and User 2** page.