



ADMINISTRATION GUIDE

Cisco SPA232D Mobility Enhanced Phone Adapter

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Getting Started

Thank you for choosing the Cisco SPA232D Mobility Enhanced Analog Telephone Adapter (ATA). This chapter provides more information about the features of the product and provides instructions about connecting the equipment and getting started in the web-based configuration utility.

Feature Overview

With the Cisco SPA232D Mobility Enhanced ATA, you can provide your analog phone and Cisco SPA302D cordless handsets with access to analog and Internet phone services through a standard RJ-11 phone port and a built-in DECT base station. The ATA supports up to five Cisco SPA302D handsets. 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.

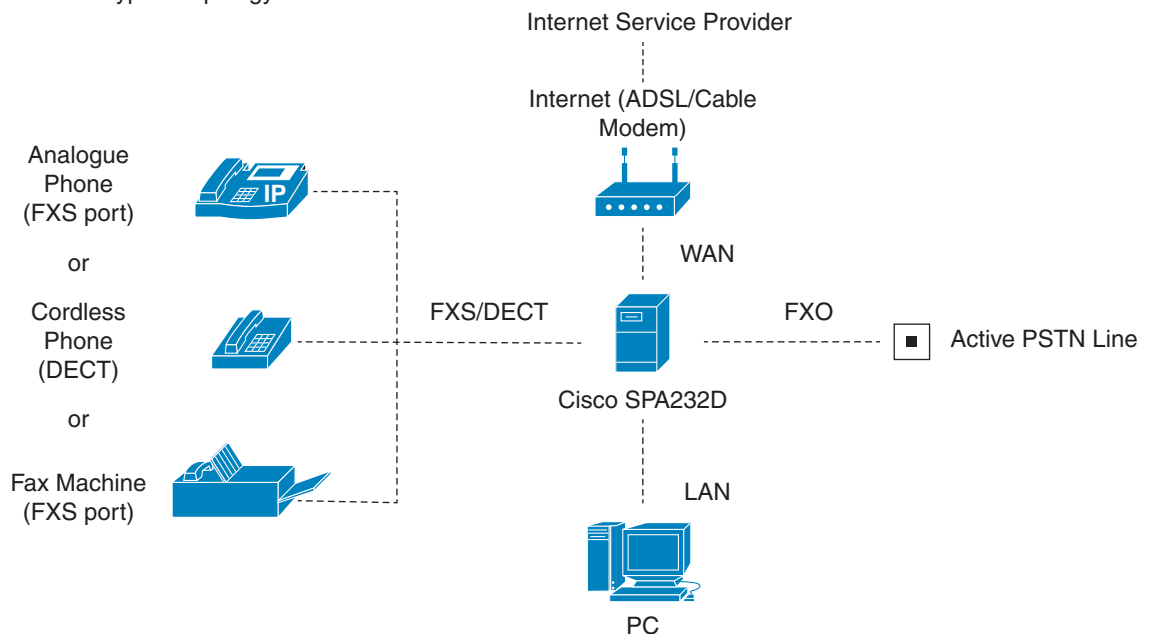
The ATA is an intelligent low-density Voice over IP (VoIP) gateway that enables carrier-class residential and business IP Telephony services delivered over broadband or high-speed Internet connections. An ATA maintains the state of each call it terminates and reacts appropriately to user input events (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 or DECT handset 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 and Cisco SPA302D handsets.

SPA232D typical topology



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 these pages: **Line 1 Settings (PHONE Port)**, **PSTN (LINE Port)**, and **DECT Line 1 - DECT Line 10**.

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 and the [DECT Line 1 - DECT Line 10](#) pages.

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 these pages: [Line 1 Settings \(PHONE Port\)](#), [PSTN \(LINE Port\)](#), and [DECT Line 1 - DECT Line 10](#).

- **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.

NOTE For more information, please refer to [Audio Configuration, page 113](#) and [Configuration Tables, page 117](#).

- **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 these pages: [Line 1 Settings \(PHONE Port\)](#), [PSTN \(LINE Port\)](#), and [DECT Line 1 - DECT Line 10](#).

- **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 these pages: [Line 1 Settings \(PHONE Port\)](#), [PSTN \(LINE Port\)](#), and [DECT Line 1 - DECT Line 10](#).

- **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 Settings \(PHONE Port\)](#) page and the [DECT Line 1 - DECT Line 10](#) pages.
- 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 225](#).

- **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 Settings \(PHONE Port\)](#) 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 these pages: [Line 1 Settings \(PHONE Port\)](#), [PSTN \(LINE Port\)](#), and [DECT Line 1 - DECT Line 10](#).

- **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 these pages: [Line 1 Settings \(PHONE Port\)](#) and [DECT Line 1 - DECT Line 10](#). 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 or a DECT line. See the *Supplementary Service Settings* section of the [User 1](#) page and the [DECT Line 1 - DECT Line 10](#) page.





Before You Begin


Before you begin the installation, make sure that you have the following equipment and services:

- An active Internet account and Voice over IP account
- Ethernet cable to connect to your broadband network device
- Phone to connect to your ATA
- Phone cable to connect your phone
- *Optional:* Uninterruptible Power Supply (UPS) to provide backup power
- *Optional:* Cisco SPA302D Mobility Enhanced Cordless Handsets

Product Features

Top Panel

Feature	Description
 Page/ Registration	<p>Steady green—One or more handsets is registered.</p> <p>Fast flashing green—The base is in registration mode. To activate registration mode, press the button for at least 7 seconds.</p> <p>Slow flashing green—The base is in paging mode or a handset is off hook. To activate paging mode to locate a handset, press the button for a few seconds; handsets ring.</p> <p>Off—No handset is registered to the base.</p>
 LINE	<p>Steady green—The line is off hook and connected to the local telephone system.</p> <p>Slow flashing green—The line is off hook.</p> <p>Off—The port is not ready.</p>
 PHONE	<p>Steady green—The device is on hook and registered to a SIP proxy.</p> <p>Slow flashing green—The device is off hook.</p> <p>Off—The port is not ready.</p>
 INTERNET	<p>Flashing green—Transmitting or receiving data through the WAN port.</p> <p>Off—No link.</p>

Feature	Description
 SYSTEM	Steady green —The system is ready. Slow flashing green —Acquiring an IP address, if applicable. (DHCP is used by default.) Fast flashing green —Upgrading the firmware. Off —There is no power or the system cannot boot up.

Back Panel



Feature	Description
RESET	Performs two functions: Restart the ATA: Press quickly (less than a second) with a paperclip or similar object. Restore the factory default settings: Press and hold for 5 to 6 seconds.
LINE (Green)	Connects to an analog phone line, using an RJ-11 phone cable.
PHONE (Gray)	Connects to an analog phone, using an RJ-11 phone cable.
ETHERNET (Yellow)	Connects to a device on your local area network (LAN), such as a computer by using an Ethernet cable.
INTERNET (Blue)	Connects to a broadband network device (DSL or cable modem) or a network router by using an Ethernet cable.
POWER	Connects to a power by using the provided adapter.

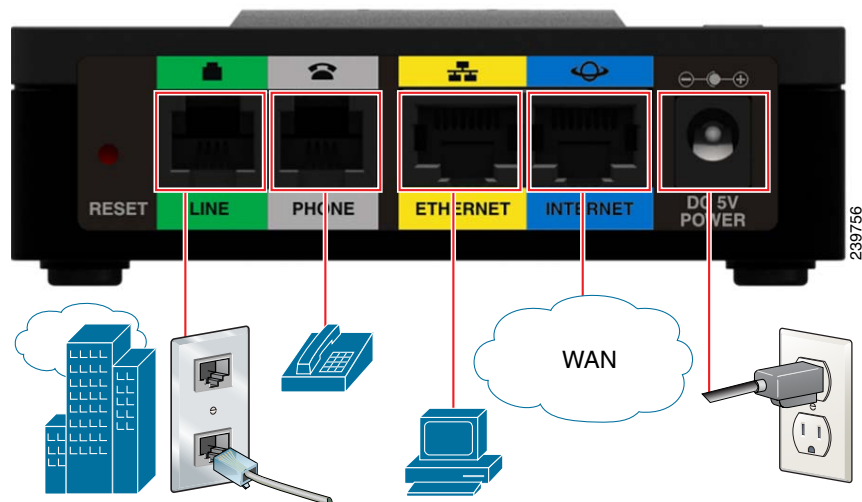
Default Settings

Parameter	Default Value
Administrator Username	admin
Administrator Password	admin
User Username	cisco
User Password	cisco
Internet Connection Type	Automatic Configuration - DHCP
LAN IP Address (Also the address for the web-based configuration utility.)	192.168.15.1
DHCP Range (DHCP server enabled by default.)	192.168.15.100-149
Netmask	255.255.255.0
PIN for handset registration, IP settings, and SIP settings	Blank

Connecting the Equipment

NOTE For wall-mounting instructions, see [Additional Information, page 18](#).

- STEP 1** Connect the provided Ethernet cable to the INTERNET (Blue) port. Connect the other end of the cable directly to your broadband network device.
- STEP 2** Connect the provided phone cable to the PHONE 1 (Gray) port. Connect the other end of the cable to your analog phone or fax machine.
- STEP 3** Optionally, connect an Ethernet network cable to the ETHERNET (Yellow) port of the ATA. Connect the other end of the cable to a device on your network, such as a computer.
- STEP 4** Connect an analog phone line to the LINE (Green) port to connect the ATA to your local telephone system.
- STEP 5** Connect the provided power adapter to the POWER port. The unit powers on.



Configuration and Management of the ATA

You can use the web-based configuration utility to set up your ATA. You also can use the built-in Interactive Voice Response (IVR) system. (See [Using the IVR for Administration, page 18.](#))

STEP 1 Connect the provided Ethernet network cable to the ETHERNET (Yellow) port of the ATA. Connect the other end of the cable to the Ethernet port of your PC.

STEP 2 Power on your computer.

NOTE: Make sure your computer's Ethernet adapter is set to obtain an IP address automatically (DHCP). For more information, refer to the Help for your operating system.

STEP 3 Start a web browser on your computer.

STEP 4 In the Address bar, enter: **192.168.15.1**

Note: 192.168.15.1 is the default local IP address of the ATA.

STEP 5 To log in for the first time, enter the default username, **admin**, and the default password, **admin**. The password is case sensitive.

NOTE: A user account allows access to limited settings and status pages. To log in as a user, enter **cisco** as the username and the password.

STEP 6 Use the *Quick Setup* page as needed to register your VoIP accounts in the fields for Line 1 and DECT Line 1.

Your VoIP service may require only a few basic parameters to successfully register the Cisco SPA232D. The *Quick Setup* page offers a shortcut to enter the basic parameters. For a more comprehensive listing of parameters, choose the *Voice* menu, and then use the links in the navigation tree.

- **Enter Proxy:** Enter the domain name or URL of the service provider's proxy server.
- **Display Name:** Enter the name of the business. This name typically is used for the Caller ID.
- **User ID:** Enter the user ID for your Internet account with this service provider.
- **Password:** Enter the password for your Internet account.
- **Dial Plan (Line 1 only):** Keep the default settings (recommended). Detailed information about the dial plan settings is available in the online Help and the administration guide.

Note: The Cisco SPA232D assigns DECT Line 1 as the default line for outgoing calls from Cisco SPA302D handsets. If needed, you can configure additional VoIP accounts as separate “DECT Lines.” To do so, choose the **Voice** menu, and then use the *DECT Line 1~10* links in the navigation tree. Use the check boxes on the *Quick Setup* page to associate the DECT Line(s) to each handset.

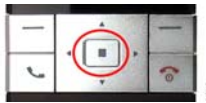
STEP 7 Click **Submit** to save your settings.

STEP 8 If you wish to change the PIN for handset registration, open the *Voice > System* page, and then enter up to four digits in the **Handset (HS) Pairing Password** field. Click **Submit** to save your settings.

Registering a Cisco SPA302D Handset

You can register Cisco SPA302D handsets to the integrated DECT base station. These handsets can be purchased separately.

STEP 1 On the Cisco SPA302D handset, press the center Select button on the 4-way navigation keypad.



STEP 2 Select **Register**.

STEP 3 Using the navigation arrows, scroll to the Settings icon and press the center navigation button to select it.



STEP 4 Select **Handset Registration**.

STEP 5 On the Cisco SPA232D, press the **Page/Registration** button and hold it down for at least seven seconds until the green status light flashes quickly.

TIP: If you press the button for fewer than seven seconds, the green status light flashes slowly, indicating the unit is in “paging” mode and is not in registration mode. Registration will not work if the unit is in paging mode.

STEP 6 The default PIN is blank, so do not enter a PIN. Press the left softkey to confirm that you want to register the handset. The “registering” message appears.

STEP 7 To verify that the handset registered to the base station, confirm that the wireless status icon is solid and that the handset ID, such as DECT1 or DECT 2, appears near the top right corner of the display screen.

Additional Information

Using the IVR for Administration

An IVR system is available to help you to configure and manage your ATA. You can use the telephone keypad to select options and to make your entries.

To access the IVR menu:

-
- STEP 1** Connect an analog phone to the PHONE port of the ATA.
 - STEP 2** Press the star key four times: ****
 - STEP 3** After the greeting plays, press the keys on the phone keypad to select your options.
 - STEP 4** Enter the code for the desired action. See the **IVR Actions** table for details.
-

TIPS:

- Enter the numbers slowly, listening for the audio confirmation before entering the next number.
- After you select an option, press the # (pound) key.
- To exit the menu, hang up the telephone or enter **3948#** to exit.
- After entering a value, such as an IP address, press the # (pound) key to indicate that 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).
- While entering a value, you can cancel the changes by pressing the * (star) key twice within half a second. Be sure to press the key quickly, or the * will be treated as a decimal point entry.
- If the menu is inactive for more than one minute, the ATA times out. You will need to re-enter the menu by pressing the star key four times: ****. Your settings take effect after you hang up the telephone or exit the IVR. The ATA may reboot at this time.

- To enter the decimal points in an IP address, press the * (star) key. For example, to enter the IP address 191.168.1.105, perform the following tasks:
 - Press these keys: 191*168*1*105.
 - Press the # (pound) key to indicate that you have finished entering the IP address.
 - Press **1** to save the IP address or press the * (star) key to cancel your entry and return to the main menu.

IVR Actions

IVR Action	Menu Option	Choices and Instructions
Enter IVR Menu	****	
Check Internet Connection Type	100	
Set Internet Connection Type	101	DHCP: 0 Static IP: 1 PPPoE: Press 2
Check Internet IP Address (WAN port)	110	
Set Static IP Address (WAN)	111	Enter the IP address by using numbers on the telephone key pad. Use the * (star) key when entering a decimal point. Note: This option is available only after you choose Static IP as the Internet Connection Type, through option 101.
Check Network Mask	120	

IVR Action	Menu Option	Choices and Instructions
Set Network Mask	121	To enter the value, press numbers on the telephone key pad. Press the * (star) key to enter a decimal point. Note: This option is available only after you choose Static IP as the Internet Connection Type, through option 101.
Check Gateway IP Address	130	
Set Gateway IP Address	131	To enter the value, press numbers on the telephone key pad. Press the * (star) key to enter a decimal point. Note: This option is available only after you choose Static IP as the Internet Connection Type, through option 101.
Check MAC Address	140	
Check Firmware Version	150	
Check Primary DNS Server Setting	160	
Set Primary DNS Server	161	To enter the value, press numbers on the telephone key pad. Press the * (star) key to enter a decimal point. Note: This option is available only after you choose Static IP as the Internet Connection Type, through option 101.
Check Internet web server port	170	

IVR Action	Menu Option	Choices and Instructions
SPA122 only: Check LAN IP address (Ethernet port)	210	
Announce Line 1 SIP Transport	1910	
Set Line 1 SIP Transport	1911	0: UDP 1: TCP 2: TLS
Check Line 2 SIP Transport	1920	
Set Line 2 SIP Transport	1921	0: UDP 1: TCP 2: TLS
Exit IVR	3948	
Allow or prevent WAN access to the administration web server The system will allow WAN access only if the default admin username and password have been changed in the Configuration Utility.	7932	1: Enable 0: Disable
Factory Reset of Unit WARNING: All non-default settings will be lost. This includes network and service provider data.	73738 "RESET"	When prompted, press 1 to confirm, or press * (star) to cancel. After you hear "Option successful," hang up the phone. The ATA reboots.
Reboot of Voice System	732668 "REBOOT"	After you hear "Option successful," hang up the phone. The ATA reboots.

Mounting the ATA

You can place the ATA on a desktop or mount it on a wall.



CAUTION

To prevent the ATA from overheating, do not operate it in an area that exceeds an ambient temperature of 104°F (40°C).

Desktop Placement

Place the ATA on a flat surface near an electrical outlet.



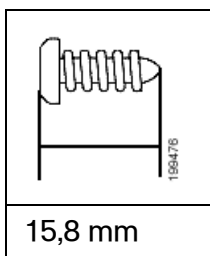
WARNING

Do not place anything on top of the ATA; excessive weight could damage it.

Wall Mounting

The ATA has two wall-mount slots on the bottom panel. To mount the ATA on a wall, you need mounting hardware (not included). Suggested hardware is illustrated (not true to scale).

Recommended hardware (not included): Two number-six pan-head tapping screws, 5/8-in. length, with anchors for sheet rock installation.

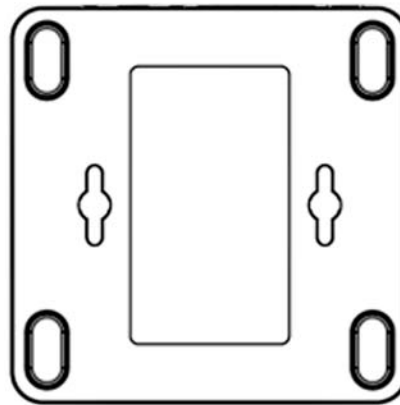


WARNING

Insecure mounting might damage the ATA or cause injury. Cisco is not responsible for damages incurred by insecure wall-mounting.

To mount the unit to the wall:

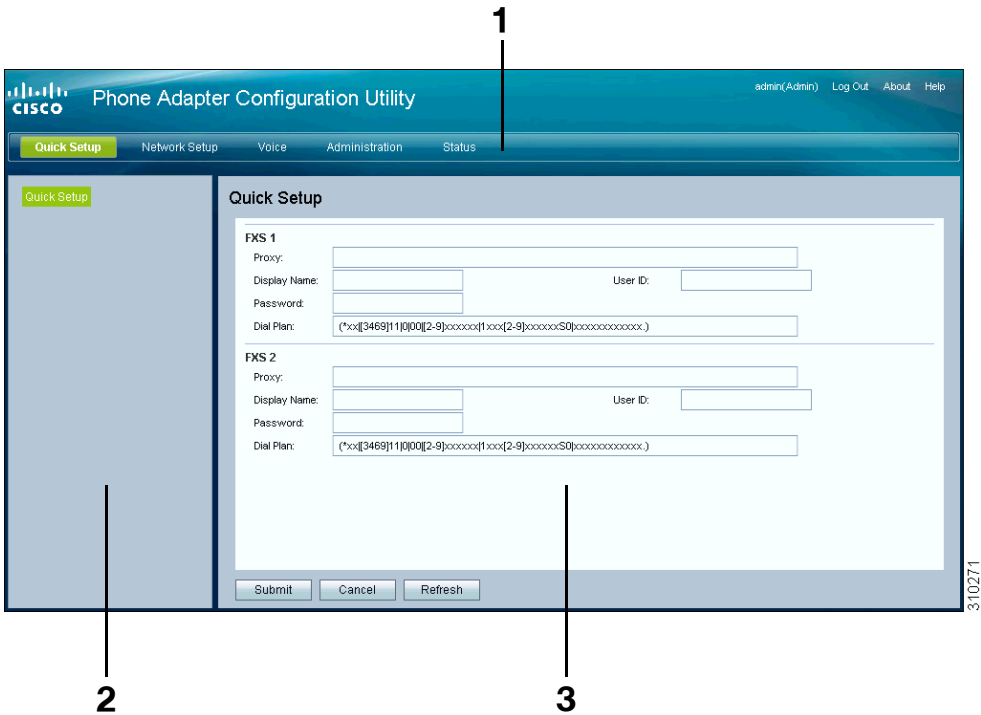
- STEP 1** Determine where you want to mount the unit. Verify that the surface is smooth, flat, dry, and sturdy.
- STEP 2** Drill two pilot holes into the surface 58 mm apart (about 2.28 in.). Make sure that the holes are at the same height above the floor so that the unit is level and secure in either of its two safety-certified orientations.
- STEP 3** Insert a screw into each hole, leaving a gap of 5 mm (0.1968 in.) between the underside of each screw head and the surface of the wall.
- STEP 4** Place the unit wall-mount slots over the screws and slide the unit down until the screws fit snugly into the wall-mount slots.



Elements of the User Interface

Before you use your ATA, become familiar with the following features of the user interface.




Screen Elements



Component	Description
1. Menu Bar (top)	Provides access to the modules of the configuration utility. Click a menu to view the options in the navigation tree.
2. Navigation Tree (left panel)	Provides access to the configuration pages within the selected module. Click a category heading to view the list of features. Click a link to open the configuration page.
3. Configuration Page (main area)	Settings for the selected feature.

Configuration Utility Icons

Many configuration pages provide the following icons for common tasks.

Icon	Description
	The Edit icon lets you edit an existing item from a list. After making your changes, click the Submit button to save your changes.
	The Add Item icon lets you add an item to a list. After you have created a new item, click the Submit button to save the new item.
	The Delete Item icon lets you delete an item from a list. After you have deleted an item, click the Submit button to save your changes.

Saving the Settings

Your settings on a configuration page are not saved until you click the **Submit** button. When you navigate to another page, any unsaved settings are abandoned. Changes cannot be saved while calls are in progress. Try again when the phones are idle.

To clear the settings without saving them, you can click the **Cancel** button.

Help

To view information about the configuration pages, click the **Help** link near the top right corner of the configuration utility. You can then use the table of contents to find topics of interest.

Logout

To exit the configuration utility, click the **Logout** link near the top right corner of the window. The *Login* page appears. You can close the browser window.

Quick Setup for Voice over IP Service

The *Quick Setup* page is displayed automatically when you first log on ATA. You can use this page to quickly configure connectivity to your provider's Voice over IP network for your analog phone and Cisco SPA302D handsets.

NOTE Connecting to your service provider's network requires Internet connectivity. With the default network settings, your ATA should have Internet connectivity when you connect a cable from the WAN port of the ATA to a port on your router or broadband network device. For more information, see [Internet Settings, page 31](#).

To open this page: Click **Quick Setup** in the menu bar.

STEP 1 Specify the settings for the phone service to be used by each type of device or line. Follow the requirements and recommendations of your service provider. The options are described below.

Device/line types:

- **Line 1:** The phone service used by an analog (FXS) phone or fax machine that is connected to the PHONE port.
- **PSTN:** The phone service used by a phone line that is connected from the LINE port to the PSTN.
- **DECT Line 1:** The phone service used by all connected Cisco SPA302D cordless handsets (when using the default settings in *DECT Handset Outgoing Line Selection* and *DECT Line Contact List* sections). You can add additional phone services for these handsets on the *Voice > DECT Line 2* to *DECT Line 10* pages.

Settings:

- **Proxy:** Enter the domain name or URL of the service provider's proxy server.
- **Display Name:** Enter the name that you want to use to identify your account. This name typically is used as your Caller ID name.
- **User ID:** Enter the user ID that is required to log in to your Internet account.

- **Password:** Enter the password that is required to log in to your Internet account.
- **Dial Plan** in (*Line* section only): Keep the default settings (recommended) or edit the dial plan to suit your site. For more information, see [Configuring Dial Plans, page 225](#).

STEP 2 DECT Handset Outgoing Line Selection: For each DECT Handset, check the boxes to choose the DECT Lines for outgoing calls. Uncheck the boxes for the lines that you do not want to use.

- If you are using only one phone service for all Cisco SPA302D handsets, simply configure the DECT Line 1 settings above and keep the default settings in this section.
- If you have multiple lines, you can select multiple lines for each handset. Alternatively, check the **All Lines** box to make all lines. The enabled options will be listed on the phone screen when the user displays the call options or holds down the green call button.
- Choose a **Default** line, which will be selected automatically for a call when the user presses the green call button.
- Optionally, if you enabled multiple lines, enable **Failover** by selecting **yes**. When this feature is enabled and a call fails through the selected line, the ATA automatically attempts to place the call over another enabled DECT line.

NOTE Cisco SPA232D now supports PSTN to DECT and DECT to PSTN outgoing line failover.

STEP 3 DECT Line Contact List: For each line, check the boxes to choose the handsets that ring when an incoming call is received. Uncheck the boxes for the handsets that you do not want to ring. Check the **All Handsets** box to ring all handsets for the specified line.

STEP 4 Click **Submit** to save your settings. The voice service will restart.

STEP 5 To verify your progress, perform the following tasks:

- a. Click **Voice** in the menu bar, and then click **Info** in the navigation tree. Verify that the Registration State is Registered for all configured lines (Line 1 Status, PSTN Line Status, and DECT 1 Status ~ DECT 10 Status).

If the line is not registered, you may need to refresh the browser several times because it can take a few seconds for the registration to complete. Also verify that your Internet Settings, including DNS server settings, are configured according to the information from your ISP. For more information, see [Internet Settings, page 31](#).

- b. Use an external phone to place an inbound call to the telephone number that was assigned by your ITSP. Verify that the phone rings and you have two-way audio on the call.
-

Configuring the Network

This chapter describes how to configure the network settings for your ATA. It includes the following sections:

- **Basic Setup**
- **Advanced Settings**
- **Application**

Basic Setup

Use the *Network Setup > Basic Setup* pages to configure your Internet connection, local network settings, and your time settings.

- **Network Service**
- **Internet Settings**
- **Network Settings for the LAN and DHCP Server**
- **Time Settings**

Network Service

Use the *Network Setup > Basic Setup > Network Service* page to configure the operating mode of the ATA.

To open this page: Click **Network Setup** in the menu bar, and then click **Basic Setup > Network Service** in the navigation tree. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

You can configure the ATA to operate in one of the following modes:

- **NAT:** Network Address Translation (NAT) is a function that allows multiple devices on a private network to share a public, routable IP address to establish connections over the Internet. To enable Voice over IP service to co-exist with NAT, some form of NAT traversal is required, either on the ATA or another network device. Use this option if your ATA connects to one network on the WAN port (10.0.0.0 for example) and to another network on the LAN port (192.168.0.0 for example). This option is selected by default and is suitable for most deployments.
- **Bridge:** Bridged mode is used if the ATA is acting as a bridge device to another router. Choose this option if your ATA bridges a network (10.0.0.0 for example) to its LAN port (with connected devices also in the 10.0.0.x range).

Internet Settings

Use the *Network Setup > Basic Setup > Internet Settings* page to set up your Internet connection.

To open this page: Click **Network** in the menu bar, and then click **Basic Setup > Internet Settings** in the navigation tree.

Enter the settings as described in the table. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

Internet Connection Type

Field	Description
Connection Type	<p>Specify the Internet addressing method that your ISP requires. Default setting: Automatic Configuration - DHCP</p> <ul style="list-style-type: none">▪ Automatic Configuration - DHCP: Use this setting if your ISP dynamically provides an IP address. No additional settings are required on this page.▪ Static IP: Use this setting if your ISP assigned a static/permanent IP address. Complete the fields that appear. See more information below.▪ PPPoE (DSL service): Some DSL-based ISPs use PPPoE (Point-to-Point Protocol over Ethernet) to establish Internet connections. If you are connected to the Internet through a DSL line, check with your ISP to see if they use PPPoE. Complete the fields that appear. See more information below.

Field	Description
Static IP Settings	<ul style="list-style-type: none">▪ Internet IP Address and Subnet Mask: Enter the IP address and subnet mask that was assigned to your account by your service provider. This address is seen by external users on the Internet.▪ Default Gateway: Enter the Gateway IP Address that was provided by your ISP. <p>If needed, you can adjust the MTU and Optional Settings, as described below.</p>
PPPoE Settings	<ul style="list-style-type: none">▪ User Name and Password: Enter the user name and password that you use to log into your ISP network through a PPPoE connection.▪ Service Name: If provided by your ISP, enter the Service Name.▪ Connect on Demand: You can configure the ATA to disconnect your Internet connection after a specified period of inactivity (Max Idle Time). If your Internet connection has been terminated due to inactivity, this feature enables the ATA to automatically re-establish your connection as soon as you attempt to access the Internet again. If you choose this option, also set the Max Idle Time.▪ Keep Alive: This option keeps you connected to the Internet indefinitely, even when your connection sits idle. If you choose this option, also set the Redial Period, which is the interval at which the ATA verified Internet connectivity. The default period is 30 seconds. <p>If needed, you can adjust the MTU and Optional Settings, as described below.</p>

Field	Description
MTU	The Maximum Transmission Unit (MTU) setting specifies the largest protocol data unit (in bytes) permitted for network transmission. Generally, a larger MTU means greater efficiency. However, a larger packet may cause delays for other traffic and is more likely to become corrupted. In most cases, you should keep the default setting, Auto, to allow the ATA to choose the appropriate MTU. To specify the MTU, select Manual, and then enter the number of bytes.

Optional Settings

Feature	Description
Host Name	The name of the ATA. The default value is the model number. Your ISP may specify a host name to use.
Domain Name	The domain name, if specified by your ISP. Otherwise, leave the field blank.
DNS Server Order	Choose the preferred method for choosing a DNS server. <ul style="list-style-type: none"> ▪ DHCP-Manual—The DNS server settings from the network server will take precedence, and your entries in the DNS fields below will be used only as a backup. ▪ Manual-DHCP—Your entries in the DNS fields below will take precedence, and the DNS server settings from the network server will be used as a backup. ▪ Manual—Your entries in the DNS fields below will be used to choose a DNS server.
Primary DNS	Enter the IP address of the primary Domain Name Service (DNS) server to use for domain name resolution. Keep the default entry, 0.0.0.0, to use the primary DNS server that is specified for the WAN connection.

Feature	Description
Secondary DNS	Enter the IP address of the secondary Domain Name Service (DNS) server to use for domain name resolution. Keep the default entry, 0.0.0.0, to use the secondary DNS server that is specified for the WAN connection.

Network Settings for the LAN and DHCP Server

Use the *Network Setup > Basic Setup > Network Settings* page to set the IP address and subnet mask for your local network. Also configure the settings for the built-in DHCP server.

To open this page: Click **Network Setup** in the menu bar, and then click **Basic Setup > Network Settings** in the navigation tree.

Enter the settings as described below. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

Router IP

Enter the **Local IP Address** and **Subnet Mask** for your local network. The default setting is 192.168.15.1 with a subnet mask of 255.255.255.0.

DHCP Server Setting

Field	Description
DHCP Server	The ATA can use the built-in DHCP server to dynamically assign IP addresses to connected devices. Click Enabled to enable the DHCP server, or click Disabled to disable this feature. Default setting: Enabled

Field	Description
IP Reservation:	<p>Click the Show DHCP Reservation button to view and manage the DHCP client list. Click the Hide DHCP Reservation button to hide the list. When the list is displayed, you can perform the following tasks:</p> <ul style="list-style-type: none"> ▪ To reserve a static IP address for a current DHCP client: Check the box for the client in the <i>Select Clients from DHCP Tables</i> list. Click Add Clients. The selected clients are added to the <i>Clients Already Reserved</i> list. These clients have static IP addresses that do not change. ▪ To add a client that is not in the Select Clients from DHCP Tables list: Type a name for the client in the Enter Client Name box. Enter an IP address for this client in the Assign IP Address box. Enter the MAC address in the following format: 00:00:00:00:00:00. Click Add. ▪ To remove a client from the Clients Already Reserved list: Check the box for the client. Click Remove.
Default Gateway	<p>Enter the IP address of the default gateway to be used by the DHCP clients.</p> <p>Default setting: 192.168.15.1 (the IP address of the ETHERNET (LAN) interface)</p>
Starting IP Address	<p>Enter the first address in the range of addresses to be assigned dynamically by the DHCP server.</p> <p>Default setting: 192.168.15.100</p>
Maximum DHCP Users	<p>Enter the maximum number of devices that can dynamically receive, or “lease,” DHCP addresses from the DHCP server.</p> <p>Default setting: 50</p> <p>IMPORTANT: Typically, the ATA can support up to five connected computers for business-related tasks such as web browsing and viewing email. The ATA is not designed to support streaming music, video, games, or other network traffic-intensive tasks.</p>

Field	Description
Client Lease Time	Enter the number of minutes that a dynamically assigned IP address can be in use, or “leased.” After this time elapses, a client device has to request a DHCP lease renewal. Use 0 to represent 1 day, 9999 never expire. Default setting: 0
Option 66	Provides provisioning server address information to hosts that request this option. Server information can be defined in one of three ways: <ul style="list-style-type: none"> ▪ None: The ATA uses its own TFTP server to source provisioning files, so it returns its own local IP address to the client. ▪ Remote TFTP Server: The ATA was configured by using this method, and received server information through Option 66 on its WAN interface. In response to client requests, it provides the remote TFTP server information. ▪ Manual TFTP Server: Allows the manual configuration of a configuration server address. While this option is typically used to provide either an IP address or a fully qualified hostname, the ATA will also accept and offer a full URL including protocol, path and filename to meet to requirements of specific clients. Default setting: None
TFTP Server	If you chose Manual TFTP Server for Option 66, enter the IP address, hostname, or URL of the TFTP server that is used to configure the ATA. Default setting: blank
Option 67	Provides a configuration/bootstrap filename to hosts that request this option. This option is used in conjunction with option 66 to allow a client to form an appropriate TFTP request for the file. Default setting: blank

Field	Description
Option 159	Provides a configuration URL to clients that request this option. An option 159 URL defines the protocol and path information by using an IP address for clients that cannot use DNS. For example: https://10.1.1.1:888/configs/bootstrap.cfg Default setting: blank
Option 160	Provides a configuration URL to clients that request this option. An option 160 URL defines the protocol and path information by using a fully qualified domain name for clients that can use DNS. For example: https://myconfigs.cisco.com:888/configs/bootstrap.cfg Default setting: blank
DNS Proxy	When enabled, the DNS proxy relays DNS requests to the current public network DNS server for the proxy, and replies as a DNS resolver to the client device on the network. Click Enabled to enable this feature, or click Disabled to disable it. If DNS proxy is disabled, then DHCP clients will be offered DNS server information by using the Static DNS servers, if defined, or by using the servers specified for the INTERNET (WAN) interface. Default setting: Enabled

Time Settings

Use the *Network Setup > Basic Setup > Time Settings* page to set the system time for the ATA and connected Cisco SPA302D handsets. By default, the system time is set automatically by using a Network Time Protocol (NTP) server. You can configure the system time manually. In addition, you can use this page to specify your time zone, enable Daylight Saving adjustments, and modify related settings.

To open this page: Click **Network Setup** in the menu bar, and then click **Basic Setup > Time Settings** in the navigation tree. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

User Manual

If you prefer to set the system manually rather than automatically obtaining the settings from an NTP server, click **User Manual** and then enter the date and time.

Field	Description
Date	Enter the date in the following order: four-digit year, month, day.
Time	Enter the time in the following order: hour (from 1 to 24), minutes, and seconds.

Time Zone

To use a time server to establish the time settings, select **Time Zone**. Then complete the fields in this section.

Field	Description
Time Zone	Choose the time zone for the site where the ATA is in operation. Default setting: (GMT-08:00) Pacific Time (USA & Canada)
Adjust Clock for Daylight Saving Changes	Check the box if you want to automatically adjust the time when Daylight Savings Time is in effect. Otherwise, uncheck the box. Default setting: Enabled
Time Server Address	To use the ATA's default Network Time Protocol (NTP) server, select Auto from the drop-down list. This is the default setting. If you want to specify the NTP server, select Manual , and then enter the NTP server address. Default setting: Auto
Resync Timer	Enter the Resync timer interval value (in seconds). This timer controls how often the ATA resynchronizes with the NTP server. Default setting: 3600 seconds

Field	Description
Auto Recovery After Reboot	Choose this option to allow the ATA to automatically reconnect to the time server after a system reboot. Default setting: Disabled

Advanced Settings

Use the *Network Setup > Advanced Settings* pages to configure features including port flow control, MAC address cloning, VPN passthrough, and VLAN.

- **Port Setting**
- **MAC Address Clone**
- **VPN Passthrough**
- **VLAN**
- **CDP & LLDP**

Port Setting

Use the *Network Setup > Advanced Settings > Port Setting* page to set the ETHERNET (LAN) port attributes.

To open this page: Click **Network Setup** in the menu bar, and then click **Advanced Settings > Port Settings** in the navigation tree. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

Field	Description
Flow Control	<p>Flow control is a mechanism that temporarily stops the transmission of data on a port. For example, a situation might arise where a sending station (computer) is transmitting data faster than some other part of the network (including the receiving station) can accept. The overwhelmed network element will halt the transmission of the sender for a specified period of time.</p> <p>Choose Enabled to enable this feature, or choose Disabled to disable this feature. Default setting: Enabled</p>
Speed Duplex	<p>Choose the duplex mode. You can select from Auto-negotiate, 10 Half, 10 Full, 100 Half and 100 Full. Cisco recommends choosing Auto-negotiate to automatically select the appropriate mode for the traffic. Use caution with other settings. Problems can result if you choose a setting that is not appropriate for the network devices.</p> <p>Default setting: Auto-negotiate</p>

MAC Address Clone

A MAC address is a 12-digit code assigned to a unique piece of hardware for identification purposes. Some ISPs require that you register a MAC address in order to access the Internet. If you previously registered your account with another MAC address, it may be convenient to assign that MAC address to your ATA. You can use the *Network Setup > Advanced Settings > MAC Address Clone* page to assign a MAC address that you previously registered with your Service Provider.

To open this page: Click **Network Setup** in the menu bar, and then click **Advanced Settings > MAC Address Clone** in the navigation tree. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

Field	Description
MAC Clone	Click Enabled to enable MAC address cloning, or click Disabled to disable this feature. Default setting: Disabled.
MAC Address	Enter the MAC address that you want to assign to your ATA. If your computer's MAC address is the address that you previously registered for your ISP account, click Clone Your PC's MAC . Your computer's MAC address appears in the <i>MAC Address</i> field. Default setting: the current MAC address of your ATA

VPN Passthrough

Use the *Network Setup > Advanced Settings > VPN Passthrough* page to configure VPN passthrough for IPsec, PPTP, and L2TP protocols. Use this feature if there are devices behind the ATA that need to set up IPsec tunnels independently. For example, a device may need to use a VPN tunnel to connect to another router on the WAN.

By default, VPN Passthrough is enabled for IPsec, PPTP, and L2TP.

To open this page: Click **Network Setup** in the menu bar, and then click **Advanced Settings > VPN Passthrough** in the navigation tree. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

Field	Description
IPsec Passthrough	Internet Protocol Security (IPsec) is a suite of protocols used to implement secure exchange of packets at the IP layer. Click Enabled to enable this feature, or click Disabled to disable it. Default setting: Enabled
PPTP Passthrough	Point-to-Point Tunneling Protocol (PPTP) allows the Point-to-Point Protocol (PPP) to be tunneled through an IP network. To disable PPTP Passthrough, select Disabled. Default setting: Enabled

Field	Description
L2TP Passthrough	Layer 2 Tunneling Protocol is the method used to enable Point-to-Point sessions via the Internet on the Layer 2 level. Click Enabled to enable this feature, or click Disabled to disable it. Default setting: Enabled

VLAN

Use the *Network Setup > Advanced Settings > VLAN* page to assign a VLAN ID to your network. For example, your call control system may require a particular voice VLAN ID.

To open this page: Click **Network Setup** in the menu bar, and then click **Advanced Settings > VLAN** in the navigation tree. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

Field	Description
Enable VLAN	Click Enabled to enable a VLAN, or click Disabled to disable this feature. Default setting: Disabled
VLAN ID	The VLAN ID can be any numeral from 1 through 4094. When VLAN is enabled, the default setting is 1.

CDP & LLDP

Device discovery protocols enable directly connected devices to discover information about each other. You may wish to enable these protocols to allow your network management system to learn about your ATA and endpoints. Use the *Network Setup > Advanced Settings > CDP & LLDP* page to specify the settings for Cisco Discovery Protocol (CDP) and the Link Layer Discovery Protocol (LLDP). When a discovery protocol is enabled, the ATA sends periodic messages to a multicast address and also listens to the periodic messages sent by other devices that use that protocol.

To open this page: Click **Network Setup** in the menu bar, and then click **Advanced Settings > CDP & LLDP** in the navigation tree. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

Field	Description
Enable CDP	CDP may be enabled on Cisco devices. Click Enabled to enable CDP, or click Disabled to disable this feature. Default setting: Disabled
Enable LLDP-MED	LLDP is a vendor-neutral device discovery protocol that may be enabled on other manufacturers' equipment. LLDP-MED is specifically for Media Endpoint Devices. Click Enabled to enable CDP, or click Disabled to disable this feature. Default setting: Disabled
Layer 2 Logging	Click Enabled to enable Layer 2 Logging, or click Disabled to disable this feature. L2 Logging is used by CDP/LLDP for debugging purpose. Default setting: Disabled

Application

Use the *Network Setup > Application* pages to support voice service and any servers that you host for public access.

- **Quality of Service (QoS)**
- **Port Forwarding**
- **DMZ**

Quality of Service (QoS)

Use the *Network Setup > Application > QoS* page to set the upstream bandwidth to suit your broadband service. This feature is enabled by default and helps to ensure that voice is prioritized during periods of heavy network traffic.

To open this page: Click **Network Setup** in the menu bar, and then click **Application > QoS** in the navigation tree.

Enter the settings as described below. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

Field	Description
QoS Policy	Click Always On to enable QoS settings at all times, or click On When Phone In Use to enable it only when there is voice traffic. Default setting: On When Phone In Use
Upstream Bandwidth	Enter the maximum available upstream bandwidth value specified by your Internet Service Provider. Default setting: 10000 kbps IMPORTANT: Do not overstate the upstream bandwidth that you receive from your service provider. Setting this value higher than the available service bandwidth can result in traffic being dropped arbitrarily in the service provider's network.

Port Forwarding

Use the *Network Setup > Application > Port Forwarding* page if you need to explicitly allow access to specific ports from external devices.

To open this page: Click **Network Setup** in the menu bar, and then click **Application > Port Forwarding** in the navigation tree.

List of Port Forwarding

To add a port forwarding rule, click **Add Entry**. To edit a port forwarding rule, select it in the list and then click the pencil icon. To remove a port forwarding rule, click the delete icon. For more information, see [Manually Adding Port Forwarding, page 46](#).

Field	Description
Number	An identification number for the port forwarding rule.
Type	The type of rule: Single Port Forwarding or Port Range Forwarding.
Status	The status of the rule: Enabled or Disabled.
Application	The application that uses this rule to access a network resource.

Port Forwarding Details

To display the details, click an entry in the *List of Port Forwarding*.

Field	Description
External Port	The port that external clients will use to set up this connection.
Internal Port	The port that the ATA uses when forwarding traffic to the internal server.
Protocol	The protocol that is used: TCP or UDP.
IP Address	The IP address of the internal server that is accessed by this rule.

Manually Adding Port Forwarding

Use this page to enter the port forwarding settings for an application.

To open this page: On the *Network Setup > Application > Port Forwarding* page, click the **Add Entry** button or the pencil icon.

Enter the settings as described below. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

Field	Description
Port Forwarding Type	Choose the type of port forwarding: <ul style="list-style-type: none">▪ Single Port Forwarding: Forwards traffic for a specified port to the same or an alternative port on the target server in the LAN.▪ Port Range Forwarding: Forwards traffic to a range of ports to the same ports on the target server in the LAN. see the Internet application's documentation for the required ports or ranges.
Application Name	For single port forwarding, choose a common application from the drop-down list (such as Telnet, or DNS). To add an application that is not on the list, choose Add a new name , and then enter the name in the Enter a Name field.
Enter a Name	If you chose Port Range Forwarding, or if you chose Add a new name in the <i>Application Name</i> list for Single Port Forwarding, enter a name to identify the application.

Field	Description
External Port, Internal Port	<p>For Single Port Forwarding, specify the ports to use. For simplicity, the internal and external port numbers will often be the same. However, different external port numbers could be used to differentiate traffic of the same application type intended for different internal servers, or to promote privacy through the use of non-standard ports.</p> <ul style="list-style-type: none"> ▪ External port: For single port forwarding, enter the port number that external clients will use to set up a connection with the internal server. ▪ Internal port: For single port forwarding, enter the port number that the ATA uses when forwarding traffic to the internal server. <p>The correct entries appear automatically if you choose a standard application from the <i>Application Name</i> list for Single Port Forwarding.</p>
Start - End Port	For Port Range Forwarding, specify the range of ports to use. Valid values are from 1 to 65535.
Protocol	Select the protocol(s) that can be forwarded: TCP, UDP, or TCP and UDP.
IP Address	<p>Enter the IP address of the local server that will receive forwarded traffic.</p> <p>To ensure correct forwarding of traffic, local servers must either be configured with a static IP address, or be assigned a reserved IP address through DHCP. Use the Interface Setup > LAN > DHCP Server page to reserve IP addresses. See Network Settings for the LAN and DHCP Server, page 34.</p>
Enabled	Check the box to enable this port forwarding rule, or uncheck the box to disable it. Default setting: Disabled

DMZ

Use the *Network Setup > Application > DMZ* page if you need to allow a local device to be exposed to the Internet for a special-purpose service.

The specified network device must have its DHCP client function disabled and must have a reserved IP address (also known as a static IP address) to ensure that it is reachable at the specified IP address. To reserve an IP address for a device, see [Network Settings for the LAN and DHCP Server, page 34](#).

NOTE A Demilitarized Zone (DMZ) is similar to Port Range Forwarding. Both features allow Internet traffic to access a resource on your private network. However, Port Range Forwarding is more secure because it only opens the ports that you specify for an application. DMZ hosting opens all the ports of one device, exposing it to the Internet.

To open this page: Click **Network Setup** on the menu bar, and then click **Application > DMZ** in the navigation tree.

Enter the settings as described below. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

Field	Description
Status	Click Enabled to enable this feature, or click Disabled to disable it. Default setting: Disabled
Private IP	Specify the local IP address of the device that can be accessed through the DMZ.

Configuring the Voice Settings

This chapter describes how to configure the voice settings and voice services for the ATA. It includes the following sections:

- **Information**
- **System**
- **SIP**
- **Provisioning**
- **Regional**
- **Line 1 Settings (PHONE Port)**
- **PSTN (LINE Port)**
- **User 1**
- **PSTN User**
- **DECT Line 1 - DECT Line 10**
- **DECT User**

NOTE For additional information, see [Appendix C, “Advanced Options for Voice Services.”](#)

Information

Use the *Voice > Information* page to view information about the ATA voice application.

To open this page: Click **Voice** on the menu bar, and then click **Information** in the navigation tree. Enter the settings as described below. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

Product Information

Field	Description
Product Name	Model number/name.
Serial Number	Product serial number.
Software Version	Software version number.
Hardware Version	Hardware version number.
MAC Address	MAC Address. For example: 8843E1657936.
Client Certificate	Status of the client certificate, which can indicate if the ATA was authorized by your ITSP.
Customization	Used for Remote Configuration by service providers who deploy the ATA to their customers. <ul style="list-style-type: none">▪ Open: Not a Remote Configuration unit. This ATA can be configured by using the configuration utility.▪ Pending: This Remote Configuration unit has not yet connected to the server to get its profile.▪ Customized: This Remote Configuration unit has received its profile from the server.

System Status

Field	Description
Current Time	Current date and time of the system; for example, 10/3/2003 16:43:00. Set the system time by using the <i>Network Setup > Time Settings</i> page.

Field	Description
Elapsed Time	Total time elapsed since the last reboot of the system; for example, 25 days and 18:12:36.
RTP Packets Sent	Total number of RTP packets sent (including redundant packets)
RTP Bytes Sent	Total number of RTP bytes sent.
RTP Packets Recv	Total number of RTP packets received (including redundant packets)
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 bytes of SIP messages sent (including retransmissions)
SIP Messages Recv	Total number of SIP messages received (including retransmissions)
SIP Bytes Recv	Total number of bytes of SIP messages received (including retransmissions)
External IP	The External IP address used for NAT mapping.

Line 1 Status

Field	Description
Hook State	The hook state of the port: On or Off.
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	The number of seconds before the next registration renewal. Indicates whether you have new voice mail waiting.
Message Waiting	Indicates Yes when a message is received.
Mapped SIP Port	Port number of the SIP port mapped by NAT.

Field	Description
Call Back Active	Indicates whether or not a call back request is in progress. Options are either yes or no .
Last Called Number	The phone number that was most recently called through this port.
Last Caller Number	The originating phone number of the call that was most recently received through this port.
Call 1 and 2 State	Indicates the state of calls, if any: <ul style="list-style-type: none">▪ Idle▪ Collecting PSTN PIN▪ Invalid PSTN PIN▪ PSTN Caller Accepted▪ Connected to PSTN
Call 1 and 2 Tone	The type of tone used by the call.
Call 1 and 2 Encoder	The codec used for encoding.
Call 1 and 2 Decoder	The codec used for decoding.
Call 1 and 2 FAX	The status of the fax passthrough mode.

Field	Description
Call 1 and 2 Type	<p>The direction of the call. May take one of the following values:</p> <ul style="list-style-type: none"> ▪ PSTN Gateway Call = VoIP-To-PSTN Call ▪ VoIP Gateway Call = PSTN-To-VoIP Call ▪ PSTN To Line 1 = PSTN call ring through and answered by Line 1 ▪ Line 1 Forward to PSTN Gateway = VoIP calls Line 1 then forwarded to PSTN GW ▪ Line 1 Forward to PSTN Number =VoIP calls Line 1 then forwarded to PSTN number ▪ Line 1 To PSTN Gateway ▪ Line 1 Fallback To PSTN Gateway
Call 1 and 2 Remote Hold	Indicates whether the far end has placed the call on hold.
Call 1 and 2 Callback	Indicates whether the call was triggered by a call back request.
Call 1 and 2 Peer Name	The name of the peer phone.
Call 1 and 2 Peer Phone	The phone number of the peer phone.
Call 1 and 2 Call Duration	The duration of the call.
Call 1 and 2 Packets Sent	The number of packets sent
Call 1 and 2 Packets Recv	The number of packets received.
Call 1 and 2 Bytes Sent	The number of bytes sent.
Call 1 and 2 Bytes Recv	The number of bytes received.

Field	Description
Call 1 and 2 Decode Latency	The number of milliseconds for decoder latency.
Call 1 and 2 Jitter	The number of milliseconds for receiver jitter
Call 1 and 2 Round Trip Delay	The number of milliseconds for delay.
Call 1 and 2 Packets Lost	The number of packets lost.
Call 1 and 2 Packet Error	The number of invalid packets received.

Custom CA Status

Field	Description
Custom CA Provisioning Status	The status of the latest custom CA (Certificate Authority) certificate download.
Custom CA Info	The successfully downloaded CA information, or "Not Installed" if no custom CA certificate was installed. Default setting: Not Installed

PSTN Line Status

Field	Description
Hook State	The hook state of the LINE port: 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	The last date and time when the line was registered.
Next Registration In	The number of seconds before the next registration renewal.

Field	Description
Last Called VoIP Number	The VoIP phone number that was most recently called through this port.
Last Called PSTN Number	The PSTN phone number that was most recently called through the LINE port
Last VoIP Caller	The originating phone number of the VoIP call that was most recently received through the LINE port.
Last PSTN Caller	The originating phone number of the PSTN call that was most recently received through the LINE port.
Last PSTN Disconnect Reason	<p>The reason for the ATA hanging up the LINE port:</p> <ul style="list-style-type: none"> ▪ PSTN Disconnect Tone ▪ PSTN Activity Timeout ▪ CPC Signal ▪ Polarity Reversal ▪ VoIP Call Failed ▪ VoIP Call Ended ▪ Invalid VoIP Destination ▪ Invalid PIN ▪ PIN Digit Timeout ▪ VoIP Dialing Timeout ▪ PSTN Gateway Call Timeout ▪ VoIP Gateway Call Timeout
PSTN Activity Timer	The time in milliseconds (ms) before the ATA disconnects the current gateway unless the PSTN side has some audio activity.
Mapped SIP Port	The port number of the SIP port mapped by NAT.

Field	Description
Call Type	<p>The type of call:</p> <ul style="list-style-type: none"> ▪ PSTN Gateway Call = VoIP-To-PSTN Call ▪ VoIP Gateway Call = PSTN-To-VoIP Call ▪ PSTN To Line 1 = PSTN call ring through and answered by Line 1 ▪ Line 1 Forward to PSTN Gateway = VoIP calls Line 1 then forwarded to PSTN GW ▪ Line 1 Forward to PSTN Number =VoIP calls Line 1 then forwarded to PSTN number ▪ Line 1 To PSTN Gateway ▪ Line 1 Fallback To PSTN Gateway
VoIP State	<p>May take one of the following values:</p> <p>Idle</p> <ul style="list-style-type: none"> ▪ Collecting PSTN Pin ▪ Invalid PSTN PIN ▪ PSTN Caller Accepted ▪ Connected to PSTN
PSTN State	<p>May take one of the following values:</p> <p>Idle</p> <ul style="list-style-type: none"> ▪ Collecting PSTN Pin ▪ Invalid PSTN PIN ▪ PSTN Caller Accepted ▪ Connected to PSTN
VoIP Tone	The tone that is being played to the VoIP call leg.
PSTN Tone	The tone that is being played to the PSTN call leg.
VoIP Peer Name	The of the party at the VoIP call leg.

Field	Description
PSTN Peer Name	The name of the party at the PSTN call leg.
VoIP Peer Number	The phone number of the party at the VoIP call leg.
PSTN Peer Number	The phone number of the party at the PSTN call leg.
VoIP Call Encoder	The audio encoder being used for the VoIP call leg.
VoIP Call Decoder	The audio decoder being used for the VoIP call leg.
VoIP Call FAX	The status of the fax passthrough mode for VoIP calls.
VoIP Call Remote Hold	Indicates whether the far end has placed the VoIP call on hold.
VoIP Call Duration	The duration of the VoIP call.
VoIP Call Packets Sent	The number of packets sent for VoIP calls.
VoIP Call Packets Recv	The number of packets received for VoIP calls.
VoIP Call Bytes Sent	The number of bytes sent for VoIP calls.
VoIP Call Bytes Recv	The number of bytes received for VoIP calls.
VoIP Call Decode Latency	The number of milliseconds for decoder latency for VoIP calls.
VoIP Call Jitter	The number of milliseconds for receiver jitter for VoIP calls.
VoIP Call Round Trip Delay	The number of milliseconds for delay for VoIP calls.
VoIP Call Packets Lost	The number of packets lost for VoIP calls.
VoIP Call Packet Error	The number of invalid packets received for VoIP calls.
VoIP Call Mapped RTP Port	The port mapped for Real Time Protocol traffic for VoIP calls.

DECT 1 ~ DECT 10 Status

Field	Description
Registration State	Indicates whether or not the line has registered with the SIP proxy: Registered, Not Registered, or Failed.
Last Registration At	The last date and time when the line was registered.
Next Registration In	The number of seconds before the next registration renewal.
Message Waiting	Indicates whether or not there are new messages: yes or no . The value automatically is set to yes when a message is received. You also can clear or set the flag manually from the <i>User 1</i> page.
Call Back Active	Indicates whether a call back request is in progress: yes or no .
Last Called Number	The phone number that was most recently called through this port.
Last Caller Number	The originating phone number of the call that was most recently received through this port.
Mapped SIP Port	Port number of the SIP port mapped by NAT.
Call 1 and 2 State	The current call state: <ul style="list-style-type: none"> ▪ Idle ▪ Collecting PSTN Pin ▪ Invalid PSTN PIN ▪ PSTN Caller Accepted ▪ Connected to PSTN
Call 1 and 2 Tone	The type of tone used by the call.
Call 1 and 2 Encoder	The codec used for encoding.
Call 1 and 2 Decoder	The codec used for decoding.

Field	Description
Call 1 and 2 FAX	The status of the fax passthrough mode.
Call 1 and 2 Type	<p>The direction of the call:</p> <ul style="list-style-type: none"> ▪ PSTN Gateway Call = VoIP-To-PSTN Call ▪ VoIP Gateway Call = PSTN-To-VoIP Call ▪ PSTN To Line 1 = PSTN call ring through and answered by Line 1 ▪ Line 1 Forward to PSTN Gateway = VoIP calls Line 1 then forwarded to PSTN GW ▪ Line 1 Forward to PSTN Number =VoIP calls Line 1 then forwarded to PSTN number ▪ Line 1 To PSTN Gateway ▪ Line 1 Fallback To PSTN Gateway
Call 1 and 2 Remote Hold	Indicates whether the far end has placed the call on hold.
Call 1 and 2 Callback	Indicates whether the call was triggered by a call back request.
Call 1 and 2 Peer Name	The name of the peer phone.
Call 1 and 2 Peer Phone	The phone number of the peer phone.
Call 1 and 2 Call Duration	The duration of the call.
Call 1 and 2 Packets Sent	The number of packets sent.
Call 1 and 2 Packets Recv	The number of packets received.
Call 1 and 2 Bytes Sent	The number of bytes sent.
Call 1 and 2 Bytes Recv	The number of bytes received.

Field	Description
Call 1 and 2 Decode Latency	The number of milliseconds for decoder latency.
Call 1 and 2 Jitter	The number of milliseconds for receiver jitter
Call 1 and 2 Round Trip Delay	The number of milliseconds for delay.
Call 1 and 2 Packets Lost	The number of packets lost.
Call 1 and 2 Packet Error	The number of invalid packets received.
Call 1 and 2 Mapped RTP Port	The port mapped for Real Time Protocol traffic for Call 1/2.

DECT Handset 1 ~ DECT Handset 10 Status

Field	Description
Handset IPEI	The unique hardware identifier of the unit, comparable to a MAC address.
Model Number	The Cisco model number of the unit.

System

Use the *Voice > System* page to configure general voice system settings and to enable logging by using a syslog server. (Logging also can be configured in the *Administration > Logging* pages. For more information, see [Logging, page 186](#).)

To open this page: Click **Voice** on the menu bar, and then click **System** in the navigation tree. Enter the settings as described below. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

Requirements for Logging

- You need a computer that is on the same subnetwork as the ATA, to capture the log files. This computer needs to be running a syslog daemon. Enter the IP address of this computer in the Syslog Server and Debug Server fields.

- You can deploy a syslog server to receive syslog messages from the ATA, which acts as a syslog client. The syslog client device uses the syslog protocol to send messages, based on its configuration, to a syslog server. The syslog messages can be accessed by reviewing the "syslog.514.log" file which resides in the same directory as the slogsrv.exe syslog server application.
- Partners can download the Syslog Server for SPA Devices by using the link below (login required):
www.cisco.com/en/US/partner/prod/collateral/voicesw/ps6788/phones/ps10499/syslog_server_for_spa_devices.zip

Enter the settings as described below. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

System Configuration

Field	Description
Restricted Access Domains	Domain of the service provider to which the ATA is connected to. It prevents the ATA from connecting to other service providers.
IVR Admin Password	Password for the administrator to manage the ATA by using the built-in IVR through a connected phone. NOTE The registered HS can also manage the ATA or deregister other HS with this password.
Network Startup Delay	The number of seconds of delay between restarting the voice module and initializing network interface. Default setting: 3

Field	Description
Handset (HS) Pairing Password	<p>Password used as self authentication for Handset registration and deregistration.</p> <p>Default setting: blank</p> <p>NOTE There are 3 options in Handset registration:</p> <ul style="list-style-type: none"> Register: For an unregistered HS, select Register and enter the Handset Pairing Passwd for authentication. Deregister: For a registered HS, select Deregister and enter the Handset Pairing Passwd for authentication. Use select softkey to deregister the HS. Deregister Others: For a registered HS, select Deregister Others. Enter <IVR Admin Passwd> for authentication. Use select softkey to deregister others.

Miscellaneous Settings

Field	Description
DNS Query TTL Ignore	<p>In DNS packages, the server will suggest a TTL value to the client; if this parameter is set to yes, the value from the server will be ignored.</p> <p>Default setting: yes</p>
Syslog Server	<p>Specify the syslog server name and port. This feature specifies the server for logging ATA system information and critical events. If both Debug Server and Syslog Server are specified, Syslog messages are also logged to the Debug Server.</p> <p>Default setting: blank</p>
Syslog Server Transport	<p>The Syslog messages can be printed to the server with the selected transport method (UDP or TLS)</p> <p>Default setting: UDP</p>
Debug Server	<p>The debug server name and port. This feature specifies the server for logging debug information. The level of detailed output depends on the debug level parameter setting.</p> <p>Default setting: blank</p>

Field	Description
Debug Server Transport	<p>The Debug information can be printed to the server with the selected transport method (UDP and TLS.)</p> <p>Default setting: UDP</p>
Debug Level	<p>Determines the level of debug information that will be generated. Select 0, 1, 2, 3 or 3+Router from the drop-down list. The higher the debug level, the more debug information will be generated. Level 0 means that no information will be collected. Levels 1, 2 & 3 generate messages related to the voice ports only. Level 3+Router generates debug content for both voice and router components.</p> <p>Default setting: 3</p>
DevTest Password	<p>Used for internal automation test; not for the user.</p>
Syslog Prefix	<p>Allows the user to prefix additional information to syslog.</p> <p>Default setting: 215</p>
DECT Codec Change	<p>Codec change between handset (HS) and base to match RTP codec.</p> <p>Default setting: yes</p>

SIP

Use the *Voice > SIP* page to configure SIP parameters and values.

To open this page: Click **Voice** on the menu bar, and then click **SIP** in the navigation tree. Enter the settings as described below. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

NOTE For a deeper understanding of these fields, refer to Request for Comments (RFC) 3261.

SIP Parameters

Field	Description
Max Forward	The maximum times a call can be forwarded. The valid range is from 1 to 255. Default setting: 70
Max Redirection	Number of times an invite can be redirected to avoid an infinite loop. Default setting: 5.
Max Auth	The maximum number of times (from 0 to 255) a request may be challenged. Default setting: 2
SIP User Agent Name	The User-Agent header used in outbound requests. If empty, the header is not included. Macro expansion of \$A to \$D corresponding to GPP_A to GPP_D allowed. Default setting: \$VERSION
SIP Server Name	The server header used in responses to inbound responses. Default setting: \$VERSION
SIP Reg User Agent Name	The User-Agent name to be used in a REGISTER request. If this value is not specified, the SIP User Agent Name parameter is also used for the REGISTER request. Default setting: blank

Field	Description
SIP Accept Language	Accept-Language header used. There is no default (this indicates that the ATA does not include this header) If empty, the header is not included. Default setting: blank
DTMF Relay MIME Type	The MIME Type used in a SIP INFO message to signal a DTMF event. Default setting: application/dtmf-relay.
Hook Flash MIME Type	The MIME Type used in a SIP INFO message to signal a hook flash event. Default setting: application/hook-flash
Remove Last Reg	Determines whether or not the ATA removes the last registration before submitting a new one, if the value is different. Select yes to remove the last registration, or select no to omit this step. Default setting: no
Use Compact Header	Determines whether or not the ATA uses compact SIP headers in outbound SIP messages. Select yes or no from the drop-down list. Select yes to use compact SIP headers in outbound SIP messages. Select no to use normal SIP headers. If inbound SIP requests contain compact headers, the 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, the ATA substitutes those headers with compact headers (if defined by RFC 261) if Use Compact Header parameter is set to yes. Default setting: no
Escape Display Name	Determines whether or not the Display Name is private. Select yes if you want the ATA to enclose the string (configured in the Display Name) in a pair of double quotes for outbound SIP messages. If the display name includes " or \, these will be escaped to \" and \\ within the double quotes. Otherwise, select no. Default setting: no

Field	Description
RFC 2543 Call Hold	Configures the type of call hold: a:sendonly or 0.0.0.0. Do not use the 0.0.0.0 syntax in a HOLD SDP; use the a:sendonly syntax. Default setting: no
Mark all AVT Packets	Select yes if you want all AVT tone packets (encoded for redundancy) to have the marker bit set for each DTMF event. Select no to have the marker bit set only for the first packet. Default setting: yes
SIP TCP Port Min	The lowest TCP port number that can be used for SIP sessions. Default setting: 5060
SIP TCP Port Max	The highest TCP port number that can be used for SIP sessions. Default setting: 5080
CTI Enable	Enables or disables the Computer Telephone Interface feature provided by some servers. Default setting: no

SIP Timer Values

Field	Description
SIP T1	RFC 3261 T1 value (round-trip time estimate), which can range from 0 to 64 seconds. Default setting: 0.5
SIP T2	RFC 3261 T2 value (maximum retransmit interval for non-INVITE requests and INVITE responses), which can range from 0 to 64 seconds. Default setting: 4
SIP T4	RFC 3261 T4 value (maximum duration a message remains in the network), which can range from 0 to 64 seconds. Default setting: 5
SIP Timer B	INVITE time-out value, which can range from 0 to 64 seconds. Default setting: 32

Field	Description
SIP Timer F	Non-INVITE time-out value, which can range from 0 to 64 seconds. Default setting: 32
SIP Timer H	H INVITE final response, time-out value, which can range from 0 to 64 seconds. Default setting: 32
SIP Timer D	ACK hang-around time, which can range from 0 to 64 seconds. Default setting: 32
SIP Timer J	Non-INVITE response hang-around time, which can range from 0 to 64 seconds. Default setting: 32
INVITE Expires	INVITE request Expires header value. If you enter 0, the Expires header is not included in the request. Range: 0–($2^{31}-1$) Default setting: 240
ReINVITE Expires	ReINVITE request Expires header value. If you enter 0, the Expires header is not included in the request. Range: 0–($2^{31}-1$) Default setting: 30
Reg Min Expires	Minimum registration expiration time allowed from the proxy in the Expires header or as a Contact header parameter. If the proxy returns a value less than this setting, the minimum value is used. Default setting: 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. Default setting: 7200
Reg Retry Intvl	Interval to wait before the ATA retries registration after failing during the last registration. Default setting: 30

Field	Description
Reg Retry Long Intvl	When registration fails with a SIP response code that does not match Retry Reg RSC, the ATA waits for the specified length of time before retrying. If this interval is 0, the ATA stops trying. This value should be much larger than the Reg Retry Intvl value, which should not be 0. Default setting: 1200
Reg Retry Random Delay	Random delay range (in seconds) to add to Register Retry Intvl when retrying REGISTER after a failure. Default setting: 0 (disabled)
Reg Retry Long Random Delay	Random delay range (in seconds) to add to Register Retry Long Intvl when retrying REGISTER after a failure. Default setting: 0 (disabled)
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. Default setting: 0, which disables the exponential backoff 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. Default setting: blank
SIT2 RSC	SIP response status code to INVITE on which to play the SIT2 Tone. Default setting: blank

Field	Description
SIT3 RSC	SIP response status code to INVITE on which to play the SIT3 Tone. Default setting: blank
SIT4 RSC	SIP response status code to INVITE on which to play the SIT4 Tone. Default setting: blank
Try Backup RSC	SIP response code that retries a backup server for the current request. Default setting: blank
Retry Reg RSC	Interval to wait before the ATA retries registration after failing during the last registration. Default setting: blank

RTP Parameters

Field	Description
RTP Port Min	Minimum port number for RTP transmission and reception. The RTP Port Min and RTP Port Max parameters should define a range that contains at least 4 even number ports, such as 100 –106. Default setting: 16384.
RTP Port Max	Maximum port number for RTP transmission and reception. Default setting: 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. Default setting: 0.030
Max RTP ICMP Err	Number of successive ICMP errors allowed when transmitting RTP packets to the peer before the ATA terminates the call. If value is set to 0, the ATA ignores the limit on ICMP errors. Default setting: 0

Field	Description
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 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. The NTP timestamp used in the SR is a snapshot of the local time for the ATA, not the time reported by an NTP server. If the 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) on the <i>Information</i> page. Default setting: 0
No UDP Checksum	Select yes if you want the ATA to calculate the UDP header checksum for SIP messages. Otherwise, select no. Default setting: no
Stats In BYE	Determines whether the ATA includes the P-RTP-Stat header or response in a BYE message. The header contains the RTP statistics of the current call. Select yes or no from the drop-down list. Default setting: yes 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>,JL=<jitter in ms>,LA=<delay in ms>,DU=<call duration ins>,EN=<encoder>,DE=<decoder>.

SDP Payload Types

Field	Description
NSE Dynamic Payload	NSE dynamic payload type. The valid range is 96-127. Default setting: 100
AVT Dynamic Payload	AVT dynamic payload type. The valid range is 96-127. Default setting: 101
INFOREQ Dynamic Payload	INFOREQ dynamic payload type. Default setting: blank
G726r32 Dynamic Payload	G726r32 dynamic payload type. Default setting: 2
G729b Dynamic Payload	G.729b dynamic payload type. The valid range is 96-127. Default setting: 99
EncapRTP Dynamic Payload	EncapRTP Dynamic Payload type. Default setting: 112
RTP-Start-Loopback Dynamic Payload	RTP-Start-Loopback Dynamic Payload type. Default setting: 113
RTP-Start-Loopback Codec	RTP-Start-Loopback Codec. Select one of the following: G711u, G711a, G726-32, G729a. Default setting: G711u
NSE Codec Name	NSE codec name used in SDP. Default setting: NSE
AVT Codec Name	AVT codec name used in SDP. Default setting: telephone-event
G711u Codec Name	G.711u codec name used in SDP. Default setting: PCMU
G711a Codec Name	G.711a codec name used in SDP. Default setting: PCMA
G726r32 Codec Name	G.726-32 codec name used in SDP. Default setting: G726-32
G729a Codec Name	G.729a codec name used in SDP. Default setting: G729a

Field	Description
G729b Codec Name	G.729b codec name used in SDP. Default setting: G729ab
EncapRTP Codec Name	EncapRTP codec name used in SDP. Default setting: encaprtpt

NAT Support Parameters

Field	Description
Handle VIA received	If you select yes, the ATA processes the received parameter in the VIA header (this value is inserted by the server in a response to any one of its requests) If you select no, the parameter is ignored. Select yes or no from the drop-down menu. Default setting: no
Handle VIA rport	If you select yes, the ATA processes the rport parameter in the VIA header (this value is inserted by the server in a response to any one of its requests) If you select no, the parameter is ignored. Select yes or no from the drop-down menu. Default setting: 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. Default setting: no
Insert VIA rport	Inserts the 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. Default setting: 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. Default setting: 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. Default setting: no

Field	Description
STUN Enable	Enables the use of STUN to discover NAT mapping. Select yes or no from the drop-down menu. Default setting: no
STUN Test Enable	If the STUN Enable feature is enabled and a valid STUN server is available, the 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 ATA detects symmetric NAT or a symmetric firewall, NAT mapping is disabled. Default setting: no
STUN Server	IP address or fully-qualified domain name of the STUN server to contact for NAT mapping discovery. Default setting: blank
EXT IP	<p>External IP address to substitute for the actual IP address of the ATA in all outgoing SIP messages. If 0.0.0.0 is specified, no IP address substitution is performed.</p> <p>If this parameter is specified, the 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.</p> <p>This option requires that you have (1) a static IP address from your Internet Service Provider and (2) an edge device with a symmetric NAT mechanism. If the ATA is the edge device, the second requirement is met. Default setting: blank</p>
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. Default setting: blank
NAT Keep Alive Intvl	Interval between NAT-mapping keep alive messages. Default setting: 15
Redirect Keep Alive	Interval between NAT Redirect keep alive messages. Default setting: 15

Linksys Key System Parameters

Field	Description
Linksys Key System	To enable operation with the Cisco SPA9000, choose yes. Otherwise, choose no. Default setting: no
Multicast Address	The multicast address for devices in the Cisco SPA9000 voice network. Default setting: 224.168.168.168:6061
Key System Auto Discovery	To enable auto-discovery of the Cisco SPA9000 voice system, choose yes. Otherwise, choose no. Default setting: yes
Key System IP Address	The IP address of the Cisco SPA9000. Default setting: blank
Force LAN Codec	If needed, specify a voice codec. Default setting: none

Provisioning

Use the *Voice > Provisioning* page to configure profiles and parameters to provision the ATA from a remote server.

NOTE Cisco SPA122, SPA112, and SPA232D supports 302/310 redirect while provisioning.

To open this page: Click **Voice** on the menu bar, and then click **Provisioning** in the navigation tree. Enter the settings as described below. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

Configuration Profile

Field	Description
Provision Enable	Controls all resync actions independently of firmware upgrade actions. Set to yes to enable remote provisioning. Default setting: yes
Resync On Reset	Triggers a resync after every reboot except for reboots caused by parameter updates and firmware upgrades. Default setting: yes
Resync Random Delay	<p>The maximum value for a random time interval that the ATA waits before making its initial contact with the provisioning server. This delay is effective only on the initial configuration attempt following power-on or reset. The delay is a pseudo-random number between zero and this value.</p> <p>This parameter is in units of 20 seconds; the default value of 2 represents 40 seconds. This feature is disabled when this parameter is set to zero.</p> <p>This feature can be used to prevent an overload of the provisioning server when a large number of devices power-on simultaneously. Default setting: 2 (40 seconds)</p>
Resync At (HHmm)	The time of day when the device tries to resync. The resync is performed each day. Used in conjunction with the Resync At Random Delay. Default setting: blank
Resync At Random Delay	Used in conjunction with the Resync At (HHmm) setting, this parameter sets a range of possible values for the resync delay. The system randomly chooses a value from this range and waits the specified number of seconds before attempting to resync. This feature is intended to prevent the network jam that would occur if all resynchronizing devices began the resync at the exact same time of day. Default setting: 600

Field	Description
Resync Periodic	<p>The time interval between periodic resyncs with the provisioning server. The associated resync timer is active only after the first successful synchronization with the server. Setting this parameter to zero disables periodic resynchronization.</p> <p>Default setting: 3600 seconds</p>
Resync Error Retry Delay	<p>Resync retry interval (in seconds) applied in case of resync failure.</p> <p>The ATA has an error retry timer that activates if the previous attempt to sync with the provisioning server fails. The ATA waits to contact the server again until the timer counts down to zero.</p> <p>This parameter is the value that is initially loaded into the error retry timer. If this parameter is set to zero, the ATA immediately retries to sync with the provisioning server following a failed attempt.</p> <p>Default setting: 3600 seconds</p>
Forced Resync Delay	<p>Maximum delay (in seconds) that the ATA waits before performing a resync.</p> <p>The ATA does not resync while one of its lines is active. Because a resync can take several seconds, it is desirable to wait until the ATA has been idle for an extended period before resynchronizing. This allows a user to make calls in succession without interruption.</p> <p>The ATA has a timer that begins counting down when all of its lines become idle. This parameter is the initial value of the counter. Resync events are delayed until this counter decrements to zero.</p> <p>Default setting: 14400 seconds</p>
Resync From SIP	<p>Enables a resync to be triggered via a SIP NOTIFY message.</p> <p>Default setting: yes</p>
Resync After Upgrade Attempt	<p>Triggers a resync after every firmware upgrade attempt.</p> <p>Default setting: yes</p>

Field	Description
Resync Trigger 1 Resync Trigger 2	Configurable resync trigger conditions. A resync is triggered when the logic equation in these parameters evaluates to TRUE. Default setting: blank
Resync Fails On FNF	Determines whether a file-not-found response from the provisioning server constitutes a successful or a failed resync. A failed resync activates the error resync timer. Default setting: yes
Profile Rule	This parameter is a profile script that evaluates to the provisioning resync command. The command is a TCP/IP operation and an associated URL. The TCP/IP operation can be TFTP, HTTP, or HTTPS. If the command is not specified, TFTP is assumed, and the address of the TFTP server is obtained through DHCP option 66. In the URL, either the IP address or the FQDN of the server can be specified. The file name can have macros, such as \$MA, which expands to the ATA MAC address. Default setting: /spa\$PSN.cfg
Profile Rule B: Profile Rule C: Profile Rule D	Defines second, third, and fourth resync commands and associated profile URLs. These profile scripts are executed sequentially after the primary Profile Rule resync operation has completed. If a resync is triggered and Profile Rule is blank, Profile Rule B, C, and D are still evaluated and executed. Default setting: blank
Log Resync Request Msg	This parameter contains the message that is sent to the Syslog server at the start of a resync attempt. Default setting: \$PN \$MAC -- Requesting resync \$SCHEME://\$SERVIP:\$PORT\$PATH
Log Resync Success Msg	Syslog message issued upon successful completion of a resync attempt. Default setting: \$PN \$MAC -- Successful resync \$SCHEME://\$SERVIP:\$PORT\$PATH
Log Resync Failure Msg	Syslog message issued after a failed resync attempt. Default setting: \$PN \$MAC -- Resync failed: \$ERR

Field	Description
Report Rule	<p>The target URL to which configuration reports are sent. This parameter has the same syntax as the Profile_Rule parameter, and resolves to a TCP/IP command with an associated URL.</p> <p>A configuration report is generated in response to an authenticated SIP NOTIFY message, with Event: report. The report is an XML file containing the name and value of all the device parameters.</p> <p>This parameter may optionally contain an encryption key. For example:</p> <pre>[--key \$K] tftp://ps.callhome.net/\$MA/rep.xml.enc</pre> <p>Default setting: blank</p>

Firmware Upgrade

Field	Description
Upgrade Enable	<p>Determines whether or not firmware upgrade operations can occur independently of resync actions.</p> <p>Default setting: yes</p>
Upgrade Error Retry Delay	<p>The upgrade retry interval (in seconds) applied in case of upgrade failure. The ATA has a firmware upgrade error timer that activates after a failed firmware upgrade attempt. The timer is initialized with the value in this parameter. The next firmware upgrade attempt occurs when this timer counts down to zero.</p> <p>Default setting: 3600 seconds</p>
Downgrade Rev Limit	<p>Enforces a lower limit on the acceptable version number during a firmware upgrade or downgrade. The ATA does not complete a firmware upgrade operation unless the firmware version is greater than or equal to this parameter.</p> <p>Default setting: blank</p>
Upgrade Rule	<p>This parameter is a firmware upgrade script with the same syntax as Profile_Rule. Defines upgrade conditions and associated firmware URLs.</p> <p>Default setting: blank</p>

Field	Description
Log Upgrade Request Msg	Syslog message issued at the start of a firmware upgrade attempt. Default setting: \$PN \$MAC -- Requesting upgrade \$SCHEME://\$SERVIP:\$PORT\$PATH
Log Upgrade Success Msg	Syslog message issued after a firmware upgrade attempt completes successfully. Default setting: \$PN \$MAC -- Successful upgrade \$SCHEME://\$SERVIP:\$PORT\$PATH -- \$ERR
Log Upgrade Failure Msg	Syslog message issued after a failed firmware upgrade attempt. Default setting: \$PN \$MAC -- Upgrade failed: \$ERR
License Keys	This field is not currently used.

CA Settings

Field	Description
Custom CA URL	The URL of a file location for a custom Certificate Authority (CA) certificate. Either the IP address or the FQDN of the server can be specified. The file name can have macros, such as \$MA, which expands to the ATA MAC address. Default setting: null

General Purpose Parameters

Field	Description
GPP A to GPP P	General purpose provisioning parameters. These parameters can be used as variables in provisioning and upgrade rules. They are referenced by prepending the variable name with a '\$' character, such as \$GPP_A. Default setting: blank

Regional

Use the *Voice > Regional* page to localize your system with the appropriate regional settings.

To open this page: Click **Voice** on the menu bar, and then click **Region** in the navigation tree. Enter the settings as described below. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

Defining Ring and Cadence and Tone Scripts

To define ring and tone patterns, the ATA uses the concept of scripts. Below is information about creating Cadence Scripts (CadScripts), Frequency Scripts (FreqScripts), and Tone Scripts (ToneScripts).

NOTE Total tone length is not configurable.

CadScript

A mini-script of up to 127 characters that specifies the cadence parameters of a signal.

Syntax: $S_1[S_2]$, where:

$S_i = D_i(\text{on}_{i,1}/\text{off}_{i,1}[\text{on}_{i,2}/\text{off}_{i,2}[\text{on}_{i,3}/\text{off}_{i,3}[\text{on}_{i,4}/\text{off}_{i,4}[\text{on}_{i,5}/\text{off}_{i,5}[\text{on}_{i,6}/\text{off}_{i,6}]]]])$ and is known as a section, $\text{on}_{i,j}$ and $\text{off}_{i,j}$ are the on/off duration in seconds of a *segment* and $i = 1$ or 2 , and $j = 1$ to 6 . D_i is the total duration of the section in seconds. All durations can have up to three decimal places to provide 1 ms resolution. The wildcard character "*" represents infinite duration. The segments within a section are played in order and repeated until the total duration is played.

Example 1: 60(2/4)

```
Number of Cadence Sections = 1
Cadence Section 1: Section Length = 60 s
Number of Segments = 1
Segment 1: On=2s, Off=4s
Total Ring Length = 60s
```

Example 2—Distinctive ring (short,short,short,long): 60(.2/.2,.2/.2,.2/.2,1/4)

```
Number of Cadence Sections = 1
Cadence Section 1: Section Length = 60s
Number of Segments = 4
Segment 1: On=0.2s, Off=0.2s
Segment 2: On=0.2s, Off=0.2s
```

```
Segment 3: On=0.2s, Off=0.2s
Segment 4: On=1.0s, Off=4.0s
Total Ring Length = 60s
```

FreqScript

A mini-script of up to 127 characters that specifies the frequency and level parameters of a tone.

Syntax: $F_1@L_1[,F_2@L_2[,F_3@L_3[,F_4@L_4[,F_5@L_5[,F_6@L_6]]]]]$

Where F_1 – F_6 are frequency in Hz (unsigned integers only) and L_1 – L_6 are corresponding levels in dBm (with up to 1 decimal places) White spaces before and after the comma are allowed (but not recommended)

Example 1—Call Waiting Tone: 440@-10

```
Number of Frequencies = 1
Frequency 1 = 440 Hz at -10 dBm
```

Example 2—Dial Tone: 350@-19,440@-19

```
Number of Frequencies = 2
Frequency 1 = 350 Hz at -19 dBm
Frequency 2 = 440 Hz at -19 dBm
```

ToneScript

A mini-script of up to 127 characters that specifies the frequency, level and cadence parameters of a call progress tone. May contain up to 127 characters.

Syntax: FreqScript; $Z_1[Z_2]$.

The section Z_1 is similar to the S_1 section in a CadScript except that each on/off segment is followed by a frequency components parameter: $Z_1 = D_1(\text{on}_{i,1}/\text{off}_{i,1}/f_{i,1}, \text{on}_{i,2}/\text{off}_{i,2}/f_{i,2}, [\text{on}_{i,3}/\text{off}_{i,3}/f_{i,3}, \text{on}_{i,4}/\text{off}_{i,4}/f_{i,4}, [\text{on}_{i,5}/\text{off}_{i,5}/f_{i,5}, \text{on}_{i,6}/\text{off}_{i,6}/f_{i,6}]]])$, where $f_{i,j} = n_1[+n_2]+n_3[+n_4[+n_5[+n_6]]])$ and $1 < n_k < 6$ indicates which of the frequency components given in the FreqScript are used in that segment; if more than one frequency component is used in a segment, the components are summed together.

Example 1—Dial tone: 350@-19,440@-19;10(*0/1+2)

```
Number of Frequencies = 2
Frequency 1 = 350 Hz at -19 dBm
Frequency 2 = 440 Hz at -19 dBm
Number of Cadence Sections = 1
Cadence Section 1: Section Length = 10 s
Number of Segments = 1
Segment 1: On=forever, with Frequencies 1 and 2
Total Tone Length = 10s
```

Example 2—Stutter tone: 350@-19,440@-19;2(.1/1+2);10(*0/1+2)

```

Number of Frequencies = 2
  Frequency 1 = 350 Hz at -19 dBm
  Frequency 2 = 440 Hz at -19 dBm
Number of Cadence Sections = 2
  Cadence Section 1: Section Length = 2s
    Number of Segments = 1
      Segment 1: On=0.1s, Off=0.1s with Frequencies 1 and 2
  Cadence Section 2: Section Length = 10s
    Number of Segments = 1
      Segment 1: On=forever, with Frequencies 1 and 2
Total Tone Length = 12s

```

Enter the settings as described below. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

Call ProgressTones

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. Default setting: 350@-5,440@-5;10(*0/1+2)
Second Dial Tone	Alternative to the Dial Tone when the user dials a three-way call. Default setting: 420@-5,520@-5;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. Default setting: 420@-4;10(*0/1)
Prompt Tone	Prompts the user to enter a call forwarding phone number. Default setting: 520@-5,620@-5;10(*0/1+2)
Busy Tone	Played when a 486 RSC is received for an outbound call. Default setting: 480@-5,620@-5;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. Default setting: 480@-5,620@-5;10(.25/.25/1+2)

Field	Description
Off Hook Warning Tone	Played when the caller has not properly placed the handset on the cradle. Off Hook Warning Tone is played when the Reorder Tone times out. Default setting: 480@-3,620@3;10(.125/.125/1+2)
Ring Back Tone	Played during an outbound call when the far end is ringing. Default setting: 440@-5,480@-5;*(2/4/1+2)
Ring Back 2 Tone	Your ATA plays this ringback tone instead of Ring Back Tone if the called party replies with a SIP 182 response without SDP to its outbound INVITE request. Default setting: the same as Ring Back Tone, except the cadence is 1s on and 1s off. Default setting: 440@-5,480@-5;*(1/1/1+2)
Confirm Tone	Brief tone to notify the user that the last input value has been accepted. Default setting: 600@-4;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. Default setting: 985@-4,1428@-4,1777@-4;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. Default setting: 914@-4,1371@-4,1777@-4;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. Default setting: 914@-4,1371@-4,1777@-4;20(.380/0/1,.380/0/2,.380/0/3,0/4/0)
SIT4 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. Default setting: 985@-4,1371@-4,1777@-4;20(.380/0/1,.274/0/2,.380/0/3,0/4/0)

Field	Description
MWI Dial Tone	Played instead of the Dial Tone when there are unheard messages in the caller's mailbox. Default setting: 350@-5,440@-5;2(.1/.1/1+2);10(*0/1+2)
Cfwd Dial Tone	Played when all calls are forwarded. Default setting: 350@-5,440@-5;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. Default setting: 600@-5;*(.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. Default setting: 350@-5;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. Default setting: 397@-5,507@-5;15(0/2/0,.2/.1/1,.1/2.1/2)
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.
Feature Invocation Tone	Played when a feature is implemented. Default setting: 350@-4;*(.1/.1/1)
Call Remind Tone	When there are 2 calls on the FXS port, and if one call is held by the UUT (Unit under test) and the other is connected, a holding tone is played on the UUT to remind the presence of the held call. Default setting: blank (the feature is not enabled)

Distinctive Ring Patterns

Field	Description
Ring1 Cadence	Cadence script for distinctive ring 1. Default setting: 60(2/4)
Ring2 Cadence	Cadence script for distinctive ring 2. Default setting: 60(.8/.4,.8/4)
Ring3 Cadence	Cadence script for distinctive ring 3. Default setting: 60(.4/.2,.4/.2,.8/4)
Ring4 Cadence	Cadence script for distinctive ring 4. Default setting: 60(.3/.2,1/.2,.3/4)
Ring5 Cadence	Cadence script for distinctive ring 5. Default setting: 1(.5/.5)
Ring6 Cadence	Cadence script for distinctive ring 6. Default setting: 60(.2/.4,.2/.4,.2/4)
Ring7 Cadence	Cadence script for distinctive ring 7. Default setting: 60(.4/.2,.4/.2,.4/4)
Ring8 Cadence	Cadence script for distinctive ring 8. Default setting: 60(0.25/9.75)

Distinctive Call Waiting Tone Patterns

Field	Description
CWT1 Cadence	Cadence script for distinctive CWT 1. Default setting: 30(.3/9.7)
CWT2 Cadence	Cadence script for distinctive CWT 2. Default setting: 30(.1/.1, .1/9.7)
CWT3 Cadence	Cadence script for distinctive CWT 3. Default setting: 30(.1/.1, .1/.1, .1/9.7)
CWT4 Cadence	Cadence script for distinctive CWT 4. Default setting: 30(.1/.1, .3/.1, .1/9.3)
CWT5 Cadence	Cadence script for distinctive CWT 5. Default setting: 1(.5/.5)

Field	Description
CWT6 Cadence	Cadence script for distinctive CWT 6. Default setting: 30(.3/.1,.3/.1,.1/9.1)
CWT7 Cadence	Cadence script for distinctive CWT 7. Default setting: 30(.3/.1,.3/.1,.1/9.1)
CWT8 Cadence	Cadence script for distinctive CWT 8. Default setting: 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. Default setting: Bellcore-r1
Ring2 Name	Name in an INVITE's Alert-Info Header to pick distinctive ring/CWT 2 for the inbound call. Default setting: Bellcore-r2
Ring3 Name	Name in an INVITE's Alert-Info Header to pick distinctive ring/CWT 3 for the inbound call. Default setting: Bellcore-r3
Ring4 Name	Name in an INVITE's Alert-Info Header to pick distinctive ring/CWT 4 for the inbound call. Default setting: Bellcore-r4
Ring5 Name	Name in an INVITE's Alert-Info Header to pick distinctive ring/CWT 5 for the inbound call. Default setting: Bellcore-r5
Ring6 Name	Name in an INVITE's Alert-Info Header to pick distinctive ring/CWT 6 for the inbound call. Default setting: Bellcore-r6
Ring7 Name	Name in an INVITE's Alert-Info Header to pick distinctive ring/CWT 7 for the inbound call. Default setting: Bellcore-r7
Ring8 Name	Name in an INVITE's Alert-Info Header to pick distinctive ring/CWT 8 for the inbound call. Default setting: Bellcore-r8

Ring and Call Waiting Tone Spec

IMPORTANT: Ring and Call Waiting tones do not work the same way on all phones. When setting ring tones, consider the following recommendations:

- Begin with the default Ring Waveform, Ring Frequency, and Ring Voltage.
- If your ring cadence doesn't sound right, or your phone doesn't ring, change the following settings:
 - Ring Waveform: Sinusoid
 - Ring Frequency: 25
 - Ring Voltage: 80Vc

Field	Description
Ring Waveform	Waveform for the ringing signal. Choices are Sinusoid or Trapezoid. Default setting: Sinusoid
Ring Frequency	Frequency of the ringing signal. Valid values are 10–100 (Hz) Default setting: 20
Ring Voltage	Ring voltage. Choices are 60–90 (V) Default setting: 85
CWT Frequency	Frequency script of the call waiting tone. All distinctive CWTs are based on this tone. Default setting: 440@-10
Synchronized Ring	If this is set to yes, when the ATA is called, all lines ring at the same time (similar to a regular PSTN line) After one line answers, the others stop ringing. Default setting: no

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. Default setting: 0.1

Field	Description
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. Default setting: 0.9
Callee On Hook Delay	Phone must be on-hook for at this time in sec. before the ATA will tear down the current inbound call. It does not apply to outbound calls. Range: 0–255 seconds. Default setting: 0
Reorder Delay	Delay after far end hangs up before reorder tone is played. 0 = plays immediately, inf = never plays. Range: 0–255 seconds. Default setting: 5.
Call Back Expires	Expiration time in seconds of a call back activation. Range: 0–65535 seconds. Default setting: 1800
Call Back Retry Intvl	Call back retry interval in seconds. Range: 0–255 seconds. Default setting: 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 ATA still considers the call as failed and keeps on retrying. Default setting: 0.5
VMWI Refresh Intvl	Interval between VMWI refresh to the device. Default setting: 0
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. Default setting: 10

Field	Description
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. Default setting: 3
CPC Delay	Delay in seconds after caller hangs up when the ATA starts removing the tip-and-ring voltage to the attached equipment of the called party. The range is 0–255 seconds. 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 be 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. Default setting: 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 the 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. Default setting: 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	Call Return Code This code calls the last caller. Default setting: *69
Call Redial Code	Redials the last number called. Default setting: *07
Blind Transfer Code	Begins a blind transfer of the current call to the extension specified after the activation code. Default setting: *98
Call Back Act Code	Starts a callback when the last outbound call is not busy. Default setting: *66
Call Back Deact Code	Cancels a callback. Default setting: *86
Call Back Busy Act Code	Starts a callback when the last outbound call is busy. Default setting: *05
Cfwd All Act Code	Forwards all calls to the extension specified after the activation code. Default setting: *72
Cfwd All Deact Code	Cancels call forwarding of all calls. Default setting: *73
Cfwd Busy Act Code	Forwards busy calls to the extension specified after the activation code. Default setting: *90
Cfwd Busy Deact Code	Cancels call forwarding of busy calls. Default setting: *91
Cfwd No Ans Act Code	Forwards no-answer calls to the extension specified after the activation code. Default setting: *92
Cfwd No Ans Deact Code	Cancels call forwarding of no-answer calls. Default setting: *93

Field	Description
Cfwd Last Act Code	Forwards the last inbound or outbound call to the number that the user specifies after entering the activation code. Default setting: *63
Cfwd Last Deact Code	Cancels call forwarding of the last inbound or outbound call. Default setting: *83
Block Last Act Code	Blocks the last inbound call. Default setting: *60
Block Last Deact Code	Cancels blocking of the last inbound call. Default setting: *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. Default setting: *64
Accept Last Deact Code	Cancels the code to accept the last outbound call. Default setting: *84
CW Act Code	Enables call waiting on all calls. Default setting: *56
CW Deact Code	Disables call waiting on all calls. Default setting: *57
CW Per Call Act Code	Enables call waiting for the next call. Default setting: *71
CW Per Call Deact Code	Disables call waiting for the next call. Default setting: *70
Block CID Act Code	Blocks caller ID on all outbound calls. Default setting: *67
Block CID Deact Code	Removes caller ID blocking on all outbound calls. Default setting: *68
Block CID Per Call Act Code	Blocks caller ID on the next outbound call. Default setting: *81
Block CID Per Call Deact Code	Removes caller ID blocking on the next inbound call. Default setting: *82

Field	Description
Block ANC Act Code	Blocks all anonymous calls. Default setting: *77
Block ANC Deact Code	Removes blocking of all anonymous calls. Default setting: *87
DND Act Code	Enables the do not disturb feature. Default setting: *78
DND Deact Code	Disables the do not disturb feature. Default setting: *79
CID Act Code	Enables caller ID generation. Default setting: *65
CID Deact Code	Disables caller ID generation. Default setting: *85
CWCID Act Code	Enables call waiting, caller ID generation. Default setting: *25
CWCID Deact Code	Disables call waiting, caller ID generation. Default setting: *45
Dist Ring Act Code	Enables the distinctive ringing feature. Default setting: *26
Dist Ring Deact Code	Disables the distinctive ringing feature. Default setting: *46
Speed Dial Act Code	Assigns a speed dial number. Default setting: *74
Paging Code	Used for paging other clients in the group. Default setting: *96
Secure All Call Act Code	Makes all outbound calls secure. Default setting: *16
Secure No Call Act Code	Makes all outbound calls not secure. Default setting: *17
Secure One Call Act Code	Makes the next outbound call secure. (It is redundant if all outbound calls are secure by default.) Default setting: *18

Field	Description
Secure One Call Deact Code	Makes the next outbound call not secure. (It is redundant if all outbound calls are not secure by default.) Default setting: *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. Default setting: blank
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. Default setting: blank
Modem Line Toggle Code	Toggles the line to a modem. Modem passthrough mode can be triggered only by pre-dialing this code. Default setting: *99
FAX Line Toggle Code	Toggles the line to a fax machine. Default setting: #99
Media Loopback Code	Use for media loopback. Default setting: *03

Field	Description
Referral Services Codes	<p>These codes tell the ATA what to do when the user places the current call on hold and is listening to the second dial tone.</p> <p>One or more *codes can be configured into this parameter, such as *98, or *97!*98!*123, etc. The maximum length is 79 characters. This parameter applies when the user places the current call on hold by pressing the hook flash button. Each *code (and the following valid target number according to current dial plan) triggers the ATA to perform a blind transfer to a target number that is prepended by the service *code.</p> <p>For example, after the user dials *98, the ATA plays a special dial tone called the Prompt Tone while waiting for the user to enter a target number (which is checked according to dial plan as in normal dialing). When a complete number is entered, the ATA sends a blind REFER to the holding party with the Refer-To target equal to *98 target_number. This feature allows the ATA to hand off a call to an application server to perform further processing, such as call park.</p> <p>The *codes should not conflict with any of the other vertical service codes internally processed by the ATA. You can empty the corresponding *code that you do not want the ATA to process.</p> <p>Default setting: blank</p>

Field	Description
Feature Dial Services Codes	<p>These codes tell the ATA what to do when the user is listening to the first or second dial tone.</p> <p>One or more *codes can be configured into this parameter, such as *72, or *72!*74!*67!*82, etc. The maximum length is 79 characters. This parameter applies when the user has a dial tone (first or second dial tone) After receiving dial tone, a user enters the *code and the target number according to current dial plan. For example, after user dials *72, the ATA plays a special tone called a Prompt tone while awaiting the user to enter a valid target number. When a complete number is entered, the 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)</p> <p>The *codes should not conflict with any of the other vertical service codes internally processed by the ATA. You can remove a corresponding *code that you do not want to the ATA to process.</p> <p>You can add a parameter to indicate which tone plays after the *code is entered, such as *72'c'*67'p'. Below is a list of allowed tone parameters (note the use of open quotes surrounding the parameter, without spaces)</p> <p>'c' = <Cfwd Dial Tone> 'd' = <Dial Tone> 'm' = <MWI Dial Tone> 'o' = <Outside Dial Tone> 'p' = <Prompt Dial Tone> 's' = <Second Dial Tone> 'x' = No tones are place, x is any digit not used above</p> <p>If no tone parameter is specified, the 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 this parameter. Instead, add the *code in the dial plan and the ATA send INVITE *73@..... as usual when user dials *73.</p> <p>Default setting: blank</p>

Vertical Service Announcement Codes

Field	Description
Service Annc Base Number	Base number for service announcements. Default setting: blank
Service Annc Extension Codes	Extension codes for service announcements. Default setting: blank

Outbound Call Codec Selection Codes

Field	Description
Prefer G711u Code	Dial prefix to make G.711u the preferred codec for the call. Default setting: *017110
Force G711u Code	Dial prefix to make G.711u the only codec that can be used for the call. Default setting: *027110
Prefer G711a Code	Dial prefix to make G.711a the preferred codec for the call. Default setting: *017111
Force G711a Code	Dial prefix to make G.711a the only codec that can be used for the call. Default setting: *027111
Prefer G726r32 Code	Dial prefix to make G.726r32 the preferred codec for the call. Default setting: *0172632
Force G726r32 Code	Dial prefix to make G.726r32 the only codec that can be used for the call. Default setting: *0272632
Prefer G729a Code	Dial prefix to make G.729a the preferred codec for the call. Default setting: *01729
Force G729a Code	Dial prefix to make G.729a the only codec that can be used for the call. Default setting: *02729
Prefer G722 Code	Dial prefix to make G.722 the preferred codec for the call. Default setting: *01722

Field	Description
Force G722 Code	Dial prefix to make G.722 the only codec that can be used for the call. Default setting: *02722

Miscellaneous

Field	Description
FXS Port Impedance	Sets the electrical impedance of the PHONE port. Choices are: 600, 900, 600+2.16uF, 900+2.16uF, 270+750 150nF, 220+850 120nF, 220+820 115nF, or 200+600 100nF. Default setting: 600. NOTE For New Zealand impedance (370+620 310nF), use 270+750 150nF.
FXS Port Input Gain	Input gain in dB, up to three decimal places. The range is 6.000 to -12.000. Default setting: -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 parameter. Default setting: -3.
DTMF Playback Level	Local DTMF playback level in dBm, up to one decimal place. Default setting: -16.0.
DTMF Twist	To gain difference between the two tone frequency. Default setting: 2
DTMF Playback Length	Local DTMF playback duration in milliseconds. Default setting: .1.
Detect ABCD	To enable local detection of DTMF ABCD, select yes. Otherwise, select no. Default setting: yes This setting has no effect if DTMF Tx Method is INFO; ABCD is always sent OOB regardless in this setting.

Field	Description
Playback ABCD	To enable local playback of OOB DTMF ABCD, select yes. Otherwise, select no. Default setting: yes
Caller ID Method	<p>The choices are described below. Default setting: Bellcore(N.Amer, China)</p> <ul style="list-style-type: none"> ▪ Bellcore (N.Amer,China): CID, CIDCW, and VMWI. FSK sent after first ring (same as ETSI FSK sent after first ring) (no polarity reversal or DTAS) ▪ DTMF (Finland, Sweden): CID only. DTMF sent after polarity reversal (and no DTAS) and before first ring. ▪ DTMF (Denmark): CID only. DTMF sent before first ring with no polarity reversal and no DTAS. ▪ 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 a device 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 a device after DTAS for CIDCW. Polarity reversal is applied only if equipment is on hook. ▪ DTMF (Denmark) with PR: CID only. DTMF sent after polarity reversal (and no DTAS) and before first ring. Default setting: Bellcore(N.Amer, China)
FXS Port Power Limit	The choices are from 1 to 8. Default setting: 3

Field	Description
Caller ID FSK Standard	The ATA supports bell 202 and v.23 standards for caller ID generation. Default setting: bell 202
Feature Invocation Method	Select the method you want to use, Default or Sweden default. Default setting: Default

Line 1 Settings (PHONE Port)

Use the *Voice > Line 1* page to configure the settings for calls through the PHONE port.

To open this page: Click **Voice** on the menu bar, and then click **Line 1** in the navigation tree. Enter the settings as described below. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

NOTE In a configuration profile, the FXS parameters must be appended with the appropriate numeral (for example, [1] or [2]) to identify the port to which the setting applies.

Line Enable

Field	Description
Line Enable	To enable this line for service, select yes. Otherwise, select no. Default setting: 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. Default setting: no
SAS DLG Refresh Intvl	If this value 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 ATA ends this call with a SIP BYE message. The range is 0 to 255 seconds (0 means that the session refresh is disabled) Default setting: 30
SAS Inbound RTP Sink	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 an FQDN or IP address of an RTP sink to be used by the 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, if specified, will appear in the m = line of the SDP. If this value is not specified or is 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. Default setting: blank

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. Default setting: no
NAT Keep Alive Enable	To send the configured NAT keep alive message periodically, select yes. Otherwise, select no. Default setting: 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. Default setting: \$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 proxy server or outbound proxy server. Default setting: \$PROXY

Network Settings

Field	Description
SIP ToS/DiffServ Value	TOS/DiffServ field value in UDP IP packets carrying a SIP message. Default setting: 0x68
SIP CoS Value	CoS value for SIP messages. Valid values are 0 through 7. Default setting: 3
RTP ToS/DiffServ Value	ToS/DiffServ field value in UDP IP packets carrying RTP data. Default setting: 0xb8
RTP CoS Value [0-7]	CoS value for RTP data. Valid values are 0 through 7. Default setting: 6

Field	Description
Network Jitter Level	Determines how jitter buffer size is adjusted by the 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. Default setting: high
Jitter Buffer Adjustment	Choose yes to enable or no to disable this feature. Default setting: yes

SIP Settings

Field	Description
SIP Transport	The TCP choice provides “guaranteed delivery”, which assures that lost packets are retransmitted. TCP also guarantees that the SIP packages are received in the same order that they were sent. As a result, TCP overcomes the main disadvantages of UDP. In addition, for security reasons, most corporate firewalls block UDP ports. With TCP, new ports do not need to be opened or packets dropped, because TCP is already in use for basic activities such as Internet browsing or e-commerce. Options are: UDP, TCP, TLS. Default setting: UDP
SIP Port	Port number of the SIP message listening and transmission port. Default setting: 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. Default setting: no
EXT SIP Port	The external SIP port number. Default setting: blank

Field	Description
Auth Resync-Reboot	If this feature is enabled, the ATA authenticates the sender when it receives the NOTIFY resync reboot (RFC 2617) message. To use this feature, select yes. Otherwise, select no. Default setting: 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. Default setting: blank
SIP Remote-Party-ID	To use the Remote-Party-ID header instead of the From header, select yes. Otherwise, select no. Default setting: yes
SIP GUID	This feature limits the registration of SIP accounts. The Global Unique ID is generated for each line for each ATA. When it is enabled, the 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. Default setting: no

Field	Description
SIP Debug Option	<p>SIP messages are received at or sent from the proxy listen port. This feature controls which SIP messages to log. The choices are described below. Default setting: none</p> <ul style="list-style-type: none"> ▪ none—No logging. ▪ 1-line—Logs the start-line only for all messages. ▪ 1-line excl. OPT—Logs the start-line only for all messages except OPTIONS requests/responses. ▪ 1-line excl. NTFY—Logs the start-line only for all messages except NOTIFY requests/responses. ▪ 1-line excl. REG—Logs the start-line only for all messages except REGISTER requests/responses. ▪ 1-line excl. OPTINTFYIREG—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. OPTINTFYIREG—Logs all SIP messages in full text except for OPTIONS, NOTIFY, and REGISTER requests/responses.
RTP Log Intvl	<p>The interval for the RTP log. Default setting: 0</p>
Restrict Source IP	<p>If configured, the ATA drops all packets sent to its SIP Ports 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) Default setting: no</p>

Field	Description
Referor Bye Delay	The number of seconds to wait before sending a BYE to the referrer to terminate a stale call leg after a call transfer.
Refer Target Bye Delay	The number of seconds to wait before sending a BYE to the refer target to terminate a stale call leg after a call transfer.
Referee Bye Delay	The number of seconds to wait before sending a BYE to the referee to terminate a stale call leg after a call transfer.
Refer-To Target Contact	To contact the refer-to target, select yes. Otherwise, select no. Default setting: no
Sticky 183	If this feature is enabled, the ATA 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. Default setting: no
Auth INVITE	When enabled, authorization is required for initial incoming INVITE requests from the SIP proxy. Default setting: no
Reply 182 On Call Waiting	When enabled, the ATA replies with a SIP182 response to the caller if it is already in a call and the line is off-hook. To use this feature select yes. Default setting: no
Use Anonymous With RPID	Determines whether or not the ATA uses “Anonymous” when Remote Party ID is requested in the SIP message. Default setting: yes
Use Local Addr In From	Use the local ATA IP address in the SIP FROM message. Default setting: no

Call Feature Settings

Field	Description
Blind Attn-Xfer Enable	Enables the 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 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. Default setting: 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. Default setting: blank
Xfer When Hangup Conf	Makes the ATA perform a transfer when a conference call has ended. Select yes or no from the drop-down menu. Default setting: yes
Conference Bridge URL	This feature supports external conference bridging for n-way conference calls ($n > 2$), instead of mixing audio locally. To use this feature, set this parameter to that of the server's name. For example: conf@myserver.com:12345 or conf (which uses the Proxy value as the domain). Default setting: blank
Conference Bridge Ports	Select the maximum number of conference call participants. The range is 3 to 10. Default setting: 3 NOTE When FXS port initiates a conference call, the caller ID on the analogue phone is updated to the Conference Bridge.

Field	Description
Enable IP Dialing	Enable or disable IP dialing. If IP dialing is enabled, one can dial [userid@] 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. Default setting: no
Emergency Number	Comma separated list of emergency number patterns. If outbound call matches one of the pattern, the ATA will disable hook flash event handling. The condition is restored to normal after the call ends. Blank signifies that there is no emergency number. Maximum number length is 63 characters. Default setting: blank
Mailbox ID	Enter the ID number of the mailbox for this line. Default setting: blank

Proxy and Registration

Field	Description
Proxy	SIP proxy server for all outbound requests. Default setting: blank
Outbound Proxy	SIP Outbound Proxy Server where all outbound requests are sent as the first hop. Default setting: blank
Use Outbound Proxy	Enables the use of an Outbound Proxy. If set to no, the Outbound Proxy and Use OB Proxy in Dialog parameters are ignored. Default setting: no

Field	Description
Use OB Proxy In Dialog	Whether to force SIP requests to be sent to the outbound proxy within a dialog. Ignored if the parameter Use Outbound Proxy is no, or the Outbound Proxy parameter is empty. Default setting: yes
Register	Enable periodic registration with the Proxy parameter. This parameter is ignored if Proxy is not specified. Default setting: 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. Default setting: no
Register Expires	Expires value in sec in a REGISTER request. The ATA will periodically renew registration shortly before the current registration expired. This parameter is ignored if the Register parameter is no. Range: 0 – (2 ³¹ – 1) sec. Default setting: 3600
Ans Call Without Reg	Allow answering inbound calls without successful (dynamic) registration by the unit. Default setting: no
Use DNS SRV	Whether to use DNS SRV lookup for Proxy and Outbound Proxy. Default setting: no
DNS SRV Auto Prefix	If enabled, the ATA will automatically prepend the Proxy or Outbound Proxy name with _sip._udp when performing a DNS SRV lookup on that name. Default setting: no

Field	Description
Proxy Fallback Intvl	After failing over to a lower priority server, the ATA waits for the specified Proxy Fallback Interval, in seconds, before retrying the highest priority proxy (or outbound proxy) servers. This parameter is useful only if the primary and backup proxy server list is provided to the ATA via DNS SRV record lookup on the server name. (Using multiple DNS A records per server name does not allow the notion of priority, so all hosts will be considered at the same priority and the ATA will not attempt to fall back after a failover.) Default setting: 3600
ProxyRedundancy Method	The method that the ATA uses to create a list of proxies returned in the DNS SRV records. If you select Normal, the list will contain proxies ranked by weight and priority. If you select Based on SRV port, the ATA also inspects the port number based on 1st proxy's port. Default setting: Normal
Mailbox Subscribe URL	The URL or IP address of the voicemail server. Default setting: blank
Mailbox Subscribe Expires	Sets subscription interval for voicemail message waiting indication. When this time period expires, the ATA sends another subscribe message to the voice mail server. Default: 2147483647

Subscriber Information

Field	Description
Display Name	Display name for caller ID. Default setting: blank
User ID	User ID for this line. Default setting: blank
Password	Password for this line. Default setting: blank

Field	Description
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. Default setting: no
Auth ID	Authentication ID for SIP authentication. Default setting: blank
Resident Online Number	This setting allows you to associate a "local" telephone number with this line using a valid Skype Online Number from Skype. Calls made to that number will ring your phone. Enter the number without spaces or special characters. Default setting: blank

Supplementary Service Subscription

The 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 ATA.

Field	Description
Call Waiting Serv	Enable Call Waiting Service. Default setting: yes
Block CID Serv	Enable Block Caller ID Service. Default setting: yes
Block ANC Serv	Enable Block Anonymous Calls Service Default setting: yes
Dist Ring Serv	Enable Distinctive Ringing Service Default setting: yes
Cfwd All Serv	Enable Call Forward All Service Default setting: yes
Cfwd Busy Serv	Enable Call Forward Busy Service Default setting: yes

Field	Description
Cfwd No Ans Serv	Enable Call Forward No Answer Service Default setting: yes
Cfwd Sel Serv	Enable Call Forward Selective Service. Configure this service in the Selective Call Forward Settings section. Default setting: yes
Cfwd Last Serv	Enable Forward Last Call Service Default setting: yes
Block Last Serv	Enable Block Last Call Service Default setting: yes
Accept Last Serv	Enable Accept Last Call Service Default setting: yes
DND Serv	Enable Do Not Disturb Service Default setting: yes
CID-Serv	Enable Caller ID Service Default setting: yes
CWCID Serv	Enable Call Waiting Caller ID Service Default setting: yes
Call Return Serv	Enable Call Return Service Default setting: yes
Call Redial Serv	Enable Call Redial Service.
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. Default setting: yes
Three Way Conf Serv	Enable Three Way Conference Service. Three Way Conference is required for Attended Transfer. Default setting: yes
Attn Transfer Serv	Enable Attended Call Transfer Service. Three Way Conference is required for Attended Transfer. Default setting: yes

Field	Description
Unattn Transfer Serv	Enable Unattended (Blind) Call Transfer Service. Default setting: yes
MWI Serv	Enable MWI Service. MWI is available only if a Voice Mail Service is set-up in the deployment. Default setting: yes
VMWI Serv	Enable VMWI Service (FSK) Default setting: yes
Speed Dial Serv	Enable Speed Dial Service. Default setting: yes
Secure Call Serv	Secure Call Service. If this feature is enabled, a user can make a secure call by entering an activation code (*18 by default) before dialing the target number. Then audio traffic in both directions is encrypted for the duration of the call. Default setting: yes Star codes are set in Vertical Service Activation Codes . To enable secure calling by default, without requiring a star code, set the user's Secure Call Setting to yes. See User 1, page 147 .
Referral Serv	Enable Referral Service. See the Referral Services Codes parameter For more information. Default setting: yes
Feature Dial Serv	Enable Feature Dial Service. See the Feature Dial Services Codes parameter For more information. Default setting: yes
Service Announcement Serv	Enable Service Announcement Service. Default setting: no
Reuse CID Number As Name	Use the Caller ID number as the caller name. Default settings: yes

Audio Configuration

Field	Description
Preferred Codec, Second Preferred Codec, Third Preferred Codec	Up to three codecs to be used for all calls from this handset, listed order of preference. The actual codec used in a call still depends on the outcome of the codec negotiation protocol. Select one of the following: G711u, G711a, G726-32, G729a, or G722. Default setting for Preferred Codec: G711u Default setting for Second and Third Preferred Codec: Unspecified
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. Default setting: yes
Use Remote Pref Codec	To use the preferred codec specified by the remote peer, select yes. Otherwise, select no. Default setting:
Codec Negotiation	Specify the codecs for codec negotiation: Default or List All. Default setting: Default
G729a Enable	To enable the use of the G.729a codec at 8 kbps, select yes. Otherwise, select no. Default setting: yes
Silence Supp Enable	To enable silence suppression so that silent audio frames are not transmitted, select yes. Otherwise, select no. Default setting: no
G726-32 Enable	To enable the use of the G.726 codec at 32 kbps, select yes. Otherwise, select no. Default setting: yes
Silence Threshold	Select the appropriate setting for the threshold: high, medium, or low. Default setting: medium
FAX V21 Detect Enable	To enable detection of V21 fax tones, select yes. Otherwise, select no. Default setting: yes

Field	Description
Echo Canc Enable	To enable the use of the echo canceller, select yes. Otherwise, select no. Default setting: yes
FAX CNG Detect Enable	To enable detection of the fax Calling Tone (CNG), select yes. Otherwise, select no. Default setting: yes
FAX Passthru Codec	Select the codec for fax passthrough, G711u or G711a. Default setting: G711u
Echo Canc Adapt Enable	To enable the echo canceller to adapt, select yes. Otherwise, select no. Default setting: yes
FAX Codec Symmetric	To force the ATA to use a symmetric codec during fax passthrough, select yes. Otherwise, select no. Default setting: yes
DTMF Process INFO	To use the DTMF process info feature, select yes. Otherwise, select no. Default setting: yes
FAX Passthru Method	Select the fax passthrough method: None, NSE, or ReINVITE. Default setting: NSE
DTMF Process AVT	To use the DTMF process AVT feature, select yes. Otherwise, select no. Default setting: yes
FAX Process NSE	To use the fax process NSE feature, select yes. Otherwise, select no. Default setting: yes
DTMF Tx Method	Select the method to transmit DTMF signals to the far end: InBand, AVT, INFO, or Auto. InBand sends DTMF by 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. Default setting: Auto

Field	Description
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. Default setting: no
DTMF Tx Mode	DTMF Detection Tx Mode is available for SIP information and AVT. Options are: Strict or Normal. Default setting: Strict for which the following are true: <ul style="list-style-type: none"> A DTMF digit requires an extra hold time after detection. The DTMF level threshold is raised to -20 dBm. The minimum and maximum duration thresholds are: <ul style="list-style-type: none"> strict mode for AVT: 70 ms normal mode for AVT: 40 ms strict mode for SIP info: 90 ms normal mode for SIP info: 50 ms
DTMF Tx Strict Hold Off Time	This parameter is in effect only when DTMF Tx Mode is set to strict, and when DTMF Tx Method is set to out-of-band; i.e. either AVT or SIP-INFO. The value can be set as low as 40 ms. There is no maximum limit. A larger value will reduce the chance of talk-off (beeping) during conversation, at the expense of reduced performance of DTMF detection, which is needed for interactive voice response systems (IVR) Default: 70 ms
FAX Enable T38	To enable the use of ITU-T T.38 standard for FAX Relay, select yes. Otherwise select no. Default setting: 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. Default setting: None

Field	Description
FAX T38 Redundancy	Select the appropriate number to indicate the number of previous packet payloads to repeat with each packet. Choose 0 for no payload redundancy. The higher the number, the larger the packet size and the more bandwidth consumed. Default setting: 1
FAX T38 ECM Enable	Select yes to enable T.38 Error Correction Mode. Otherwise select no. Default setting: yes
FAX Tone Detect Mode	This parameter has three possible values: <ul style="list-style-type: none"> ▪ caller or callee: The ATA will detect FAX tone whether it is callee or caller ▪ caller only: The ATA will detect FAX tone only if it is the caller ▪ callee only: The ATA will detect FAX tone only if it is the callee Default setting: caller or callee.
Symmetric RTP	Enable symmetric RTP operation. If enabled, the ATA 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 ATA sends RTP to the destination as indicated in the inbound SDP. Default setting: no
Fax T38 Return to Voice	When this feature is enabled, upon completion of the fax image transfer, the connection remains established and reverts to a voice call using the previously designated codec. Select yes to enable this feature, or select no to disable it. Default setting: no

Configuration Tables

G3 NSE-Based Passthrough FAX Call

Parameter	Value	Description
FAX V21 Detect Enable	Yes	
FAX CNG Detect Enable	Yes	If CNG detection is not needed, set it to No .
FAX Tone Detect Mode	Caller or Callee	
FAX Passthru Method	NSE	
FAX Process NSE	Yes	
FAX Passthru Codec	G.711u or G.711a	Depends on the passthrough codec used.
FAX Disable ECAN	No	
Echo Canc Enable	Yes	
FAX Enable T38	No	
Modem Line	No	

SG3/V.34 NSE-Based Passthrough FAX Call

Parameter	Value	Description
FAX V21 Detect Enable	Yes	
FAX CNG Detect Enable	Yes	If CNG detection is not needed, set it to No .
FAX Tone Detect Mode	Caller or Callee	

Parameter	Value	Description
FAX Passthru Method	NSE	
FAX Process NSE	Yes	
FAX Passthru Codec	G.711u or G.711a	Depends on the passthrough codec used.
FAX Disable ECAN	No	
Echo Canc Enable	Yes	
FAX Enable T38	No	
Modem Line	No	

SIP Protocol-Based passthrough FAX Call

Parameter	Value	Description
FAX V21 Detect Enable	Yes	
FAX CNG Detect Enable	Yes	If CNG detection is not needed, set it to No .
FAX Tone Detect Mode	Caller or Callee	
FAX Passthru Method	ReINVITE	
FAX Process NSE	No	
FAX Passthru Codec	G.711u or G.711a	Depends on the passthrough codec used.
FAX Disable ECAN	No	
Echo Canc Enable	Yes	
FAX Enable T38	No	
Modem Line	No	

SG3/V.34 SIP Protocol-Based passthrough FAX Call

Parameter	Value	Description
FAX V21 Detect Enable	Yes	
FAX CNG Detect Enable	Yes	If CNG detection is not needed, set it to No .
FAX Tone Detect Mode	Caller or Callee	
FAX Passthru Method	ReINVITE	
FAX Process NSE	No	
FAX Passthru Codec	G.711u or G.711a	Depends on the passthrough codec used.
FAX Disable ECAN	No	
Echo Canc Enable	Yes	
FAX Enable T38	No	
Modem Line	No	

SIP Protocol-Based T.38 Relay FAX Call

Parameter	Value	Description
FAX V21 Detect Enable	Yes	
FAX CNG Detect Enable	Yes	If CNG detection is not needed, set it to No .
FAX Tone Detect Mode	Caller or Callee	
FAX Passthru Method	ReINVITE	
FAX Disable ECAN	No	

Parameter	Value	Description
Echo Canc Enable	Yes	
FAX Enable T38	No	
Modem Line	No	
FAX T38 Return to Voice	No	If voice call back is needed after T38 fac call, set it to No .

T38 Relay FAX Failover to SIP Protocol-Based Passthrough FAX call

Parameter	Value	Description
FAX V21 Detect Enable	Yes	
FAX CNG Detect Enable	Yes	If CNG detection is not needed, set it to No .
FAX Tone Detect Mode	Caller or Callee	
FAX Passthru Method	ReINVITE	
FAX Passthru Codec	G.711u or G.711a	Depends on the passthrough codec used.
FAX Disable ECAN	No	If connected to SG3 fax machine, set parameter to "Yes".
Echo Canc Enable	Yes	
FAX Enable T38	No	
Modem Line	No	
FAX T38 Return to Voice	No	

G3 Passthrough FAX Call With <FAX Line Toggle Code

Parameter	Value	Description
FAX Disable ECAN	No	
Echo Canc Enable	Yes	
Modem Line	No	
FAX Line Toggle Code	#99	<p>Set it to any preferred code.</p> <p>NOTE Pre-dial #99 before dialing. Fax call made in this method does not detect fax tone or send out any fax request messages (like NSE or Re-INVITE message) to the remote gateway, which causes interoperability issue with remote side.</p> <p>Other fax parameters are not relevant to the call.</p>

SG3/V.34 Passthrough FAX Call With <FAX Line Toggle Code

Parameter	Value	Description
FAX Disable ECAN	Yes	
Echo Canc Enable	Yes	
Modem Line	No	
FAX Line Toggle Code	#99	<p>Set it to any preferred code.</p> <p>NOTE Pre-dial #99 before dialing. Fax call made in this method does not detect fax tone or send out any fax request messages (like NSE or Re-INVITE message) to the remote gateway, which causes interoperability issue with remote side.</p> <p>Other fax parameters are not relevant to the call.</p>

Passthrough Modem Call With <Modem Line Toggle Code>

Parameter	Value	Description
Modem Line	No	If Modem Line Toggle Code is used to make the modem call, do not set it to Yes
Modem Line Toggle Code	*99	Set it to any preferred code. NOTE Pre-dial *99 before dialing. Other fax parameters are not relevant to the call.
FAX Disable ECAN	Yes	

Passthrough Modem Call With <Modem Line>

Parameter	Value	Description
Modem Line	Yes	
FAX Disable ECAN	Yes	

NOTE Other fax parameters are not relevant to the call.

Passthrough Modem Call With Manual Configuration

Parameter	Value	Description
Preferred Codec	G.711u or G.711a	Depends on the passthrough codec used.
FAX V21 Detect Enable	No	
FAX CNG Detect Enable	No	

Parameter	Value	Description
Echo Canc Enable	No	
Silence Supp Enable	No	
FAX Enable T38	No	
FAX Process NSE	No	
Modem Line	No	
Jitter Buffer Adjustment	No	
Network Jitter Level	High	

Dial Plan

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

Each parameter is separated by a semi-colon (;)

Example 1:

```
*1xxxxxxxxxx<:@fwdnat.pulver.com:5082;uid=jsmith;pwd=xy z
```

Example 2:

```
*1xxxxxxxxxx<:@fwd.pulver.com;nat;uid=jsmith;pwd=xyz
```

The syntax for a dial plan expression is described in the table below.

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

Dial Plan Entry	Functionality
xxxxxxxxxxxxx.	Everything else

Gateway Accounts

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 1408xxxxxx<:@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). Default setting: blank
GW1/2/3/4 NAT Mapping Enable	If enabled, the ATA uses NAT mapping when contacting Gateway 1. Default setting: no
GW1/2/3/4 Auth ID	This value is the authentication user-id to be used by the ATA to authenticate itself to Gateway 1. Default setting: blank
GW1/2/3/4 Password	This value is the password to be used by the ATA to authenticate itself to Gateway 1. Default setting: blank

VoIP Fallback to PSTN section

Field	Description
Auto PSTN Fallback	If enabled, the ATA automatically routes all calls to the PSTN gateway when the Line 1 proxy is down (registration failure or network link down). Default setting: yes

FXS Port Polarity Configuration

Field	Description
Idle Polarity	Polarity before a call is connected: Forward or Reverse. Default setting: Forward
Caller Conn Polarity	Polarity after an outbound call is connected: Forward or Reverse. Default setting: Forward.
Callee Conn Polarity	Polarity after an inbound call is connected: Forward or Reverse. Default setting: Forward

PSTN (LINE Port)

Use the *Voice > PSTN* page to configure the settings for calls through the LINE (PSTN) port.

To open this page: Click **Voice** on the menu bar, and then click **PSTN** in the navigation tree. Enter the settings as described below. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

Field	Description
PSTN Line Enable	To enable this line for service, select yes. Otherwise, select no. Default setting: yes

Field	Description
Incoming Handset List	The devices that ring when an incoming call is received. Default setting: fxs,1,2,3,4,5,6,7,8,9,10

Network Settings

Field	Description
SIP ToS/DiffServ Value	TOS/DiffServ field value in UDP IP packets carrying a SIP message. Default setting: 0x68
SIP CoS Value	CoS value for SIP messages. Valid values are 1 through 7. Default setting: 3
RTP ToS/DiffServ Value	ToS/DiffServ field value in UDP IP packets carrying RTP data. Default setting: 0xb8
RTP CoS Value	CoS value for RTP data. Valid values are 1 through 7. Default setting: 6
Network Jitter Level	Determines how jitter buffer size is adjusted by the ATA device. 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. Default setting: low
Jitter Buffer Adjustment	Controls how the jitter buffer should be adjusted. Select the appropriate setting: up and down, up only, down only, or disable. Default setting: yes

SIP Settings

Field	Description
SIP Transport	The TCP choice provides “guaranteed delivery,” which assures that lost packets are retransmitted. TCP also guarantees that the SIP packages are received in the same order that they were sent. As a result, TCP overcomes the main disadvantages of UDP. In addition, for security reasons, most corporate firewalls block UDP ports. With TCP, new ports do not need to be opened or packets dropped, because TCP is already in use for basic activities such as Internet browsing or e-commerce. Options are: UDP, TCP, TLS. The default is UDP. Default setting: UDP
SIP Port	Port number of the SIP message listening and transmission port. Default setting: 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. Default setting: no
EXT SIP Port	The external SIP port number. Default setting: 5061
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. Default setting: no
EXT SIP Port	The external SIP port number. Default setting: blank
Auth Resync-Reboot	If this feature is enabled, the ATA device authenticates the sender when it receives the NOTIFY resync reboot (RFC 2617) message. To use this feature, select yes. Otherwise, select no. Default setting: yes

Field	Description
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. Default setting: no
SIP GUID	The Global Unique ID is generated for each line for each device. When it is enabled, the ATA device 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. Default setting: no

Field	Description
SIP Debug Option	<p>SIP messages are received at or sent from the proxy listen port. This feature controls which SIP messages to log. Choices are as follows:</p> <ul style="list-style-type: none"> ▪ none—No logging. ▪ 1-line—Logs the start-line only for all messages. ▪ 1-line excl. OPT—Logs the start-line only for all messages except OPTIONS requests/responses. ▪ 1-line excl. NTFY—Logs the start-line only for all messages except NOTIFY requests/responses. ▪ 1-line excl. REG—Logs the start-line only for all messages except REGISTER requests/responses. ▪ 1-line excl. OPTINTFYIREG—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. OPTINTFYIREG—Logs all SIP messages in full text except for OPTIONS, NOTIFY, and REGISTER requests/responses. <p>Default setting: none</p>
RTP Log Intvl	<p>The interval for the RTP log.</p> <p>Default setting: 0</p>

Field	Description
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 ATA 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 <i>Proxy</i> (or <i>Outbound Proxy</i> if <i>Use Outbound Proxy</i> is yes). Default setting: no
Referor Bye Delay	Controls when the ATA device 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. Default setting: 4
Refer Target Bye Delay	For the Refer Target Bye Delay, enter the appropriate period of time in seconds. Default setting: 0
Referee Bye Delay	For the Referee Bye Delay, enter the appropriate period of time in seconds. Default setting: 0
Refer-To Target Contact	To contact the refer-to target, select yes. Otherwise, select no. Default setting: 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. Default setting: no
Auth INVITE	When enabled, authorization is required for initial incoming INVITE requests from the SIP proxy. Default setting: no

Field	Description
Use Anonymous with RPID	When set to yes, use “anonymous” in the SIP message when remote party ID is requested in the SIP message. Default setting: yes
Use Local Addr in FROM	The IP address of the local address enclosed in the FROM of the SIP message. Default setting: no

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. Default setting: no
NAT Keep Alive Enable	To send the configured NAT keep alive message periodically, select yes. Otherwise, select no. Default setting: 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. Escape sequence of %xx is also accepted. For example, %0d%0a is unescaped into \r\n (CRLF). Default setting: \$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. Default setting: \$PROXY

Proxy and Registration

Field	Description
Proxy	SIP proxy server for all outbound requests. Default setting: blank
Outbound Proxy	SIP Outbound Proxy Server where all outbound requests are sent as the first hop. Default setting: blank
Use Outbound Proxy	Enable the use of <i>Outbound Proxy</i> . If set to no, the <i>Outbound Proxy</i> parameter and <i>Use OB Proxy in Dialog</i> is ignored. Default setting: no
Use OB Proxy In Dialog	Whether to force SIP requests to be sent to the outbound proxy within a dialog. Ignored if the <i>Use Outbound Proxy</i> parameter is no, or if the <i>Outbound Proxy</i> parameter is empty. Default setting: yes
Register	Enable periodic registration with the <i>Proxy</i> . This parameter is ignored if the <i>Proxy</i> parameter is not specified. Default setting: 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. Default setting: yes
Register Expires	Allow answering inbound calls without successful (dynamic) registration by the unit. If proxy responded to REGISTER with a smaller Expires value, the ATA will renew registration based on this smaller value instead of the configured value. If registration failed with an Expires too brief error response, the ATA will retry with the value given in the Min-Expires header in the error response. Default setting: 3600

Field	Description
Ans Call Without Reg	Expires value in sec in a REGISTER request. ATA will periodically renew registration shortly before the current registration expired. This parameter is ignored if the <i>Register</i> parameter is no. Range: 0 – (231 – 1) sec Default setting: yes
Use DNS SRV	If required by your provider, check this box to use DNS SRV lookup for Proxy and Outbound Proxy. Default setting: no
DNS SRV Auto Prefix	If enabled, the ATA will automatically prepend the Proxy or Outbound Proxy name with <i>_sip._udp</i> when performing a DNS SRV lookup on that name. Default setting: no
Proxy Fallback Intvl	This parameter sets the delay (sec) after which the ATA 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 ATA 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 ATA will not attempt to fall back after a fail over). Default setting: 3600
Proxy Redundancy Method	The ATA makes an internal list of proxies returned in DNS SRV records. In normal mode this list will contain proxies ranked by weight and priority. If the parameter <i>Based on SRV port</i> is configured, the ATA creates a list in normal mode first, and then inspects the port numbers based on the 1 st proxy's port on the list. Default setting: Normal

Subscriber Information

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

Field	Description
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. Default setting: no
Auth ID	The Authentication ID for SIP authentication.

Audio Configuration

NOTE A codec resource is considered as allocated if it has been included in the SDP codec list of an active call, even though it eventually may not be the one chosen for the connection. So, if the G.729a codec is enabled and included in the codec list, that resource is tied up until the end of the call whether or not the call actually uses G.729a. If the G.729a resource is already allocated and since only one G.729a resource is allowed per device, no other low-bit-rate codec may be allocated for subsequent calls; the only choices are G.711a and G.711u. On the other hand, two G.723.1/G.726 resources are available per device. Therefore it is important to disable the use of G.729a in order to guarantee the support of two simultaneous G.723/G.726 codec.

Field	Description
Preferred Codec	Preferred codec for all calls. (The actual codec used in a call still depends on the outcome of the codec negotiation protocol.) Select one of the following: G.711u, G.711a, G.726-32, or G.729a. Default setting: G.711u
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. Default setting: yes
G.729a Enable	To enable the use of the G.729a codec at 8 kbps, select yes. Otherwise, select no. Default setting: no

Field	Description
Silence Supp Enable	To enable silence suppression so that silent audio frames are not transmitted, select yes. Otherwise, select no. Default setting: no
G726-32 Enable	To enable the use of the G726 codec at 32 kbps, select yes. Otherwise, select no. Default setting: no
Echo Canc Enable	To enable the use of the echo canceller, select yes. Otherwise, select no. Default setting: yes
FAX V21 Detect Enable	To enable detection of V21 fax tones, select yes. Otherwise, select no. Default setting: no
Echo Canc Adapt Enable	To enable the echo canceller to adapt, select yes. Otherwise, select no. Default setting: yes
FAX CNG Detect Enable	To enable detection of the fax Calling Tone (CNG), select yes. Otherwise, select no. Default setting: no
Echo Supp Enable	To enable the use of the echo suppressor, select yes. Otherwise, select no. Default setting: no
FAX Passthru Codec	Select the codec for fax passthrough, G711u or G711a. Default setting: G711u
DTMF Process INFO	To use the DTMF process info feature, select yes. Otherwise, select no. Default setting: yes
FAX Codec Symmetric	To force the ATA device to use a symmetric codec during fax passthrough, select yes. Otherwise, select no. Default setting: yes
DTMF Process AVT	To use the DTMF process AVT feature, select yes. Otherwise, select no. When set to no, the AVT (RFC2833) payload type is not be included in outbound SDP. Default setting: yes

Field	Description
FAX Passthru Method	Select the fax passthrough method: None, NSE, or ReINVITE. Default setting: 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. Default setting: Auto
FAX Process NSE	To use the fax process NSE feature, select yes. Otherwise, select no. Default setting: yes
Symmetric RTP	Enable symmetric RTP operation. If enabled, the ATA 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 ATA sends RTP to the destination as indicated in the inbound SDP. Default setting: yes
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. Default setting: no

Dial Plans

Field	Description
Dial Plan 1/2/3/4/5/6/7/8	Enter up to eight dial plans into this dial plan pool. You can associate a dial plan with a VoIP Caller or a PSTN Caller by referencing the index number (1~8). See VoIP-To-PSTN Gateway Setup , VoIP Users and Passwords (HTTP Authentication) , and PSTN-To-VoIP Gateway Setup . For information about the dial plan syntax, see Configuring Dial Plans, page 225 . Default setting: (xx.)

VoIP-To-PSTN Gateway Setup

Field	Description
VoIP-To-PSTN Gateway Enable	Choose yes to enable or choose no to disable the VoIP-To-PSTN Gateway functionality. Default setting: yes
VoIP Caller Auth Method	The method to authenticate a VoIP Caller to access the PSTN gateway. Choose from none, PIN, or HTTP Digest. Default setting: none
VoIP PIN Max Retry	The number of times that a VoIP caller can attempt to enter a PIN, if the VoIP Caller Auth Method is set to PIN. Default setting: 3
One Stage Dialing	Choose yes to enable or choose no to disable one-stage dialing. This setting applies if the VoIP Caller Auth Method is none or HTTP Digest, or if caller is in the Access List. Default setting: yes
Line 1 VoIP Caller DP	The index number of the dial plan to use when the VoIP Caller is calling from Line 1 of the same ATA during normal operation (in other words, not due to fallback to PSTN service when Line 1 VoIP service is down). the Authentication is skipped for Line 1 VoIP caller. Default setting: 1
VoIP Caller Default DP	The index number of the dial plan to use when the VoIP Caller is not authenticated. Dial plans are configured in the Dial Plans section. Default setting: 1
Line 1 Fallback DP	The index number of the dial plan to use when the VoIP Caller is calling from Line 1 of the same ATA due to fallback to PSTN service when Line 1 VoIP service is down. Default setting: none

Field	Description
VoIP Caller ID Pattern	<p>A comma-separated list of caller phone number patterns that is used to allow or block access to the PSTN gateway based on the caller ID. If the caller ID does not match a specified pattern, access is rejected, regardless of the authentication method. This comparison is applied before the access list is applied. If this parameter is blank (not specified), all callers are considered for VoIP service.</p> <ul style="list-style-type: none"> Use ? to match any single digit. Use * to match any number of digits. <p>Example: 1408*, 1512???1234</p> <p>In the above example, the caller ID either must start with 1408 or must be an 11-digit numbering starting with 1512 and ending with 1234.</p> <p>Default setting: blank</p>
VoIP Access List	<p>A comma-separated list of number patterns that is used to allow or block access to the PSTN gateway based on the source IP address. If the IP address matches a specified pattern, service is allowed without further authentication.</p> <p>Example: 192.168.**, 66.43.12.1??.</p> <p>In the above example, the source IP address either must begin with 192.168 or must be in the range of 66.43.12.100-199.</p> <p>Default setting: blank</p>
VoIP Caller 1/2/3/4/5/6/7/8 PIN	<p>A PIN number that a VoIP caller can use to access the PSTN gateway, when the VoIP Caller Auth Method is set to PIN.</p> <p>Default setting: blank</p>
VoIP Caller 1/2/3/4/5/6/7/8 DP	<p>The index number of the dial plan to use upon successful entry of the corresponding VoIP Caller PIN. Dial plans are configured in the Dial Plans section.</p> <p>Default setting: 1</p>

VoIP Users and Passwords (HTTP Authentication)

Field	Description
VoIP User 1/2/3/4/ 5/6/7/8 Auth ID	<p>A user ID that a VoIP Caller can use for authentication by using the HTTP Digest method (in other words, by embedding an Authorization header in the SIP INVITE message sent to the ATA. If the credentials are missing or incorrect, the ATA 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 ATA.</p> <p>NOTE: If the caller specifies an authentication user-id that does not match any of the VoIP User Auth ID's, the INVITE will be rejected with a 403 response. Default setting: blank.</p>
VoIP User 1/2/3/4/ 5/6/7/8 Password	<p>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 Default setting: blank.</p>
VoIP User 1/2/3/4/ 5/6/7/8 DP	<p>For up to 8 VoIP users, specify the index of the dial plan to be used after successful authentication. Dial plans are configured in the Dial Plans section. If authentication is disabled, the default dial plan is used for all unknown VoIP users. Default setting: 1.</p>

PSTN-To-VoIP Gateway Setup

Field	Description
PSTN-To-VoIP Gateway Enable	<p>Select yes to enable or select no to disable PSTN-To-VoIP Gateway functionality. Default setting: yes</p>
PSTN Caller Auth Method	<p>The method to authenticate a PSTN Caller to access the VoIP gateway. Choose from none or PIN. Default setting: none</p>

Field	Description
PSTN Ring Thru Line 1	<p>To enable ring through to Line 1 based on caller number patterns, choose yes. Otherwise choose no.</p> <p>Note: For more information about PSTN Caller number patterns, see PSTN Caller ID Pattern.</p> <p>Default setting: yes</p>
PSTN PIN Max Retry	<p>The number of times that a PSTN caller can attempt to enter a PIN number, if the authentication method is set to PIN.</p> <p>Default setting: 3</p>
PSTN CID for VoIP CID	<p>Choose yes or no.</p> <p>Default setting: no</p>
PSTN CID Number Prefix	<p>A dialing prefix, if needed, to add to the caller ID number on the PBX to ensure that a callback goes to the correct number.</p> <p>Default setting: blank</p>
PSTN Caller Default DP	<p>The index number of the dial plan that is used when the PSTN Caller Auth Method is set to none. Dial plans are configured in the Dial Plans section.</p> <p>Default settings: 1</p>
Line 1 Signal Hook Flash to PSTN	<p>Specify the operation of the hook flash on the analog phone when a PSTN-to-VoIP call is active. Choose Disabled or Double Hook Flash.</p> <p>Default setting: Disabled</p>
PSTN CID Name Prefix	<p>The prefix to add to the caller ID name that is sent to the PBX. Enter the characters to add to the caller ID name.</p> <p>Default setting: blank</p>

Field	Description
PSTN Caller ID Pattern	<p>A comma-separated list of phone number patterns that is used to allow or block access to the VoIP gateway based on the caller ID. If the caller ID does not match a specified pattern, access is rejected, regardless of the authentication method. This comparison is applied before the access list is applied. If this parameter is blank (not specified), all callers are considered for VoIP service.</p> <ul style="list-style-type: none"> Use ? to match any single digit. Use * to match any number of digits. <p>Example: 1408*, 1512???1234</p> <p>In the above example, the caller ID either must start with 1408 or must be an 11-digit numbering starting with 1512 and ending with 1234.</p> <p>Default setting: blank</p>
PSTN Access List	<p>A comma-separated list of number patterns that is used to allow or block access to the VoIP gateway based on the destination IP address. If the destination IP address matches a specified pattern, service is allowed without further authentication.</p> <p>Example: 192.168.**, 66.43.12.1??.</p> <p>In the above example, the IP address either must begin with 192.168 or must be in the range of 66.43.12.100-199.</p> <p>The default is blank.</p>
PSTN Caller 1/2/3/4/5/6/7/8 PIN	<p>A PIN number that allows a PSTN caller to access to the VoIP gateway. Calls will be subject to the dial plan specified by the corresponding PSTN Caller DP setting (see below). These settings apply when the PSTN Caller Authentication Method parameter is set to PIN.</p> <p>Default setting: blank</p>
PSTN Caller 1/2/3/4/5/6/7/8 DP	<p>The index number of the dial plan to use upon successful entry of the corresponding PSTN Caller PIN. Dial plans are configured in the Dial Plans section.</p> <p>Default setting: 1</p>

PSTN Timer Values (sec)

Field	Description
VoIP Answer Delay	The number of seconds to wait before auto-answering an inbound VoIP call for the FXO account. The range is 0-255. Default setting: 0
VoIP PIN Digit Timeout	After a VoIP caller is prompted for a PIN or enters a digit, the number of seconds to wait for an entry. The range is 0-255. Default setting: 10
PSTN Answer Delay	After an inbound PSTN call starts ringing, the number of seconds to wait before auto-answering the call. The range is 0-255. Default setting: 16
PSTN PIN Digit Timeout	After a PSTN caller is prompted for a PIN or enters a digit, the number of seconds to wait for an entry. The range is 0-255. Default setting: 10
PSTN-To-VoIP Call Max Dur	The limit on the duration of a PSTN-To-VoIP Gateway Call. Unit is in seconds. 0 means unlimited. The range is 0-2147483647. Default setting: 0
PSTN Ring Thru Delay	After a PSTN call starts ringing, the number of seconds to wait before ring through to Line 1. In order for Line 1 to have the caller ID information, this value must be greater than the time required to complete the PSTN caller ID delivery. The range is 0-255. Default setting: 1
VoIP-To-PSTN Call Max Dur	The limit on the duration of a VoIP-To-PSTN Gateway Call. Unit is in seconds. 0 means unlimited. The range is 0-2147483647. Default setting: 0
PSTN Ring Thru CWT Delay	When a call is active and a new PSTN call starts ringing, the number of seconds to wait before ring through to Line 1 with a Call Waiting Tone. Default setting: 3

Field	Description
VoIP DLG Refresh Intvl	The interval between (SIP) Dialog refresh messages sent by the ATA to detect if the VoIP call-leg is still up. If this value is set to 0, the VoIP call-leg status will not be checked by the ATA. The refresh message is a SIP ReINVITE, and the VoIP peer must response with a 2xx response. If the VoIP peer does not reply or the response is not greater than 2xx, the ATA will disconnect both call legs automatically. The range is 0-255. Default setting: 0
PSTN Ring Timeout	After a ring burst, the number of seconds to wait before concluding that PSTN ring has ceased. The range is 0-255. Default setting: 5
PSTN Dialing Delay	After hook, the number of seconds to wait before dialing a PSTN number. The range is 0-255. Default setting: 1
PSTN Dial Digit Len	The on/off time when the Gateway transmits digits through the Line (FXO) port. The syntax is <i>on-time/off-time</i> , expressed in seconds. The permitted range is 0.05 to 3.00 (up to two decimal places only). Default setting: .1/.1
PSTN Hook Flash Len	The length of the hook flash in seconds. Default setting: .25

PSTN Disconnect Detection

Field	Description
Detect CPC	Choose yes to enable or choose no to disable this feature. CPC is a brief removal of tip-and-ring voltage. If enabled, the ATA will disconnect both call legs when this signal is detected during a gateway call. Default setting: yes

Field	Description
Detect Polarity Reversal	Choose yes to enable or choose no to disable this feature. If enabled, the ATA will disconnect both call legs when this signal is detected during a gateway call. If it is a PSTN gateway call, the first polarity reversal is ignored and the second one triggers the disconnection. For VoIP gateway call, the first polarity reversal triggers the disconnection. Default setting: yes
Detect PSTN Long Silence	Choose yes to enable or choose no to disable this feature. If enabled, the ATA will disconnect both call legs when the PSTN side has no voice activity for a duration longer than the length specified in the Long Silence Duration parameter during a gateway call. Default setting: no
Detect VoIP Long Silence	Choose yes to enable or choose no to disable this feature. If enabled, the ATA will disconnect both call legs when the VoIP side has no voice activity for a duration longer than the length specified in the Long Silence Duration parameter during a gateway call. Default setting: no
PSTN Long Silence Duration	This value is minimum length of PSTN silence (or inactivity) in seconds to trigger a gateway call disconnection if Detect Long Silence is enabled. Default setting: 30
VoIP Long Silence Duration	This value is minimum length of VoIP silence (or inactivity) in seconds to trigger a gateway call disconnection if Detect Long Silence is enabled. Default setting: 30
PSTN 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. Default setting: medium
Min CPC Duration	Specify the minimum duration of a low tip-and-ring voltage (below 1V) for the Gateway to recognize it as a CPC signal or PSTN line removal. Default setting: 0.2

Field	Description
Detect Disconnect Tone	<p>Choose yes to enable or choose no to disable this feature. If enabled, the ATA 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 <i>Disconnect Tone</i> parameter, which depends on the region of the PSTN service.</p> <p>Default setting: yes</p>
Disconnect Tone	<p>This value is the tone script which describes to the ATA 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.</p> <p>Default setting: 480@-30,620@-30;4(.25/.25/1+2)</p> <p>Restrictions:</p> <ul style="list-style-type: none">▪ 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 1 segment set is allowed▪ Total duration of the segment set is interpreted as the minimum duration of the tone to trigger detection▪ 6 segments of on/off time (seconds) can be specified. A 10% margin is used to validated cadence characteristics of the tone.

Field	Description
Disconnect Tone (continued)	Disconnect Tone Script values: US—480@-30,620@-30;4(.25/.25/1+2) UK—400@-30,400@-30; 2(3/0/1+2) France—440@-30,440@-30; 2(0.5/0.5/1+2) Germany—440@-30,440@-30; 2(0.5/0.5/1+2) Netherlands—425@-30,425@-30; 2(0.5/0.5/1+2) Sweden—425@-10; 10(0.25/0.25/1) Norway—425@-10; 10(0.5/0.5/1) Italy—425@-30,425@-30; 2(0.2/0.2/1+2) Spain—425@-10; 10(0.2/0.2/1,0.2/0.2/1,0.2/0.6/1) Portugal—425@-10; 10(0.5/0.5/1) Poland—425@-10; 10(0.5/0.5/1) Denmark—425@-10; 10(0.25/0.25/1) New Zealand—400@-15; 10(0.25/0.25/1) Australia—425@-13; 10(0.375/0.375/1)

International Settings

Field	Description
FXO Country Setting	The country of deployment. This setting applies the relevant regional settings for PSTN calls. Default setting: USA
Tip Ring Voltage Adjustment	Voltage adjustment. The choices are 3.1V, 3.2V, 3.35V, and 3.5V. Default setting: 3.5V.
Ring Frequency Min	The lower limit of the ring frequency used to detect the ring signal. Default setting: 10
SPA To PSTN Gain	dB of digital gain (or attenuation if negative) to be applied to the signal sent from the ATA to the PSTN side. The range is -15 to 12. Default setting: 0
Ring Frequency Max	The higher limit of the ring frequency used to detect the ring signal. Default setting: 100

Field	Description
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 ATA. The range is -15 to 12. Default setting: 0
Ring Validation Time	The minimum signal duration required by the Gateway for recognition as a ring signal. Default setting: 256 ms
Ring Indication Delay	Choose from {0, 512, 768, 1024, 1280, 1536, 1792} (ms). Default setting: 512ms
Ring Timeout	Choose from {0, 128, 256, 384, 512, 640, 768, 896, 1024, 1152, 1280, 1408, 1536, 1664, 1792, 1920} (ms). Default setting: 640 ms
Ring Threshold	Choose from {13.5–16.5, 19.35–2.65, 40.5–49.5} (Vrms). Default setting: 13.5-16.5 Vrms
Line-In-Use Voltage	The voltage threshold at which the ATA 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). Default setting: 30

User 1

Use the *Voice > User 1* page to set the user preferences for the calls through the PHONE port1 .

To open this page: Click **Voice** on the menu bar, and then click **User 1** in the navigation tree. Enter the settings as described below. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

Call Forward Settings

Field	Description
Cfwd All Dest	Forward number for Call Forward All Service. Default setting: blank

Field	Description
Cfwd Busy Dest	Forward number for Call Forward Busy Service. Same as Cfwd All Dest. Default setting: blank
Cfwd No Ans Dest	Forward number for Call Forward No Answer Service. Same as Cfwd All Dest. Default setting: blank
Cfwd No Ans Delay/Forward No Ans Delay	Delay in sec before Call Forward No Answer triggers. Same as Cfwd All Dest. Default setting: 20

Selective Call Forward Settings

Field	Description
Cfwd Sel1-8 Caller	<p>Caller number pattern to trigger Call Forward Selective service. When the caller's phone number matches the entry, the call is forwarded to the corresponding Cfwd Selective Destination (Cfwd Sel1-8 Dest).</p> <ul style="list-style-type: none"> Use ? to match any single digit. Use * to match any number of digits. <p>Example: 1408*, 1512???1234</p> <p>In the above example, a call is forwarded to the corresponding destination if the caller ID either starts with 1408 or is an 11-digit numbering starting with 1512 and ending with 1234.</p> <p>Default setting: blank</p>
Cfwd Sel1-8 Dest	<p>The destination for the corresponding Call Forward Selective caller pattern (Cfwd Sel1-8 Caller).</p> <p>Default setting: blank</p>
Cfwd Last Caller	<p>The number of the last caller; this caller is actively forwarded to the Cfwd Last Dest via the Call Forward Last service. For more information, see Vertical Service Activation Codes.</p> <p>Default setting: blank</p>

Field	Description
Cfwd Last Dest	The destination for the Cfwd Last Caller.
Block Last Caller	The number of the last caller; this caller is blocked via the Block Last Caller Service. For more information, see Vertical Service Activation Codes . Default setting: blank
Accept Last Caller	The number of the last caller; this caller is accepted via the Accept Last Caller Service. For more information, see Vertical Service Activation Codes . Default setting: blank

Speed Dial Settings

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

Supplementary Service Settings (User)

Field	Description
CW Setting	Call Waiting on/off for all calls. Default setting: yes
Block CID Setting	Block Caller ID on/off for all calls. Default setting: no
Block ANC Setting	Block Anonymous Calls on or off. Default setting: no
DND Setting	DND on or off. Default setting: no
CID Setting	Caller ID Generation on or off. Default setting: yes
CWCID Setting	Call Waiting Caller ID Generation on or off. Default setting: yes

Field	Description
Dist Ring Setting	Distinctive Ring on or off. Default setting: yes
Secure Call Setting	<p>If yes, all outbound calls are secure calls by default, without requiring the user to dial a star code first. Default setting: no</p> <ul style="list-style-type: none"> ▪ If Secure Call Setting is set to yes, all outbound calls are secure. However, a user can disable security for a call by dialing *19 before dialing the target number. ▪ If Secure Call Setting is set to No, the user can make a secure outbound call by dialing *18 before dialing the target number. ▪ A user cannot force inbound calls to be secure or not secure; that depends on whether the caller has security enabled or not. <p>Note: This setting is applicable only if Secure Call Serv is set to yes on the line interface. See Line 1 Settings (PHONE Port), page 99.</p>
Message Waiting	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. Default setting: no
Accept Media Loopback Request	<p>Controls how to handle incoming requests for loopback operation. Default setting: automatic</p> <ul style="list-style-type: none"> ▪ never: Never accepts loopback calls; replies 486 to the caller. ▪ automatic: Automatically accepts the call without ringing. ▪ manual: Rings the phone first, and the call must be picked up manually before loopback starts. Default setting: Automatic

Field	Description
Media Loopback Mode	The loopback mode to assume locally when making call to request media loopback. Choices are: Source and Mirror. Default setting: source NOTE If the ATA answers the call, the mode is determined by the caller.
Media Loopback Type	The loopback type to use when making call to request media loopback operation. Choices are Media and Packet. Default setting: media Note that if the ATA answers the call, then the loopback type is determined by the caller (the ATA always picks the first loopback type in the offer if it contains multiple type)

Distinctive Ring Settings

Field	Description
Ring1 - 8 Caller	Caller number pattern to play Distinctive Ring/CWT 1, 2, 3, 4, 5, 6, 7, or 8. 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. The distinctive rings are set on the Regional page. Default setting: blank

Ring Settings

Field	Description
Default Ring	Default ringing pattern, 1–8, for all callers. Default setting: 1
Default CWT	Default CWT pattern, 1–8, for all callers. Default setting: 1
Hold Reminder Ring	Ring pattern for reminder of a holding call when the phone is on-hook. Default setting: 8
Call Back Ring	Ring pattern for call back notification. Default setting: 7

Field	Description
Cfwd Ring Splash Len	Duration of ring splash when a call is forwarded (0 – 10.0s) Default setting: 0
Cblk Ring Splash Len	Duration of ring splash when a call is blocked (0 – 10.0s) Default setting: 0
VMWI Ring Policy	The parameter controls when a ring splash is played when a the VM server sends a SIP NOTIFY message to the ATA indicating the status of the subscriber's mail box. Three settings are available. Default setting: New VM Available <ul style="list-style-type: none"> ▪ New VM Available: Ring as long as there new voicemail messages. ▪ New VM Becomes Available: Ring at the point when the first new voicemail message is received. ▪ New VM Arrives: Ring when the number of new voicemail messages increases.
VMWI Ring Splash Len	Duration of ring splash when new messages arrive before the VMWI signal is applied (0 – 10.0s) Default setting: 0
Ring On No New VM	If enabled, the ATA plays a ring splash when the voicemail server sends SIP NOTIFY message to the ATA indicating that there are no more unread voice mails. Some equipment requires a short ring to precede the FSK signal to turn off VMWI lamp. Default setting: no

PSTN User

Use the *Voice > PSTN User* page to set the user preferences for calls through the LINE port.

To open this page: Click **Voice** on the menu bar, and then click **PSTN** in the navigation tree. Enter the settings as described below. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

PSTN-To-VoIP Selective Call Forward Settings

Field	Description
Cfwd Sel1-8 Caller	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 ATA will not auto-answers the call.
Cfwd Sel1-8 Dest	Eight VoIP destinations to forward a PSTN caller matching the <i>Cfwd Sel x Caller parameter</i> . If this entry is blank, the PSTN caller is blocked for VoIP service.
Cfwd Last Caller	The Caller number that is actively forwarded to Cfwd Last Dest by using the Call Forward Last activation code. Default setting: blank
Cfwd Last Dest	Forward number for the Cfwd Last Caller parameter. Default setting: blank
Block Last Caller	ID of caller blocked via the Block Last Caller service. Default setting: blank
Accept Last Caller	ID of caller accepted via the Accept Last Caller service. Default setting: blank

PSTN-To-VoIP Speed Dial Settings

Field	Description
Speed Dial 2-9	The VoIP number to call when the PSTN caller dials the specified digit. Default setting: blank

PSTN Ring Thru Line 1 Distinctive Ring Settings

Field	Description
Ring1-8 Caller	Eight PSTN Caller Number Patterns such that the corresponding ring will be used to ring through Line 1 if the PSTN caller matches this pattern. The ring patterns are configured on the <i>Voice > Regional</i> page. For more information, see Distinctive Ring Patterns, page 85 . Default setting: blank

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 Cfg}. If Follow Line Cfg is selected, the ring is determined by the distinctive ring settings for Line 1. The ring patterns are configured on the <i>Voice > Regional</i> page. For more information, see Distinctive Ring Patterns, page 85 . Default setting: 2

DECT Line 1 - DECT Line 10

Use the *Voice > DECT Line 1~DECT Line 10* pages to configure the settings for calls using Cisco SPA302D handsets.

To open this page: Click **Voice** on the menu bar, and then click **DECT Line 1~10** in the navigation tree. Enter the settings as described below. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

NOTE When a DECT line receives an incoming from a RU and if no handset is registered to this line or all handsets attached to this line are turned off, the base responds to the RU in 9 seconds with "486 Busy Here."

General

Field	Description
Line Enable	To enable this line for service, select yes. Otherwise, select no. Default setting: 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. Default setting: no
SAS DLG Refresh Intvl	If this value 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 ATA ends this call with a SIP BYE message. The range is 0 to 255 seconds (0 means that the session refresh is disabled) Default setting: 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 value is used by the 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 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. Default setting: blank

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. Default setting: no
NAT Keep Alive Enable	To send the configured NAT keep alive message periodically, select yes. Otherwise, select no. Default setting: 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. Escape sequence of %xx is also accepted. For example, %0d%0a is unescaped into \r\n (CRLF). Default setting: \$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. Default setting: \$PROXY

Network Settings

Field	Description
SIP ToS/DiffServ Value	TOS/DiffServ field value in UDP IP packets carrying a SIP message. Default setting: 0x68
SIP CoS Value	CoS value for SIP messages. Valid values are 0 through 7. Default setting: 3
RTP ToS/DiffServ Value	ToS/DiffServ field value in UDP IP packets carrying RTP data. Default setting: 0xb8
RTP CoS Value [0-7]	CoS value for RTP data. Valid values are 0 through 7. Default setting: 6

Field	Description
Network Jitter Level	Determines how jitter buffer size is adjusted by the 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. Default setting: high
Jitter Buffer Adjustment	Choose yes to enable or no to disable this feature. Default setting: yes

SIP Settings

Field	Description
SIP Transport	The TCP choice provides “guaranteed delivery”, which assures that lost packets are retransmitted. TCP also guarantees that the SIP packages are received in the same order that they were sent. As a result, TCP overcomes the main disadvantages of UDP. In addition, for security reasons, most corporate firewalls block UDP ports. With TCP, new ports do not need to be opened or packets dropped, because TCP is already in use for basic activities such as Internet browsing or e-commerce. Options are: UDP, TCP, TLS. Default setting: UDP
SIP Port	Port number of the SIP message listening and transmission port. Default setting: 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. Default setting: no
EXT SIP Port	The external SIP port number. Default setting: blank

Field	Description
Auth Resync-Reboot	If this feature is enabled, the ATA authenticates the sender when it receives the NOTIFY resync reboot (RFC 2617) message. To use this feature, select yes. Otherwise, select no. Default setting: 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. Default setting: blank
SIP Remote-Party-ID	To use the Remote-Party-ID header instead of the From header, select yes. Otherwise, select no. Default setting: yes
SIP GUID	This feature limits the registration of SIP accounts. The Global Unique ID is generated for each line for each ATA. When it is enabled, the 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. Default setting: no

Field	Description
SIP Debug Option	<p>SIP messages are received at or sent from the proxy listen port. This feature controls which SIP messages to log. The choices are described below.</p> <p>Default setting: none</p> <ul style="list-style-type: none"> ▪ none—No logging. ▪ 1-line—Logs the start-line only for all messages. ▪ 1-line excl. OPT—Logs the start-line only for all messages except OPTIONS requests/responses. ▪ 1-line excl. NTFY—Logs the start-line only for all messages except NOTIFY requests/responses. ▪ 1-line excl. REG—Logs the start-line only for all messages except REGISTER requests/responses. ▪ 1-line excl. OPTINTFYIREG—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. OPTINTFYIREG—Logs all SIP messages in full text except for OPTIONS, NOTIFY, and REGISTER requests/responses.
RTP Log Intvl	<p>The interval for the RTP log.</p> <p>Default setting: 0</p>

Field	Description
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 ATA 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) Default setting: no
Referor Bye Delay	Controls when the 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. Default setting: 4
Refer Target Bye Delay	For the Refer Target Bye Delay, enter the appropriate period of time in seconds. Default setting: 0
Referee Bye Delay	For the Referee Bye Delay, enter the appropriate period of time in seconds. Default setting: 0
Refer-To Target Contact	To contact the refer-to target, select yes. Otherwise, select no. Default setting: no
Sticky 183	If this feature is enabled, the ATA 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. Default setting: no
Use Anonymous With RPID	When set to yes, use "anonymous" in the SIP message. Default setting: yes
Use Local Addr In From	Use the local ATA IP address in the SIP FROM message. Default setting: no

Field	Description
Auth INVITE	When enabled, authorization is required for initial incoming INVITE requests from the SIP proxy. Default setting: no
Reply 182 On Call Waiting	When enabled, the ATA replies with a SIP182 response to the caller if it is already in a call and the line is off-hook. To use this feature select yes. Default setting: no

Call Feature Settings

Field	Description
Blind Attn-Xfer Enable	Enables the 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 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. Default setting: 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. Default setting: blank
Xfer When Hangup Conf	Makes the ATA perform a transfer when a conference call has ended. Select yes or no from the drop-down menu. Default setting: yes
Conference Bridge URL	This feature supports external conference bridging for n-way conference calls ($n > 2$), instead of mixing audio locally. To use this feature, set this parameter to that of the server's name. For example: conf@mysefver.com:12345 or conf (which uses the Proxy value as the domain). Default setting: blank
Conference Bridge Ports	Select the maximum number of conference call participants. The range is 3 to 10. Default setting: 3

Field	Description
Voice Mail Number	The phone number for the voice mail system. Default setting: blank
Mailbox ID	Enter the ID number of the mailbox for this line. Default setting: blank

Proxy and Registration

Field	Description
Proxy	SIP proxy server for all outbound requests. Default setting: blank
Outbound Proxy	SIP Outbound Proxy Server where all outbound requests are sent as the first hop. Default setting: blank
Use Outbound Proxy	Enable the use of <i>Outbound Proxy</i> . If set to no, the <i>Outbound Proxy</i> parameter and <i>Use OB Proxy in Dialog</i> is ignored. Default setting: no
Use OB Proxy In Dialog	Whether to force SIP requests to be sent to the outbound proxy within a dialog. Ignored if the <i>Use Outbound Proxy</i> parameter is no, or if the <i>Outbound Proxy</i> parameter is empty. Default setting: yes
Register	Enable periodic registration with the <i>Proxy</i> . This parameter is ignored if the <i>Proxy</i> parameter is not specified. Default setting: 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. Default setting: yes

Field	Description
Register Expires	Allow answering inbound calls without successful (dynamic) registration by the unit. If proxy responded to REGISTER with a smaller Expires value, the ATA will renew registration based on this smaller value instead of the configured value. If registration failed with an Expires too brief error response, the ATA will retry with the value given in the Min-Expires header in the error response. Default setting: 3600
Ans Call Without Reg	Expires value in sec in a REGISTER request. ATA will periodically renew registration shortly before the current registration expired. This parameter is ignored if the <i>Register</i> parameter is no. Range: 0 – (231 – 1) sec Default setting: yes
Use DNS SRV	If required by your provider, check this box to use DNS SRV lookup for Proxy and Outbound Proxy. Default setting: no
DNS SRV Auto Prefix	If enabled, the ATA will automatically prepend the Proxy or Outbound Proxy name with _sip._udp when performing a DNS SRV lookup on that name. Default setting: no
Proxy Fallback Intvl	This parameter sets the delay (sec) after which the ATA 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 ATA 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 ATA will not attempt to fall back after a fail over). Default setting: 3600

Field	Description
Proxy Redundancy Method	<p>The ATA makes an internal list of proxies returned in DNS SRV records. In normal mode this list will contain proxies ranked by weight and priority.</p> <p>If the parameter <i>Based on SRV port</i> is configured, the ATA creates a list in normal mode first, and then inspects the port numbers based on the 1st proxy's port on the list. Default setting: Normal</p>
Voice Mail Server	The URL or IP address of the voice mail server.
Mailbox Subscribe Expires	<p>The subscription interval for voicemail message waiting indication. When this time period expires, the ATA sends another subscribe message to the voice mail server. Default: 2147483647</p>

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	<p>To use the authentication ID and password for SIP authentication, select yes. Otherwise, select no to use the user ID and password. Default setting: no</p>
Auth ID	The Authentication ID for SIP authentication.
Directory Number	The number for this line.
Resident Online Number	<p>This setting allows you to associate a "local" telephone number with this line using a valid Skype Online Number from Skype. Calls made to that number will ring your phone. Enter the number without spaces or special characters. Default setting: blank</p>

Supplementary Service Subscription

The 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 ATA.

Field	Description
Call Waiting Serv	Enable Call Waiting Service. Default setting: yes
Block CID Serv	Enable Block Caller ID Service. Default setting: yes
Block ANC Serv	Enable Block Anonymous Calls Service Default setting: yes
Dist Ring Serv	Enable Distinctive Ringing Service Default setting: yes
Cfwd All Serv	Enable Call Forward All Service Default setting: yes
Cfwd Busy Serv	Enable Call Forward Busy Service Default setting: yes
Cfwd No Ans Serv	Enable Call Forward No Answer Service Default setting: yes
Cfwd Sel Serv	Enable Call Forward Selective Service. Default setting: yes
Cfwd Last Serv	Enable Forward Last Call Service Default setting: yes
Block Last Serv	Enable Block Last Call Service Default setting: yes
Accept Last Serv	Enable Accept Last Call Service Default setting: yes
DND Serv	Enable Do Not Disturb Service Default setting: yes

Field	Description
CID-Serv	Enable Caller ID Service Default setting: yes
CWCID Serv	Enable Call Waiting Caller ID Service Default setting: yes
Call Return Serv	Enable Call Return Service Default setting: yes
Call Redial Serv	Enable Call Redial Service.
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. Default setting: yes
Three Way Conf Serv	Enable Three Way Conference Service. Three Way Conference is required for Attended Transfer. Default setting: yes
Attn Transfer Serv	Enable Attended Call Transfer Service. Three Way Conference is required for Attended Transfer. Default setting: yes
Unattn Transfer Serv	Enable Unattended (Blind) Call Transfer Service. Default setting: yes
MWI Serv	Enable MWI Service. MWI is available only if a Voice Mail Service is set-up in the deployment. Default setting: yes
VMWI Serv	Enable VMWI Service (FSK) Default setting: yes
Speed Dial Serv	Enable Speed Dial Service. Default setting: yes

Field	Description
Secure Call Serv	<p>Secure Call Service. If this feature is enabled, a user can make a secure call by entering an activation code (*18 by default) before dialing the target number. Then audio traffic in both directions is encrypted for the duration of the call.</p> <p>Default setting: yes</p> <p>For more information about star code settings, see Vertical Service Activation Codes, page 90. To enable secure calling by default, without requiring a star code, set the user's Secure Call Setting to yes. See User 1, page 147.</p>
Referral Serv	<p>Enable Referral Service. See the Referral Services Codes parameter For more information.</p> <p>Default setting: yes</p>
Feature Dial Serv	<p>Enable Feature Dial Service. See the Feature Dial Services Codes parameter For more information.</p> <p>Default setting: yes</p>
Service Announcement Serv	<p>Enable Service Announcement Service.</p> <p>Default setting: no</p>

Audio Configuration

NOTE A codec resource is considered as allocated if it has been included in the SDP codec list of an active call, even though it eventually may not be the one chosen for the connection. So, if the G.729a codec is enabled and included in the codec list, that resource is tied up until the end of the call whether or not the call actually uses G.729a. If the G.729a resource is already allocated and since only one G.729a resource is allowed per device, no other low-bit-rate codec may be allocated for subsequent calls; the only choices are G.711a and G.711u. On the other hand, two G.723.1/G.726 resources are available per device. Therefore it is important to disable the use of G.729a in order to guarantee the support of two simultaneous G.723/G.726 codec.

Field	Description
Preferred Codec, Second Preferred Codec, Third Preferred Codec	Up to three codecs to be used for all calls from this handset, listed order of preference. The actual codec used in a call still depends on the outcome of the codec negotiation protocol. Select one of the following: G.711u, G.711a, G.726-32, G.729a, or G.722. Default setting for Preferred Codec: G.711u Default setting for Second and Third Preferred Codec: Unspecified
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. Default setting: yes
Use Remote Pref Codec	To use the preferred codec specified by the remote peer, select yes. Otherwise, select no. Default setting:
Codec Negotiation	Specify the codecs for codec negotiation: Default or List All. Default setting: Default
Silence Supp Enable	To enable silence suppression so that silent audio frames are not transmitted, select yes. Otherwise, select no. Default setting: no

Field	Description
Silence Threshold	Select the appropriate setting for the threshold: high, medium, or low. Default setting: medium
G729a Enable	To enable the use of the G729a codec at 8 kbps, select yes. Otherwise, select no. Default setting: no
Echo Canc Enable	To enable the use of the echo canceller, select yes. Otherwise, select no. Default setting: yes
G726-32 Enable	To enable the use of the G726 codec at 32 kbps, select yes. Otherwise, select no. Default setting: no
G722 Enable	To enable the use of the G722 codec at 32 kbps, select yes. Otherwise, select no. Default setting: no
DTMF Process INFO	To use the DTMF process info feature, select yes. Otherwise, select no. Default setting: yes
DTMF Process AVT	To use the DTMF process AVT feature, select yes. Otherwise, select no. When set to no, the AVT (RFC2833) payload type is not be included in outbound SDP. Default setting: yes
DTMF Tx Method	Select the method to transmit DTMF signals to the far end: InBand, AVT, INFO, Auto, InBand+INFO, or AVT+INFO. InBand sends DTMF using the audio path. AVT sends DTMF as AVT events. INFO uses the SIP INFO method. Auto uses InBand or AVT based on the outcome of codec negotiation. Default setting: Auto

Field	Description
DTMF Tx Mode	<p>DTMF Detection Tx Mode is available for SIP information and AVT. Options are: Strict or Normal. Default setting: Strict for which the following are true:</p> <ul style="list-style-type: none"> A DTMF digit requires an extra hold time after detection. The DTMF level threshold is raised to -20 dBm. <p>The minimum and maximum duration thresholds are:</p> <ul style="list-style-type: none"> strict mode for AVT: 70 ms normal mode for AVT: 40 ms strict mode for SIP info: 90 ms normal mode for SIP info: 50 ms
Hook Flash Tx Method	<p>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. Default setting: None</p>
Symmetric RTP	<p>Enable symmetric RTP operation. If enabled, the ATA 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 ATA sends RTP to the destination as indicated in the inbound SDP. Default setting: yes</p>

Dial Plan

Field	Description
Dial Plan	<p>The allowed number patterns for outbound calls. For information about the dial plan syntax, see Configuring Dial Plans, page 225. Default setting: (*xxl[3469]11l0l00l[2-9]xxxxxxl1xxx[2-9]xxxxxxS0lxxxxxxxxxxxxxx.)</p>

Field	Description
PSTN Fallback Dial Plan	The dial plan used when PSTN fallback is enabled and in use. Default setting: (S0<:@gw0>)
Enable IP Dialing	Enable or disable IP dialing. If IP dialing is enabled, one can dial [userid@] 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. Default setting: no
Emergency Number	Comma separated list of emergency number patterns. If outbound call matches one of the pattern, the ATA will disable hook flash event handling. The condition is restored to normal after the call ends. Blank signifies that there is no emergency number. Maximum number length is 63 characters. Default setting: blank
Incoming Handset List	The devices that ring when an incoming call is received. Default setting: fxs,1,2,3,4,5,6,7,8,9,10

Call Forward Settings

Field	Description
Cfwd All Dest	Forward number for Call Forward All Service. Default setting: blank
Cfwd Busy Dest	Forward number for Call Forward Busy Service. Same as Cfwd All Dest. Default setting: blank
Cfwd No Ans Dest	Forward number for Call Forward No Answer Service. Same as Cfwd All Dest. Default setting: blank

Field	Description
Cfwd No Ans Delay	Delay in sec before Call Forward No Answer triggers. Same as Cfwd All Dest. Default setting: 20

Supplementary Service Settings

Field	Description
Secure Call Setting	<p>If yes, all outbound calls are secure calls by default, without requiring the user to dial a star code first. Default setting: no</p> <ul style="list-style-type: none"> ▪ If Secure Call Setting is set to yes, all outbound calls are secure. However, a user can disable security for a call by dialing *19 before dialing the target number. ▪ If Secure Call Setting is set to No, the user can make a secure outbound call by dialing *18 before dialing the target number. ▪ A user cannot force inbound calls to be secure or not secure; that depends on whether the caller has security enabled or not. <p>Note: This setting is applicable only if Secure Call Serv is set to yes on the line interface. See Line 1 Settings (PHONE Port), page 99.</p>
Message Waiting	<p>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. Default setting: no</p>

Field	Description
Accept Media Loopback Request	<p>Controls how to handle incoming requests for loopback operation. Default setting: automatic</p> <ul style="list-style-type: none">▪ never: Never accepts loopback calls; replies 486 to the caller.▪ automatic: Automatically accepts the call without ringing.▪ manual: Rings the phone first, and the call must be picked up manually before loopback starts. Default setting: Automatic
Media Loopback Mode	<p>The loopback mode to assume locally when making call to request media loopback. Choices are: Source and Mirror. Default setting: source</p> <p>NOTE If the ATA answers the call, the mode is determined by the caller.</p>
Media Loopback Type	<p>The loopback type to use when making call to request media loopback operation. Choices are Media and Packet. Default setting: media</p> <p>Note that if the ATA answers the call, then the loopback type is determined by the caller (the ATA always picks the first loopback type in the offer if it contains multiple type)</p>

DECT User

Use the *Voice > DECT User* page to set the user preferences for calls using Cisco SPA302D handsets.

To open this page: Click **Voice** on the menu bar, and then click **DECT User** in the navigation tree. Enter the settings as described below. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

General

Field	Description
Call Park Enable	Enables or disables Call Park. Default setting: No
Call Pickup Enable	Enables or disables Call Pickup. Default setting: No
Call Group Pickup Enable	Enables or disables Group Pickup. Default setting: No

Handset 1

Field	Description
Handset Name	<p>Name of the handset can be configured either from the SPA232D web GUI or from the handset GUI. The name can support only alphabets and numbers and the maximum length is 10.</p> <p>Default setting: Handset 1.</p> <p>Similary for Handset 2, 3, 4, and 5, the default setting is Handset 2, 3, 4, 5, respectively.</p> <p>NOTE When multiple handset names are to be changed, they are sent to CSS at a time with a format like this:</p> <p>5# 1: Name1#2: Name2#3: Name3#4: Name4#5: Name5</p> <p>For example: If 5 handsets are requesting name provisioning simultaneously, and their names are "Hank", "White", "Pinkman", "Gus" and "Skyler" from handset1 to handset5 respectively. Then the format as following:</p> <p>5# 1:Hank#2:White#3:Pinkman#4:Gus#5:Skyler</p> <p>If the handset name is changed from handset in Settings>Handset Settings>Handset Name, it will be sent back to base and updated on the web page.</p>
Outgoing DECT Lines	<p>A comma-separated list of the index numbers (1~10) for the lines that are available from this handset for an outgoing call. These lines will be listed on the phone screen when the user displays the call options or holds down the green call button.</p> <p>Example: 1,2,8</p> <p>In this example, a user can select DECT line 1, 2, or 8 for an outbound call.</p> <p>Default setting: 1</p> <p>Note: You also can choose these lines from the DECT Handset Outgoing Line Selection section of the <i>Quick Setup</i> page.</p>

Field	Description
Failover	<p>When this feature is enabled and a call fails through the selected line, the ATA automatically attempts to place the call over another enabled DECT/PSTN line. Select yes to enable this feature or select no to disable it.</p> <p>Default setting: no</p> <p>NOTE Cisco SPA232D now supports PSTN to DECT and DECT to PSTN outgoing line failover.</p>
Deregister	<p>To deregister a handset, select yes. After you submit the settings and the voice module reboots, then the handset is deregistered. At that point, this parameter is reset to the default value.</p> <p>Default setting: no</p>
Bound IPEI	<p>Enter the device's IPEI number (a unique hardware identifier comparable to a MAC address) if you want to bind this device to the specified handset ID, such as Handset 3. The IPEI can be found in the Settings > Phone Info menu on the handset.</p> <p>Default setting: blank</p>

Field	Description
Handset Auto Update	<p>To enable handset auto update select yes, otherwise select no.</p> <p>Default setting: yes</p> <p>NOTE When enabled, one of the following events triggers software update notification from SPA232D to handset:</p> <ul style="list-style-type: none">▪ SPA232D reboots▪ New handset subscribes to SPA232D▪ Registered handset comes in range <p>a. Five minutes after the firmware update notification from SPA232D, the handset gets the version number from SPA232D when it is idle.</p> <p>b. If the version retrieved from SPA232D is different from its local version number, the HS displays a message Software Update in 30 Seconds with 2 options—Update and Delay.</p> <ul style="list-style-type: none">▪ If Delay is selected, the firmware update window pops up again in 1 hour.▪ If Update is selected, the handset performs the upgrade immediately. <p>c. If no soft-key is selected, the handset upgrades automatically in 30 seconds.</p> <p>If the upgrade fails for some reason, a splash window displays "System busy. Will retry in 5 minutes."</p>

Administration Settings

This chapter describes the administrative settings for the ATA. It includes the following sections:

- **Management**
- **Logging**
- **Diagnostics**
- **Factory Defaults**
- **Firmware Upgrade**
- **Configuration Management**
- **Reboot**

Management

Use the *Management* pages to manage web access to the configuration utility and to enable protocols for remote configuration and network management.

- **Web Access Management**
- **TR-069**
- **SNMP**
- **User List (Password Management)**
- **Bonjour**
- **Reset Button**

Web Access Management

Use the *Administration > Management > Web Access Management* page to configure the settings for access to the administration of the ATA.

To open this page: Click **Administration** in the menu bar, and then click **Management > Web Access Management** in the navigation tree. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

Web Access

Field	Description
Admin Access	<p>This feature controls access to the configuration utility from devices that are connected via the ETHERNET (LAN) port.</p> <p>Click Enabled to enable this feature, or click Disabled to disable it. The default setting is Enabled. If you administer and configure the ATA from a computer that is connected to the LAN, this feature must be enabled.</p>
Web Utility Access	<p>Select the protocol to use for access to the configuration utility from a device on the WAN. Choose HTTP and/or HTTPS. For secure Internet access, select HTTPS. The default value is HTTP.</p>

Remote Access

Field	Description
Remote Management	<p>Allows access to the configuration utility from a device that is on the WAN side of the ATA. For example, you could connect from another subnet in your office or from your home computer.</p> <p>Click Enabled to enable this feature, or click Disabled to disable it. The default setting is Disabled. The other fields in this section of the page are available only if you enable this feature.</p> <p>If you attempt to enable this feature while using the default administrator login credentials, you will be prompted to change the credentials. Click OK to acknowledge the warning message. Use the <i>Administration > Management > User List</i> page to change administrator password. For more information, see User List (Password Management), page 185.</p>
Web Utility Access	<p>Select the protocol to use for access to the configuration utility from a device on the WAN side of the ATA. Choose HTTP and/or HTTPS. For secure Internet access, select HTTPS. The default value is HTTP.</p> <p>Include the specified protocol when you enter the address in your web browser. For example, with the HTTPS protocol, a WAN IP address of 203.0.113.50, and the default Remote Management Port of 80, you would enter: <i>https://203.0.113.50:80</i></p>
Remote Upgrade	<p>If you enabled Remote Management, choose whether or not to allow firmware upgrades from a device on the WAN side of the ATA. Click Enabled to enable this feature, or click Disabled to disable it. The default value is Disabled.</p> <p>You can change this setting only when your computer is connected to the configuration utility from the LAN.</p>

Field	Description
Allowed Remote IP Address	You can use this feature to limit access to the configuration utility based on the IP address of a device. Choose Any IP Address to allow access from any external IP address. To specify an external IP address or range of IP addresses, select the second radio button and then enter the desired IP address or range. The default setting is Any IP Address.
Remote Management Port	<p>Enter the port number to use for access to the configuration utility from a device on the WAN side of the ATA. The default port number is 80.</p> <p>Include the specified port when you enter the address in your web browser. For example, with the HTTPS protocol, a WAN IP address of 203.0.113.50, and the default Remote Management Port of 80, you would enter: <i>https://203.0.113.50:80</i></p>

TR-069

Use the *Administration > Management > TR-069* page to configure communication with an Auto-Configuration Server (ACS) via TR-069 CPE WAN Management Protocol (CWMP). TR-069 (Technical Report 069) provides a common platform to manage all voice devices and other customer-premises equipment (CPE) in large-scale deployments. It provides the communication between the CPE and the ACS.

To open this page: Click **Administration** in the menu bar, and then click **Management > TR-069** in the navigation tree.

Enter the settings as described below. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

Field	Description
Status	Click Enabled to enable remote provisioning, or click Disabled to disable this feature. The default setting is Disabled.

Field	Description
ACS URL	The URL for the ACS. The format should be http(s)://xxx.xxx.xxx.xxx:port or xxx.xxx.xxx.xxx:port. The xxx.xxx.xxx.xxx is the domain name or IP address of the ACS server. Both the IP address and the port number are required.
ACS Username	The username for the ACS. The default username is the Organization Unit Identifier (OUI). This value is required and must match the username configured on the ACS.
ACS Password	The password for the ACS. This value is required and must match the password configured on the ACS.
Connection Request Port	The port to use for connection requests
Connection Request Username	The username for connection requests. This value must match the Connection Request Username configured on the ACS.
Connection Request Password	The password for connection requests. This value must match the Connection Request Password configured on the ACS.
Periodic Inform Interval	If Periodic Inform is enabled, the duration, in seconds, between CPE attempts to connect to the ACS. The default value is 86400 seconds.
Periodic Inform Enable	Click Enabled to enable CPE connection requests to the ACS, or click Disabled to disable this feature.
Request Download	If applied, ACS may call the Download RPC after it receives the request from the ATA.

SNMP

Use the *Administration > Management > SNMP* page to set up Simple Network Management Protocol (SNMP) for the ATA.

SNMP is a network protocol that allows network administrators to manage, monitor, and receive notifications of critical events as they occur on the network. The ATA supports SNMPv2 and SNMPv3. It acts as an SNMP agent that replies to SNMP commands from SNMP Network Management Systems. It supports the standard SNMP get, next, and set commands. It also generates SNMP traps to notify the SNMP manager when configured alarm conditions occur. Examples include reboots, power cycles, and INTERNET (WAN) events.

To open this page: Click **Administration** in the menu bar, and then click **Management > SNMP** in the navigation tree.

Enter the settings as described below. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

Settings for SNMPv2

Field	Description
Enabled, Disabled	Click Enabled to enable this feature, or click Disabled to disable it. The default setting is Disabled.
Trusted IP	Choose Any to allow access from any IP address (not recommended). Click Address to specify the IP address and subnet mask of a single SNMP manager or trap agent that can access the ATA through SNMP.
Get/Trap Community	Enter a community string for authentication for SNMP GET commands. The default value is public.
Set Community	Enter a community string for authentication for SNMP SET commands. The default value is private.

Settings for SNMPv3

Field	Description
Enabled, Disabled	Click Enabled to enable this feature, or click Disabled to disable it. The default setting is Disabled.
R/W User	Enter the user name for SNMPv3 authentication. The default value is v3rwuser.
Auth-Protocol	Choose the SNMPv3 authentication protocol from the drop-down list (HMAC-MD5 or HMAC-SHA).
Auth-Password	Enter the authentication password.
PrivProtocol	Choose a privacy authentication protocol from the drop-down list (None or CBC-DES). If you select CBCDES, the privKey encrypts the data portion of the message that is being sent.
Privacy Password	Enter the key for the authentication protocol to use.

Trap Configuration

Field	Description
IP Address	The IP Address of the SNMP manager or trap agent.
Port	The SNMP trap port used by the SNMP manager or trap agent to receive the trap messages. Valid entries are 162 or 1025~65535. The default value is 162.
SNMP Version	The SNMP version in use by the SNMP manager or trap agent. Choose a version from the list.

User List (Password Management)

Use the *Administration > Management > User List* page to manage the two user accounts for the configuration utility. The administrator-level account has the default username **admin** and password **admin**. The user-level account has access to modify a limited set of features. This account has the default username **cisco** and password **cisco**.

For the IVR, no user password is required; the user simply presses # when prompted. The default administrator password is 1234#. You can configure these passwords on the [System](#) page.

To open this page: Click **Administration** in the menu bar, and then click **Management > User List** in the navigation tree.

To update a password:

-
- STEP 1** In the *User List* table, click the pencil icon for the account that you want to update.
- STEP 2** On the *User Account* page, enter the username and password, as described below.
- **Username:** Enter a username.
 - **Old Password (administrator account only):** Enter the existing password. The default administrator password is **admin**. The default guest password is **cisco**.
 - **New Password:** Enter up to 32 characters for your new password.
 - **Confirm New Password:** Enter the new password again, to confirm.
- STEP 3** After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.
-

Bonjour

Use the *Administration > Management > Bonjour* page to enable or disable Bonjour. Bonjour is a service discovery protocol that locates network devices such as computers and servers on your LAN. It may be required by network management systems that you use. When this feature is enabled, the ATA periodically multicasts Bonjour service records to its entire local network to advertise its existence.

To open this page: Click **Administration** in the menu bar, and then click **Management > Bonjour** in the navigation tree.

Click **Enabled** to enable this feature, or click **Disabled** to disable it. The default setting is Enabled. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

Reset Button

To open this page: Click **Administration** in the menu bar, and then click **Management > Reset Button** in the navigation tree.

Click **Enabled** to enable the reset button, or click **Disabled** to disable it. The default setting is Enabled. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

Logging

The ATA allows you to record incoming, outgoing, and DHCP lists for various events that occur on your network. The Incoming Log displays a temporary list of the source IP addresses and destination port numbers for the incoming Internet traffic. The Outgoing Log displays a temporary list of the local IP addresses, destination URLs/IP addresses, and service/port numbers for the outgoing Internet traffic.

See these topics:

- [Log Module](#)
- [Log Setting](#)
- [Log Viewer](#)

Log Module

Use the *Administration > Log > Log Module* page to enable and configure logging.

To open this page: Click **Administration** in the menu bar, and then click **Log > Log Module** in the navigation tree.

NOTE

- As a best practice, Cisco recommends that you enable logging only when needed, and disable logging when you finish the investigation. Logging consumes resources and can impact system performance.
- If you want to enable email or syslog server logging, first specify the email or syslog server settings on the *Log Setting* page.

Enter the settings as described below. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

Field	Description
Status	Click Enabled to enable logging, or click Disabled to disable logging. The default setting is Disabled.
Log - Enable	Check the box in the heading row to enable logging for all services (kernel and system). Alternatively, check the box for kernel or system to enable logging for that service.
Service	The type of service to include: kernel or system.

Field	Description
Priority	<p>The types of events to include in the log. The lowest level of logging is Emergency, which is limited to messages about high impact events. The highest level of logging is Debugging, which includes all message types from Emergency upward.</p> <ul style="list-style-type: none">▪ Emergency: Messages about events, such as an imminent system crash, that make the system unusable. Typically this type of message is broadcast to all users.▪ Alert: Messages about conditions, such as a corrupted system database, that require immediate corrective action.▪ Critical: Messages about serious conditions, such as a disk failure.▪ Error: Messages about conditions that require corrective action but are not critical.▪ Warning: Warnings about possible issues.▪ Notification: Messages about normal but significant conditions that may require attention.▪ Information: Messages that provide information only.▪ Debugging: Messages that are used to debug programs.
Local	<p>Check the box in the heading row to include all services in the local logs that can be viewed in the Log Viewer. Alternatively, check the box for kernel or system to include that service in the local log.</p>
E-Mail	<p>Check the box in the heading row to include all services in the emailed logs, if configured on the <i>Log Setting</i> page. Alternatively, check the box for kernel or system to include that service in the emailed log.</p>

Field	Description
Syslog Server	Check the box in the heading row to include all services in the log file that is transmitted to the syslog server. Alternatively, check the box for kernel or system to include that service in the log file.

Log Setting

If logging is enabled on the *Administration > Log > Log Module* page, the ATA can periodically send the log file to a server or to an email address. Use the *Log Setting* page to enter the information for your syslog server and email account.

NOTE For information about enabling and configuring logging, see [Log Module, page 187](#).

To open this page: Click **Administration** in the menu bar, and then click **Log > Log Setting** in the navigation tree.

Enter the settings as described below. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

Local

Field	Description
Log Size	Enter the maximum size of the log file in kilobytes. Valid values are from 128 to 1024.

Syslog Server

Field	Description
IP Address	Enter the IP address of the syslog server where the messages will be sent.
Port	Enter the port to use on the server. Valid values are from 1 to 65535.
TLS Enabled	Check the box if enabled.

E-Mail

When logging is enabled, you can send logs to an email address by using SMTP.

NOTE Service providers' requirements vary. Be aware that some providers do not allow SMTP email from a free account. Other providers may require a user to log on to a new mailbox before sending emails. For accurate information, read the support documentation from your provider. In your provider's support or help system, search for information about SMTP server settings.

Field	Description
Sender	If you wish to send log entries to an email account, complete all of the fields in this section. Enter a valid email address to identify the sender of the email. Example: user1@company.com
Receiver	Enter a valid email address where the email will be sent. Example: user2@company.com
SMTP Server	Enter the IP address or domain name of the mail server that you will use to send this email. Example: smtp.gmail.com
SMTP Port	Enter the port to use on the SMTP server. The default value is 25. Use the port specified by your email server administrator or service provider.
Subject	Enter a brief description for the subject line of the email. Example: Log from My ATA
Number of Logs	Enter the number of log entries to include in the email. The valid range is 10 to 200.
Interval	Enter the interval, in minutes, at which to send emails. The valid range is 1 to 1440 (24 hours).
Username	Enter the username for the email account that will be used to send these emails. Use the format required by your service provider. Usually it is the full email address. Example: user1@company.com.
Password	Enter the password for the email account that will be used to send these emails.

Log Viewer

If logging is enabled on the *Administration > Log > Log Module* page, you can use the *Log Viewer* page view the logs online and to download the system log file to your computer. You can limit the contents of the log by choosing the types of entries to include and by specifying keywords.

NOTE For information about enabling and configuring logging, see [Log Module, page 187](#).

To open this page: Click **Administration** in the menu bar, and then click **Log > Log Viewer** in the navigation tree.

Field	Description
Download Log	Click this button to download the contents of the log as a file on your computer. In the dialog box, you can open the file or save it. The file can be opened in a text editor such as Notepad.
Clear Log	Click this button to remove all entries from the log.
Display	Choose the type of content to display: All, kernel, or system.
Filter	Enter a keyword to filter the log entries that appear in the viewer. The page will display only the entries that include the keyword.

Diagnostics

The ATA includes two built-in diagnostic tools:

- **Ping Test**
- **Traceroute Test**

Ping Test

Use the *Administration > Diagnostics > Ping Test* page to test connectivity between the ATA and a destination.

To open this page: Click **Administration** in the menu bar, and then click **Diagnostics > Ping Test** in the navigation tree.

-
- STEP 1** Enter the IP address or domain name that you want to ping.
- STEP 2** Enter a packet size in bytes. The range is 32 to 65500 bytes.
- STEP 3** Choose the number of times to send the ping request (5, 10, or Unlimited).
- STEP 4** Click **Start to Ping** to start the test. After the test is complete, the test results appear on the page. While the ping test is running, you can click **Stop** to abandon the test.

The test results indicate the number of packets sent and received, the percentage of packet loss, and the round-trip speed.

- STEP 5** Click **Close** to close the test results and display the *Ping Test* form.
-

Traceroute Test

Use the *Administration > Diagnostics > Traceroute* page to view the route between the ATA and a destination.

To open this page: Click **Administration** in the menu bar, and then click **Diagnostics > Traceroute Test** in the navigation tree.

-
- STEP 1** Enter the IP address or domain name of the destination.
- STEP 2** Click **Start to Traceroute** to start the test. The results appear on the page and are refreshed every 5 seconds. During the test, you can click **Stop** to abandon the test.

The results display up to 30 hops.

STEP 3 Click **Close** to close the results and display the *Traceroute Test* form.

Factory Defaults

Use the *Administration > Factory Defaults* page to reset the ATA to the default configuration. Alternatively, press and hold the RESET button for 20 seconds. All user-changeable non-default settings will be lost. This may include network and service provider data.

To open this page: Click **Administration** in the menu bar, and then click **Factory Defaults** in the navigation tree.

You can perform the following tasks:

- **Restore Router Factory Defaults:** Choose **Yes** to remove any custom data (router) settings that you have configured. The default settings will be restored when you click **Submit**.
- **Restore Voice Factory Defaults:** Choose **Yes** to remove any custom settings that you configured on the *Voice* pages of the configuration utility. The default settings will be restored when you click **Submit**.

Fast Factory Reset

RC Unit:

STEP 1 Trigger Factory Reset.

STEP 2 Reset Router after 3 seconds.

STEP 3 Send command 16 to Reset Voice after 3 seconds.

STEP 4 Perform sp default, send /etc/flat_profile.xml to Voice via vsock after 5 seconds.

STEP 5 Reboot after 30 seconds.

Non-RC Unit:

-
- STEP 1** Trigger Factory Reset.
 - STEP 2** Reset Router after 3 seconds.
 - STEP 3** Send command 16 to Reset Voice after 3 seconds.
 - STEP 4** Reboot after 10 seconds.

Firmware Upgrade

Use the *Administration > Firmware Upgrade* page to upgrade the firmware on the ATA. It is not necessary to upgrade unless you are experiencing problems with the ATA or if the new firmware has a feature that you want to use. Before upgrading the firmware, download the firmware upgrade file for the ATA at: www.cisco.com/go/smallbizvoicegateways

To open this page: Click **Administration** in the menu bar, and then click **Firmware Upgrade** in the navigation tree.

-
- STEP 1** Click **Browse** and select the location of the upgrade file that you downloaded.
 - STEP 2** Click the **Upgrade** button to upgrade the firmware.
-



CAUTION Upgrading the firmware may take several minutes. Until the process is complete, DO NOT turn off the power, press the hardware reset button, or click the Back button in your current browser.

Configuration Management

Use the *Administration > Config Management* pages to backup and restore the configuration settings for the ATA.

- **Backup Configuration**
- **Restore Configuration**

Backup Configuration

Use the *Administration > Config Management > Backup Configuration* page to back up the ATA configuration settings to a file. You can then later restore these same settings to the ATA.

To open this page: Click **Administration** in the menu bar, and then click **Config Management > Backup Configuration** in the navigation tree.

Click the **Backup** button to save the configuration information of the ATA. When the dialog box appears, choose a location where you want to save the .cfg file. **Tip:** Rename the file with a name that includes the date and time when you did the backup.

Restore Configuration

Use the *Administration > Config Management > Restore Configuration* page to restore the ATA configuration settings from a previous backup. It is recommended that you back up your current configuration settings before you restore a configuration.

To open this page: Click **Administration** in the menu bar, and then click **Config Management > Restore Configuration** in the navigation tree.

STEP 1 Click **Browse** to locate the .cfg file on your computer.

STEP 2 Click **Restore** to restore the settings from the selected file.

Reboot

Use the *Administration > Reboot* page to power cycle the ATA (if necessary) from the configuration utility. Alternatively, accomplish this task by pressing the RESET button.

To open this page: Click **Administration** in the menu bar, and then click **Reboot** in the navigation tree.

Click the **Reboot** button to power cycle the ATA. When the warning message appears, read the information, and then click **OK** to reboot the ATA, or click **Cancel** to abandon the operation. The ATA and any connected devices will lose network connectivity during this operation.

NOTE To avoid reboot upon every resync (when a same parameter appears in multiple profile files), the changed parameters are saved in a temporary table and applied to flash after processing all the profile files.

For example:

—profile A : <_Proxy_1_>1.1.1.1<_Proxy_1_>

—profile B: <_Proxy_1_>2.2.2.2<_Proxy_1_>

After all the profile files are processed, Proxy_1 parameter in the flash is saved with 2.2.2.2 and then reboot once. (Proxy_1 from profile A is overwritten and takes no effect.)

Save the parameter to the table if it is not in the table or is different with current flash value, otherwise remove from the table.

Viewing the Status and Statistics

This chapter describes how to view the status and statistics for the ATA. It includes the following sections:

- **System Information**
- **Interface Information**
- **Internet Status**
- **Port Statistics**
- **DHCP Server Information**

System Information

Use the *Status > System Information* page to view information about the ATA and its current settings.

To open this page: Click **Status** on the menu bar, and then click **System Information** in the navigation tree.

Field	Description
Model	The model number and product description.
Hardware Version	The hardware version number.
Boot Version	The boot firmware version number.
Firmware Version	The current firmware version.
Recovery Firmware	The version number of the recovery firmware.
WAN MAC Address	The MAC address of the WAN interface.

Field	Description
Host Name	The host name of the ATA.
Domain Name	The domain name of the ATA.
Serial Number	The serial number of the ATA.
Current Time	Time that is set on the ATA.

Interface Information

Use the *Status > Interface Information* page to view information for the WAN interface (INTERNET port) and the LAN interface (ETHERNET port).

To open this page: Click **Status** on the menu bar, and then click **Interface Information** in the navigation tree.

Interface List

Field	Description
Interface	The name of the interface: WAN or LAN.
Connect Type	The type of connection configured for the interface.
IP Address	The IP address of the interface.
Subnet Mask	The subnet mask of the interface.
MAC Address	The MAC address of the interface.

Port List

Field	Description
Interface	The name of the interface: WAN or LAN.
TX (pkts)	The number of packets transmitted from this port.
RX (pkts)	The number of packets received by this port.

Field	Description
Status	The status of the port, showing whether the port is connected to a device or disconnected.
Clear TX & RX	Click this button to reset to 0 the count of TX and RX packets.

Internet Status

Use the *Status > Internet Status* page to view information about the WAN interface (INTERNET port).

To open this page: Click **Status** on the menu bar, and then click **Internet Status** in the navigation tree.

Field	Description
Link Status	The status of the INTERNET (WAN) interface, showing whether the port is connected or disconnected.
IP Address	The IP address of the INTERNET (WAN) interface.
Netmask	The subnet mask for the INTERNET (WAN) interface.
Gateway	The IP address of the default gateway.
Host Name	The host name of the ATA.
Domain Name	The domain name of the ATA.
MTU Type	The method for setting the MTU: Auto or Manual.
MTU Size	The largest protocol data unit (in bytes) permitted for network transmission.
DNS 1-3 (if applicable)	IP addresses for up to three DNS servers that are used for name resolution.

Port Statistics

Use the *Status > Port Statistics* page to view information about the port activity on the WAN interface (INTERNET port) and the LAN interface (ETHERNET port).

To open this page: Click **Status** on the menu bar, and then click **Port Statistics** in the navigation tree.

Field	Description
Input (pkts)	The number of packets received by the port.
Output (pkts)	The number of packets transmitted by the port.
Input Errors	The number of receive errors for incoming traffic.
Input Broadcasts	The number of broadcast messages received by the interface.
Output Broadcasts	The number of broadcast messages sent by the interface.
Input Multicasts	The number of multicast messages received by the interface.
Output Multicasts	The number of multicast messages sent by the interface.

Memory Information

Use the *Status > Memory Information* page to view information about the total and free memory available.

To open this page: Click **Status** on the menu bar, and then click **Memory Information** in the navigation tree.

Field	Description
Memory Total	The total memory available on the ATA is 284 16 kB.
Memory Free	The memory available for later usage is 8876 kB.

DHCP Server Information

Use the *Status > DHCP Server Information* page to view information about the DHCP server and clients.

To open this page: Click **Status** on the menu bar, and then click **DHCP Server Information** in the navigation tree.

DHCP Pool Information

Field	Description
Client Name	The host name of the DHCP client.
IP Address	The IP address leased to the client.
MAC Address	The MAC address of the DHCP client.
Expires Time	The remaining time in the current DHCP lease, shown in HH:MM:SS (hours:minutes:seconds) format. The page is periodically updated with the new value as the timer counts down.
Interface	The interface through which the client is connected.

DHCP Server Details

Field	Description
DHCP Server	The status of the DHCP server: Enabled or Disabled.
IP Address / Mask	The IP address and subnet mask for the ETHERNET (LAN) interface.
DNS Proxy	The setting for the DNS proxy service: Enabled or Disabled.
Maximum DHCP Users	The maximum number of clients that can lease an IP address from the DHCP server.
IP Address Range	The range of IP addresses that can be dynamically assigned by the DHCP server.

Field	Description
Client Lease Time	The maximum amount of time, in minutes, that a client can lease a dynamically assigned IP address.
Static DNS	The IP addresses of up to three DNS servers to be used by DHCP clients.
Option 66	The setting for Option 66, which provides provisioning server address information to hosts requesting this option. The ATA may be set to None (internal), Remote TFTP Server, or Manual TFTP Server.
TFTP Server	The IP address, hostname, or URL of the TFTP server used for provisioning.
Option 67	The configuration/bootstrap filename that is provided to hosts that request this option.
Option 159	The configuration URL that is provided to clients that request this option.
Option 160	The configuration URL that is provided to clients that request this option.

Frequently Asked Questions

Q. I cannot connect to the Internet through the ATA.

- STEP 1** Make sure that the ATA is powered on. The Power/Sys LED should be solid green and not flashing.

If the Power LED is flashing, then power off all of your network devices, including the modem, the ATA, and the connected devices. Wait for 30 seconds. Then power on each device in the following order:

- Cable or DSL modem
- ATA
- Connected Devices

- STEP 2** Check the cable connections. Ensure that the cable in the INTERNET (WAN) port is securely connected to the device that provides your Internet access, such as your modem or ADSL line. On the Cisco SPA122, check the cable connection for the ETHERNET (LAN) port.

- STEP 3** Check the settings on the *Network Setup > Internet Settings* page. Verify that you entered the settings specified by your Internet Service provider.
-

Q. I upgraded my firmware and now the ATA is not working properly.

If the ATA is not working properly after an upgrade, you may need to perform a factory reset. Use the *Administration > Factory Defaults* page to reset the ATA to the default configuration. Alternatively, press and hold the RESET button for 20 seconds. All user-changeable non-default settings will be lost. This may include network and service provider data.

Q. I cannot use the DSL service to connect manually to the Internet.

After you have installed the ATA, it will automatically connect to your service provider's network, so you no longer need to connect manually.

Q. There is no dial tone, and the Phone 1 or 2 LED is not solid green.

-
- STEP 1** Make sure the telephone is connected to the appropriate port, PHONE 1 or 2.
- STEP 2** Disconnect the RJ-11 telephone cable from the PHONE port, and then reconnect it.
- STEP 3** Make sure your telephone is set to its tone setting (not pulse).
- STEP 4** Make sure your network has an active Internet connection. Try to access the Internet, and check to see if the ATA WAN LED is flashing green. If you do not have a connection, then power off all of your network devices, including the modem, the ATA, and the computers. Wait 30 seconds. Then power on each device in the following order:
- Cable or DSL modem
 - ATA
 - Computers and other devices
- STEP 5** Verify the settings on the *Quick Setup* page. Verify that you entered the account information and settings required by your service provider. On the *Voice > Info* page, *Line 1* or *Line 2 Status* section, verify that the Registration State is registered. If the line is not registered, check with your ITSP to determine if additional settings are required.
-

Q. When I place an Internet phone call, the audio breaks up.

Consider the following possible causes and solutions:

- **Network activity**
There may be heavy network activity, particularly if you are running a server or using a file sharing program. Try to limit network or Internet activity during Internet phone calls. For example, if you are running a file sharing program, files may be uploaded in the background even though you are not downloading any files, so make sure you exit the program before making Internet phone calls.
- **Bandwidth**
There may insufficient bandwidth available for your Internet phone call. You may want to test your bandwidth by using one of the bandwidth tests available online. If necessary, access your Internet phone service account and reduce the bandwidth requirements for your service. For more information, refer to the website of your ITSP.

Q. When I open a web browser, I am prompted for a username and password. How can I bypass this prompt?

Launch the web browser and perform the following steps (these steps are specific to Internet Explorer but are similar for other browsers):

STEP 1 Select **Tools > Internet Options**.

STEP 2 Click the **Connections** tab.

STEP 3 Select **Never dial a connection**.

STEP 4 Click **OK**.

Q. The DSL telephone line does not fit into the ATA WAN (Internet) port.

The ATA does not replace your modem. You need your DSL modem in order to use the ATA. Connect your telephone line to the DSL modem.

Q. My modem does not have an Ethernet port.

If your modem does not have an Ethernet port, then it is a modem for traditional dial-up service. To use the ATA, you need a cable/DSL modem and a high-speed Internet connection.

Q. The ATA does not have a coaxial port for the cable connection.

The ATA does not replace your modem. You need your cable modem in order to use the ATA. Connect your cable connection to the cable modem.

Q. A firmware upgrade (or downgrade) failed.

You can run a firmware recovery procedure to either retry the failed upgrade or install an earlier version of the firmware.

STEP 1 Reboot the ATA.

STEP 2 Connect your computer to the ATA or to a switch that is connected to the ATA.

STEP 3 Start a web browser, and enter the LAN IP address of the ATA. The default LAN IP address is 192.168.15.1.

STEP 4 Log in to the ATA. The default user name and password are both **cisco**. The password is case sensitive. After you log in, the *Recovery Firmware* page appears.

STEP 5 Click **Browse**, select the firmware to install, and then click **Start to Upgrade**.

Using the IVR for Administration

An IVR system is available to help you to configure and manage your ATA. Use a telephone keypad to select options and to make your entries.

To access the IVR menu:

-
- STEP 1** Connect an analog phone to a PHONE port of the ATA.
 - STEP 2** Press the star key four times: ****
 - STEP 3** When challenged for a password, log in as an administrator by using the default administrator's password of **1234#** or log in as the PHONE port's user by pressing **#**.
 - STEP 4** Enter the code for the desired action. See the **IVR Actions** table for details.
-

TIPS:

- Enter the numbers slowly, listening for the audio confirmation before entering the next number.
- After you select an option, press the **#** (pound) key.
- To exit the menu, hang up the telephone or enter **3948#** to exit.
- After entering a value, such as an IP address, press the **#** (pound) key to indicate that you have finished your selection. Then proceed as needed:
 - To save a setting, press **1**.
 - To review a setting, press **2**.
 - To re-enter a setting, press **3**.
 - To cancel your entry and return to the main menu, press ***** (star).
- While entering a value, you can cancel the changes by pressing the ***** (star) key twice within half a second. Be sure to press the key quickly, or the ***** will be treated as a decimal point entry.

- If the menu is inactive for more than one minute, the IVR times out. You will need to re-enter the IVR menu by pressing the star key four times: ****. Your settings take effect after you hang up the telephone or exit the IVR. The ATA may reboot at this time.
- To enter the decimal points in an IP address, press the * (star) key. For example, to enter the IP address 191.168.1.105, perform the following tasks:
 - Press these keys: 191*168*1*105
 - Press the # (pound) key to indicate that you have finished entering the IP address.
 - Press 1 to save the IP address or press the * (star) key to cancel your entry and return to the main menu.

IVR Actions

IVR Action	Menu Option	Choices and Instructions
Enter IVR Menu	****	
Check Internet Addressing Method	100	
Set Internet Addressing Method	101	DHCP: 0 Static IP: 1 PPPoE: Press 2
Check Internet IP Address (INTERNET port)	110	
Set Static IP Address (INTERNET port)	111	Enter the IP address by using numbers on the telephone key pad. Use the * (star) key when entering a decimal point. Note: This option is available only after you choose Static IP as the Internet Connection Type, through option 101.
Check Network Mask	120	

IVR Action	Menu Option	Choices and Instructions
Set Network Mask	121	<p>To enter the value, press numbers on the telephone key pad. Press the * (star) key to enter a decimal point.</p> <p>Note: This option is available only after you choose Static IP as the Internet Connection Type, through option 101.</p>
Check Gateway IP Address	130	
Set Gateway IP Address	131	<p>To enter the value, press numbers on the telephone key pad. Press the * (star) key to enter a decimal point.</p> <p>Note: This option is available only after you choose Static IP as the Internet Connection Type, through option 101.</p>
Check MAC Address	140	
Check Firmware Version	150	
Check Primary DNS Server Setting	160	
Set Primary DNS Server	161	<p>To enter the value, press numbers on the telephone key pad. Press the * (star) key to enter a decimal point.</p> <p>Note: This option is available only after you choose Static IP as the Internet Connection Type, through option 101.</p>
Check INTERNET web server port	170	
Check LAN IP address (ETHERNET port)	210	
Announce Line 1 SIP Transport	1910	

IVR Action	Menu Option	Choices and Instructions
Set Line 1 SIP Transport	1911	0: UDP 1: TCP 2: TLS
Check Line 2 SIP Transport	1920	
Set Line 2 SIP Transport	1921	0: UDP 1: TCP 2: TLS
Exit IVR	3948 (Spells EXIT on the phone keypad)	
Allow or prevent WAN access to the administration web server	7932	1: Enable 0: Disable
Factory Reset of Unit WARNING: All non-default settings will be lost. This includes network and service provider data.	73738 (Spells RESET on the phone keypad)	When prompted, press 1 to confirm, or press * (star) to cancel. After you hear “Option successful,” hang up the phone. The ATA reboots. NOTE: This action is equivalent to Pressing and holding the RESET button for 10 seconds.
Reboot of Voice System	732668 (Spells REBOOT on the phone keypad)	After you hear “Option successful,” hang up the phone. The ATA reboots. NOTE: This action is equivalent to Pressing and immediately releasing the RESET button.

IVR Action	Menu Option	Choices and Instructions
User Factory Reset of Unit WARNING: All user-changeable non-default settings will be lost. This may include network and service provider data.	877778	When prompted, press 1 to confirm, or press * (star) to cancel. After you hear “Option successful,” hang up the phone. The ATA reboots.

Advanced Options for Voice Services

This appendix provides additional information about configuring advanced options for voice services.

STEP 1

- **Optimizing Fax Completion Rates**
- **VoIP-to-PSTN and PSTN-to-VoIP Calling**
- **Call Scenarios**
- **Configuring Dial Plans**

Optimizing Fax Completion Rates

Issues can occur with fax transmissions over IP networks, even with the T.38 standard, which is supported by the ATA. You can adjust several settings on your ATA to optimize your fax completion rates.

STEP 1 Ensure that you have enough bandwidth for the uplink and the downlink.

- For G.711 fallback, it is recommend to have approximately 100 kbps.
- For T.38, allocate at least 50 kbps.

STEP 2 Click **Voice** in the menu bar, and then click **Line 1** or **Line 2** in the navigation tree.

STEP 3 In the *Network Settings* section, enter the following settings:

- **Network Jitter Level:** very high
- **Jitter Buffer Adjustment:** no

STEP 4 In the *Supplementary Service Subscription* section, enter the following settings:

- **Call Waiting Serv:** no

- **Three Way Call Serv:** no

STEP 5 In the *Audio Configuration* section, enter the following settings to support T.38 fax:

- **Preferred Codec:** G.711u (USA) or G.711a (rest of the world)
- **Use pref. codec only:** yes
- **Silence Supp Enable:** no
- **Echo Canc Enable:** no
- **FAX Passthru Method:** ReINVITE

STEP 6 Click **Submit** to save your settings or click **Cancel** to abandon the unsaved settings.

STEP 7 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
```

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

STEP 8 If you are using a Cisco media gateway, make sure the Cisco gateway is correctly configured for T.38 with the dial peer. For example:

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

Fax Troubleshooting

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

STEP 1 Verify that your fax machine is set to a speed between 7200 and 14400.

STEP 2 Send a test fax in a controlled environment between two ATAs.

STEP 3 Determine the success rate.

STEP 4 Monitor the network and record the statistics for jitter, loss, and delay.

STEP 5 If faxes fail consistently, capture a copy of the configuration as described below. You can then send this file to Technical Support.

a. In your web browser, enter the path for the configuration file:

```
http://<ATA_Local_IP_Address>/admin/config.xml&xuser=  
<admin_user>&xpassword=<admin_password>
```

b. On the **File** menu, choose **Save As**, and save the file with a filename such as *MyConfiguration.xml*.

STEP 6 To enable logging, go to the *Voice > System* page, and set the IP address of your syslog and/or debug server. Set the Debug Level to 3. For more information, see [System, page 60](#).

NOTE: You can also capture data using a sniffer trace.

STEP 7 Identify the type of fax machine connected to the ATA.

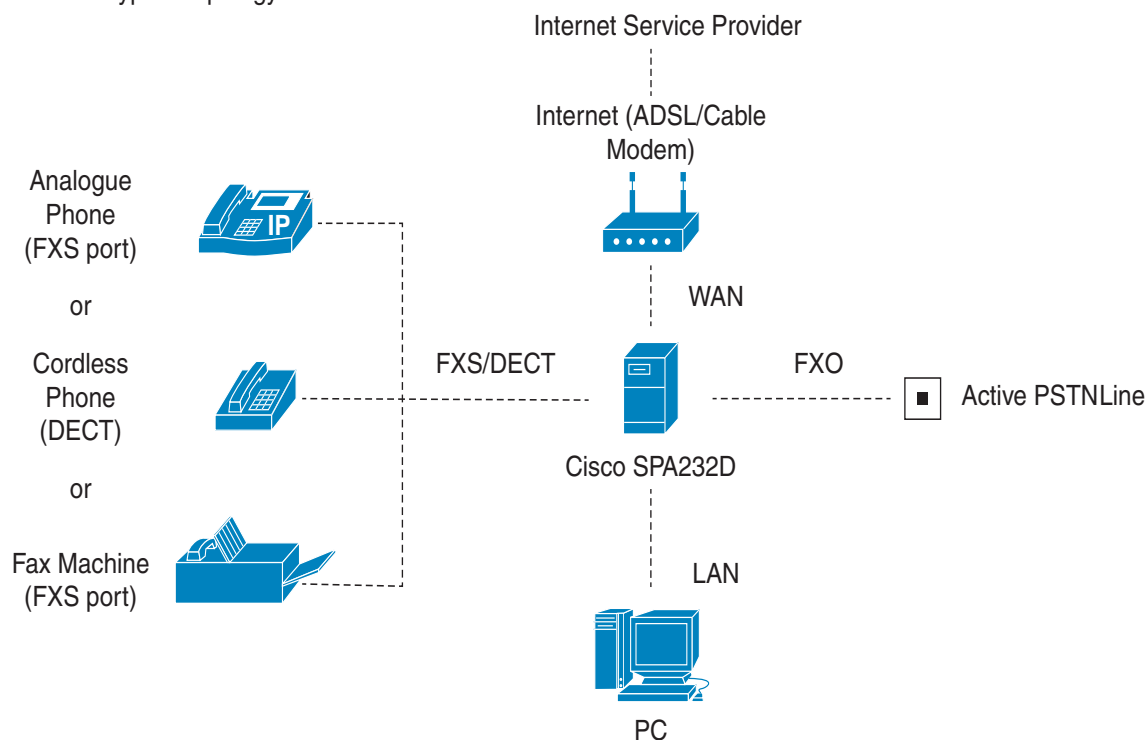
STEP 8 Contact technical support:

- If you are an end user of VoIP products, contact the reseller or service provider that supplied the equipment.
- If you are an authorized Cisco partner, contact Cisco technical support. For contact options, see: www.cisco.com/go/sbsc

VoIP-to-PSTN and PSTN-to-VoIP Calling

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 and Cisco SPA302D Mobility Enhanced Cordless Telephone Handsets.

SPA232D typical topology



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.

How VoIP-To-PSTN Calls Work

To obtain PSTN services through the Cisco SPA302D, the VoIP caller establishes a connection with the PSTN Line by way of a standard SIP INVITE request addressed to the PSTN Line.

One-Stage Dialing

One-stage dialing allows a call to be started over VoIP and then immediately get a dial tone on the PSTN. When you take a phone off hook and dial a number, the call is automatically routed to the VoIP or the PSTN, based on the dial plan.

Optionally, you can enable HTTP Digest Authentication. In this case, the ATA challenges the INVITE with a 401 response if it does not have a valid Authorization header. The Authorization header should include a `<User IDn>` parameter, where `n` refers to one of eight VoIP user accounts that can be configured on the ATA device. The credentials are computed based on the corresponding password using Message Digest 5 (MD5). The `<User IDn>` parameter must match one of the VoIP accounts stored on the ATA device. You can configure these settings on the [PSTN \(LINE Port\)](#) page.

Two-Stage Dialing

In two-stage dialing, the LINE port goes 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 IDn>` 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 ATA (as if no user-id is given) if one-stage dialing is disabled.

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).

You also can enable PIN authentication. In this case, the VoIP caller is prompted to enter a PIN number after the ATA 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 ATA. A dial plan can be selected for each PIN number. If the caller enters a wrong PIN or the ATA times out waiting for more PIN digits, the ATA tears down the call immediately with a BYE request.

The call scenarios may involve the following types of callers:

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

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.

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.

You can configure these settings in the *VoIP-To-PSTN Gateway Setup* section of the **PSTN (LINE Port)** page.

How PSTN-To-VoIP Calls Work

For PSTN-to-VoIP calls, the basic PSTN-to-VoIP call flow is as follows:

1. When a PSTN call comes in to the ATA device and is unanswered (after a configurable number of rings), then the ATA takes the LINE port off hook.
2. The ATA device plays dial tone.
3. The PSTN caller enters the target telephone number. The collected digits are processed by the default dial plan.

You can add PIN authentication to the basic flow:

1. When a PSTN call comes in to the ATA and is unanswered (after a configurable number of rings), then the ATA takes the LINE port off hook.
2. The ATA prompts the caller to enter the PIN number followed by the # key.
3. The ATA compares the PIN to the configured PSTN PIN values.
 - If the PIN matches one of the configured PSTN PIN values, then the ATA plays dial tone. The caller enters the telephone number and the collected digits are processed by the dial plan associated with the PIN number. (These dial plans are configured on the *Voice > PSTN* page, *Dial Plans* section.)

- If the PIN does not match one of the configured PSTN PIN values, then the ATA device plays the reorder tone and then takes the LINE port on-hook.

NOTE You can configure these settings in the *PSTN Timer Values (sec)* section of the **PSTN (LINE Port)** page. You can configure caller authentication in the *Gateway Setup* sections of the **PSTN (LINE Port)** page.

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. When the call terminates, the LINE port goes on-hook so the PSTN line is available for use. The ATA 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 LINE port.
- There is no voice activity for a configurable period of time in either direction at the LINE port.

When any of the above conditions occur, the ATA takes the LINE port on hook and sends a BYE request to end the VoIP call leg. On the other hand, when the ATA device receives a SIP BYE from the VoIP during a call, it takes the LINE port on hook to end the PSTN call leg.

In addition, the ATA device 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 ATA device ends both call legs.

You can configure these settings in the *PSTN Disconnect Detection* section of the **PSTN (LINE Port)** page.

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 that the user enters. The ATA allows flexible call routing with four sets of gateway parameters and configurable dial plans. You can configure this feature in the **Gateway Accounts** section of the **Line 1 Settings (PHONE Port)** page.

You can specify Gateways 1 to 4 in a dial plan by using the identifiers gw1, gw2, gw3, or gw4. Also, gw0 represents the internal PSTN gateway via the LINE 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

The following table lists some examples.

Example	Description
<code><9, :>xx.<:@gw1</code>	Dial 9 to start outside dial tone, followed by one or more digits, and route the call to Gateway 1.
<code>[93]11<:@gw0></code>	Route 911 and 311 calls to the local PSTN gateway
<code><8, :1408>xxxxxxx<:@pstn.cisco.com:5061;usr=joe;pwd=joe_pwd;nat></code>	Dial 8 to start outside dial tone, prepend 1408 followed by seven digits, and route the call to pstn.cisco.com:5061, with user-id = joe, and pwd = bell_pwd, and enable NAT mapping
<code><8, :1408>xxxxxxx<:@gw2:5061;usr="Alex Bell";pwd="anything";nat=no></code>	Dial 8 to start outside dial tone, prepend 1408 followed by seven digits, and route the call to Gateway 2, but use the given port, user-id, and password, and no pstn.cisco.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 ATA, the FXS port is connected to the LINE port. In this case, the telephone attached to the FXS port is electrically connected to the PSTN service via the LINE port. When power is applied to the ATA device, the FXS port is disconnected from the LINE port. However, if the PSTN line is in use when the power is applied to the ATA device, the relay is not flipped until the PSTN line is released. This feature ensures that the ATA device 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), the ATA device can be configured to automatically route all outbound calls to the internal gateway using the parameter listed below.

You can configure this setting on the *Voice > Line 1* page.

Sharing One VoIP Account Between the PHONE and LINE Ports

Both the PHONE port (Line 1) and LINE port (PSTN) can receive incoming calls for a single VoIP account. Consider the following points:

- 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.

- When using the Forward-To-GW0 feature, you can 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 ATA device after the caller is forwarded to gw0.

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

The *Symmetric RTP* parameter is used to send audio RTP to the source IP and port of the inbound RTP packets. This facilitates NAT traversal. You can configure these settings in the **SIP 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 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 the following specifications:

- Number of frequency components
- Frequency and amplitude of each component
- Cadence information

When one VoIP account is shared between the FXS and PSTN Lines, the following parameters are recommended to be set. You can configure these settings on the **Regional** page.

Call Progress Tone	Description
VoIP PIN Tone	This tone is played to prompt a VoIP caller to enter a PIN number.

Call Progress Tone	Description
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 ATA device collects digits from the VoIP caller and processes the number using the dial plan. The ATA device plays the <i>Outside Dial Tone</i> 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 ATA device 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**
- **VoIP to PSTN Call With and Without Authentication**
- **Call Forwarding to PSTN Gateway**

PSTN to VoIP Call with and Without Ring-Thru

The PSTN caller calls the PSTN line connected to the LINE port. Ring-Thru is disabled. After the call rings for a delay equal 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, the caller is prompted to dial the VoIP number. A dial plan is selected according to the PIN number entered by the caller. If authentication is disabled, the default PSTN dial plan is used. Note that the dial plan choice cannot be 0 for a PSTN caller.

NOTE A *PSTN Access List* in terms of Caller ID (ANI) patterns can be configured into the ATA device 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**
- **Using HTTP Digest Authentication**
- **Without Authentication**

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 ATA device then prompts the VoIP caller for a PIN. When a valid PIN is entered, the ATA 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 ATA device).

If the PSTN Line is busy (off-hook, ringing, or PSTN line not connected) when the VoIP caller calls, the ATA device replies with 503. If the PIN number is invalid or entered after the VoIP call leg is connected, the ATA device 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 ATA device 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 ATA device PSTN gateway. When the VoIP caller calls the PSTN Line, the ATA device 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 ATA device.

If the credentials are incorrect, the ATA device challenges the INVITE again. If the auth-ID does not exist in the ATA device configuration, the ATA device replies 403 to the INVITE. If the target number is invalid according to the corresponding dial plan, the ATA device also replies 403 to the INVITE. Again, if the PSTN Line is busy at the time of the call, the ATA device 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**
- **Forward-All to the PSTN gateway**
- **Forward to a Particular PSTN Number**
- **Forward-On-Busy to PSTN Gateway or Number**

You can configure Call Forward settings on the **User 1** page.

Forward-On-No-Answer to the PSTN Gateway

In this scenario, Line 1 is configured to *Cfwd No Ans Dest* to the PSTN Gateway. The scenario is implemented by setting User 1 to forward to gw0 on no answer, with *Cfwd No Ans 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 *Cfwd All Dest* parameter 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 ATA device 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.

Configuring Dial Plans

Dial plans determine how dialed digits are interpreted and transmitted. They also determine whether the dialed number is accepted or rejected. You can use a dial plan to facilitate dialing or to block certain types of calls such as long distance or international. This section includes information that you need to understand dial plans, as well as procedures for configuring your own dial plans.

To edit a dial plan, click **Voice** on the menu bar, and then click **Line 1** or **Line 2** in the navigation tree. Scroll down to the *Dial Plan* section, and then enter the digit sequences in the **Dial Plan** field. For more information and examples, see the following topics:

- [Digit Sequences](#)
- [Acceptance and Transmission of the Dialed Digits](#)
- [Dial Plan Timer \(Off-Hook Timer\)](#)
- [Interdigit Long Timer \(Incomplete Entry Timer\)](#)
- [Interdigit Short Timer \(Complete Entry Timer\)](#)
- [Resetting the Control Timers](#)

Digit Sequences

A dial plan contains a series of digit sequences, separated by the pipe character: | The entire collection of sequences is enclosed within parentheses. Each digit sequence within the dial plan includes a series of elements, which are individually matched to the keys that the user presses.

NOTE White space is ignored, but may be used for readability.

Digit Sequence	Function
0 1 2 3 4 5 6 7 8 9 0 * #	Enter any of these characters to represent a key that the user must press on the phone keypad.
x	Enter x to represent any character on the phone keypad.
[sequence]	Enter characters within square brackets to create a list of accepted key presses. The user can press any one of the keys in the list. <ul style="list-style-type: none">▪ Numeric range: For example, you would enter [2-9] to allow the user to press any one digit from 2 through 9.▪ Numeric range with other characters: For example, you would enter [35-8*] to allow the user to press 3, 5, 6, 7, 8, or *.

Digit Sequence	Function
.	Enter a period for element repetition. The dial plan accepts zero or more entries of the digit. For example, 01. allows users to enter 0, 01, 011, 0111, and so on.
(period)	
<dialled:substituted>	<p>Use this format to indicate that certain dialed digits are replaced by other characters when the sequence is transmitted. The dialed digits can be zero or more characters.</p> <p>EXAMPLE 1: <8:1650>xxxxxxxx</p> <p>When the user presses 8 followed by a seven digit number, the system automatically replaces the dialed 8 with 1650. If the user dials 85550112, the system transmits 16505550112.</p> <p>EXAMPLE 2: <:1>xxxxxxxxxxx</p> <p>In this example, no digits are replaced. When the user enters a 10-digit string of numbers, the number 1 is added at the beginning of the sequence. If the user dials 9725550112, the system transmits 19725550112.</p>
,	Enter a comma between digits to play an “outside line” dial tone after a user-entered sequence.
(comma)	<p>EXAMPLE: 9, 1xxxxxxxxxxx</p> <p>An “outside line” dial tone is sounded after the user presses 9, and the tone continues until the user presses 1.</p>
!	Enter an exclamation point to prohibit a dial sequence pattern.
(exclamation point)	<p>EXAMPLE: 1900xxxxxxxx!</p> <p>The system rejects any 11-digit sequence that begins with 1900.</p>
*xx	Enter an asterisk to allow the user to enter a 2-digit star code.

Digit Sequence	Function
S0 or L0	Enter S0 to reduce the short inter-digit timer to 0 seconds, or enter L0 to reduce the long inter-digit timer to 0 seconds.

Digit Sequence Examples

The following examples show digit sequences that you can enter in a dial plan.

In a complete dial plan entry, sequences are separated by a pipe character (|), and the entire set of sequences is enclosed within parentheses.

EXAMPLE: ([1-8]xx | 9, xxxxxxxx | 9, <:1>[2-9]xxxxxxxxxx | 8, <:1212>xxxxxxxx | 9, 1 [2-9] xxxxxxxx | 9, 1 900 xxxxxxxx ! | 9, 011xxxxxx. | 0 | [49]11)

NOTE Red text is used to highlight the elements that are explained in the examples.

- **Extensions on your system**

([1-8]xx | 9, xxxxxxxx | 9, <:1>[2-9]xxxxxxxxxx | 8, <:1212>xxxxxxxx | 9, 1 [2-9] xxxxxxxx | 9, 1 900 xxxxxxxx ! | 9, 011xxxxxx. | 0 | [49]11)

[1-8]xx Allows a user dial any three-digit number that starts with the digits 1 through 8. If your system uses four-digit extensions, you would instead enter the following string: [1-8]xxx

- **Local dialing with seven-digit number**

([1-8]xx | 9, xxxxxxxx | 9, <:1>[2-9]xxxxxxxxxx | 8, <:1212>xxxxxxxx | 9, 1 [2-9] xxxxxxxx | 9, 1 900 xxxxxxxx ! | 9, 011xxxxxx. | 0 | [49]11)

9, xxxxxxxx After a user presses 9, an external dial tone sounds. The user can then dial any seven-digit number, as in a local call.

- **Local dialing with 3-digit area code and a 7-digit local number**

([1-8]xx | 9, xxxxxxxx | 9, <:1>[2-9]xxxxxxxxxx | 8, <:1212>xxxxxxxx | 9, 1 [2-9] xxxxxxxx | 9, 1 900 xxxxxxxx ! | 9, 011xxxxxx. | 0 | [49]11)

9, <:1>[2-9]xxxxxxxxxx This example is useful where a local area code is required. After a user presses 9, an external dial tone sounds. The user must enter a 10-digit number that begins with a digit 2 through 9. The system automatically inserts the 1 prefix before transmitting the number to the

carrier.

- **Local dialing with an automatically inserted 3-digit area code**

```
( [1-8]xx|9, xxxxxxx|9, <:1>[2-9]xxxxxxxx|8, <:1212>xxxxxx|9, 1 [2-9]
xxxxxxxx|9, 1 900 xxxxxxx !|9, 011xxxxx.|0|[49]11 )
```

8, <:1212>xxxxxx This example is useful where a local area code is required by the carrier but the majority of calls go to one area code. After the user presses 8, an external dial tone sounds. The user can enter any seven-digit number. The system automatically inserts the 1 prefix and the 212 area code before transmitting the number to the carrier.

- **U.S. long distance dialing**

```
( [1-8]xx|9, xxxxxxx|9, <:1>[2-9]xxxxxxxx|8, <:1212>xxxxxx|9, 1 [2-9]
xxxxxxxx|9, 1 900 xxxxxxx !|9,011xxxxx.|0|[49]11 )
```

9, 1 [2-9] xxxxxxxxx After the user presses 9, an external dial tone sounds. The user can enter any 11-digit number that starts with 1 and is followed by a digit 2 through 9.

- **Blocked number**

```
( [1-8]xx|9, xxxxxxx|9, <:1>[2-9]xxxxxxxx|8, <:1212>xxxxxx|9, 1 [2-9]
xxxxxxxx|9, 1 900 xxxxxxx !|9, 011xxxxx.|0|[49]11 )
```

9, 1 900 xxxxxxx ! This digit sequence is useful if you want to prevent users from dialing numbers that are associated with high tolls or inappropriate content, such as 1-900 numbers in the United States. After the user press 9, an external dial tone sounds. If the user enters an 11-digit number that starts with the digits 1900, the call is rejected.

- **U.S. international dialing**

```
( [1-8]xx|9, xxxxxxx|9, <:1>[2-9]xxxxxxxx|8, <:1212>xxxxxx|9, 1 [2-9]
xxxxxxxx|9, 1 900 xxxxxxx !|9, 011xxxxx.|0|[49]11 )
```

9, 011xxxxxx. After the user presses 9, an external dial tone sounds. The user can enter any number that starts with 011, as in an international call

from the United States.

▪ **Informational numbers**

([1-8]xx | 9, xxxxxxx | 9, <:1>[2-9]xxxxxxxx | 8, <:1212>xxxxxxx | 9, 1 [2-9]xxxxxxxx | 9, 1 900 xxxxxxx ! | 9, 011xxxxxx. | **0 | [49]11**)

0 | [49]11 This example includes two digit sequences, separated by the pipe character. The first sequence allows a user to dial 0 for an operator. The second sequence allows the user to enter 411 for local information or 911 for emergency services.

Acceptance and Transmission of the Dialed Digits

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. When a terminating event occurs, the ATA either accepts the user-dialed sequence and initiates a call, or else rejects the sequence as invalid. The user hears the reorder (fast busy) tone if the dialed sequence is invalid.

The following table explains how terminating events are processed.

Terminating Event	Processing
The dialed digits do not match any sequence in the dial plan.	The number is rejected.
The dialed digits exactly match one sequence in the dial plan.	<ul style="list-style-type: none">▪ If the sequence is allowed by the dial plan, the number is accepted and is transmitted according to the dial plan.▪ If the sequence is blocked by the dial plan, the number is rejected.

Terminating Event	Processing
A timeout occurs.	<p>The number is rejected if the dialed digits are not matched to a digit sequence in the dial plan within the time specified by the applicable interdigit timer.</p> <ul style="list-style-type: none"> ▪ The Interdigit Long Timer applies when the dialed digits do not match any digit sequence in the dial plan. Default setting: 10 seconds ▪ The Interdigit Short Timer applies when the dialed digits match one or more candidate sequences in the dial plan. Default setting: 3 seconds
The user presses the # key.	<ul style="list-style-type: none"> ▪ If the sequence is complete and is allowed by the dial plan, the number is accepted and is transmitted according to the dial plan. ▪ If the sequence is incomplete or is blocked by the dial plan, the number is rejected.

Dial Plan Timer (Off-Hook Timer)

You can think of the Dial Plan Timer as “the off-hook timer.” This timer starts counting when the phone goes off hook. If no digits are dialed within the specified number of seconds, the timer expires and the null entry is evaluated. Unless you have a special dial plan string to allow a null entry, the call is rejected. Default setting: 5

Syntax for the Dial Plan Timer

`(PS<:n> | dial plan)`

- **s**: The number of seconds; if no number is entered after **P**, the default timer of 5 seconds applies.
- **n**: (optional): The number to transmit automatically when the timer expires; you can enter a valid number. No wildcard characters are allowed because the number will be transmitted as shown. If you omit the number substitution, `<n>`, then the user hears a reorder (fast busy) tone after the specified number of seconds.

Examples for the Dial Plan Timer

NOTE Red text is used to highlight the elements that are explained in the examples.

- **Allow more time for users to start dialing after taking a phone off hook.**

(P9 | (9,8<:1408>[2-9]xxxxxx | 9,8,1[2 9]xxxxxxxxxx | 9,8,011xx. | 9,8,xx.|[1-8]xx)

P9 After taking a phone off hook, a user has 9 seconds to begin dialing. If no digits are pressed within 9 seconds, the user hears a reorder (fast busy) tone. By setting a longer timer, you allow more time for users to enter the digits.

xx This code allows the entry of one or more digits. Do not use a single x, allowing 0 or more digits. This setting will produce unwanted results especially if you are deploying timers.

- **Create a hotline for all sequences on the System Dial Plan**

(P9<:23> | (9,8<:1408>[2-9]xxxxxx | 9,8,1[2-9]xxxxxxxxxx | 9,8,011xx. | 9,8,xx.|[1-8]xx)

P9<:23> After taking the phone off hook, a user has 9 seconds to begin dialing. If no digits are pressed within 9 seconds, the call is transmitted automatically to extension 23.

- Create a hotline on a line button for an extension

(P0 <:1000>)

With the timer set to 0 seconds, the call is transmitted automatically to the specified extension when the phone goes off hook.

Interdigit Long Timer (Incomplete Entry Timer)

You can think of this timer as the “incomplete entry” timer. This timer measures the interval between dialed digits. It applies as long as the dialed digits do not match any digit sequences in the dial plan. Unless the user enters another digit within the specified number of seconds, the entry is evaluated as incomplete, and the call is rejected. Default setting: 10 seconds

NOTE This section explains how to edit a timer as part of a dial plan. Alternatively, you can modify the Control Timer that controls the default interdigit timers for all calls. See [Resetting the Control Timers, page 234](#).

Syntax for the Interdigit Long Timer

L:*s*, (*dial plan*)

s: The number of seconds; if no number is entered after **L:**, the default timer of 5 seconds applies. The timer sequence appears to the left of the initial parenthesis for the dial plan.

Example for the Interdigit Long Timer

L:15, (9,8<:1408>[2-9]xxxxxx | 9,8,1[2-9]xxxxxxxxxx | 9,8,011xx. | 9,8,xx.[1-8]xx)

L:15, This dial plan allows the user to pause for up to 15 seconds between digits before the Interdigit Long Timer expires.

Interdigit Short Timer (Complete Entry Timer)

You can think of this timer as the “complete entry” timer. This timer measures the interval between dialed digits. It applies when the dialed digits match at least one digit sequence in the dial plan. Unless the user enters another digit within the specified number of seconds, the entry is evaluated. If it is valid, the call proceeds. If it is invalid, the call is rejected. Default setting: 3 seconds

Syntax for the Interdigit Short Timer

SYNTAX 1: **S:***s*, (*dial plan*)

Use this syntax to apply the new setting to the entire dial plan within the parentheses.

SYNTAX 2: *sequence Ss*

Use this syntax to apply the new setting to a particular dialing sequence.

s: The number of seconds; if no number is entered after **S**, the default timer of 5 seconds applies.

Examples for the Interdigit Short Timer

Set the timer for the entire dial plan.

S:6, (9,8<:1408>[2-9]xxxxxx | 9,8,1[2-9]xxxxxxxxxx | 9,8,011xx. | 9,8,xx.[1-8]xx)

S:6, While entering a number with the phone off hook, a user can pause for up to 15 seconds between digits before the Interdigit Short Timer expires.

Set an instant timer for a particular sequence within the dial plan.

(9,8<:1408>[2-9]xxxxxx | **9,8,1[2-9]xxxxxxxxxxS0** | 9,8,011xx. | 9,8,xx.[1-8]xx)

9,8,1[2-9]xxxxxxxxxS0 With the timer set to 0, the call is transmitted automatically when the user dials the final digit in the sequence.

Resetting the Control Timers

You can use the following procedure to reset the default timer settings for all calls.

- NOTE** To edit a timer setting only for a particular digit sequence or type of call, you can edit the dial plan. See [Digit Sequences, page 226](#).
-
- STEP 1** Log in to the configuration utility. If prompted, enter the administrative logon provided by the Service Provider. The default username and password are both **admin**.
- STEP 2** Under the Voice menu, click **Regional**.
- STEP 3** In the *Control Timer Values* section, enter the desired values in the *Interdigit Long Timer* field and the *Interdigit Short Timer* field. See the definitions at the beginning of this section.
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Where to Go From Here

Cisco provides a wide range of resources to help you and your customer obtain the full benefits of the ATA.

Support

Cisco Small Business Support Community	www.cisco.com/go/smallbizsupport
Online Technical Support and Documentation (Login Required)	www.cisco.com/support
Cisco Small Business Support and Resources	www.cisco.com/go/smallbizhelp

Downloads and Documentation

Firmware	www.cisco.com/go/software
Cisco Small Business Voice Gateways Documentation	www.cisco.com/go/smallbizvoicegateways
Open Source Documentation	Follow the Release Notes link at www.cisco.com/go/smallbizvoicegateways

Cisco Small Business

Cisco Partner Central for Small Business (Partner Login Required)	www.cisco.com/web/partners/sell/smb
Cisco Small Business Home	www.cisco.com/smb