Cisco ATA 191 and ATA 192 Analog Telephone Adapter Administration Guide for Multiplatform Firmware

First Published: 2018-02-05

Last Modified: 2023-08-31

Americas Headquarters
Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
http://www.cisco.com
Tel: 408 526-4000
800 553-NETS (6387)
Fax: 408 527-0883

THE SPECIFICATIONS AND INFORMATION REGARDING THE PRODUCTS IN THIS MANUAL ARE SUBJECT TO CHANGE WITHOUT NOTICE. ALL STATEMENTS, INFORMATION, AND RECOMMENDATIONS IN THIS MANUAL ARE BELIEVED TO BE ACCURATE BUT ARE PRESENTED WITHOUT WARRANTY OF ANY KIND, EXPRESS OR IMPLIED. USERS MUST TAKE FULL RESPONSIBILITY FOR THEIR APPLICATION OF ANY PRODUCTS.

THE SOFTWARE LICENSE AND LIMITED WARRANTY FOR THE ACCOMPANYING PRODUCT ARE SET FORTH IN THE INFORMATION PACKET THAT SHIPPED WITH THE PRODUCT AND ARE INCORPORATED HEREIN BY THIS REFERENCE. IF YOU ARE UNABLE TO LOCATE THE SOFTWARE LICENSE OR LIMITED WARRANTY, CONTACT YOUR CISCO REPRESENTATIVE FOR A COPY.

The following information is for FCC compliance of Class A devices: This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to part 15 of the FCC rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio-frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference, in which case users will be required to correct the interference at their own expense.

The following information is for FCC compliance of Class B devices: This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If the equipment causes interference to radio or television reception, which can be determined by turning the equipment off and on, users are encouraged to try to correct the interference by using one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

Modifications to this product not authorized by Cisco could void the FCC approval and negate your authority to operate the product.

The Cisco implementation of TCP header compression is an adaptation of a program developed by the University of California, Berkeley (UCB) as part of UCB's public domain version of the UNIX operating system. All rights reserved. Copyright © 1981, Regents of the University of California.

NOTWITHSTANDING ANY OTHER WARRANTY HEREIN, ALL DOCUMENT FILES AND SOFTWARE OF THESE SUPPLIERS ARE PROVIDED "AS IS" WITH ALL FAULTS. CISCO AND THE ABOVE-NAMED SUPPLIERS DISCLAIM ALL WARRANTIES, EXPRESSED OR IMPLIED, INCLUDING, WITHOUT LIMITATION, THOSE OF MERCHANTABILITY, FITNESS FOR A PARTICULAR PURPOSE AND NONINFRINGEMENT OR ARISING FROM A COURSE OF DEALING, USAGE, OR TRADE PRACTICE.

IN NO EVENT SHALL CISCO OR ITS SUPPLIERS BE LIABLE FOR ANY INDIRECT, SPECIAL, CONSEQUENTIAL, OR INCIDENTAL DAMAGES, INCLUDING, WITHOUT LIMITATION, LOST PROFITS OR LOSS OR DAMAGE TO DATA ARISING OUT OF THE USE OR INABILITY TO USE THIS MANUAL, EVEN IF CISCO OR ITS SUPPLIERS HAVE BEEN ADVISED OF THE POSSIBILITY OF SUCH DAMAGES.

Any Internet Protocol (IP) addresses and phone numbers used in this document are not intended to be actual addresses and phone numbers. Any examples, command display output, network topology diagrams, and other figures included in the document are shown for illustrative purposes only. Any use of actual IP addresses or phone numbers in illustrative content is unintentional and coincidental.

All printed copies and duplicate soft copies of this document are considered uncontrolled. See the current online version for the latest version.

Cisco has more than 200 offices worldwide. Addresses and phone numbers are listed on the Cisco website at www.cisco.com/go/offices.

CONTENTS

CHAPTER 1

Get Started

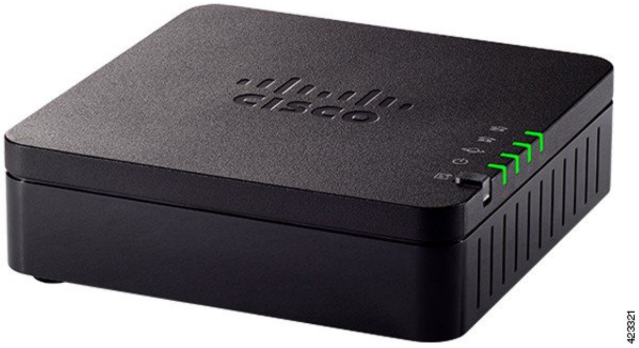
Your Analog Telephone Adapter

The ATA 191 and ATA 192 analog telephone adapters are telephony-device-to-Ethernet adapters that allow regular analog phones to operate on IP-based telephony networks. Both models support two voice ports, each with an independent phone number. Both have an RJ-45 10/100BASE-T data port while the ATA 192 has an additional Ethernet port.

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 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.

Figure 1: Cisco Analog Telephone Adapter



ATA 191 and ATA 192 Top Panel

The following figure shows the different LEDs and buttons found on the top of your ATA.

Figure 2: ATA 191 and ATA 192 Top Panel

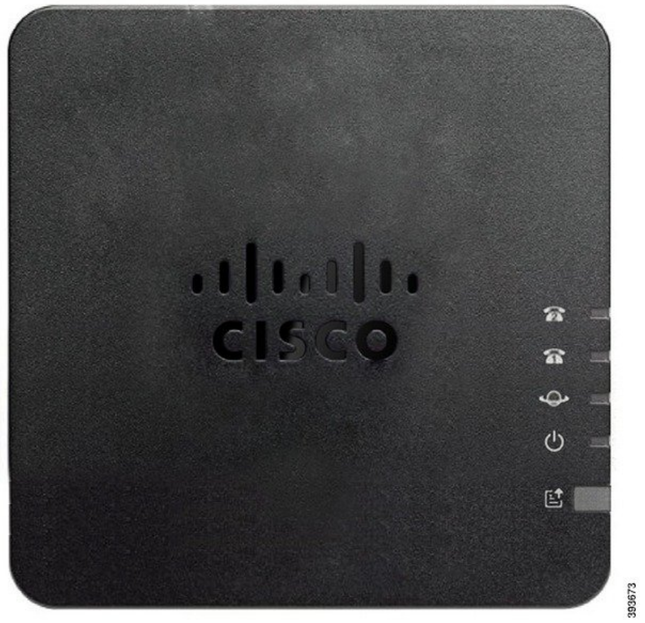


Table 1: ATA 191 and ATA 192 Top Panel Items

Item	Description
Power LED ⏻	**Steady green:** System booted up successfully and is ready for use. **Slow flashing green:** System is booting up. **Fast flashing green three times, then repeats:** System failed to boot up. **Fast flashing green:** The LED behaviour occurs in the following situations: • System detects a factory reset. To perform a factory reset, press and hold the **RESET** button for about 10 seconds. • A factory reset is performed successfully. **Off:** Power is off.
Network LED 🌐	**Flashing green:** Data transmission or reception is in progress through the WAN port. **Off:** No link.

Item	Description
Phone 1 LED Phone 2 LED ☎ ☎	**Steady green:** On hook. **Slow flashing green:** Off hook. **Fast flashing green three times, then repeats:** The analog device failed to register. **Fast flashing green:** A factory reset is performed successfully. **Off:** The port is not configured.
Problem Report Tool (PRT) Button 📤	Press this button to create a problem report using the Problem Report Tool. **Note** This button is not a power button. When you press this button, a problem report is generated and uploaded to a server for the system administrator.
Problem Report Tool (PRT) LED 📤	**Flashing amber:** The PRT is preparing the data for the problem report. **Fast Flashing amber:** The PRT is sending the problem report log to the HTTP server. **Solid green for five seconds, then off:** The PRT report was sent successfully. **Fast flashing green:** A factory reset is performed successfully. **Flashing red:** The PRT report failed. Press the PRT button once to cancel the flashing, then press again to trigger a new PRT.

Problem Report Tool Button

The Problem Report Tool (PRT) button is on the ATA top panel. Press the PRT button, and a log file is prepared and uploaded to the server for troubleshooting your network.

You can instruct your analog phone users to press the PRT button on the ATA device to start the PRT log file process.

One of the following must be completed to upload the PRT log file from the ATA:

- Set up the HTTP server to upload the PRT log file from the ATA.
- Configure the customer support upload URL to best suit your needs, and apply it to the ATA.

ATA 191 and ATA 192 Back Panel

The following figures shows the different ports and buttons found on the back of your ATA.

Figure 3: ATA 191 Back Panel

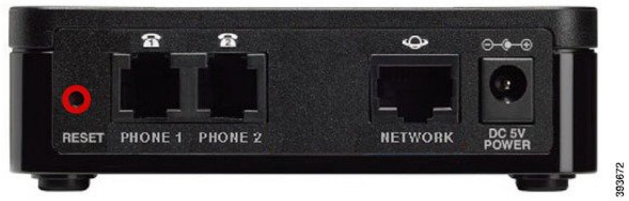


Figure 4: ATA 192—Back Panel

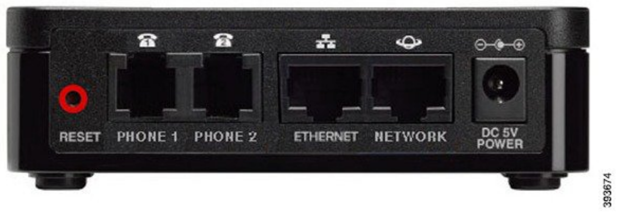


Table 2: ATA 191 and ATA 192 Back Panel Items

Item	Description
RESET	To restart the ATA, use a paper clip or similar object to press this button briefly. To restore the factory default settings, press and hold for about 10 seconds. The LED behaviour for the factory reset: 1. After you press and hold the button for about 10 seconds, the **Power LED** is fast flashing green. 2. After the factory reset is performed successfully, all LEDs are fast flashing green for about 5 seconds.
PHONE 1	Use an RJ-11 phone cable to connect an analog phone or fax machine.
PHONE 2	Use an RJ-11 phone cable to connect a second analog phone or fax machine.
ETHERNET (ATA 192 only)	Use an Ethernet cable to connect your ATA to a device on your network, such as a computer.
NETWORK	Use an Ethernet cable to connect to the network.
DC 5V POWER	Use the power adapter that was provided to connect to a power source.

Install Your Cisco ATA

You can use either Category 3/5/5e/6 cabling for 10-Mbps connections, but you must use Category 5/5e/6 for 100-Mbps connections.

Procedure

Step 1 Connect the power supply to the Cisco DC Adapter port.

Step 2 Connect a straight-through Ethernet cable from the network to the network port on the ATA. Each ATA ships with one Ethernet cable in the box.

ATA Voice Quality

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

Supported Codecs

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

Table 3: Supported Codecs

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 anyof the available codecs.
G.726-32	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 anyof the available codecs.
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

An average SIP Proxy Server can handle tens of thousands of subscribers. A backup server allows an active server to be temporarily switched out for maintenance. The ATA supports the use of backup servers to minimize or eliminate service disruption.

A simple way to support proxy redundancy is to specify a SIP Proxy Server. The ATA sends a DNS NAPTR or SRV query to the DNS server. If configured, the DNS server returns SRV records that contain a list of servers for the domain, with their hostnames, priority, listening ports, and so forth. The ATA tries to contact the servers in the order of the priority. The server with a lower number has a higher priority. Up to 6 NAPTR records and 12 SRV records are supported in a query. And an SRV record can associate with up to 10 A records.

When the ATA fails to communicate with the primary server, the ATA can failover to a lower-priority server. If configured, the ATA can restore the connection back to the primary. Failover and failback support switches between servers with different SIP transport protocols. The ATA doesn't perform failback to the primary server during an active call until the call ends and the failback conditions are met.

Example of Resource Records from the DNS Server

```
as1bsoft     3600    IN NAPTR 50   50  "s"  "SIPS+D2T"     ""  _sips._tcp.tlstest
             3600    IN NAPTR 90   50  "s"  "SIP+D2T"      ""  _sip._tcp.tcptest
             3600    IN NAPTR 100  50  "s"  "SIP+D2U"      ""  _sip._udp.udptest

_sips._tcp.tlstest  SRV 1 10 5061 srv1.sipurash.com.
                    SRV 2 10 5060 srv2.sipurash.com.
_sip._tcp.tcptest   SRV 1 10 5061 srv3.sipurash.com.
                    SRV 2 10 5060 srv4.sipurash.com.
_sip._udp.udptest   SRV 1 10 5061 srv5.sipurash.com.
                    SRV 2 10 5060 srv6.sipurash.com.

srv1     3600    IN    A   1.1.1.1
srv2     3600    IN    A   2.2.2.2
srv3     3600    IN    A   3.3.3.3
srv4     3600    IN    A   4.4.4.4
srv5     3600    IN    A   5.5.5.5
srv6     3600    IN    A   6.6.6.6
```

The following example shows the priority of the servers from the perspective of the ATA.

```
Priority       IP Address     SIP Protocol    Status
1st            1.1.1.1            TLS            UP
2nd            2.2.2.2            TLS            UP
3rd            3.3.3.3            TCP            UP
4th            4.4.4.4            TCP            UP
5th            5.5.5.5            UDP            UP
6th            6.6.6.6            UDP            UP
```

The ATA always sends SIP messages to the available address with the top priority and with the status UP in the list. In the example, the ATA sends all the SIP messages to the address 1.1.1.1. If the address 1.1.1.1 in the list is marked with the status DOWN, then the ATA communicates with 2.2.2.2 instead. The ATA can restore the connection back to 1.1.1.1 when the specified failback conditions are met. For more details about failover and failback, see SIP Proxy Failover, on page 6 and SIP Proxy Fallback, on page 7.

SIP Proxy Failover

The ATA performs a failover in any of these cases:

- The ATA sends SIP messages and doesn't get responses from the server.
- The server responds with a code that matches the specified code in **Try Backup RSC**.
- The ATA gets a TCP disconnection request.

We strongly recommend that you set the **Auto Register When Failover** to **yes** when **SIP Transport** is set to **AUTO**.

ATA Failover Behavior

When the ATA fails to communicate with the currently connected server, it refreshes the server list status. The unavailable server is marked with the status DOWN in the server list. The ATA tries to connect to the top-priority server with the status UP in the list.

In the following example, the addresses 1.1.1.1 and 2.2.2.2 aren’t available. The ATA sends SIP messages to 3.3.3.3, which has the top priority among the servers with the status UP.

```
Priority       IP Address     SIP Protocol    Status
1st            1.1.1.1            TLS          DOWN
2nd            2.2.2.2            TLS          DOWN
3rd            3.3.3.3            TCP          UP
4th            4.4.4.4            TCP          UP
5th            5.5.5.5            UDP          UP
6th            6.6.6.6            UDP          UP
```

In the following example, there are two SRV records from the DNS NAPTR response. For each SRV record, there are three A records (IP addresses).

```
Priority       IP Address     SIP Protocol    Server            Status
1st            1.1.1.1            UDP         SRV1              DOWN
2nd            1.1.1.2            UDP         SRV1              UP
3rd            1.1.1.3            UDP         SRV1              UP
4th            2.2.2.1            TLS         SRV2              UP
5th            2.2.2.2            TLS         SRV2              UP
6th            2.2.2.3            TLS         SRV2              UP
```

Let's assume that the ATA failed to connect to 1.1.1.1 and then registered to 1.1.1.2. When 1.1.1.2 goes down, ATA behavior depends on the setting of **Proxy Fallback Intvl**.

- When **Proxy Fallback Intvl** is set to **0**, the ATA tries with the addresses in this order: 1.1.1.1, 1.1.1.3, 2.2.2.1, 2.2.2.2, 2.2.2.3.
- When **Proxy Fallback Intvl** is set to a value other than zero, the ATA tries with the addresses in this order: 1.1.1.3, 2.2.2.1, 2.2.2.2, 2.2.2.3.

SIP Proxy Fallback

The proxy fallback requires a value other than zero specified in the **Proxy Fallback Intvl** field under the **Proxy and Registration** section in the ATA administration web page. If you set this field to `0`, the SIP proxy failback feature is disabled.

The time when the ATA triggers a failback depends on the ATA configuration and the SIP transport protocols in use.

To enable the ATA to perform failback between different SIP transport protocols, set **SIP Transport** to **AUTO** under the **Proxy and Registration** section from **Voice** > **Line (n)** of the ATA administration web page.

Failback from a UDP Connection

The failback from a UDP connection is triggered by SIP messages. In the following example, the ATA first failed to register to 1.1.1.1 (TLS) at the time T1 since there's no response from the server. When SIP Timer F expires, the ATA registers to 2.2.2.2 (UDP) at the time T2 (T2=T1+SIP Timer F). The current connection is on 2.2.2.2 via UDP.

```
Priority       IP Address     SIP Protocol    Status
1st            1.1.1.1            TLS          DOWN        T1 (Down time)
2nd            2.2.2.2            UDP          UP
3rd            3.3.3.3            TCP          UP
```

The ATA has the following configuration:

```
<Proxy_Fallback_Intvl_n_ ua="na">60</Proxy_Fallback_Intvl_n_>
<Register_Expires_n_ ua="na">3600</Register_Expires_n_>
<SIP_Timer_F ua="na">16</SIP_Timer_F>
```

where *n* is the extension number.

The ATA refreshes the registration at time T2 (T2=(3600-16)*78%). The ATA checks the address list for the availability of the IP addresses and the down time. If T2-T1 > = 60, the failed server 1.1.1.1 resumes back to UP and the list is updated to the following. The ATA sends SIP messages to 1.1.1.1.

```
Priority       IP Address     SIP Protocol    Status
1st            1.1.1.1            TLS           UP
2nd            2.2.2.2            UDP           UP
3rd            3.3.3.3            TCP           UP
```

Failback from a TCP or TLS Connection

The failback from a TCP or TLS connection is triggered by the parameter **Proxy Fallback Intvl**. In the following example, the ATA failed to register to 1.1.1.1 (UDP) at the time T1 and thus registered to 2.2.2.2 (TCP). The current connection is on 2.2.2.2 via TCP.

```
Priority       IP Address     SIP Protocol    Status
1st            1.1.1.1            UDP          DOWN        T1 (Down time)
2nd            2.2.2.2            TCP          UP
3rd            3.3.3.3            TLS          UP
```

The ATA has the following configuration:

```
<Proxy_Fallback_Intvl_n_ ua="na">60</Proxy_Fallback_Intvl_n_>
<Register_Expires_n_ ua="na">3600</Register_Expires_n_>
<SIP_Timer_F ua="na">16</SIP_Timer_F>
```

where *n* is the extension number.

The proxy fallback interval (60 seconds) counts down from T1. The ATA triggers proxy failback at the time of T1+60. If you set the proxy fallback interval to 0 in this example, the ATA keeps the connection on 2.2.2.2.

Other ATA Voice Quality Features

Silence Suppression and Comfort Noise Generation

Voice Activity Detection (VAD) with Silence Suppression reduces the bandwidth needed for a single call, making it possible for your network to support more calls overall. VAD distinguishes between speech and non-speech signals, and Silence Suppression removes the natural silences that occur in a conversation. The IP bandwidth is used only to transmit speech.

Comfort Noise Generation provides white noise when nobody is talking so you know that your call is still connected.

Modem and Fax Pass-Through

The following applies to 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.
- A CED/CNG tone or an NSE event triggers FAX pass-through mode.
- Echo canceller is automatically disabled for Modem passthrough mode.
- Echo canceller is disabled for FAX pass-through if FAX Disable ECAN (Line 1 or 2 tab) is set to “Yes” for that line. In this 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. Out-of-band DTMF transmission is disabled during modem or fax passthrough.

Adaptive Jitter Buffer

The ATA can buffer incoming voice packets to minimize the impact of variable network delays. This process is known as jitter buffering. The size of the jitter buffer adjusts to changing network conditions. The ATA has a Network Jitter Level control setting for each line of service. The jitter level determines how aggressively the ATA tries to shrink the jitter buffer over time to achieve a lower overall delay. If the jitter level is higher, it shrinks more gradually. If jitter level is lower, it shrinks more quickly. You can use the default settings or configure this feature in the Network Settings section of the "Voice Settings Configuration" chapter.

Adjustable Audio Frames Per Packet

This feature allows you to set the number of audio frames contained in one RTP packet. Packets can be adjusted to contain from 1 to 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 action enhances 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 and alert you to a call's status. Parameters for each type of tone, such as a dial tone, 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 you to hear the call progress tones (such as ringing) that are generated from the far-end network.

Echo Cancellation

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

Hook Flash Events

The ATA signals 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 your service provider, you may need to disable Call Waiting Service, Three Way Conference Service, or Three Way Call Service. These three features could prevent the signaling of a hook flash event to the softswitch. You can configure these settings in the Supplementary Service Subscription section of the Line 1 and Line 2 Settings (PHONE 1 and PHONE 2) page.
- The Hook Flash setting determines the time period required for hook flash detection. It is in the Control Timer Values section of the SIP page.

Configurable Dial Plan with Interdigit Timers

The ATA has three configurable interdigit timers:

- The initial timeout—signals that a phone is taken off hook.
- A long timeout—signals the end of a dialed string.
- A short timeout—signals that more digits are expected.

Polarity Control

The ATA allows the polarity to be set when a call is connected and when a call is disconnected. This feature is required to support some pay phone system and answering machines. You can configure these settings in the FXS Port Polarity Configuration section of the Line 1 and Line 2 Settings (PHONE 1 and PHONE 2) page.

Calling Party Control

Calling Party Control (CPC) momentarily removes the voltage between the tip and the ring signals, signaling that the calling party has hung up. This feature is useful for auto-answer equipment. You can configure these settings in the Control Timer Values section of the Regional page.

Encryption of SIP Messages using SIP over TLS

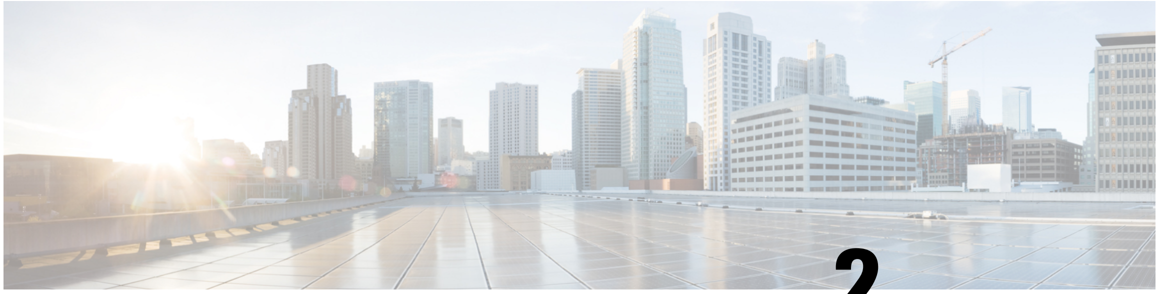
You can enable SIP over Transport Layer Security (TLS) to encrypt the SIP messages between the service provider and your business. SIP over TLS relies on the TLS protocol to encrypt the signaling messages. You can configure the SIP Transport parameter in the SIP Settings section of the Line 1 and Line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

Secure Calling Using SRTP

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

DNS NAPTR Support

The Line 1 and Line 2 (PHONE 1 and PHONE2) can select the SIP transport protocol (TPC, UDP, or TLS) automatically based on the Name Authority Pointer (NAPTR) records on the DNS server. Typically, the line uses the protocol with the highest priority in the records.



CHAPTER 2

New and Changed Information

New and Changed for Firmware Release 11.2(4)

Revision	New and Changed
Updated the `FAX Enable T38` parameter for the feature *Outbound Fax Refinement*	Audio Configuration, on page 87
Updated the topics for the feature *OPTIONS Support in NAT Keep Alive Messages*	NAT Settings, on page 74
Added the missing topic for the feature *Log Management for Users*	Crash Dump, on page 107
Updated the topics for the feature *Default Web Access Protocol Changes to HTTPS*	Access the Phone Web Interface, on page 19 Cisco ATA 192 Web Access Fields, on page 98 Cisco ATA 191 Web Access Fields, on page 97 Cisco ATA 192 Remote Access Fields, on page 98 Troubleshoot Your Fax, on page 128
Updated the topics for the feature *Increased Maximum Number of A Records for SRV Record*	SIP Proxy Redundancy, on page 6

New and Changed for Firmware Release 11.2(3)

Revision	New and Changed
Added the topic for the `HTTP Proxy Support` feature	HTTP Proxy (ATA 191 and 192), on page 34

Revision	New and Changed
Updated the topics to add the value range of the parameters `Call Back Delay`, `VMWI Refresh Intvl`, `DTMF Playback Length`, `DTMF Playback Level`, and `DTMF Twist`	Control Timer Values (Sec), on page 62 Miscellaneous, on page 71

New and Changed for Firmware Release 11.2(2)

Revision	New and Changed
Updated the parameter `SIP Transport`	SIP Settings, on page 76
Updated the topic to add the new feature `DNS NAPTR Support`	Other ATA Voice Quality Features, on page 9
Updated the topic to add more examples	SIP Proxy Redundancy, on page 6
Added the topic for the SIP proxy redundancy feature	SIP Proxy Failover, on page 6 SIP Proxy Fallback, on page 7
Updated the topic to add the parameter `Auto Register When Failover`	Proxy and Registration, on page 81
Updated the topic to mention the maximum number of NAPTR records and SRV records supported in a query	SIP Proxy Redundancy, on page 6
Added a new task topic on how to secure a line by automatic SRTP/RTP selection	Set up a Secure Line, on page 78
Updated the Call Feature Settings table with a new parameter Secure Call Option	Call Feature Settings, on page 79

New and Changed for Firmware Release 11.2(1)

Revision	New and Changed
Updated the navigation path	Debug Log Module, on page 103
Added the topics for the E911 feature	Emergency Calls, on page 135 Emergency Call Support Background, on page 135 Emergency Call Support Terminology, on page 135 Configure the ATA to Make Emergency Calls, on page 136 E911 Geolocation Configuration, on page 80

Revision	New and Changed
Updated RTP Parameters table to add the Call Statistics parameter	RTP Parameters, on page 45
Added a topic for the reporting End-of-Call Statistics feature	Call Statistics are Not Available in the Server, on page 120
Added a task for the remote PRT generation feature	Generate a Problem Report Remotely, on page 106
Updated the topic for the remote PRT generation feature	PRT Viewer, on page 105
Added a topic how to address the issue when ATA 11.1.0MSR3-9 and older doesn't upgrade	ATA with Firmware Release 11.1.0MSR3-9 and Older Doesn't Upgrade, on page 108

CHAPTER 3

Quick Setup for Voice over IP Service

Set Up Voice over IP

The Quick Setup page displays when you log in to the ATA web page for the first time. Use this page to connect your phone to your provider's Voice over IP network.

Note You need an Internet connection to link with your service provider's network. With the default network settings, your ATA has Internet connectivity, if the WAN port connects to a port on your router.

Procedure

Step 1 For Line 1 and Line 2, enter the settings for the phone services used by the phones or fax connected to the PHONE1 and PHONE2 ports.

- **Proxy:** Enter the IP address of the service provider's proxy server.
- **Display Name:** Enter the name or DN 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.

Step 2 Click **Submit** to save your settings. The voice service restarts.

Step 3 To verify your progress, perform the following tasks:

a) Check if phone LED is a steady green, indicating that the phone has registered.

If the line is not registered, 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.

b) Use an external phone to call the phone number assigned to you by your ISP. Verify that the phone rings and you have two-way audio on the call.

CHAPTER 4

Network Configuration

Web-Based Configuration Utility

Your phone system administrator can allow you to view the phone statistics and modify some or all the parameters. This section describes the features of the phone that you can modify with the phone web user interface.

Access the Phone Web Interface

If your service provider has disabled access to the configuration utility, contact the service provider before proceeding.

Procedure

Step 1 Ensure that the computer can communicate with the phone. No VPN in use.

Step 2 Start a web browser.

Step 3 Enter the IP address of the phone in your web browser address bar.

- User or Admin Access: **`https://<ip address>:<port>/`**, and then enter the username and password.

For example, `https://10.64.84.147/`

Allow Web Access to the ATA

To view the ATA parameters, enable the configuration profile. To make changes to any of the parameters, you must be able to change the configuration profile. Your system administrator might have disabled the option to make the ATA web user interface viewable or writable.

For more information, see the *Cisco ATA 191 and 192 Multiplatform Firmware Provisioning Guide*

Before you begin

Access the phone administration web page. See Access the Phone Web Interface, on page 19.

Procedure

Step 1 Click **System**.

Step 2 In the **System Configuration** section, set **Enable Web Server** to **Yes**.

Step 3 To update the configuration profile, click **Submit All Changes** after you modify the fields in the phone web user interface.

The phone reboots and the changes are applied.

Step 4 To clear all changes that you made during the current session (or after you last clicked **Submit All Changes**), click **Undo All Changes**. Values return to their previous settings.

Basic Setup

Use the **Network Setup** > **Basic Setup** pages to configure your Internet connection, local network settings (ATA 192 only), and your time settings.

Network Service (ATA 192 Only)

Use the **Network Setup** > **Basic Setup** > **Network Service** page to configure the operating mode of the ATA 192.

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) allows multiple devices on a private network to share a public, routable IP address. In order for theVoice 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 and to another network on the LAN port. 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 to its LAN port (with connected devices also in the 10.0.0x range).
- **Monitor Network Drop on Internet Port Only:** This parameter is used for how to report the link state when both Ethernet(LAN) and Network(WAN) ports are connected.
    - **Off**: If you unplug and plug the WAN cable, meanwhile the LAN port state is still UP, then the ATA doesn't perform any operation.
    - **On**: If you unplug and then plug the WAN cable, then the ATA triggers a warm reboot even though the LAN port state is still UP. In this case, the ATA will try to register again.

Note

- The parameter takes effect only when you select the **Bridge** mode.
- The parameter doesn't take effect when only the WAN port is connected.

Basic Settings

Use the **Network Setup** > **Basic Settings** page to set up your basic network settings.

Table 4: Basic Settings

Field	Description
Domain Name	The domain name, if specified by your ISP. Otherwise, leave the field blank.
Host Name	The name of the ATA. The default value is the model number. Your ISP may specify a host name to use.
Stack mode	Choose the stack mode for network; there are three modes can be set: IPv4 only, Pv6 Only, or Dual.
Signaling Preference	Choose the SIP packet preference, either IPv4 or IPv6.
Media Preference	Choose the RTP packet preference, either is IPv4 or IPv6.

IPv4 Settings

Use the **Network Setup** > **Basic Setup** > **IPv4 Settings** page to set up your IPv4 connection.

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.

Table 5: Internet Connection Type

Field	Description
Connection Type	Specify the Internet addressing method that your ISP requires. Default setting: Automatic Configuration - DHCP • **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 IP address. Complete the fields that appear. • **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.

Field	Description
Static IP Settings	• **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. If needed, you can adjust the MTU and Optional Settings.
PPPoE Settings	• **User Name and Password:** Enter the user name and password that you use to log in to 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 timed out, this feature also enables the ATA to re-establish your connection when 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. If needed, you can adjust the MTU and Optional Settings.
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. Usually, you keep the default setting to allow the ATA to choose the appropriate MTU. To specify the MTU, select Manual, and then enter the number of bytes.

Table 6: Optional Settings

Field	Description
DNS Server Order	Choose the preferred method for choosing a DNS server. • **DHCP-Manual—**The DNS server settings from the network server takes precedence, and your entries in the DNS fields are used only as a backup. • **Manual-DHCP—**Your entries in the DNS fields take precedence, and the DNS server settings from the network server are used as a backup. • **Manual—**Your entries in the DNS fields are used to choose a DNS server.
Primary DNS	Set the Primary DNS for IPv4.
Secondary DNS	Set the Secondary DNS for IPv4.

IPv6 Settings

Use the **Network Setup** > **Basic Setup** > **IPv6 Settings** page to set up your IPv6 connection.

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.

Table 7: IPv6 Settings

Field	Description
Internet Connection Type	Specify the Internet addressing method that your ISP requires. Default setting: Automatic Configuration - DHCP 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 IP address. Complete the following fields: • Internet IPv6 Address and Prefix Length—Enter the IPv6 address and prefix length that was assigned to your account by your service provider. The public sees this address. • Default Gateway—Enter the Gateway IPv6 Address that was provided by your ISP. 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 following fields: • User Name and Password—Enter the username and password that you use to log in to 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 timed out, this feature enables the ATA to automatically reconnect when you try 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.

Table 8: Optional Settings

Field	Description
DNS Server Order	Choose the preferred method for choosing a DNS server. • DHCP-Manual—The DNS server settings from the network server takes precedence, and your entries in the DNS fields are used only as a backup. • Manual-DHCP—Your entries in the DNS fields take precedence, and the DNS server settings from the network server are used as a backup. • Manual—Your entries in the DNS fields are used to choose a DNS server.
Allow Auto Configuration.	Enable if you want to allow Auto Configuration.
Primary DNS	Set the Primary DNS for IPv6.
Secondary DNS	Set the Secondary DNS for IPv6.

IPv4 LAN Settings (ATA 192 Only)

Use the **Network Setup** > **Basic Setup** > **IPv4 LAN Settings** page to set the IP address and subnet mask for your local network. Also configure the settings for the built-in DHCP server (ATA 192 only).

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: Enable

Field	Description
IP Reservation	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: • To reserve a static IP address for a current DHCP client: Check the box for the client in the **Select Clients from DHCP Tables** list. Click**Add Clients**. The selected clients are added to the *Clients Already Reserved* list. These clients have static IP addresses that do not change. • To add a client that isn't 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	Enter the IP address of the default gateway to be used by the DHCP clients. Default setting: 192.168.15.1 (the IP address of the ETHERNET (LAN) interface)
Starting IP Address	Enter the first address in the range of addresses assigned dynamically by the DHCP server. Default setting: 192.168.15.100
Maximum DHCP Users	Enter the maximum number of devices that can dynamically receive, or “lease,” DHCP addresses from the DHCP server. Default setting: 50 **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.
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

Field	Description
Option 66	Provides provisioning server address information to hosts that request this option. Server information can be defined in one of three ways: • **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. This option is used to provide either an IP address or a fully qualified hostname. But the ATA also accepts and offers a full URL including protocol, path, and filename to meet the 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. Default setting: blank
Option 67	Provides a configuration or bootstrap filename to hosts that request this option. This option is used with option 66 to allow a client to form an appropriate TFTP request for the file. Default setting: blank
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. It also 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 are offered DNS server information by using the Static DNS servers or by using the servers specified for the INTERNET (WAN) interface.

IPv6 LAN Setting (ATA 192 Only)

Use the **Network Setup** > **Basic Setup** > **IPv6 LAN Settings** page to set up your IPv6 LAN connection.

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.

Table 9: Internet Connection Type

Field	Description
DHCP Server	Click **Enabled** to enable the DHCP server, or click **Disabled** to disable this feature. Default setting: Enable
Address Assign Type	Choose the address assign type: SLAAC/DHCPv6.
DHCPv6 Delegation	Choose whether to support DHCPv6 delegation, if Yes, user can’t configure **IPv6 Address Prefix**.
IPv6 Address Prefix	Set the IPv6 address prefix for IPv6 LAN interface, the Prefix Length is fixed to 64.
IPv6 Address Length	Set the IPv6 address prefix length for IPv6 LAN interface. Range:1-112
IPv6 Static DNS	Set the IPv6 Static DNS.
LAN IPv6 Address	Display the LAN IPv6 address information.

Time Settings

Use the **Network Setup** > **Basic Setup** > **Time Settings** page to set the system time for the ATA. 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.

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

Note If the ATA doesn't receive the response from the NTP server, then the system time for the ATA uses the build date and time of the firmware.

User Manual

If you prefer to set the system manually, click **User Manual** and then enter the date and time.

Table 10: Time Settings

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.

Table 11: Time Zone Settings

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.
Time Server Address.	To use the ATA's default Network Time Protocol (NTP) server, select Auto from the drop-down list. 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
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 (ATA 192 Only)

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

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

Table 12: Port Settings

Field	Description
Flow Control	Flow control is a mechanism that temporarily stops the transmission of data on a port. For example, a device is transmitting data faster than some other part of the network can accept it. The overwhelmed network element halts the transmission of the sender for a specified time. Choose **Enabled** to enable this feature, or choose **Disabled** to disable this feature. Default setting: Enabled

Field	Description
Speed Duplex	Choose the duplex mode. You can select from Auto-negotiate, 10 Half, 10 Full, 100 Half and 100 Full. Cisco recommends that 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. Default setting: Auto-negotiate

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.

After making changes, click Submit to save your settings, or click Cancel to redisplay the page with the saved settings.

Table 13: MAC Address Clone 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 *MAC Address* field. Default setting: the current MAC address of your ATA

VPN Passthrough (ATA 192 Only)

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 require an independent IPsec tunnel. 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.

After making changes, click Submit to save your settings, or click Cancel to redisplay the page with the saved settings.

Table 14: VPN Passthrough 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
L2TP Passthrough	Layer 2 Tunneling Protocol is the method used to enable Point-to-Point sessions using 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.

After making changes, click Submit to save your settings, or click Cancel to redisplay the page with the saved settings.

Table 15: VLAN 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 to 4094. When VLAN is enabled, the default setting is 1.

CDP and 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 enabled, the ATA sends messages to a multicast address and listens to the messages sent by other devices using the protocol.

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

Application

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

Quality of Service (QoS) (ATA 192 Only)

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.

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

Table 16: QoS Settings

Field	Description
QoS Policy	Click **Always On** to enable QoS settings always, 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: 100000 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 (ATA 192 Only)

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

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.

Table 17: Port Forwarding Settings

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.

Field	Description
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**.

Table 18: Port Settings

Field	Description
External Port	The port that external clients 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 accessed by this rule.

Manually Add Port Forwarding (ATA 192 Only)

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

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

Table 19: Port Forwarding Settings

Field	Description
Port Forwarding Type	Choose the type of port forwarding: • **Single Port Forwarding**: Forwards traffic for a specified port to the same or an alternate 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 Application Name list for Single Port Forwarding, enter a name to identify the application.

Field	Description
External Port, Internal Port	For Single Port Forwarding, specify the ports to use. For simplicity, the internal and external port numbers are often the same. Different external port numbers could be used to differentiate traffic of the same application type intended for different servers, or for privacy by using non-standard ports. • **External port**: For single port forwarding, enter the port number that external clients 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. The correct entries appear automatically if you choose a standard application from the Application Name list for Single Port Forwarding.
Start - End Port	For Port Range Forwarding, specify the range of ports to use. Valid values are from 1 to 65535.
Protocol	Select the protocols that can be forwarded: TCP, UDP, or TCP and UDP.
IP Address	Enter the IP address of the local server that receives forwarded traffic. For 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.
Enabled	Check the box to enable this port forwarding rule, or uncheck the box to disable it. Default setting: Disabled

DMZ (ATA 192 Only)

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

The specified network device must have its DHCP client function disabled. It must also have a reserved IP address to ensure that it is reachable at the specified IP address.

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.

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

Table 20: DMZ Settings

Field	Description
Staus.	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.

HTTP Proxy (ATA 191 and 192)

Use the **Network Setup** > **Application** > **HTTP Proxy** page to set up a proxy server for the ATA to enhance security. A proxy server acts as a firewall between the ATA and Internet. After successful configuration, the ATA connects to Internet through the proxy server which protects the ATA from cyber attack.

You can set up a proxy server by either using an automatic configuration script or by manually configuring the host server (hostname or IP address) and port of the proxy server.

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

When the ATA is configured with the HTTP proxy feature, the feature applies to all the applications that use the HTTP protocol. The applications include the following:

- Profile Rule B, C, and D
- Log Resync Request, Success, and Failure Msg
- Report Rule
- Upgrade Rule
- Custom CA URL
- E911 Request URL
- EDOS RC Server
- TR69 (ACS URL)
- PRT Upload URL

Table 21: HTTP Proxy Settings

Field	Description
Proxy Mode	Choose the HTTP proxy mode: • **Auto**: The ATA automatically retrieves a Proxy Auto-Configuration (PAC) file to select a proxy server. In this mode, you can determine whether to use Web Proxy Auto-Discovery (WPAD) protocol to retrieve a PAC file or manually enter a valid URL of the PAC file. For details about the fields, see Use Auto Discovery (WPAD) and PAC URL. • **Manual**: You must manually specify a server (hostname or IP address) and a port of a proxy server. For details about the fields, see Proxy Host and Proxy Port. • **Off**: You disable the HTTP proxy feature on the ATA. Default setting: Off
Use Auto Discovery	Click **Yes** to use the Web Proxy Auto-Discovery (WPAD) protocol to retrieve a PAC file automatically. Click **No** to specify the URL of a PAC file manually. For details about the field, see PAC URL. Default setting: Yes
PAC URL	Specify the URL of a PAC file. For example, `http://proxy.department.branch.example.com` TFTP, HTTP, and HTTPS are supported. The field must be configured when the **Proxy Mode** is **Auto** and **Use Auto Discovery** is **No**. Default setting: Blank
Proxy Server Requires Authentication	Enter the authentication credentials (username and password) if the proxy server requires. This field is configured according to the actual behaviour of the proxy server. For details about the fields, see Username and Password. Default setting: No
Proxy Host	IP address or hostname of the proxy host server for the ATA to access. For example: `proxy.example.com` The scheme (http:// or https://) is not required. Default setting: Blank
Proxy Port	Port number of the proxy host server. The value range is from 2 to 65535. Default setting: 3128

Field	Description
Username	Username for a credential user on the proxy server. If **Proxy Mode** is set to **Manual** and **Proxy Server Requires Authentication** is set to **Yes**, you must configure the field. Default setting: Blank
Password	Password of the specified username for the proxy authentication purpose. If **Proxy Mode** is set to **Manual** and **Proxy Server Requires Authentication** is set to **Yes**, you must configure the field. Default setting: Blank

CHAPTER 5

Voice Settings Configuration

Information

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

Product Information

Table 22: Product Information

Field	Description
Product Name	The product name of ATA.
Serial Number	The serial number of ATA.
Software Version	The software version of ATA.
Hardware Version	The hardware version of ATA.
MAC Address	The mac address of ATA.
Client Certificate	The client certificate of ATA.
Customization	The customization of ATA.

System Status

Table 23: System Status Settings

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 Network Setup > Time Settings page.
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 and Line 2 Settings (PHONE 1 and PHONE 2)

Use the **Voice** > **Line 1** and **Voice** > **Line 2** pages to configure the settings for calls through the PHONE 1 and PHONE 2 ports.

Enter the settings as described. 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 include an appropriate numeral for identifying the port receiving the setting.

Custom CA Status

Table 24: CA Status Settings

Field	Description
Custom CA Provisioning Status	The status of the latest custom CA (Certificate Authority) certificate download.

Field	Description
Custom CA Info	The successfully downloaded CA information, or “Not Installed” if no custom CA certificate was installed. Default setting: Not Installed

Provision Status

Table 25: Provision Status Settings

Field	Description
Provisioning Profile	Profile rule setting Default setting: Empty
Provision Status	Indicate the status of last provisioning Default setting: Empty
Provisioning Failure Reason	Reason for failure Default setting: Empty

System

Use the **Voice** > **System** page to configure general voice system settings and to enable logging by using a syslog server. Logging can also be configured in the **Administration** > **Logging** pages.

System Configuration

Table 26: System Settings

Field	Description
Restricted Access Domains	Domain that Cisco IP phones responds to SIP messages only from the identified servers. Applicable to Line 1.
IVR Admin Passwd	Password for the administrator to manage the ATA by using the built-in IVR through a connected phone.
Network Startup Delay	The number of seconds of delay between restarting the voice module and initializing network interface. Default setting: 3

Miscellaneous Settings

Table 27: Miscellaneous Settings

Field	Description
DNS Query TTL Ignore	In DNS packages, the server suggests a TTL value to the client. If this parameter is set to Yes, the value from the server is ignored. Default setting: No

SIP

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

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

Table 28: SIP Parameters Settings

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

Field	Description
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
SIP Reg Starting Sequence Number:	Defines the SIP Reg message Sequence Number. Default setting: Blank
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 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 if the ATA uses compact SIP headers in outbound SIP messages. 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 headers when generating the response regardless of the Use Compact Header parameter. If inbound SIP requests contain normal headers, the ATA substitutes those headers with compact headers as defined by RFC 261 when Use Compact Header is set to Yes. Default setting: No
Escape Display Name:	Determines if 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 out bound SIP messages. If the display name includes " or \, these will be escaped to \" and \\ within the double quotes. Otherwise, select **No**. Default setting: No
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: Yes

Field	Description
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
AVT Packet Size:	Indicates the AVT Packet size according to value set in ptime or fixed 10ms. Default setting: ptime
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

Keep Referee When REFER Failed:	Set this parameter to **Yes** to configure the phone to handle NOTIFY sipfrag messages.
	You can also configure this parameter in the configuration file:
	`<Keep_Referee_When_REFER_Failed ua="na">Yes`
	`</Keep_Referee_When_REFER_Failed>`

Caller ID Header:	Provides the option to take the caller ID from PAID-RPID-FROM,P-ASSERTEDIDENTITY, REMOTE-PARTY-ID, or FROM header. Default setting: PAID-RPID-FROM

SIP Timer Values

Table 29: SIP Timer Values Settings

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

Field	Description
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
SIP Timer F	Non-INVITE time-out value, which can range from 0 to 64 seconds. Default setting: 16
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
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 must be larger than the Reg Retry Intvl value, which cannot be 0. Default setting: 1200

Field	Description
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). 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

Table 30: Response Status Code Settings

Field	Description
SIT1 RSC	SIP response status code for the appropriate Special Information Tone (SIT). 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
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

Table 31: 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 to 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
RTP Tx Packet Size Follows Remote SDP	Enable the Remote pair RTP Packet Size. Default setting: Yes
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
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 extra 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 *Information* 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

Field	Description
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>,JI=<jitter in ms>,LA=<delay in ms>,DU=<call duration ins>,EN=<encoder>,DE=<decoder>.

Field	Description
Call Statistics	Specifies whether the phone sends end-of-call statistics within SIP messages when a call terminates or is put on hold. You can select **Yes** to enable the phone to send end-of-call statistics in Session Initiation Protocol (SIP) messages (BYE and re-INVITE messages). The phone sends call statistics to the other party of the call when the call terminates or when the call is on hold. The statistics include: • Real-time Transport Protocol (RTP) packets sent or received • Total bytes sent or received • Total number of lost packets • Delay jitter • Round-trip delay • Call duration The call statistics are sent as headers in SIP BYE messages and SIP BYE response messages (200 OK and re-INVITE during hold). For audio sessions, the headers are RTP-RxStat and RTP-TxStat. • Rtp-Rxstat -> Data received • Rtp-Txstat -> Data transmitted Example of call statistics in a SIP BYE message: From: xxxx User-Agent: xxxx Call-ID: xxxx **Rtp-Rxstat**: Dur=13,Pkt=408,Oct=97680,LatePkt=8,LostPkt=0,AvgJt=0,VQMetrics="CCR=0.0017;ICR=0.0000;ICRmx=0.0077;CS=2; SCS=0;VoRxCodec=PCMU;CID=4;VoPktSizeMs=30;VoPktLost=0;VoPktDis=0;VoOneWayDelayMs=28;maxJitter=1;MOScq=4.2;MOSlq=3.2;networkType=CPU;rtpbitrate=61587;rtcpbitrate=0" **Rtp-Txstat**: Dur=13,Pkt=417,Oct=100080,tvqMetrics="TxCodec=PCMU;rtpbitrate=61587;rtcpbitrate=0" You can also use the Call_Statistics parameter in the phone configuration file to enable this feature. `<Call_Statistics ua="na">Yes</Call_Statistics>` Default: Yes Options: Yes and No

SDP Payload Types

Table 32: SDP Payloads

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

Field	Description
G729a Codec Name	G.729a codec name used in SDP. Default setting: G729a
G729b Codec Name	G.729b codec name used in SDP. Default setting: G729ab
EncapRTP Codec Name	EncapRTP codec name used in SDP. Default setting: encaprtp

NAT Support Parameters

Table 33: NAT Support Parameters

Field	Description
Handle VIA received.	If you select **Yes**, the ATA processes the received parameter in the VIA header. The server inserts this value in a response to any one of its requests. If you select **No**, the parameter is ignored. 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. 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	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. If this parameter is specified, the ATA assumes this IP address when generating SIP messages and SDP. However, the results of STUN and VIA received parameter processing supersede this statically configured value. 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
EXT RTP Port Min	External port mapping number of the RTP Port Min. number. If this number 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	Enables or disables NAT Redirect keep alive messages. Default setting: No

Provisioning

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

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

Configuration Profile

Table 34: Configuration Profile Settings

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:	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. 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. This feature can be used to prevent an overload of the provisioning server when many devices power-on simultaneously. Default setting: 2 (40 seconds)
Resync At (HHmm):	The time of day when the device tries to resync. The resync is performed each day. Used with the Resync At Random Delay. Default setting: blank
Resync At Random Delay:	Used 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
Resync Periodic:	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. Default setting: 3600

Field	Description
Resync Error Retry Delay:	Resync retry interval (in seconds) applied if there is a resync failure. 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. 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. Default setting: 3600
Forced Resync Delay:	Maximum delay (in seconds) that the ATA waits before performing a resync. 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. It allows you to make calls in succession without interruption. The ATA has a timer that begins counting down when all lines become idle. This parameter is the initial value of the counter. Resync events are delayed until this counter decrements to zero. Default setting: 14400
Resync From SIP:	Enables a resync to be triggered with a SIP NOTIFY message. Default setting: yes
Resync After Upgrade Attempt:	Triggers a resync after every firmware upgrade attempt. Default setting: yes
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 filename can have macros, such as $MA, which expands to the ATA MAC address. Default setting: /spa$PSN.cfg

Field	Description
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 Rules B, C, and D are still evaluated and executed. Default setting: blank
DHCP Option To Use:	DHCP Options, delimited by commas, retrieves firmware and profiles. Default setting: 66.160.159.150
Transport Protocol:	Transport Protocol retrieves firmware and profiles. If none is selected, TFTP is assumed and the IP address of the TFTP server is obtained from the DHCP server. Default setting: https
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
Report Rule:	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. 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. This parameter may optionally contain an encryption key. For example: [--key $K] tftp://ps.callhome.net/$MA/rep.xml.enc Default setting: blank

Firmware Upgrade

Table 35: Firmware Upgrade Settings

Field	Description
Upgrade Enable.	Determines if the firmware upgrade operations occur independently of resync actions. Default setting: yes

Field	Description
Upgrade Error Retry Delay.	The upgrade retry interval (in seconds) applied if there is an 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. Default setting: 3600
Downgrade Rev Limit.	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. Default setting: blank
Upgrade Rule.	This parameter is a firmware upgrade script with the same syntax as Profile_Rule. Defines upgrade conditions and associated firmware URLs. Default setting: blank
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

CA Settings

Table 36: 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: blank

General Purpose Parameters

Table 37: General Purpose Settings

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.

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

Ring, Cadence, and Tone Scripts

To define ring and tone patterns, the ATA uses the concept of scripts. In the next sections, you will find information about creating Cadence Scripts (CadScripts), Frequency Scripts (FreqScripts), and Tone Scripts (ToneScripts).

CadScript

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

Syntax: S1[;S2], where:

Si=Di(oni,1/offi,1[,oni,2/offi,2[,oni,3/offi,3[,oni,4/offi,4[,oni,5/offi,5,oni,6/offi,6]]]]]) and is known as a section, oni,j and offi,j are the on/off duration in seconds of a segment and i = 1 or 2, and j = 1 to 6. Di 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 specifics the frequency and level parameters of a tone.

Syntax: F1@L1[,F2@L2[,F3@L3[,F4@L4[,F5@L5[,F6@L6]]]]]

Where F1–F6 are frequency in Hz (unsigned integers only) and L1–L6 are corresponding levels in dBm (with up to 1 decimal place). White spaces are allowed before and after the comma, but they are 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: ToneScript;Z1[;Z2].

The section Z1 is similar to the S1 section in a CadScript except that each on/off segment is followed by a frequency components parameter: Z1 = D1(oni,1/offi,1/fi,1[,oni,2/offi,2/fi,2[,oni,3/offi,3/fi,3[,oni,4/offi,4/fi,4[,oni,5/offi,5/fi,5[,oni,6/offi,6/fi,6]]]]]), where fi,j = n1[+n2]+n3[+n4[+n5[+n6]]]]] and 1 < nk < 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/.1/1+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

Call Progress Tones

Table 38: Call Progress Settings

Field	Description
Dial Tone	Prompts you to enter a phone number. Reorder Tone is played automatically when Dial Tone or any of its alternatives times out. Default setting: 350@-19,440@-19;10(*/0/1+2)
Second Dial Tone	Alternative to the Dial Tone when you dial a three-way call. Default setting: 420@-19,520@-19;10(*/0/1+2)
Outside Dial Tone	Alternative to the Dial Tone. It prompts you to enter an external phone number, as opposed to an internal extension. A comma character in the dial plan triggers it. Default setting: 420@-16;10(*/0/1)
Prompt Tone	Prompts you to enter a call forwarding phone number. Default setting: 520@-19,620@-19;10(*/0/1+2)
Busy Tone	Played when a 486 RSC is received for an outbound call. Default setting: 480@-19,620@-19;10(.5/.5/1+2)

Field	Description
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@-19,620@-19;10(.25/.25/1+2)
Off Hook Warning Tone	Played when the caller has not properly placed the handset on the cradle. Off Hook Warning Tone is played when the Reorder Tone times out. Default setting: 480@-10,620@0;10(.125/.125/1+2)
Ring Back Tone	Played during an outbound call when the far end is ringing. Default setting: 440@-19,480@-19;*(2/4/1+2)
Ring Back 2 Tone	Your ATA plays this 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@-19,480@-19;*(1/1/1+2)
Confirm Tone	Brief tone to notify you that the last input value has been accepted. Default setting: 600@-16;1(.25/.25/1)
SIT1 Tone	Alternative to the Reorder Tone played when an error occurs as a caller makes an outbound call. The RSC to trigger this tone is configurable on the SIP screen. Default setting: 985@-16,1428@-16,1777@-16;20(.380/0/1,.380/0/2,.380/0/3,0/4/0)
SIT2 Tone	Alternative to the Reorder Tone played when an error occurs as a caller makes an outbound call. The RSC to trigger this tone is configurable on the SIP screen. Default setting: 914@-16,1371@-16,1777@-16;20(.274/0/1,.274/0/2,.380/0/3,0/4/0)
SIT3 Tone	Alternative to the Reorder Tone played when an error occurs as a caller makes an outbound call. The RSC to trigger this tone is configurable on the SIP screen. Default setting: 914@-16,1371@-16,1777@-16;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@-16,1371@-16,1777@-16;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@-19,440@-19;2(.1/.1/1+2);10(*/0/1+2)
Cfwd Dial Tone	Played when all calls are forwarded. Default setting: 350@-19,440@-19;2(.2/.2/1+2);10(*/0/1+2)
Holding Tone	Informs the local caller that the far end has placed the call on hold. Default setting: 600@-19;*(.1/.1/1,.1/.1/1,.1/9.5/1)
Conference Tone	Played to all parties when a three-way conference call is in progress. Default setting: 350@-19;20(.1/.1/1,.1/9.7/1)
Secure Call Indication Tone	Played when a call has been successfully switched to secure mode. Play it for a short period - less than 30 seconds - and at a reduced level - less than 19 dBm - so it doesn't interfere with the call. Default setting: 397@-19,507@-19;15(0/2/0,.2/.1/1,.1/2.1/2)
Feature Invocation Tone	Played when a feature is implemented. Default setting: 350@-16;*(.1/.1/1)
Call Remind Tone	The holding tone is played on the Phone ports during the active call to remind you of the held call. Default setting: blank

Distinctive Ring Patterns

Table 39: Distinctive Ring Settings

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)

Field	Description
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

Table 40: Distinctive Call Waiting Tones

Field	Description
CWT1 Cadence	Cadence script for distinctive CWT 1. Default setting: *(.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)
CWT6 Cadence	Cadence script for distinctive CWT 6. Default setting: 30(.1/.1,.3/.2,.3/9.1)
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

Table 41: Distinctive Ring/CWT Patterns

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: 80

Table 42: Ring and Call Waiting Tones

Field	Description
Ring Waveform	Waveform for the ringing signal. Choices are Sinusoid or Trapezoid. Default setting: Trapezoid
Ring Frequency	Frequency of the ringing signal. Valid values are 15–50 (Hz) Default setting: 20
Ring Voltage	Ringing voltage. Choices are 30–90 (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)

Table 43: Control Timer Values

Field	Description
Hook Flash Timer Min.	Minimum on-hook time before off-hook qualifies as hook flash. Less than this value, and the on-hook event is ignored. Range: 0.1–0.4 seconds. Default setting: 0.1
Hook Flash Timer.	Max Maximum on-hook time before off-hook qualifies as hook flash. More than this value, and the on-hook event is treated as on hook (no hook-flash event). Range: 0.4–1.6 seconds. Default setting: 0.9

Field	Description
Callee On Hook Delay.	Phone must be on-hook for time before the ATA tears 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. Range: 0–65 seconds. Default setting: 0.5
VMWI Refresh Intvl.	Interval between VMWI refresh to the device. Range: 0–65535 seconds. 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
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

Field	Description
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 is played after a configurable delay. If CPC is enabled, dial tone will be played when tip-to-ring voltage is restored. Resolution is 1 second. 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: 0.5

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.

Table 44: Vertical Service Activation Codes

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 active 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

Field	Description
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
Cfwd Last Act Code.	Forwards the last inbound or outbound call to the number that you specify 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

Field	Description
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
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

Field	Description
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
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, you 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, you 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.	These codes tell the ATA what to do when you place the active call on hold and is listening to the second dial tone. One or more *codes can be configured into this parameter, such as *98, or *97|*98|*123, and so on. The maximum length is 79 characters. This parameter applies when you place the active 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. For example, after you dial *98, the ATA plays the Prompt Tone while waiting for you 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. 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. Default setting: blank

Field	Description
Feature Dial Services Codes.	These codes let the ATA know what to do when you are listening to the first or second dial tone. One or more *codes can be configured into this parameter, such as *72, or *72|*74|*67|*82, and so on. The maximum length is 79 characters. This parameter applies when you have a dial tone (first or second dial tone). After receiving dial tone, you enters the *code and the target number according to current dial plan. For example, after you dial *72, the ATA plays a special tone called a Prompt tone while awaiting you 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). 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. 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). '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 placed, x is any digit not used above. 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 you dial *73. Default setting: blank

Vertical Service Announcement Codes

Table 45: Vertical Service Announcement Codes

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

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

Outbound Call Codec Selection Codes

Table 46: 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

Miscellaneous

Table 47: Miscellaneous Settings

Field	Description
FXS Port Impedance:	Sets the electrical impedance of the PHONE port. Choices are: • 600 • 900 • 600+2.16uF • 900+2.16uF • 220+850||120nF • 220+820||115nF • 200+600||100nF Default setting: 600
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. Range: -30–0. Default setting: -16.0
DTMF Twist:	To gain difference between the two tone frequency. Range: 0–5. Default setting: 2
DTMF Playback Length:	Local DTMF playback duration in milliseconds. Range: 0–65 seconds. Default setting: 0.1
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.
Playback ABCD:	To enable local playback of OOB DTMF ABCD, select **Yes**. Otherwise, select **No**. Default setting: Yes

Field	Description
Caller ID Method:	Your choices are: • 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
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 and Line 2 Settings (PHONE 1 and PHONE 2)

Use the **Voice** > **Line 1** and **Voice** > **Line 2** pages to configure the settings for calls through the PHONE 1 and PHONE 2 ports.

Enter the settings as described. 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 include an appropriate numeral for identifying the port receiving the setting.

General

Table 48: General Settings

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

Streaming Audio Server (SAS)

Table 49: Streaming Audio Server Settings

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	A non-zero value is the interval in which the streaming audio server sends out session refresh (SIP re-INVITE) messages to determine if the connection is 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

Field	Description
SAS Inbound RTP Sink	This parameter works 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 appears in the c = line and the port number appears 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 are used in the SDP to tell the SAS client not to send any RTP to this SAS line. If a non-zero value is specified, then a=sendrecv and the SAS client streams 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 streams RTP packets to the SAS line. Default setting: blank

NAT Settings

Table 50: 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 sent periodically to maintain the current NAT mapping. Valid values are: **$NOTIFY**, **$REGISTER**, and **$OPTIONS**. • $NOTIFY: A NOTIFY message is sent to keep NAT alive. • $REGISTER: A REGISTER message without contact is sent. • $OPTIONS: An OPTIONS message is sent. Default setting: $NOTIFY
NAT Keep Alive Dest	Destination receiving the 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

Table 51: 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 [0-7]	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
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

Table 52: SIP Settings

Field	Description
SIP Transport	Select the protocol for SIP messages: • UDP • TCP • TLS • AUTO 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 don't need to be opened or packets dropped for activities such as Internet browsing or e-commerce. **AUTO** allows the ATA to select the appropriate protocol automatically, based on the NAPTR records on the DNS server.
SIP Port	Port number of the SIP message listening and transmission port. Default setting: 5060 for PHONE1 and 5061 for PHONE2
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 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

Field	Description
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
RTP Log Intvl	The interval for the RTP log. Default setting: 0
Restrict Source IP.	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 doesn't match the IP addresses resolved from the configured Proxy (or Outbound Proxy if Use Outbound Proxy is Yes). Default setting: No
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). Default setting: 4
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. Default setting: 0
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. 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
Auth INVITE.	When enabled, authorization is required for initial incoming INVITE requests from the SIP proxy. Default setting: No

Field	Description
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 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
Broadsoft ALTC.	Set whether the SIP is the Broadsoft ALTC. Options are: Yes or No. Default setting: No

Set up a Secure Line

You can configure a line to only accept secure calls. If the line is configured to only accept secure calls, then any calls the line makes will be secure.

Before you begin

- Access the phone adapter administration web page. See Access the Phone Web Interface.
- Enable **Secure Call Serv** under **Supplementary Service Subscription** section from **Voice** > **Line(n)**.
- SIP transport with TLS can be set statically on the phone adapter administration web page or automatically with information in the DNS NAPTR records. If the SIP transport parameter is set for the ATA line as TLS, it only allows SRTP. If the SIP transport parameter is set to AUTO, the phone adapter performs a DNS query to get the transport method.

Procedure

Step 1 Select **Voice** > **Line(n)** , where n is the line number that represents PHONE 1 or PHONE 2.

Step 2 In the section **Call Feature Settings** , set the paramter **Secure Call Option** as described in Call Feature Settings, on page 79.

Step 3 Click **Submit**.

Call Feature Settings

Table 53: Call Feature Parameters

Field	Description
Blind Attn-Xfer Enable	Enables the ATA to perform an attended transfer operation by ending the active 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 active 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
Enable IP Dialing.	Enable or disable IP dialing. If IP dialing is enabled, you can dial [userid@] a.b.c.d[:port], where '@', '.', and ':' are dialed by entering *, user-id must be numeric and a, b, c, d must be between 0 and 255; the 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

Field	Description
Emergency Number	Comma separated list of emergency number patterns. If outbound call matches one of the patterns, the ATA disables 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
Feature Key Sync	Allows the phone to synchronize with the call server. If Do Not Disturb or Call Forwarding settings are changed on the phone, the changes are also made on the server. If changes are made on the server, they are propagated to the phone. Default setting: no
Secure Call Option	Configures a line to only accept secure calls. Select any of the options: Optional: Retains the current secure call option for the phone adapter. Strict: Allows SRTP only when SIP transport is set to TLS and if the ATA receives an unsecure call, the call fails. Allows RTP only when SIP transport is UDP/TCP and if the ATA receives an unsecure call, the call fails. Default setting: Optional

E911 Geolocation Configuration

Table 54: E911 Geolocation Configuration

Field	Description
Company UUID	The Universally Unique Identifier (UUID) assigned to the customer by the emergency call services provider. For example: `19c8168c-a366-44b5-853c-960fcaa19592` Allowed values: Maximum identifier length is 128 characters. Default setting: Blank

Field	Description
Primary Request URL	URL of the primary location server that provides the emergency call services. The location server returns an HELD response to the phone with the requested location URI that is tied to the user phone IP address. This parameter must be in the form of a valid HTTP or HTTPS URL. Allowed values: A valid URL not exceeding 255 characters. Default setting: Blank
Secondary Request URL	URL of the backup server to obtain the user's phone location. If the primary request URL fails, ATA tries to send the secondary request URL to the emergency call services provider. This parameter must be in the form of a valid HTTP or HTTPS URL. Allowed values: A valid URL not exceeding 255 characters. Default setting: Blank

Proxy and Registration

Table 55: Proxy and Registration Parameters

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
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

Field	Description
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 – (231 – 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
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. If the value is 0, the SIP proxy fallback feature is disabled. Range: 0 –65535 sec Default setting: 3600
Proxy Redundancy 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 proxiesranked 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

Field	Description
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 setting: 2147483647
Auto Register When Failover	Controls the fallback duration. • **no**: the fallback happens immediately and automatically. If the Proxy Fallback Intvl is exceeded, all the new SIP messages go to the primary proxy. • **yes**: the fallback happens only when current registration expires, which means only a REGISTER message can trigger fallback. For example, when the value for Register Expires is 3600 seconds and Proxy Fallback Intvl is 600 seconds, the fallback is triggered 3600 seconds later and not 600 seconds later. When the value for Register Expires is 600 seconds and Proxy Fallback Intvl is 1000 seconds, the fallback is triggered at 1200 seconds. After successfully registering back to primary server, all the SIP messages go to primary server. Default setting: yes

Subscriber Information

Table 56: Subscriber Information Parameters

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
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

Field	Description
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
SIP URI	The parameter by which the user agent will identify itself for this line. If this field is blank, the actual URI used in the SIP signaling should be automatically formed as: sip:UserName@Domain Where UserName is the username given for this line in the User ID, and Domain is the domain given for this profile in the User Agent Domain. If the User Agent Domain is an empty string, then the IP address of the phone should be used for the domain. If the URI field is not empty, but if a SIP or SIPS URI that contains no @ character, then the actual URI used in the SIP signaling should be automatically formed by appending this parameter with an @ character followed by the IP address of the device.

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.

Table 57: Supplementary Service Subscription Settings

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

Field	Description
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. 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. Default setting: yes
Call Back Serv	Enable Call Back Service. Default setting: yes
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

Field	Description
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
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 and User 2, on page 91.
Referral Serv	Enable Referral Service. See the Referral Services Codes parameter in Vertical Service Activation Codes, on page 64 for more information. Default setting: yes
Feature Dial Serv	Enable Feature Dial Service. See the Feature Dial Services Codes parameter in Vertical Service Activation Codes, on page 64 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
CONFCID Serv	Enable Caller ID during conference call. Default settings: yes

Audio Configuration

Table 58: Audio Configuration Settings

Field	Description
Preferred Codec	Preferred codec for all calls. (The actual codec used in a call still depends on the outcome of the codec negotiation protocol.) Select one of the following: • **G711u** • **G711a** • **G726-32** • **G729a** Default setting: G711u.
Second Preferred Codec	If the first codec fails, then second preferred codec is tried. Default setting: blank
Third Preferred Codec	If the second codec fails, then third preferred codec is tried. Default setting: blank
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: no
Codec Negotiation	When set to **Default**, the Cisco IP phone responds to an Invite with a 200 OK response advertising the preferred codec only. When set to **List All**, the Cisco IP phone responds listing all the codecs that the phone supports. 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

Field	Description
FAX V21 Detect Enable	To enable detection of V21 fax tones, select **yes**. Otherwise, select **no**. Default setting: yes
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
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
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

Field	Description
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: • • 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: • strict mode for AVT and SIP: the value set in DTMF Tx Strict Hold Off Time • normal mode for AVT: 40 ms • normal mode for SIP: 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 not set to inband; that is, either AVT or 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 setting: 70 ms
FAX Enable T38	To enable the use of ITU-T T.38 standard for FAX Relay, select **yes**. Otherwise select **no**. • **no**: The ATA can parse only one "m=" line of the SDP packet. If the ATA receives multiple "m=" lines contained in the SDP packet from a provider, an outbound FAX failure might occur. This issue typically occurs when the first "m=" line specifies an invalid port number "0" while the second "m=" line specifies a valid port. To avoid this issue, set **FAX Enable T38** to **yes** and **FAX Passthru Method** to **ReINVITE**. In the aforementioned situation, the ATA can parse the second "m=" line successfully. • **yes**: The ATA can parse the first two "m=" lines of the SDP packet. It ignores the other "m=" lines. Default setting: no
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: • **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
Modem Line	Enable an alternate method to make the modem call without Modem Line Toggle Code pre-dialing. Default setting: no
RTP to Proxy in Remote Hold	Enable to send RTP to proxy when line is held by remote side. Default setting: no

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|xxxxxxxxxxxx.)
```

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.

Table 59: Dial Plan Settings

Dial Plan Entry	Functionality
*xx	Allow arbitrary 2 digit star code
[3469]11	Allow x11 sequences
0	Operator
00	Int’l Operator
[2-9]xxxxxx	US local number
1xxx[2-9]xxxxxx	US 1 + 10-digit long distance number
xxxxxxxxxxxx.	Everything else

FXS Port Polarity Configuration

Table 60: FXS Port Polarity Settings

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

User 1 and User 2

Use the **Voice** > **User 1** and **Voice** > **User2** pages to set the user preferences for the calls through the PHONE 1 and PHONE 2 ports.

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

Table 61: 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
Cfwd No Ans Delay	Delay in sec before Call Forward No Answer triggers. Default setting: 20

Selective Call Forward Settings

Table 62: Selective Call Forward Settings

Field	Description
Cfwd Sel1-8 Caller	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). • • Use ? to match any single digit. • • Use * to match any number of digits. **Example:** 1408*, 1512???1234 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. Default setting: blank
Cfwd Sel1-8 Dest	The destination for the corresponding Call Forward Selective caller pattern (Cfwd Sel1-8 Caller). Default setting: blank
Cfwd Last Caller	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, on page 64. Default setting: blank

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, on page 64. 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, on page 64. Default setting: blank

Speed Dial Settings

Table 63: 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

Table 64: Supplementary Service Settings

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

Field	Description
CWCID Setting	Call Waiting Caller ID Generation on or off. Default setting: yes
Dist Ring	Setting Distinctive Ring on or off. Default setting: yes
Secure Call Setting	If yes, all outbound calls are secure calls by default, without requiring the user to dial a star code first. Default setting: no • 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. **Note** This setting is applicable only if Secure Call Serv is set to yes on the line interface. See Line 1 and Line 2 Settings (PHONE 1 and PHONE 2), on page 38.
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	Controls how to handle incoming requests for loopback operation. • **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	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.

Field	Description
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)
CONFCID Setting	Enables or disables the CONFCID. Default setting: yes

Distinctive Ring Settings

Table 65: Distinctive Ring Parameters

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. See Regional, on page 55. Default setting: blank

Ring Settings

Table 66: Ring Parameters

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
Cfwd Ring Splash Len	Duration of ring splash when a call is forwarded (0 – 10.0s) Default setting: 0

Field	Description
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 • **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

CHAPTER 6

Administration Settings

Management

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

Web Access Management

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

Cisco ATA 191 Web Access Fields

Access to the Cisco ATA 191 web page is enabled by default. Admin Access allows you to manage the configuration from a computer in your office network, and Web Utility Access allows you to connect from a computer on a different subnet or on the Internet.

To access the ATA web page, launch a web browser and enter the URL in the address bar. The URL must include the specified protocol, the WAN IP address of the ATA, and the specified port number. For example, with the HTTPS protocol, a WAN IP address of 203.0.113.50, and port 443, you would enter: https://203.0.113.50:443

Table 67: Cisco ATA 191 Web Access Settings

Field	Description
Admin Access	This feature controls access to the ATA web page from devices that are connected via the ETHERNET (LAN) port. 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.
Web Utility Access	Select the protocol to use for access to the ATA web page from a device on the WAN. Choose **HTTP**, **HTTPS**, or both entries. For secure Internet access, select HTTPS. The default value is HTTPS.
Remote Management Port	Enter the port number to use for access to the ATA web page from a device on the WAN side of the ATA. The default port number is 443 for HTTPS, 80 for HTTP. 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 443, you would enter: https://203.0.113.50:443

Cisco ATA 192 Web Access Fields

Access to the Cisco ATA 192 web page is enabled by default. Admin Access allows you to manage the configuration from a computer in your office network, and Web Utility Access allows you to connect from a computer on a different subnet or on the Internet.

Table 68: Cisco ATA 192 Web Access Management Settings

Field	Description
Admin Access	This feature controls access to the ATA web page from devices that are connected via the ETHERNET (LAN) port. 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.
Web Utility Access	Select the protocol to use for access to the ATA web page from a device on the WAN. Choose **HTTP** and/or **HTTPS**. For secure Internet access, select HTTPS. The default value is HTTPS.

Cisco ATA 192 Remote Access Fields

In addition to the ATA web page access, the Cisco ATA 192 provides more features of Remote Management.

Table 69: Remote Access Settings

Field	Description
Remote Management	Allows access to the ATA web page 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. 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. 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 **Administration** > **Management** > **User List** page to change the administrator password. For more information, see User List (Password Management), on page 102.
Web Utility Access	Select the protocol to use for access to the ATA web page from a device on the WAN side of the ATA. Choose **HTTP** and/or **HTTPS**. The default value is **HTTPS**. 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 443, you would enter: `https://203.0.113.50:443`
Remote Upgrade	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. You can change this setting only when your computer is connected to the configuration utility from the LAN.
Allowed Remote IP Address	You can use this feature to limit access to the ATA web page 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	Enter the port number to use for access to the ATA web page from a device on the WAN side of the ATA. The default port number is 443 for HTTPS, 80 for HTTP. 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 443, you would enter: https://203.0.113.50:443

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.

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.

Table 70: TR-069 Settings

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

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.

SNMP Settings

Table 71: SNMP Parameters

Field	Description
Enabled, Disabled	Click **Enabled** to enable this feature, or click **Disabled** to disable it. The default setting is Disabled.
Trusted IPv4	Choose **Any** to allow access from any IPv4 address (not recommended). Click **Address** to specify the IPv4 address and subnet mask of a single SNMP manager or trap agent that can access the ATA through SNMP.
Trusted IPv6	Choose **Any** to allow access from any IPv6 address (not recommended). Click **Address** to specify the IPv6 address and prefix length 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.

SNMPv3 Settings

Table 72: SNMPv3 Parameters

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.

Field	Description
Privacy Password	Enter the key for the authentication protocol to use.

Trap Configuration

Table 73: Trap Parameters

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 is162.
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 ATA web page. The user-level account has access to modify a limited set of features.

For the IVR, you can configure these passwords on the System page.

Update a Password

Procedure

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.
- New Password—Enter your new password. The password must contain 8 to 32 characters.
- 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.

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

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.

SSH

Use the **Administration** > **Management** > **SSH** to configure SSH related setting.

Table 74: SSH Settings

Field	Description
User Name	Set SSH login user name.
Password	Set SSH login password.
SSH Access	Set the SSH access to enable or disable.

Log

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.

Debug Log Module

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

- As a best practice, we recommend that you enable logging only when needed, and disable logging when you finish the investigation. Logging consumes resources and can impact system performance.
- In this page, you can select the modules which you want to see debug messages in all severity levels.

Debug Log Setting

If Debug Log Server is enabled on the **Administration** > **Log** > **Debug Log Server** page, the ATA will send the debug messages to one server.

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.

Table 75: Debug Log Setting

Field	Description
Debug Log Size	Enter the maximum size of the log file in kilobytes. Valid values are from 128 to 1024.
IPv4 Address	Enter the IPv4 address of the debug log server where the messages will be sent.
IPv6 Address	Enter the IPv6 address of the debug log server where the messages will be sent.
Port	Enter the port to use on the server. Valid values are from 1 to 65535.

Debug Log Viewer

If logging is enabled on the **Administration** > **Log** > **Debug Log Viewer** 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.

For information about enabling and configuring logging, see Debug Log Module, on page 103.

Table 76: Debug Log Settings

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.
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.

Event Log Setting

Use the **Administration** > **Log** > **Event Log Setting** page to collect required event logs. Event log messages are sent via SYSLOG protocol using UDP transport type.

Use the Event Log Setting when troubleshooting. Four event categories are defined:

- DEV—Device information. A message is sent once device boot-up and network connectivity are ready.
- SYS—System-related information. A message is sent once while device boot-up and network connectivity are ready.
- CFG—Status of provision and configuration file change. A message is sent every time the provision service restarts due to configuration or network status change.

- REG—Registration status for each line. A message is sent every time registration status changes.

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.

Table 77: Event Log Settings

Field	Description
Address	Set the Event log server address.
Port	Set the Event log server port. Default value: 514
Flag	Set the Event log flag, it's a bitwise value. Setting list is as below: • `<Dev>:1(0x01)` • `<SYS>:2(0x01<<1)` • `<CFG>:4(0x01<<2)` • `<REG>:8(0x01<<3)` Default value: 15 (All events)

PRT Viewer

Use the **Administration** > **Log** > **PRT Viewer** to generate and download Problem Report Tools (PRT) files.

To generate a problem report remotely, see *Generate a Problem Report Remotely*.

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

Table 78: Problem Report Tool Settings

Field	Description
PRT Upload URL	Set the PRT log upload URL
PRT Upload Method	Set the PRT log upload method, **POST** or **PUT**. Default: POST
PRT Max Timer	Set the PRT max timer, valid range is 15-1440 minutes Disabled: 0 Default: 0
Problem Report Tools Logs	List the PRT file which is generated by user on ATA.
Generate PRT	Click this button to generate and download the contents of the PRT as a file on your computer. In the dialog box, you can open the file or save it.

Generate a Problem Report Remotely

You can initiate a problem report remotely. To do this, initiate a `SIP-NOTIFY` message from the server to the ATA, with the Event specified as `prt-gen`. and ATA response 200 OK to the server. Then

The workflow of the PRT generation is:

- The server sends NOTIFY to the ATA.
- ATA response 200 OK to the server, and sends PRT file to the upload server.
- The PRT upload server response 200 OK to the ATA.

Before you begin

- ATA registers successfully
- PRT Upload URL is configured

Procedure

Step 1 In the **Administration** > **Log** > **PRT Viewer** section, enter the **PRT Upload URL** parameter to specifiy the server to which you want to send the PRT. For example: **`http://10.74.133.94:9090`**.

The line is provisioned correctly with a valid SIP account.

Step 2 Click **Submit All Changes**.

PCM Viewer

Use the **Administration** > **Log** > **PCM Viewer** to download and view PCM.

The ATA allows you to capture the PCM log file while a user offhook to start a call.

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

Table 79: Log Viewer Settings

Field	Description
PCM Capture Enable	Enable or disable capture PCM.
Duration	Enter the PCM capture duration in seconds. The valid range is 20 to 300.
PCM File List	List the PCM file which is captured by user. Click **Refresh** to refresh the PCM Memory Dump File. Click **Download** to download the dump file on your computer.

CSS Dump

Use the **Administration** > **Log** > **CSS Dump** page to set and download CSS dump file.

Table 80: CSS Dump Settings

Field	Description
Auto Crash Dump	Set whether the ATA creates a crash dump file automatically when it occurs an error. Click **Enabled** to enable the feature, click **Disabled** to disable it. Default setting: Disabled
Manual Trigger Key(**##)	Set whether the user can manually trigger the creation of the CSS dump by pressing ****##** on the phone keypad. Click **Enabled** to enable the feature, click **Disabled** to disable it. Default setting: Disabled
CSS Dump List	List the CSS file which is captured by user. Click **Refresh** to refresh the CSS Memory Dump File. Click **Download** to download the dump file on your computer.

Crash Dump

Use the **Administration** > **Log** > **Crash Dump** page to set and download crash dump file.

Table 81: CSS Dump Settings

Field	Description
Runtime log to flash	Set whether the runtime log can be stored in the flash memory. Click **Enabled** to enable the feature, click **Disabled** to disable it. Default setting: Disabled
Crash Dump File	Display the captured crash dump file. Click the file name to download it on your computer. Click **Refresh** to refresh the crash dump file.

Factory Defaults

Use the **Administration** > **Factory Defaults** ATA web page to reset the ATA to the default configuration.

Alternatively, press and hold the **RESET** button for 10 seconds.

After the factory reset is performed successfully, all LEDs are fast flashing green.

All user-changeable non-default settings will be lost. This may include network and service provider data.

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 ATA web page. The default settings will be restored when you click **Submit**.

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.

⚠ **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.

Before you begin

Before upgrading the firmware, download the firmware upgrade file for the ATA.

Procedure

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.

ATA with Firmware Release 11.1.0MSR3-9 and Older Doesn't Upgrade

ATA with firmware version 11.1.0MSR3-9 and older fails to upgrade to the later releases if the server is implemented as HTTP chunk mode and the chunk-size is greater than 16K.

To bypass this upgrade failure, we recomment you to use this external HTTP server.

```
http://52.26.82.54/ata/ATA19x.11-1-0MPP0414-004.img
```

Configuration Management

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

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.

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. We recommend that you back up your current configuration settings before you restore a configuration.

Procedure

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 from the ATA web page. Another way to do it is by pressing the **Reset** > **Reboot** button.

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.

CHAPTER 7

Status and Statistics

System Information

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

Table 82: System Settings

Field	Description
Model	The model number and product description.
Product ID	The product ID of the ATA.
VID	The VID of the ATA
Serial Number	The serial number of the ATA.
Hardware Revision	The hardware version number.
Boot Version	The boot firmware version number.
Boot Partition	The boot partition of the ATA.
Firmware Version	The current firmware version.
Internet MAC Address	The MAC address of the WAN interface.
Host Name	The host name of the ATA.
Domain Name	The domain name of the ATA.
Current Time	Time that is set on the ATA.

Field	Description
Time Zone	Time zone that is set on the ATA.

Interface Information

Use the **Status** > **Interface Information** page to view information for the WAN interface (INTERNET port) and on ATA 192 only, the LAN interface (ETHERNET port).

IPv4 Interface List

Field	Description
Interface	The name of the interface: WAN or LAN (ATA 192 only).
Connect Type	The type of connection configured for the interface.
IP Address	The IPv4 address of the interface.
Subnet Mask	The subnet mask of the interface.
MAC Address	The MAC address of the interface.

IPv6 Interface List

Field	Description
Interface	The name of the interface: WAN or LAN (ATA 192 only).
Connect Type	The type of connection configured for the interface.
IP Address	The IPv6 address of the interface.
Prefix Length	The Prefix length of the interface.
MAC Address	The MAC address of the interface.

Port List (ATA 192 only)

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.
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 the count of TX and RX packets to zero.

Network Status

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

Table 83: Basic Interface Detail

Field	Description
Link Status	The status of the INTERNET (WAN) interface, showing whether the port is connected or disconnected.
Host Name	The host name of the ATA.
Domain	The domain name of the ATA.

Table 84: IPv4 Interface Detail

Field	Description
IP Address	The IPv4 address of the INTERNET (WAN) interface.
Subnet Mask	The subnet mask for the INTERNET (WAN) interface.
Gateway	The IPv4 address of the default gateway.
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)	IPv4 addresses for up to three DNS servers that are used for name resolution.

Table 85: IPv6 Interface Detail

Field	Description
IP Address	The IPv6 address of the INTERNET (WAN) interface.
Prefix Length	The Prefix Length for the INTERNET (WAN) interface.
Gateway	The IPv6 address of the default gateway.
DNS 1-2 (if applicable)	IPv6 addresses for up to three DNS servers that are used for name resolution.

Table 86: VLAN Information

Field	Description
CDP	The CDP status is enable or disable.

Field	Description
CDP VLAN ID	The CDP VLAN ID of the ATA.
IVR VLAN ID	The IVR VLAN ID of the ATA.
Active Vlan ID	The Active VLAN ID of the ATA.

Port Statistics (ATA 192 Only)

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).

Table 87: Port Statistics Settings

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 memory use.

Table 88: Memory Information Interface Detail

Field	Description
MemTotal	The total memory of ATA.
MemFree	The free memory of ATA.
refresh	Refresh the latest memory information.

DHCP Server Information (ATA 192 Only)

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

IPv4 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.

IPv6 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 DHCPv6 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.

IPv4 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.

IPv6 DHCP Server Details

Field	Description
DHCPv6 Server	Display the DHCPv6 Server status.
Address Assign Type	Display the DHCPv6 Server Address assign type.
DHCPv6 Delegation	Display the DHCPv6 Server delegation is yes or no.
IPv6 Address Prefix	Display the DHCPv6 address prefix.
IPv6 Address Prefix Length	Display the DHCPv6 address prefix length.
IPv6 Static DNS	Display the DHCPv6 Static DNS.
IPv6 Active DNS1	Display the DHCPv6 Active DNS1.
IPv6 Active DNS2	Display the DHCPv6 Active DNS2.
IPv6 LAN Address	Display the DHCPv6 LAN address.

CHAPTER 8

Frequently Asked Questions

I Can't Connect to the Internet Through the ATA

Procedure

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:

a. Cable or DSL modem

b. ATA

c. 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 ATA 192, 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.

I Upgraded my Firmware and the ATA doesn't Work 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 10 seconds. All user-changeable non-default settings will be lost. This may include network and service provider data.

I Can't 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.

There is no Dial Tone, and the Phone 1 or 2 LED is not Solid Green

Procedure

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:

a. Cable or DSL modem

b. ATA

c. 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.

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.

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).

Procedure

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

Step 2 Click the **Connections** tab.

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

Step 4 Click **OK**.

The DSL Telephone Line Does Not Fit in 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.

My Modem Doesn't 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.

The ATA Doesn't Have a Coaxial port for the Cable Connection

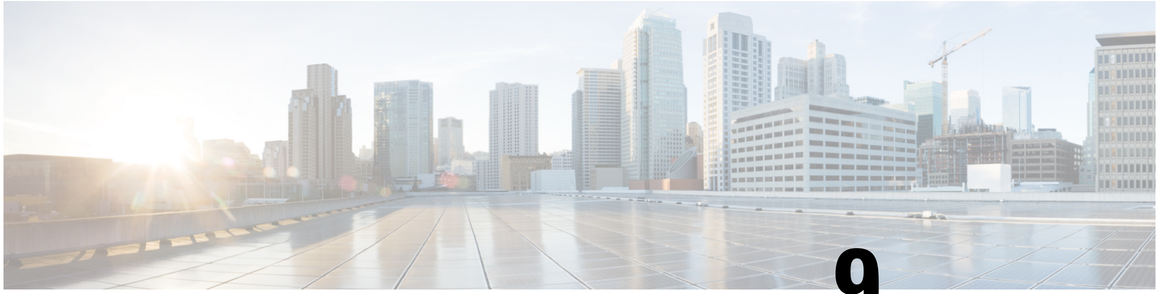
The ATA does not replace your modem. You need your cable modem in order to use the ATA. Connect your cable connection to the cable modem.

Call Statistics are Not Available in the Server

When Call Statistics are not available in the server, check the following:.

- Ensure the **Call Statistics** parameter is set to **Yes** in the web based configuration utility of ATA19x. You can check this parameter from **Voice** > **SIP** > **RTP Parameters**
- In the configuration file, the parameter must have value:

```
<Call_Statistics ua="na">Yes</Call_Statistics>
```



CHAPTER 9

IVR for Administration

Use 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.

Procedure

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.
- Log in as the PHONE port's user.

Step 4 Enter the code for the desired action.

IVR 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

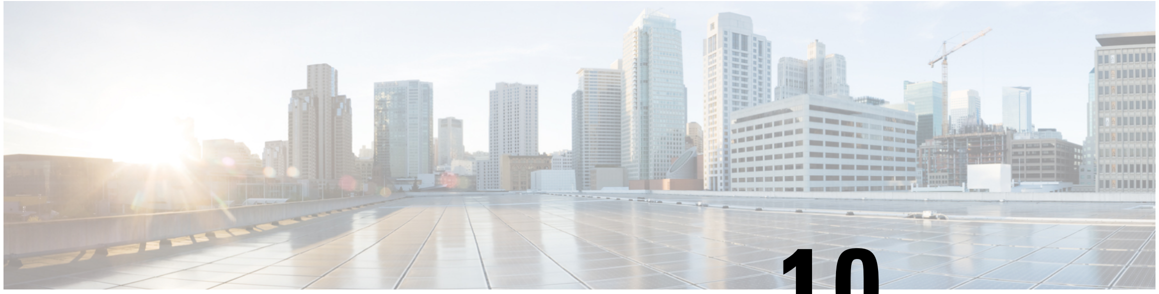
Table 89: IVR Settings

IVR Action	Menu Option	Choices and Instructions
Enter IVR Menu	****	
Check Internet Addressing Method	100	
Check Internet6 Addressing Method	600	
Set Internet Addressing Method	101	**0**—DHCP **1**—Static IP **2**—PPoE
Check Stack Mode	102	**0**—IPv4 **1**—IPv6 **2**—Dual
Set Stack Mode	103	**0**—IPv4 **1**—IPv6 **2**—Dual
Set Internet6 Addressing Method	601	**0**—DHCP **1**—Static IP **2**—PPPoE

IVR Action	Menu Option	Choices and Instructions
Check IPv6 Auto Configuration	607	**0**—Disable **1**—Enable
Set IPv6 Auto Configuration	606	**0**—Disable **1**—Enable
Check Internet IP Address (INTERNET port)	110	
Check Internet6 Address (INTERNET port)	610	
Set Static IP Address (INTERNET port)	111	Enter the IP address by using numbers on the telephone key pad. Use the ***** (star) key to a decimal point. **Note** This option is available only after you choose Static IP as the Internet Connection Type, through option 101.
Set Static IPv6 Address (INTERNET port)	611	Available only in static IPv6 mode
Check Network Mask	120	
Check IPv6 Prefix length	620	
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.
Set Static IPv6 Prefix length	621	Available only in static IPv6 mode
Check Gateway IP Address	130	
Check Gateway IPv6 Address	630	

IVR Action	Menu Option	Choices and Instructions
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.
Set Gateway IPv6 Address	631	Available only in static IPv6 mode
Check MAC Address	140	
Check Firmware Version	150	
Check Primary DNS Server Setting	160	
Check Primary IPv6 DNS Server Setting	660	
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.
Set Primary IPv6 DNS Server	661	
Check INTRNET web server port	170	
ATA 192 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

IVR Action	Menu Option	Choices and Instructions
Exit IVR	3948 (Spells EXIT on the phone keypad)	
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.
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.
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.



CHAPTER 10

Advanced Options for Phone Services

Optimize Fax Completion Rates

Issues can occur with fax transmissions over IP networks, even with the T.38 standard. Use the following task to help avoid any issues.

Procedure

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

- For G.711 fallback, we recommend 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 that 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
```

Troubleshoot Your Fax

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

Procedure

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. You can then send this file to Technical Support.

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

```
https://<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 or debug server. Set the Debug Level to 3. For more information, see System, on page 39.

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 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 https://www.cisco.com/go/sbc.

Dial Plan Configuration

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.

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.

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.

Table 90: Digit Sequences

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. • 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
. (period)	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.
<dialed:substituted>	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. **EXAMPLE 1: <8:1650>xxxxxxx** 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. **EXAMPLE 2: <:1>xxxxxxxxxx** 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.
, (comma)	Enter a comma between digits to play an “outside line” dial tone after a user-entered sequence. **EXAMPLE: 9, 1xxxxxxxxxx** An “outside line” dial tone is sounded after the user presses 9, and the tone continues until the user presses 1.
! (exclamation point)	Enter an exclamation point to prohibit a dial sequence pattern. **EXAMPLE: 1900xxxxxxx!** The system rejects any 11-digit sequence that begins with 1900.
*xx	Enter an asterisk to allow the user to enter a 2-digit star code.
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, xxxxxxx | 9, <:1>[2-9]xxxxxxxxx | 8, <:1212>xxxxxxx | 9, 1 [2-9] xxxxxxxxx | 9, 1 900 xxxxxxx ! | 9, 011xxxxxx. | 0 | [49]11)

- Extensions on your system

  ([1-8]xx | 9, xxxxxxx | 9, <:1>[2-9]xxxxxxxxx | 8, <:1212>xxxxxxx | 9, 1 [2-9] xxxxxxxxx | 9, 1 900 xxxxxxx ! | 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, xxxxxxx | 9, <:1>[2-9]xxxxxxxxx | 8, <:1212>xxxxxxx | 9, 1 [2-9] xxxxxxxxx | 9, 1 900 xxxxxxx ! | 9, 011xxxxxx. | 0 | [49]111)

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

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

  8, <:1212>xxxxxxx This is 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]xxxxxxxxx |8, <:1212>xxxxxxx | 9, 1 [2-9] xxxxxxxxx | 9, 1 900 xxxxxxx ! | 9,011xxxxxx. | 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]xxxxxxxxx |8, <:1212>xxxxxxx | 9, 1 [2-9] xxxxxxxxx | 9, 1 900 xxxxxxx ! |9, 011xxxxxx. | 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]xxxxxxxxx | 8, <:1212>xxxxxxx | 9, 1 [2-9] xxxxxxxxx | 9, 1 900 xxxxxxx ! | 9, 011xxxxxx. | 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]xxxxxxxxx | 8, <:1212>xxxxxxx | 9, 1 [2-9] xxxxxxxxx | 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 you dial 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, the set of candidates diminishes until only one or none are valid. When a terminating event occurs, the ATA either accepts the dialed sequence and initiates a call, or else rejects the sequence as invalid. You hear the reorder (fast busy) tone if the dialed sequence is invalid.

The following table explains how terminating events are processed.

Table 91: Terminating Events

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.	• 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.
A timeout occurs.	The number is rejected if the dialed digits don''t matched a digit sequence in the dial plan within the time specified by the interdigit timer. • 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
You press the # key.	• If the sequence is complete and permitted 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

- 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]xxxxxxxxx | 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]xxxxxxxxx | 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

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 Reset the Control Timers, on page 134.

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]xxxxxxxxx | 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]xxxxxxxxx | 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]xxxxxxxxxS0 | 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.

Reset the Control Timers

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

To edit a timer setting only for a particular digit sequence or type of call, you can edit the dial plan. See Digit Sequences, on page 129.

Procedure

Step 1 Log in to the ATA web page. If prompted, enter the administrative login provided by the Service Provider.

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.

Emergency Calls

Emergency Call Support Background

Emergency call service providers can register an ATA's location for each IP-based phone in a company. The location information server (LIS) transfers the emergency response location (ERL) to the ATA. The ATA stores its location during registration, after the ATA restarts. The location entry can specify the street address, building number, floor, room, and other office location information.

When you place an emergency call, the ATA transfers the location to the call server. The call server forwards the call and the location to the emergency call service provider. The emergency call service provider forwards the call and a unique call-back number (ELIN) to the emergency services. The emergency service or public safety answering point (PSAP) receives the ATA's location. The PSAP also receives a number to call you back, if the call disconnects.

See Emergency Call Support Terminology, on page 135 for the terms used to describe emergency calls from the phone.

The phone requests new location information for the following activities:

- You register the ATA with the call server.
- You or the user restarts the ATA and the ATA was previously registered with the call server.
- You change the network interface used in the SIP registration.
- You change the IP address of the ATA.

If both of the location servers do not send a location response, the phone resends the location request every two minutes.

Emergency Call Support Terminology

The following terms describe emergency call support for the ATA.

- Emergency Location ID Number (ELIN)–A number used to represent one or more ATA lines that locate the person who dialed emergency services.
- Emergency Response Location (ERL)–A logical location that groups a set of ATA lines.
- HTTP Enabled Location Delivery (HELD)–An encrypted protocol that obtains the PIDF-LO location for the ATA from a location information server (LIS).
- Location Information Server (LIS)–A server that responds to a SIP-based ATA HELD request and provides the ATA location using a HELD XML response.

- Emergency Call Service Provider–The company that responds to an ATA HELD request with the ATA's location. When you make an emergency call (which carries the ATA's location), a call server routes the call to this company. The emergency call service provider adds an ELIN and routes the call to the emergency services (PSAP). If the call is disconnected, the PSAP uses the ELIN to reconnect with the ATA used to make the emergency call.
- Public Safety Answering Point (PSAP)–Any emergency service (for example, fire, police, or ambulance) joined to the Emergency Services IP Network.
- Universally Unique Identifier (UUID)–A 128-bit number used to uniquely identify a company using emergency call support.

Configure the ATA to Make Emergency Calls

Before you begin

- Obtain the E911 Geolocation Configuration URLs and the company identifier for the ATA from your emergency call services provider. You can use the same Geolocation URLs and company identifier for line 1 and line 2 (PHONE 1 and PHONE 2).
- Access the phone adapter administration web page. See Access the Phone Web Interface, on page 19.

Procedure

Step 1 Select **Voice** > **Line n**, where *n* is the line number that represents PHONE 1 or PHONE 2.

Step 2 In the section **Call Feature Settings**, set the parameter **Emergency Number** as described in Call Feature Settings, on page 79.

Step 3 In the section **E911 Geolocation Configuration**, set the parameters **Company UUID**, **Primary Request URL**, and **Secondary Request URL** as described in E911 Geolocation Configuration, on page 80.

Step 4 Click **Submit**.