



Cisco ATA 190 Analog Telephone Adapter Administration Guide for SIP

Version 1.1

First Published: May 7, 2014

Last Updated: July 27, 2015

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Text Part Number: OL-31821-02

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GLOSSARY



Preface

Overview

The *Cisco Analog Telephone Adapter 190 Administration Guide for SIP (Version 1.0)* provides the information that you need to install, configure, and manage the Cisco ATA 190 Analog Telephone Adapter (ATA 190) on a Session Initiation Protocol (SIP) network.

Audience

This guide is intended for service providers and network administrators who administer Voice over IP (VoIP) services using the ATA 190. Most of the tasks described in this guide are not intended for end users of the ATA 190. Many of these tasks affect the ability of the ATA 190 to function on the network, and require an understanding of IP networking and telephony concepts.

Organization

This manual is organized as follows:

Chapter 1, “Cisco ATA 190 Analog Telephone Adapter Overview”	Provides descriptions of hardware and software features of the ATA 190 along with a brief overview of the Session Initiation Protocol (SIP).
Chapter 2, “Preparing to Install the ATA 190 on Your Network”	Provides information on the interactions between the ATA 190, Cisco Unified Communications Manager, and other devices. Also describes options for powering the ATA 190.
Chapter 3, “ATA 190 Installation”	Provides information on how to connect the ATA 190 hardware and load the QED and firmware files.
Chapter 4, “Configuring the ATA 190”	Provides information on how to configure the ATA 190 to operate with Session Initiation Protocol (SIP).

Chapter 5, “Configuring Fax Services”	Provides instructions for configuring both ports of the ATA 190 to support fax transmission.
Chapter 6, “Using SIP Supplementary Services”	Provides end-user information about pre-call and mid-call supplementary services, such as call forward, call return, call forwarding, and conference calling.
Appendix A, “ATA 190 Specifications”	Provides physical specifications for the ATA 190.
Appendix B, “Voice Menu Codes”	Provides a quick-reference list of the voice configuration menu options for the ATA 190.
Appendix C, “ATA 190 Country-Specific Tones and Cadences”	Provides tone parameters for various countries.
Appendix D, “Troubleshooting and Maintenance”	Provides basic testing and troubleshooting procedures for the ATA 190.
Glossary	Provides definitions of commonly used terms.

Related Documentation

For more information about the ATA 190 or Cisco Unified Communications Manager, refer to the following publications:

Cisco ATA 190 Analog Telephone Adapter

- RFC 3261 (*SIP: Session Initiation Protocol*)
- RFC 2543 (*SIP: Session Initiation Protocol*)
- RFC 768 (*User Datagram Protocol*)
- RFC 2198 (*RTP Payload for Redundant Audio Data*)
- RFC 2833 (*RTP Payload for DTMF Digits, Telephony Phones and Telephony Signals*)
- RFC 2327 (*SDP: Session Description Protocol*)
- RFC 4730 (*A Session Initiation Protocol (SIP) Event Package for Key Press Stimulus (KPML)*)
- RFC 3515 (*The Session Initiation Protocol (SIP) Refer Method*)
- *Read Me First - ATA Boot Load Information*
- *Cisco ATA 190 Analog Telephone Adapter At a Glance*
- *Regulatory Compliance and Safety Information for the Cisco ATA 190*
- *Cisco ATA 190 Analog Telephone Adapter Release Notes*

Cisco Unified Communications Manager

These publications are available at the following URL:

<http://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-call-manager/tsd-products-support-series-home.html>

Obtaining Documentation, Obtaining Support, and Security Guidelines

For information on obtaining documentation, obtaining support, providing documentation feedback, security guidelines, and also recommended aliases and general Cisco documents, see the monthly What's New in Cisco Product Documentation, which also lists all new and revised Cisco technical documentation, at:

<http://www.cisco.com/en/US/docs/general/whatsnew/whatsnew.html>

Cisco Product Security Overview

This product contains cryptographic features and is subject to United States and local country laws governing import, export, transfer and use. Delivery of Cisco cryptographic products does not imply third-party authority to import, export, distribute or use encryption. Importers, exporters, distributors and users are responsible for compliance with U.S. and local country laws. By using this product you agree to comply with applicable laws and regulations. If you are unable to comply with U.S. and local laws, return this product immediately.

Further information regarding U.S. export regulations may be found at

http://www.access.gpo.gov/bis/ear/ear_data.html.

Document Conventions

This document uses the following conventions:

Convention	Description
boldface font	Commands and keywords are in boldface .
<i>italic font</i>	Arguments for which you supply values are in <i>italics</i> .
[]	Elements in square brackets are optional.
{ x y z }	Alternative keywords are grouped in braces and separated by vertical bars.
[x y z]	Optional alternative keywords are grouped in brackets and separated by vertical bars.
string	A nonquoted set of characters. Do not use quotation marks around the string or the string will include the quotation marks.
screen font	Terminal sessions and information the system displays are in <code>screen font</code> .
boldface screen font	Information you must enter is in boldface screen font .
<i>italic screen font</i>	Arguments for which you supply values are in <i>italic screen font</i> .
^	The ^ symbol represents the key labeled Control—For example, the key combination ^D in a screen display means hold down the Control key while you press the D key.
< >	Nonprinting characters, such as passwords are in angle brackets.

**Note**

Means *reader take note*. Notes contain helpful suggestions or references to material not covered in the publication.

**Caution**

Means *reader be careful*. In this situation, you might do something that could result in equipment damage or loss of data.

**Warning**

Means danger. You are in a situation that could cause bodily injury. Before you work on any equipment, be aware of the hazards involved with electrical circuitry and be familiar with standard practices for preventing accidents.



Cisco ATA 190 Analog Telephone Adapter Overview

This section describes the hardware and software features of the Cisco ATA 190 Analog Telephone Adapter (ATA 190). The section includes a brief overview of the Session Initiation Protocol (SIP).

The ATA 190 analog telephone adapters are handset-to-Ethernet adapters that allow regular analog phones to operate on IP-based telephony networks. The ATA 190 support two voice ports, each with an independent phone number. The ATA 190 also has an RJ-45 10/100BASE-T data port.

This section covers these topics:

- [Session Initiation Protocol Overview, page 1-2](#)
- [Hardware Overview, page 1-4](#)
- [Software Features, page 1-5](#)
- [Installation and Configuration Overview, page 1-9](#)

Figure 1-1 *Cisco Analog Telephone Adapter*



Session Initiation Protocol Overview

Session Initiation Protocol (SIP) is the Internet Engineering Task Force (IETF) standard for real-time calls and conferencing over Internet Protocol (IP). SIP is an ASCII-based, application-layer control protocol (defined in RFC3261) that can be used to establish, maintain, and terminate multimedia sessions or calls between two or more endpoints.

Like other Voice over IP (VoIP) protocols, SIP is designed to address the functions of signaling and session management within a packet telephony network. Signaling allows call information to be carried across network boundaries. Session management provides the ability to control the attributes of an end-to-end call.

**Note**

SIP for the ATA 190 is compliant with RFC2543.

This section contains these topics:

- [SIP Capabilities, page 1-2](#)
- [Components of SIP, page 1-2](#)

SIP Capabilities

SIP provides these capabilities:

- Determines the availability of the target endpoint. If a call cannot be completed because the target endpoint is unavailable, SIP determines whether the called party is already on the phone or did not answer in the allotted number of rings. SIP then returns a message indicating why the target endpoint was unavailable.
- Determines the location of the target endpoint. SIP supports address resolution, name mapping, and call redirection.
- Determines the media capabilities of the target endpoint. Using the Session Description Protocol (SDP), SIP determines the lowest level of common services between endpoints. Conferences are established using only the media capabilities that all endpoints support.
- Establishes a session between the originating and target endpoint. If the call can be completed, SIP establishes a session between the endpoints. SIP also supports midcall changes, such as adding another endpoint to the conference or changing the media characteristic or codec.
- Handles the transfer and termination of calls. SIP supports the transfer of calls from one endpoint to another. During a call transfer, SIP establishes a session between the transferee and a new endpoint (specified by the transferring party). SIP also terminates the session between the transferee and the transferring party. At the end of a call, SIP terminates the sessions between all parties. Conferences can consist of two or more users and can be established using multicast or multiple unicast sessions.

Components of SIP

SIP is a peer-to-peer protocol. The peers in a session are called User Agents (UAs). A user agent can function in one of these roles:

- User agent client (UAC)—A client application that initiates the SIP request.

- User agent server (UAS)—A server application that contacts the user when a SIP request is received and returns a response on behalf of the user.

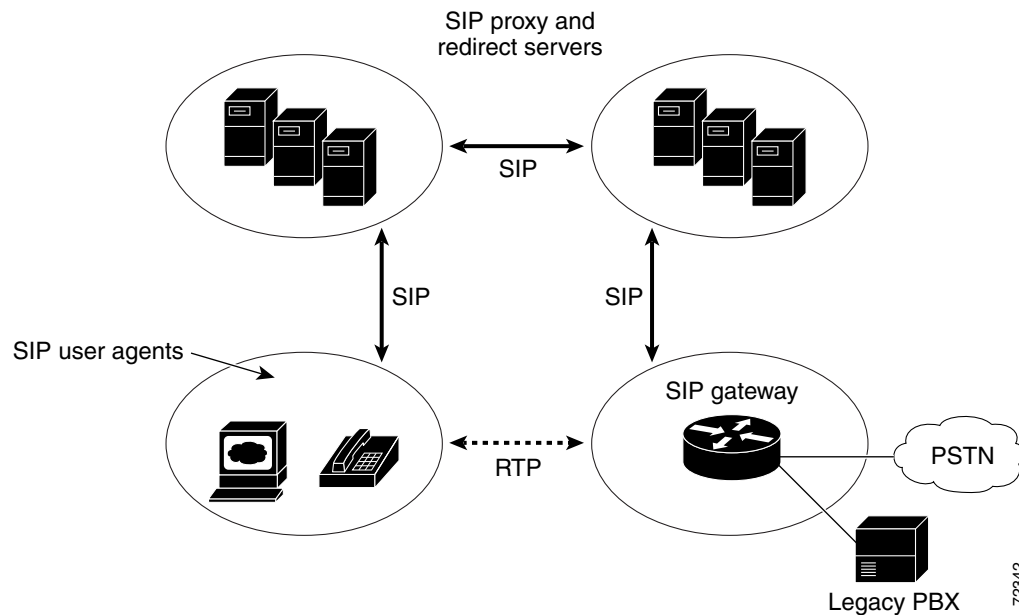
Typically, a SIP endpoint is capable of functioning as both a UAC and a UAS, but functions only as one or the other per transaction. Whether the endpoint functions as a UAC or a UAS depends on the UA that initiated the request.

From an architectural standpoint, the physical components of a SIP network can also be grouped into two categories—Clients and servers. [Figure 1-2](#) illustrates the architecture of a SIP network.

**Note**

SIP servers can interact with other application services, such as Lightweight Directory Access Protocol (LDAP) servers, a database application, or an extensible markup language (XML) application. These application services provide back-end services such as directory, authentication, and billable services.

Figure 1-2 SIP Architecture



SIP Clients

SIP clients include:

- Gateways—Provide call control. Gateways provide many services, the most common being a translation function between SIP conferencing endpoints and other terminal types. This function includes translation between transmission formats and between communications procedures. In addition, the gateway also translates between audio and video codecs and performs call setup and clearing on both the LAN side and the switched-circuit network side.
- Phones—Can act as either a UAS or UAC. The ATA 190 can initiate SIP requests and respond to requests.

SIP Servers

SIP servers include:

- Proxy server—The proxy server is an intermediate device that receives SIP requests from a client and then forwards the requests on the client's behalf. Proxy servers receive SIP messages and forward them to the next SIP server in the network. Proxy servers can provide functions such as authentication, authorization, network access control, routing, reliable request retransmission, and security.
- Redirect server—Receives SIP requests, strips out the address in the request, checks its address tables for any other addresses that may be mapped to the address in the request, and then returns the results of the address mapping to the client. Redirect servers provide the client with information about the next hop or hops that a message should take, then the client contacts the next hop server or UAS directly.
- Registrar server—Processes requests from UACs for registration of their current location. Registrar servers are often colocated with a redirect or proxy server.

Hardware Overview

The ATA 190 is a compact, easy to install device. [Figure 1-3](#) shows the rear panel of the ATA 190.

Figure 1-3 ATA 190—Rear View



The unit provides these connectors:

- 5V DC power connector.
- Two RJ-11 FXS (Foreign Exchange Station) ports—The ATA 190 supports two independent RJ-11 phone ports that can connect to any standard analog phone device. Each port supports either voice calls or fax sessions, and both ports can be used simultaneously.
- The ATA 190 has one network port—An RJ-45 10/100BASE-T data port to connect an Ethernet-capable device, such as a computer, to the network.



Note

The ATA 190 network port performs autonegotiation for duplex and speed. It supports speed of 10/100 Mbps and full-duplex.

Software Features

The ATA 190 supports these protocols, services, and methods:

- [Secure Real-Time Transport Protocol, page 1-5](#)
- [Fax Passthrough, page 1-5](#)
- [Transport Layer Security Protocol, page 1-5](#)
- [T.38 Fax Relay, page 1-5](#)
- [Voice Codecs Supported, page 1-6](#)
- [Other Supported Protocols, page 1-6](#)
- [ATA 190 SIP Services, page 1-6](#)
- [Modem Standards, page 1-7](#)
- [Fax Services, page 1-7](#)
- [Methods Supported, page 1-7](#)
- [Supplementary Services, page 1-8](#)

Secure Real-Time Transport Protocol

Secure Real-Time Transport Protocol secures voice conversations on the network and provides protection against replay attacks.



Note

Currently, ATA 190 does not support secure conference call. A 2-way secure call is supported.

Fax Passthrough

Name Signaling Event (NSE) -based and re-INVITE-based passthrough are simply transport of fax communications using the G.711a/u codec.

Transport Layer Security Protocol

Transport Layer Security (TLS) is a cryptographic protocol that secures data communications such as email on the Internet. TLS is functionally equivalent to Secure Sockets Layer (SSL).

T.38 Fax Relay

The T.38 fax relay feature enables devices to use fax machines to send files over the IP network. In general, when a fax is received, it is converted to an image, then sent to the T.38 fax device. When the target T.38 fax device receives this image, the device converts the image back to an analog fax signal. T.38 fax relays configured with voice gateways decode or demodulate the fax signals before they are transported over IP. With the SIP call control protocol, the T.38 fax relay is indicated by Session Description Protocol (SDP) entries in the initial SIP INVITE message. After the initial SIP INVITE message, the call is established to switch from voice mode to T.38 mode. Cisco Unified Communications Administration allows you to configure a SIP profile that supports T.38 fax communication.

Voice Codecs Supported

The ATA 190 supports these voice codecs (check your other network devices for the codecs they support):

- G.711 mu-law
- G.711 A-law
- G.729a
- G.729ab

Other Supported Protocols

The ATA 190 supports these additional protocols:

- 802.1Q VLAN tagging
- Cisco Discovery Protocol (CDP)
- Domain Name System (DNS)
- Dynamic Host Configuration Protocol (DHCP)
- Internet Control Message Protocol (ICMP)
- Internet Protocol (IP)
- Secure Real-Time Transport Protocol (SRTP)
- Transmission Control Protocol (TCP)
- Trivial File Transfer Protocol (TFTP)
- User Datagram Protocol (UDP)
- Transport Layer Security (TLS)

ATA 190 SIP Services

These services include these features:

- IP address assignment—DHCP-provided or statically configured
- ATA 190 configuration by Cisco Unified Communications Manager configuration interface
- VLAN configuration
- Cisco Discovery Protocol (CDP)
- Low-bit-rate codec selection
- User authentication
- Configurable tones (ringback tone, reorder tone, dialing tone, outside dialing tone, busy tone, call waiting tone)
- Dial plan and PLAR
- SIP proxy server redundancy
- Privacy features
- User-configurable, call waiting, permanent default setting

- Comfort noise during silent period when using G.711u/a and G.729ab
- Caller ID format
- Ring frequency/voltage adjustment
- Hookflash detection timing configuration
- Type of Service (ToS) configuration for audio and signaling Ethernet packets
- Debugging and diagnostic tools

Modem Standards

The ATA 190 supports the following modem standards:

- V.90
- V.92
- V.44
- K56Flex
- ITU-T V.34 Annex 12
- ITU-T V.34
- V.32bis
- V.32
- V.21
- V.22
- V.23

Fax Services

The ATA 190 supports two modes of fax services:

- Fax pass-through mode: Receiver-side Called Station Identification (CED) tone detection with automatic G.711A-law or G.711μ-law switching.
- T.38 Fax Relay mode: The T.38 fax relay feature enables devices to use fax machines to send files over the IP network. In general, when a fax is received, it is converted to an image, then sent to the T.38 fax device. When the target T.38 fax device receives this image, the device converts the image back to an analog fax signal. T.38 fax relays configured with voice gateways decode or demodulate the fax signals before they are transported over IP.

**Note**

Success of fax transmission depends on network conditions and fax modem response to these conditions. The network must have reasonably low network jitter, network delay, and packet loss rate.

Methods Supported

The ATA 190 supports these methods. For more information, see RFC3261 (*SIP: Session Initiation Protocol*).

- REGISTER
- REFER
- INVITE
- BYE
- CANCEL
- NOTIFY
- OPTIONS
- ACK
- SUBSCRIBE

Supplementary Services

SIP supplementary services are services that you can use to enhance your phone service. For information on how to use these services, see [Chapter 6, “Using SIP Supplementary Services.”](#)

The ATA 190 supports these SIP supplementary services:

- Caller ID
- Call-waiting caller ID
- Voice mail indication
- Making a conference call
- Call waiting
- Call forwarding
- Calling-line identification
- Unattended transfer
- Attended transfer
- Shared Line
- SpeedDial
- Meet-Me Conference
- Call Pickup/Group Call Pickup
- Redial
- Secure Call
- C-Barge

Installation and Configuration Overview

[Table 1-1](#) provides the basic steps required to install and configure the ATA 190 to make it operational in a typical SIP environment where many ATA 190s must be deployed.

Table 1-1 **Overview of the Steps Required to Install and Configure the ATA 190 and Make it Operational**

Action	Reference
1. Plan the network and ATA 190 configuration.	
2. Install the Ethernet connection.	
3. Install and configure the other network devices.	
4. Install the ATA 190 but do not power up the ATA 190 yet.	Install the ATA 190, page 3-3
5. Power up the ATA 190.	



Preparing to Install the ATA 190 on Your Network

The ATA 190 enables you to communicate using voice over a data network. To provide this capability, the ATA 190 depends upon and interacts with several other key Cisco Unified IP Telephony and Network components, including Cisco Unified Communications Manager, DNS and DHCP servers, TFTP servers, media resources, and so on.

This chapter focuses on the interactions between the ATA 190, Cisco Unified Communications Manager, DNS and DHCP servers, TFTP servers, and switches. It also describes options for powering the ATA 190.

For related information about voice and IP communications, see this URL:

<http://www.cisco.com/c/en/us/products/unified-communications/index.html>

This chapter provides an overview of the interaction between the ATA 190 and other key components of the Voice over IP (VoIP) network. It includes these topics:

- [Understanding Interactions with Other Cisco Unified IP Communications Products, page 2-1](#)
- [Providing Power to the ATA 190, page 2-2](#)
- [Understanding Phone Configuration Files, page 2-3](#)
- [Understanding the ATA 190 Startup Process, page 2-4](#)
- [Adding the ATA 190 to the Cisco Unified Communications Manager Database, page 2-5](#)
- [Determining the MAC Address of an ATA 190, page 2-6](#)

Understanding Interactions with Other Cisco Unified IP Communications Products

To function in the IP telephony network, the ATA 190 must be connected to a networking device, such as a Cisco Catalyst switch. You must also register the ATA 190 with a Cisco Unified Communications Manager system before sending and receiving calls.

This section includes information on [Understanding ATA 190 Interaction with Cisco Unified Communications Manager, page 2-2](#).

Understanding ATA 190 Interaction with Cisco Unified Communications Manager

Cisco Unified Communications Manager is an open and industry-standard call processing system. Cisco Unified Communications Manager software sets up and tears down calls between analog phones that are connected to the ATA 190, and thus integrates traditional PBX functionality with the corporate IP network. Cisco Unified Communications Manager manages the components of the IP telephony system: the phones, the access gateways, and the resources necessary for features such as call conferencing and route planning. Cisco Unified Communications Manager also provides:

- Firmware for devices
- Authentication and encryption (if configured for the telephony system)
- Configuration and CTL files via the TFTP service
- Phone registration
- Call preservation, so that a media session continues if signaling is lost between the primary Communications Manager and a phone

For information about configuring Cisco Unified Communications Manager to work with the IP devices described in this chapter, see *Cisco Unified Communications Manager Administration Guide*, *Cisco Unified Communications Manager System Guide*, and *Cisco Unified Communications Manager Security Guide*.

Providing Power to the ATA 190

The ATA 190 is powered with external power. External power is provided through a separate power supply.

The following sections provide more information about powering an ATA 190:

- [Power Guidelines, page 2-2](#)
- [Power Outage, page 2-2](#)
- [Understanding Phone Configuration Files, page 2-3](#)

Power Guidelines

The following power type and guideline applies to external power for the ATA 190:

- Power Type—External power (provided through the Universal AC external power supply).
- Guidelines—The ATA 190 uses the Universal AC power supply 100/240V.

Power Outage

Your accessibility to emergency service through the phone depends the phone being powered. If there is an interruption in the power supply, Service and Emergency Calling Service dialing will not function until power is restored. In the case of a power failure or disruption, you may need to reset or reconfigure equipment before using the Service or Emergency Calling Service dialing.

Understanding Phone Configuration Files

Configuration files for a phone are stored on the TFTP server and define parameters for connecting to Cisco Unified Communications Manager. In general, any time you make a change in Cisco Unified Communications Manager that requires the ATA 190 line to be reset, a change is automatically made to the phone configuration file. If a system reset or restart is required, both lines must reset or restart at the same time.

Configuration files also contain information about which image load the ATA 190 should be running. If this image load differs from the one that is currently loaded on an ATA 190, the phone contacts the TFTP server to request the required load files. (These files are digitally signed to ensure the authenticity of the file source.)

If the device security mode in the configuration file is set to Authenticated and the CTL file on the ATA 190 has a valid certificate for Cisco Unified Communications Manager, the phone establishes a TLS connection to Cisco Unified Communications Manager. Otherwise, the ATA 190 establishes a TCP/UDP connection. Users can go to **Voice > Line > SIP Settings** on the ATA 190 web GUI, where the SIP Transport should correspond to the transport type in the Phone Security Profile in Cisco Unified Communications Manager.

If you configure security-related settings in Cisco Unified Communications Manager Administration, the phone configuration file contains sensitive information. To ensure the privacy of a configuration file, configure it for encryption. For detailed information, see [Configuring Encrypted Phone Configuration Files](#) in *Cisco Unified Communications Manager Security Guide*.

An ATA 190 accesses a default configuration file named XMLDefault.cnf.xml only when the ATA 190 cannot get configuration file named ATA<mac_address>.cnf.xml, where *mac_address* is the MAC address of the phone.

If autoregistration is not enabled and you did not add the ATA 190 to the Cisco Unified Communications Manager database, the ATA 190 does not attempt to register with Cisco Unified Communications Manager.

If the ATA 190 has registered before, the ATA 190 accesses the configuration file named ATA<mac_address>.cnf.xml, where *mac_address* is the MAC address of the phone.

For the ATA 190, the TFTP server generates these SIP configuration files:

- SIP IP Phone:
 - For unsigned and unencrypted files—ATA<mac>.cnf.xml
 - For signed files—ATA<mac>.cnf.xml.sgn
 - For signed and encrypted files—ATA<mac>.cnf.xml.enc.sgn
 - Maximum number of dial patterns is 10
 - Maximum length of each dial pattern is 30

The filenames are derived from the MAC Address in the Phone Configuration window of Cisco Unified Communications Manager Administration. The MAC address uniquely identifies the phone. For more information, see the *Cisco Unified Communications Manager Administration Guide*.

For more information about how the phone interacts with the TFTP server, see the [Cisco Unified Communications Manager System Guide](#), [Cisco TFTP](#) section.

Understanding the ATA 190 Startup Process

When connecting to the VoIP network, the ATA 190 goes through a standard startup process, as described in [Table 2-1](#). Depending on your specific network configuration, not all these process steps may occur on your ATA 190.

Table 2-1 ATA 190 Startup Process

Task	Purpose	Related Topics
1.	Obtaining Power. The ATA 190 uses external power.	See Providing Power to the ATA 190, page 2-2 .
2.	Loading the Stored Image. The Cisco ATA 190 has nonvolatile flash memory in which it stores firmware images and user-defined preferences. At startup, the ATA 190 runs a bootstrap loader that loads an ATA 190 image stored in flash memory. Using this image, the ATA 190 initializes its software and hardware.	
3.	Obtaining an IP Address. If the Cisco ATA 190 is using DHCP to obtain an IP address, the device queries the DHCP server to obtain one. If you are not using DHCP in your network, assign static IP addresses to each device locally.	
4.	Requesting the CTL file. The TFTP server stores the CTL file. This file contains the certificates necessary for establishing a secure connection between the device and Cisco Unified Communications Manager.	See the <i>Cisco Unified Communications Manager Security Guide</i> , Configuring the Cisco CTL Client .
5.	Requesting the Configuration File. The TFTP server has configuration files, which define parameters for connecting to Cisco Unified Communications Manager and other information for the ATA 190.	See Understanding Phone Configuration Files, page 2-3 .
6.	Contacting Cisco Unified Communications Manager. The configuration file defines how the ATA 190 communicates with Cisco Unified Communications Manager and provides a device with its load ID. After obtaining the file from the TFTP server, the device attempts to make a connection to the highest priority Cisco Unified Communications Manager on the list. If the security profile of the device is configured for secure signaling (encrypted or authenticated), and the Cisco Unified Communications Manager is set to Mixed (security) mode, the device makes a TLS connection. Otherwise, it makes a nonsecure TCP/UDP connection.	See Understanding Phone Configuration Files, page 2-3 .

Adding the ATA 190 to the Cisco Unified Communications Manager Database

Before installing the ATA 190, choose a method for adding the devices to the Cisco Unified Communications Manager database. These sections describe the methods:

- [Adding the ATA 190 with Auto-Registration, page 2-5](#)
- [Adding the ATA 190 with Cisco Unified Communications Manager Administration, page 2-6](#)

Table 2-2 provides an overview of these methods for adding the ATA 190 to the Cisco Unified Communications Manager database.

Table 2-2 *Methods for Adding the ATA 190 to the Cisco Unified Communications Manager Database*

Method	Requires MAC Address?	Notes
Autoregistration	No	<ul style="list-style-type: none"> • Results in automatic assignment of directory numbers. • Not available when mixed mode is enabled.
Using the Cisco Unified Communications Manager Administration	Yes	Requires phones to be added individually.

Adding the ATA 190 with Auto-Registration

By enabling autoregistration before you begin installing the ATA 190, you can:

- Automatically add devices without first gathering MAC addresses from the ATA 190.
- Automatically add an ATA 190 to the Cisco Unified Communications Manager database when you physically connect the phone to your IP telephony network. During autoregistration, Cisco Unified Communications Manager assigns the next available sequential directory number to the phone.
- If you want to change settings, quickly enter devices into the Cisco Unified Communications Manager database and modify any settings, such as the directory numbers, from Cisco Unified Communications Manager.
- Move autoregistered devices to new locations and assign them to different device pools without affecting their directory numbers.



Note

Support exists for autoregistration for several devices in the Unified CM at the same time.

Autoregistration is disabled by default. Sometimes, you may not want to use autoregistration. For example, if you want to assign a specific directory number to the phone or if you plan to use secure connection with Cisco Unified Communications Manager as described in the *Cisco Unified Communications Manager Security Guide*. For information about enabling autoregistration, see the Enabling Auto-Registration in the *Cisco Unified Communications Manager Administration Guide*.

**Note**

For mixed mode, autoregistration is automatically disabled and cannot be changed. For nonsecure mode, autoregistration is disabled by default but can be enabled manually.

Related Topics

[Adding the ATA 190 with Cisco Unified Communications Manager Administration, page 2-6](#)

Adding the ATA 190 with Cisco Unified Communications Manager Administration

You can add the ATA 190 individually to the Cisco Unified Communications Manager database using Cisco Unified Communications Manager Administration. To do so, first obtain the MAC address for each device.

For information about determining a MAC address, see [Determining the MAC Address of an ATA 190, page 2-6](#).

After you have collected MAC addresses, in Cisco Unified Communications Manager Administration, choose **Device > Phone** and click **Add New** to begin.

**Note**

ATA 190 has two FXS ports, and each port has its MAC address. The first ATA 190 port uses the MAC address and the second ATA 190 port uses the shifted MAC address (example, AABBCDDDEEFF to BBCCDDEEFF01). You can add two devices from the Unified CM administration page.

For complete instructions and conceptual information about Cisco Unified Communications Manager, see the *Cisco Unified Communications Manager Administration Guide* and the *Cisco Unified Communications Manager System Guide*.

Related Topics

[Adding the ATA 190 with Auto-Registration, page 2-5](#)

Determining the MAC Address of an ATA 190

Several of the procedures that are described in this document require you to determine the MAC address of an ATA 190. You can determine the MAC address for a device in any of these ways:

- Look at the MAC label on the back of the device.
- Go to **Voice > Information** on the web page of the device and check the MAC address.



ATA 190 Installation

This section describes how to connect the ATA 190 hardware and configure the ATA 190 by loading the QED and firmware files. Install the QED file first, then install the firmware file.

Cisco ATA Rear Panel Connections

Figure 3-1 Cisco ATA Rear Panel



RESET—Using a paper clip or similar object, press this button briefly to restart the unit. Press and hold for 10 seconds to restore the factory default settings.

PHONE 1/PHONE 2—Connection to Analog telephones or fax.

NETWORK—Connection to IP network.

POWER—Connection to 5V power adapter.

Network Requirements

The ATA 190 acts as an endpoint on an IP telephony network. The following equipment is required:

- Call Control system
- Voice packet gateway—Required if you are connecting to the Public Switched Telephone Network (PSTN). A gateway is not required if an analog key system is in effect.
- Ethernet connection

Safety Recommendations

To ensure general safety, follow these guidelines:

- Do not get this product wet or pour liquids into this device.
- Do not open or disassemble this product.
- Do not perform any action that creates a potential hazard to people or makes the equipment unsafe.
- Use only the power supply that comes with the ATA 190.



Warning

Ultimate disposal of this product should be handled according to all national laws and regulations.



Warning

Read the installation instructions before you connect the system to its power source.



Warning

The plug-socket combination must always be accessible because it serves as the main disconnecting device.



Warning

Do not work on the system or connect or disconnect cables during periods of lightning activity.



Warning

To avoid electric shock, do not connect safety extra-low voltage (SELV) circuits to telephone-network voltage (TNV) circuits. LAN ports contain SELV circuits, and WAN ports contain TNV circuits. Some LAN and WAN ports both use RJ-45 connectors. Use caution when connecting cables.

For translated warnings, see the *Regulatory Compliance and Safety Information for the Cisco ATA 190* manual.

What the ATA 190 Package Includes

The ATA 190 package contains the following items:

- Cisco ATA 190 Analog Telephone Adapter
- *Cisco ATA 190 Analog Telephone Adapter at a Glance*
- *Cisco ATA 190 Analog Telephone Adapter (SIP) Product Safety and User Manual*
- Regulatory Compliance and Safety Information for the ATA 190
- 5V power adapter with appropriate country clip
- Ethernet cable



Note

The ATA 190 is intended for use only with the 5V DC power adapter that comes with the unit.

Install the ATA 190

To install an ATA 190, follow these steps:

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 190. Each ATA 190 ships with one Ethernet cable in the box. |
-

**Note**

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.

Attach a Phone to the ATA 190

You can attach one or two phones to an ATA 190 by connecting them to a phone port of the ATA 190 with an RJ11 cable. The ATA 190 PHONE1 and PHONE2 LEDs light as solid green when there is activity on that port.

Verify the ATA 190 Startup Process

After the ATA 190 has power connected to it, it begins its startup process by cycling through these steps:

1. The Power LED is on.
2. The Network LED is flashing (when there is data traffic on Network port connected to a WAN port).
The ATA 190 is launching its application in this step.
3. Network LED is on.
4. After the Phone1 and Phone2 register with Cisco Unified CM successfully, the corresponding LEDs are on.
5. All the LEDs are on.

If the ATA 190 flash memory is erased or the load is corrupted, the ATA enters a recovery mode where it can restore the image by manual upgrading.

When you go offhook on the phone, you see the phone LED begin to flash, and you hear dial tone. The ATA 190 has completed the startup process.

Configure Startup Network Settings

If you are not using DHCP in your network, configure these network settings on the ATA 190 after installing the device on the network:

- IP subnet information (subnet mask and gateway)
- TFTP server IP address

You may also configure these optional settings as necessary:

Administration VLAN ID

Collect this information and see instructions in [Appendix B, “Voice Menu Codes.”](#)

Configure Security on the ATA 190

The security features protect against several threats, including threats to the identity of the phone and to data. These features establish and maintain authenticated communication streams between the phone and the Cisco Unified Communications Manager server, and digitally sign files before they are delivered.

For more information about the security features, see the *Cisco Unified Communications Manager Security Guide*.

You can start the installation of a Locally Significant Certificate (LSC) from the Security Configuration menu on the phone. This menu also lets you update or remove an LSC.

You can start the installation of a Locally Significant Certificate (LSC) on the device profile from Cisco Unified Communications Manager. Use the **Device > Phone > Phone Configuration** menu option. This menu option also lets you update or remove an LSC.

Before you begin, make sure that the appropriate Cisco Unified Communications Manager and the CAPF security configurations are complete:

- On Cisco Unified Communications Operating System Administration, verify that the CAPF certificate has been installed
- The CAPF is running and configured

See the *Cisco Unified Communications Manager Security Guide* for more information.



Note

If you want to update LSC, use *reset to factory default* from [Appendix B, “Voice Menu Codes.”](#)



Configuring the ATA 190

Use Cisco Unified Communications Manager Administration to configure telephony features and assign users.

This chapter provides an overview of these configuration and setup procedures. Cisco Unified Communications Manager documentation provides detailed instructions for these procedures.

Telephony Features Available for the ATA 190

[Table 4-1](#) lists the supported telephony features, many of which you configure by using Cisco Unified Communications Manager Administration.

Table 4-1 **Telephony Features for the ATA 190**

Feature	Description	Configuration Reference
Audible Message Waiting Indicator	<p>A stutter tone from the handset or speakerphone indicates that a user has one or more new voice messages on a line.</p> <p>Note The stutter tone is line-specific. You hear it only when using the line with the waiting messages.</p>	<p>For more information, refer to:</p> <ul style="list-style-type: none">• <i>Cisco Unified Communications Manager Administration Guide</i>, Message Waiting Configuration.• <i>Cisco Unified Communications Manager System Guide</i>, Voice Mail Connectivity to Cisco Unified Communications Manager.
cBarge	<p>Allows a user to join a nonprivate call on a shared phone line. cBarge adds a user to a call and converts it into a conference, allowing the user and other parties to access conference features.</p> <p>The ATA 190 supports Barge on a conference bridge.</p>	<p>For more information, refer to:</p> <ul style="list-style-type: none">• <i>Cisco Unified Communications Manager Administration Guide</i>, Cisco Unified IP Phone Configuration.• <i>Cisco Unified Communications Manager System Guide</i>, Cisco Unified IP Phones.• <i>Cisco Unified Communications Manager Features and Services Guide</i>, Barge and Privacy.

Table 4-1 **Telephony Features for the ATA 190 (continued)**


Feature	Description	Configuration Reference
Call forward	Allows users to redirect incoming calls to another number. Call forward options include Call Forward All, Call Forward Busy, and Call Forward No Answer.	For more information, refer to: <ul style="list-style-type: none"> • <i>Cisco Unified Communications Manager Administration Guide</i>, Directory Number Configuration. • <i>Cisco Unified Communications Manager System Guide</i>, Cisco Unified IP Phones.
Call pickup	Allows users to redirect a call that is ringing on another phone within their pickup group to their phone.	For more information, refer to the <i>Cisco Unified Communications Manager Features and Services Guide</i> , Call Pickup .
Call waiting	Indicates (and allows users to answer) an incoming call that rings while on another call. Displays incoming call information on the phone screen.	For more information, refer to the <i>Cisco Unified Communications System Guide</i> , Understanding Directory Numbers .
Caller ID	Displays caller identification such as a phone number, name, or other descriptive text on the phone screen.	For more information, refer to: <ul style="list-style-type: none"> • <i>Cisco Unified Communications Manager Administration Guide</i>, Cisco Unified IP Phone Configurations. • <i>Cisco Unified Communications Manager System Guide</i>, Understanding Route Plans. • <i>Cisco Unified Communications Manager Features and Services Guide</i>, Call Display Restrictions. • <i>Cisco Unified Communications Manager Administration Guide</i>, Directory Number Configuration.
Conference	<ul style="list-style-type: none"> • Allows a user to talk simultaneously with multiple parties by calling each participant individually. Conference features include Adhoc Conference, cBarge, and Meet-Me. • Allows a noninitiator in a standard (ad hoc) conference to add or remove participants. 	For more information, refer to: <ul style="list-style-type: none"> • <i>Cisco Unified Communications Manager System Guide</i>, Cisco Unified IP Phones. <div>  <p>Note Be sure to inform your users whether these features are activated.</p> </div>
Direct transfer	Allows users to connect two calls to each other (without remaining on the line).	For more information, refer to the <i>Cisco Unified Communications Manager System Guide</i> , Cisco Unified IP Phones .

Table 4-1 **Telephony Features for the ATA 190 (continued)**

Feature	Description	Configuration Reference
Forced authorization codes (FAC)	Controls the types of calls that certain users can place.	For more information, refer to the <i>Cisco Unified Communications Manager Features and Services Guide</i> , Client Matter Codes and Forced Authorization Codes .
Group call pickup	Allows a user to answer a call that is ringing on a directory number in another group.	For more information, refer to the <i>Cisco Unified Communications Manager Features and Services Guide</i> , Call Pickup .
Hold/Resume	Allows the user to move a connected call between an active state and a held state. Note No support for resuming a call from a shared line party.	For more information, refer to: Requires no configuration, unless you want to use music on hold. See Music on hold in this table for information.
Meet-Me conference	Allows a user to host a Meet-Me conference in which other participants call a predetermined number at a scheduled time.	For more information, refer to <i>Cisco Unified Communications Manager Administration Guide</i> , Meet-Me Number/Pattern Configuration .
Message Waiting	Defines directory numbers for message-waiting on and message-waiting off indicator. A directly connected voice-messaging system uses the specified directory number to set or to clear a message-waiting indication for a particular Cisco Unified IP Phone.	For more information refer to: <ul style="list-style-type: none">• <i>Cisco Unified Communications Manager Administration Guide</i>, Message Waiting Configuration.• <i>Cisco Unified Communications Manager System Guide</i>, Voice Mail Connectivity to Cisco Unified Communications Manager.
Music on hold	Plays music while callers are on hold.	For more information, refer to <i>Cisco Unified Communications Manager Features and Services Guide</i> , Music On Hold .
Privacy	Prevents users who share a line from adding themselves to a call.	For more information refer to: <ul style="list-style-type: none">• <i>Cisco Unified Communications Manager Administration Guide</i>, Cisco Unified IP Phone Configuration.• <i>Cisco Unified Communications Manager System Guide</i>, Cisco Unified IP Phones.• <i>Cisco Unified Communications Manager Features and Services Guide</i> Barge and Privacy.
Redial	Allows users to call the most recently dialed phone number by pressing the *# feature code.	Requires no configuration.

Table 4-1 **Telephony Features for the ATA 190 (continued)**

Feature	Description	Configuration Reference
Shared line	Allows a user to have several devices that share the same phone number or allows a user to share a phone number with a coworker.	For more information, refer to the <i>Cisco Unified Communications Manager System Guide</i> , Understanding Directory Numbers .
Speed dialing	Allows users to speed dial a phone number by entering * and an assigned index code (1 to 199) on the phone keypad. Example: Press *199 to dial the phone number with index code 199. Users assign index codes on Line configuration from the Cisco Unified Communications Manager Device window.	For more information, refer to: <ul style="list-style-type: none"> • <i>Cisco Unified Communications Manager Administration Guide</i>, Cisco Unified IP Phone Configuration. • <i>Cisco Unified Communications Manager System Guide</i>, Cisco Unified IP Phones.
Time Zone Update	Updates the device with time zone changes.	For more information, refer to the <i>Cisco Unified Communications Manager Administration Guide</i> , Date/Time Group Configuration .
Voice-messaging system	Enables callers to leave messages if calls are unanswered.	For more information refer to: <ul style="list-style-type: none"> • <i>Cisco Unified Communications Manager Administration Guide</i>, Cisco Voice-Mail Port Configuration. • <i>Cisco Unified Communications Manager System Guide</i>, Voice Mail Connectivity to Cisco Unified Communications Manager.

Configuring Product Specific Configuration Parameters

Cisco Unified Communications Manager Administration allows you to set some product-specific configuration parameters for the ATA 190. [Table 4-2](#) lists the configuration windows and their paths to configure the parameters.

Table 4-2 **Configuration Information**

Configuration Window	Path
Phone Configuration window	Device > Phone ; Product Specific Configuration portion of window

[Table 4-3](#) lists the configuration parameters you can set using Cisco Unified Communications Manager Administration. You can set the configuration parameters using the configuration window in [Table 4-2](#).


Note

Some ATA 190 parameters are set from port 1 only. Setting the parameters from port 2 has no effect. Set the following parameters from port 1 only: IVR Password, CDP, Impedance, Input/Output Audio Level, Timers, and Call Sequence.

Table 4-3 Configuration Parameters for the ATA 190

Parameter	Description
Auto Barge	Auto Barge adds a user to an active call. An offhook phone automatically adds the user (initiator) to the shared line call (target), and the users currently on the call receive a tone (if configured). Barge supports conference bridges.
Cisco Discovery Protocol (CDP):	Enable or disable the CDP function of the ATA 190
Echo Cancellation	Enable or Disable the use of echo canceler.
Fax Mode	<p>The Cisco ATA 190 supports these fax modes:</p> <ul style="list-style-type: none"> • Fax Pass-Through—Allows fax and modem traffic to pass through a voice port via re-INVITE method (codec can be g711ulaw or g711alaw). • NSE Fax Pass-through g711ulaw—Allows fax traffic to pass through a voice port via NSE method by codec g711ulaw. • NSE Fax Pass-through g711alaw—Allows fax traffic to pass through a voice port via NSE method by codec g711alaw. • T.38 Fax Relay—Allows for a more robust protocol for fax transmission over packet networks.
Fax Error Correction Mode Override	<p>You can set the fax error correction mode override values to one of the following settings:</p> <ul style="list-style-type: none"> • Default • On • Off
FAX Disable ECAN	Set this parameter to yes to automatically disable Echo Canceler when FAX tone is detected.
Modem Line	If you set this parameter to yes , the call is treated as a modem call. The ATA 190 tunes VAD, Jitter buffer, and echo canceler automatically.
Fax T38 Return To Voice	Set this parameter yes if voice callback is needed after the T.38 fax is completed.
IVR Password	ATA 190 IVR password. The default specifies 24726.
Input Audio Level	Gain value of Network-to-Phone
Output Audio Level	Gain value of Phone-to-Network
Impedance	The ATA 190 provides multiple impedance values, such as 600 ohm for use in the United States.
Caller ID	<ul style="list-style-type: none"> • BT FSK • Bellcore FSK • ETSI FSK
Call Sequence	<ul style="list-style-type: none"> • Bellcore FSK • ETSI FSK

Table 4-3 Configuration Parameters for the ATA 190 (continued)

Parameter	Description
Offhook Validation Timer (50-1000ms)	Indicates the time to validate an offhook event. Note This parameter is reserved but does take effect now.
Onhook Validation Timer (50-1000ms)	Indicates the time to validate an onhook event
Hookflash Timer (100 to 1500 ms)	Indicates the time to validate a hookflash event
Onhook Delay Timer (0 to 155 ms)	Indicates the time to delay an onhook event. Note This parameter does not take effect.
Reorder Delay (0-30s)	Delay after far end hangs up before reorder tone is played.
RTP Packet Time (10-160ms)	Packet size in milliseconds for RTP.

**Note**

User can access the ATA 190 web page and perform limited configuration. Two access modes exist: Admin mode and User mode.

- The default username/password for Admin mode is **admin/admin**.
- The default username/password for User mode is **cisco/cisco**.

In Admin mode, most information and settings are available.

A user can modify the username/password on the **Administration > Management > User List** page.

In User mode, only limited information and access are available.

Adding Users to Cisco Unified Communications Manager

Adding users to Cisco Unified Communications Manager allows you to display and maintain information about users and allows each user to perform these tasks:

- Access the corporate directory and other customized directories from an ATA 190.
- Create a personal directory.
- Set up speed dial and call forwarding numbers.
- Subscribe to services that are accessible from an ATA 190.

You can add users to Cisco Unified Communications Manager using this method:

- To add users individually, choose **User Management > End User** from Cisco Unified Communications Manager Administration.

Refer to the *Cisco Unified Communications Manager Administration Guide* for more information about adding users. Refer to the *Cisco Unified Communications Manager System Guide* for details about user information.

Second Line Can Be Disabled

Starting with version 1.2.1, the user can choose to disable the second port on a Cisco ATA 190 box. By doing so, the customer can save on the cost of a license.

Users can simply enable or disable the second port on the ATA 190 web page.

After an enable or disable operation, the ATA 190 resets so that the change can take effect.

When disabled, the ATA 190 does not retrieve the configuration file from the TFTP server for this port.

Configuration

1. On the Cisco ATA 190 web page, go to **Voice > Line 2 > Line Enable**.
2. Choose **yes** to enable or **no** to disable.
3. Click **Submit**. Wait for 40 seconds for the change to take effect.



Configuring Fax Services

The ATA 190 provides two modes of fax services that are capable of internetworking with Cisco IOS gateways over IP networks. These modes are called *fax pass-through mode* and *T.38 fax relay mode*.

With *fax pass-through mode*, the ATA 190 encodes fax traffic within the G.711 voice codec and passes it through the Voice Over IP (VoIP) network as though the fax were a voice call.

With *T.38 fax relay mode*, the ATA 190 supports the transmission of faxes, in real time, between two standard fax terminals communicating over SIP networks. T.38 fax relay mode provides a more reliable and error-free method of sending faxes over an IP network

Using Fax Mode

You can choose the preferred fax mode on the phone configuration page of the Unified CM administration page. From the fax mode pull-down window, choose one of the following modes:

- Fax pass-through
- T.38 fax relay
- NSE Fax pass-through—g711ulaw
- NSE Fax pass-through—g711alaw

You can set the Fax Error correction mode override values. From the fax mode pull-down window, choose one of the following modes:

- On
- Off
- Default

Fax Modem Standards

The ATA 190 supports the following fax modem standards:

- ITU-T V.34
- ITU-T V.34 Annex 12
- K56flex
- V.21
- V.22

- V.23
- V.32
- V.32bis
- V.44
- V.90
- V.92

**Note**

V.34 is not supported for T.38 relay fax.

Fax Modem Speeds

The ATA 190 supports the following fax modem speeds:

- 33.6 kb/s
- 31.2 kb/s
- 28.8 kb/s
- 26.4 kb/s
- 24 kb/s
- 21.6 kb/s
- 19.2 kb/s
- 16.8 kb/s
- 14.4 kb/s
- 12 kb/s
- 9.6 kb/s
- 7.2 kb/s
- 4.8 kb/s
- 2.4 kb/s

**Note**

The speeds that are only used in V.34 do not apply for fax using T.38 relay.



Using SIP Supplementary Services

SIP supplementary services are services that you can use to enhance your telephone service. These services include call forward, redial, call forwarding, and conference calling.

Common Supplementary Services

The supplementary services described in this section, and their configuration and implementation, depend on the system of the country in which the service is activated. For information about your country's implementation of services, contact your local Cisco equipment provider.

This section contains the following topics:

- [Attended Transfer, page 6-2](#)
- [Call Pickup, page 6-2](#)
- [Caller ID, page 6-2](#)
- [Call-Waiting Caller ID, page 6-2](#)
- [Call Hold, page 6-2](#)
- [Group Call Pickup, page 6-3](#)
- [Meet-Me Conference, page 6-3](#)
- [Privacy, page 6-3](#)
- [Shared Line, page 6-3](#)
- [Speed Dial, page 6-4](#)
- [Redial, page 6-4](#)
- [Unattended Transfer, page 6-4](#)
- [Voice Mail Indication, page 6-5](#)
- [Voice-Messaging System, page 6-5](#)
- [Making a Conference Call in the United States, page 6-5](#)
- [Making a Conference Call in Sweden, page 6-5](#)
- [Call Waiting in the United States, page 6-6](#)
- [Call Waiting in Sweden, page 6-6](#)
- [About Call Forwarding, page 6-6](#)

- [Call Forward All in US Mode, page 6-6](#)
- [Call Forward All in Sweden Mode, page 6-7](#)

Attended Transfer

This feature allows a user to transfer an existing call to another phone number after first consulting with the dialed party before the user hangs up. Perform the following steps to complete an attended transfer:

Procedure

-
- | | |
|---------------|---|
| Step 1 | Press the flash button on the analog phone to put the existing party on hold and get a dial tone. |
| Step 2 | Dial the phone number to which the existing party is being transferred. |
| Step 3 | When the callee answers the phone, you may consult with the callee and then transfer the existing party by hanging up your phone. |
-

Call Pickup

Allows you to answer a call that is ringing on another phone within your call pickup group. Perform the following steps to use the call pickup feature:

Procedure

-
- | | |
|---------------|----------------------------|
| Step 1 | Pick up the phone handset. |
| Step 2 | Press **3 . |
-

Caller ID

When the phone rings, the ATA 190 sends a Caller ID signal to the phone between the first and second ring (with name, phone number, time, and date information, if these are available).

Call-Waiting Caller ID

The ATA 190 plays a call waiting tone, then sends an off-hook Caller ID signal to the phone immediately after the first tone burst.

The ATA 190 sends the name, phone number, time, and date information, if these are available.

Call Hold

This feature allows the user to place an active state in a held state.

Group Call Pickup

Allows you to answer a call on a phone that is outside your call pickup group by:

- Using a group pickup number
- Dialing the ringing phone's group number

Perform the following steps to use the group call pickup feature:

Procedure

Step 1 Pick up the phone handset.

Step 2 Press ****4** > *group ID* > #.

Meet-Me Conference

This feature allows a user to host a Meet-Me conference in which other participants call a predetermined number at a scheduled time. Perform the following steps to complete a meet-me conference:

Procedure

Step 1 Pick up the phone handset.

Step 2 Press ****5** > *room ID* > #.

Step 3 Other parties can join a Meet-Me conference by dialing the meeting number directly.

Privacy

This feature prevents users who share a line from adding themselves to a call. Perform the following steps to enable or disable the privacy feature:

Procedure

Step 1 Pick up the phone handset.

Step 2 During an active call, press ****8** to enable the privacy feature.

Step 3 During an active call, press ****9** to disable the privacy feature.

Shared Line

This feature allows a user to have multiple phones that share the same phone number or allows a user to share a phone number with a coworker.

The feature enables the phone lines to barge into an existing call:

- If auto barge is enabled, off hook triggers C-barge.
- If auto barge is disabled, pressing "***6" triggers C-barge.

Speed Dial

This feature allows users to speed dial a phone number by entering an assigned index code (from *1 to *199) on the phone keypad.

Redial

Allows users to call the most recently dialed phone number by pressing the *# buttons.

Unattended Transfer

This feature allows a user to transfer an existing call to another phone number without waiting for the dialed party to answer before the user hangs up. Two methods exist for performing an unattended transfer:

- [Semi-unattended Transfer, page 6-4](#)
- [Fully Unattended Transfer, page 6-4](#)

Semi-unattended Transfer

Perform the following steps to complete a semi-unattended transfer:

Procedure

-
- | | |
|---------------|---|
| Step 1 | Press the flash button on the analog phone to put the other party on hold and get a second dial tone. |
| Step 2 | Dial the phone number to which you want to transfer the other party. |
| Step 3 | Wait for at least one ringback tone, then hang up your phone to transfer the other party. |
-

Fully Unattended Transfer

Perform the following steps to complete a fully unattended transfer:

Procedure

-
- | | |
|---------------|--|
| Step 1 | Press the flash button on the analog phone to put the other party on hold and get a second dial tone. |
| Step 2 | Press #90 (the transfer service activation code) on your phone keypad, then enter the phone number to which you want to transfer the other party, then press # . |

- Step 3** Hang up your phone.
-

Voice Mail Indication

This feature allows the ATA 190 to play an intermittent dial tone if there is a new message in the user's voice mail box.

Voice-Messaging System

This feature enables callers to leave messages if calls are unanswered or the recipient to access voice messages. Perform the following steps to access the voice-messaging system:

Procedure

- Step 1** Pick up the phone handset.
- Step 2** Press *0.
-

Making a Conference Call in the United States

Procedure

- Step 1** Dial the first number.
- Step 2** When the person you called answers, perform hook flash on the phone handset. This action puts the first person you called on hold and you receive a second dial tone.
- Step 3** Dial the second person and speak normally when that person answers.
- Step 4** To conference with both persons at the same time, perform a hook flash.
-

Making a Conference Call in Sweden

Procedure

- Step 1** Dial the first number.
- Step 2** When the person you called answers, perform hook flash on the phone handset. This action puts the first person you called on hold and a dial tone sounds.
- Step 3** Dial the second person and speak normally when that person answers.
- Step 4** Perform a hook flash, then press **2** on your phone keypad to return to the first person. You can continue to switch back and forth between the two callers.

- Step 5** To conference with both callers at the same time, perform a hook flash, then press **3** on the phone keypad. Once you conference all three callers, the only way to drop a caller is for that caller to hang up.
-

Call Waiting in the United States

If someone calls you while you are speaking on the phone, you can answer by performing a hook flash. You cannot conference in all three callers.

Call Waiting in Sweden

If someone calls you while you are speaking on the phone, you can answer by performing a hook flash then pressing **2** on your phone keypad. You can also perform a hook flash then press **3** later during the call to create a conference call.

Performing a hook flash then pressing **1** hangs up the first caller and answers the second call. If there is no answer after one minute, the caller receives a busy signal.

To enable call waiting for Sweden, go to the ATA 190 configuration web page and change the Call Sequence to **ETSI FSK for Sweden**.

**Note**

In ETSI mode, the user must pick up the call waiting rather than start the conference service. The user cannot trigger the call conference service directly, when there is a call waiting.

About Call Forwarding

ATA 190 supports these call forwarding types: Call Forward All, Call Forward Busy, Call Forward No Answer.

This feature is implemented by configuration in Cisco Unified Communications Manager. The administrator can configure the forward destination in Cisco Unified CM to specify the type and destination of call forwarding. Then, Cisco Unified CM forwards incoming calls without delivering SIP INVITE messages to the configured line.

ATA 190 can dial a feature code followed by the call forward destination to activate/deactivate call forward all.

Call Forward All in US Mode

Forward Unconditional

Press **#72** on your phone keypad; enter the number you want to forward call to; then press **#** again.

Canceling Call Forwarding

To cancel call forwarding, press **#73** on your phone keypad

Call Forward All in Sweden Mode

Forward Unconditional

Press ***21*** on your phone keypad; enter the number you want to forward calls to; then press #. To cancel, press **#21#**.



ATA 190 Specifications

Physical Specifications

Table A-1 **Physical Specifications**

Description	Specification
Regulatory compliance	FCC (Part 15 Class B), CE, ICES-003, A-Tick certification, Restriction of Hazardous Substances (RoHS), and UL
Power supply	DC input voltage: 5V DC at 2.0A maximum power consumption: 5W Switching type (100-240V): Automatic Power adapter: 100-240V and 50-60 Hz (26-34 VA) input with 1.8m cord
Indicator lights and LEDs	Phone 1, phone 2, network, and power
Documentation	Quick Start Guide Administration Guide (online) Provisioning Guide (online)
Dimensions (W x H x D)	3.98 x 3.98 x 1.10 in. (101 x 101 x 28mm)
Unit weight	5.40 oz (153 g)

Electrical Specifications

Table A-2 *Electrical Specifications*

Description	Specification
Power	0.25 to 12W (idle to peak)
DC input voltage	5.0 VDC at 2.0A maximum
Power adapter	Universal AC/DC ~4.05 x 1.93 x 1.31 in. (~10.3 x 4.9 x 3.35 cm) ~4.23 oz (120 g) for the AC-input external power adapter ~4.9 ft (1.5 m) DC cord 6 ft (1.8 m) cord UL/cUL, CE approved Class I adapter

Environmental Specifications

Table A-3 *Environmental Specifications*

Description	Specification
Operating temperature	32 to 113° F (0 to 45° C)
Nonoperating temperature	–13 to 158° F (–25 to 70° C)
Operating humidity	10% to 90% noncondensing
Storage humidity	10% to 90% noncondensing

Physical Interfaces

Table A-4 *Physical Interfaces*

Description	Specification
Ethernet	One RJ-45 connector, IEEE 802.3 100BaseT standard
Analog phone	Two RJ-11 FXS voice ports
Power	5 VDC power connector

Ringing Characteristics

Table A-5 *Ringing Characteristics*

Description	Specification
Tip/ring interfaces for each RJ-11 FXS port (SLIC)	
Ring voltage	70V _{RMS} (typical, balanced ringing only)
Ring frequency	20 Hz
Ring waveform	Trapezoidal with 1.2 to 1.6 crest factor
Ring load	1400 ohm + 40μF
Ringer equivalence number (REN)	Up to 5 REN per RJ-11 FXS port
Loop impedance	Up to 200 ohms (plus 430-ohm maximum phone DC resistance)
On-hook/off-hook characteristics	
On-hook voltage (tip/ring)	−47V
Off-hook current	24 mA (nominal)
RJ-11 FXS port terminating impedance option	The ATA 190-I1 provides multiple impedance, such as 600 ohm for American SKU, 900 ohm for European SKU, 220 ohm (820 ohm 120nF) for Australian SKU, and so on.

Software Specifications

Table A-6 *Software Specifications (All Protocols)*

Description	Specification
Call progress tones	Configurable for two sets of frequencies and single set of on/off cadence
Dual-tone multifrequency (DTMF)	DTMF tone detection and generation
Fax	Fax pass-through and T.38 fax relay mode. V34 fax is supported for pass-through mode. Success of fax transmissions up to 33.6 kb/s depends on network conditions, and fax modem/fax machine tolerance to those conditions. The network must have reasonably low network jitter, network delay, and packet-loss rate.

Table A-6 Software Specifications (All Protocols) (continued)

Description	Specification
Line-echo cancellation	<ul style="list-style-type: none"> Echo canceler for each port 8 ms echo length Nonlinear echo suppression (ERL > 28 dB for frequency = 300 to 2400 Hz) Convergence time = 250 ms ERLE = 10 to 20 dB Double-talk detection
Out-of-band DTMF	RFC 2833 AVT tones for SIP Note Cannot transmit RFC 2833 and in-band signaling, simultaneously.
Configuration	<ul style="list-style-type: none"> DHCP (RFC 2131) Web configuration via built-in web server Basic boot configuration (RFC 1350 TFTP Profiling) Dial plan configuration Cisco Discovery Protocol
Quality of Service	<ul style="list-style-type: none"> Class-of-service (CoS) bit-tagging (802.1P) Type-of-service (ToS) bit-tagging
Security	Encryption for TFTP configuration files
Voice coder-decoders (codecs)	<ul style="list-style-type: none"> G.729A, G.729AB G.711A-law G.711μ-law
Voice features	<ul style="list-style-type: none"> Voice activity detection (VAD) Comfort noise generation (CNG) Dynamic jitter buffer (adaptive)
Voice-over-IP (VoIP) protocols	SIP (RFC 3261)

SIP Compliance Reference Information

Information on how the ATA 190 complies with the IETF definition of SIP as described in RFC 2543 is found at the following URL:

<http://www.ietf.org/rfc/rfc2543.txt>



Voice Menu Codes

This section contains information on accessing the Interactive Voice Response (IVR) and a quick-reference list of the voice configuration menu options for the ATA 190.

Accessing the IVR and Configuring Your Phone Setting

**Note**

- After setting the IP mode to static IP, user must go on-hook to make it effective. Then, the IP address, subnet mask, and default gateway can be configured
 - To set static IP, network cable must be connected.
-

To access the IVR and configure your phone settings, follow these steps:

**Note**

You can change the IVR password on the Device window of Cisco Unified Communications Manager.

Procedure

Step 1 To access the IVR, go off-hook on the phone connected to PHONE1/PHONE2.

Step 2 Press ***** from the phone keypad.

The IVR prompts for a password.

**Note**

The ATA 190 allows you to enter only numerical values for the password. The default password is 24726.

Step 3 Enter the IVR password by pressing the number keypad, followed by #.

You are at the IVR main configuration menu.

Step 4 Follow the voice prompts on the IVR. See [Table B-1](#) for information on navigating the IVR.

Step 5 To return to the main configuration menu, press *.

Step 6 To exit the IVR, end the call.

Table B-1 describes the various options in the IVR Configuration Menu.

Table B-1 Navigating the IVR Configuration Menu

Action	IVR Code	Navigating Notes
Show IP address	110	
Configure IP address	111	Available in static IP mode only.
Show subnet mask	120	
Configure subnet mask	121	Available in static IP mode only.
Show default gateway	130	
Configure default gateway	131	Available in static IP mode only.
Show TFTP server address	220	
Configure TFTP server address	221	
Show VLAN	230	
Configure VLAN	231	To disable VLAN, set VLAN ID to 4095. To enable VLAN, set VLAN ID from 1 to 4094.
Factory Reset	73738	
Check Internet Addressing Method	100	Answers with 0, the default option (DHCP).
Set Internet Addressing Method	101	DHCP: 0; Static IP: 1.



ATA 190 Country-Specific Tones and Cadences

Mechanism

The administrator can upload an XML file named `g3-tones.xml` that describes the tones and cadences to the directory on the Cisco Unified Communications Manager TFTP server. The directory name is actually a locale name, such as **Australia**.

During provisioning, device knows the network locale setting and tries to download `[locale name]/g3-tones.xml` from the Cisco Unified CM TFTP server. For example, if network locale is set to **Australia**, the path is **Australia/g3-tones.xml**.

Link Tone File With Device

There are two methods to achieve this purpose:

1. In Cisco Unified Communications Manager, navigate to **System > Device Pool**, and set the Network Locale value to specify locale option.
2. In Cisco Unified Communications Manager, navigate to **Device > Phone**. On the device window, set the value of Network Locale, which overwrites the value that is set in method 1.



Note

For method 2, network locale from **Device > Phone** is not configurable currently because there are only two choices: none and United States. There is a known issue for Cisco Unified Communications Manager that countries other than the United States cannot be selected from this menu. Method 2 has a higher priority than method 1.

Notes for Tone Configuration

- Only ATA 190 Line1 network locale setting is applied. This means that the line2 network locale always applies the line1 option, even if the value of line2 network locale configured on the Cisco Unified Communications Manager web page differs from the line1 value.
- Only these tones can be configured:
 - Ringback tone
 - Reorder tone
 - Dialing tone

- Outside dialing tone
- Busy tone
- Call waiting tone

Any tone specification that appears in the tone profile but is either not supported or has invalid data fields (even if the tone is supported) is ignored.

- **Example:** A tone profile includes a valid reorder tone specification, an invalid busy tone specification (has invalid data fields) and a recording tone spec (not supported), only the reorder tone spec would be applied.
- The name of the XML that describes tones and cadences can only be **g3-tones.xml** now. (New implementations may be added to try other files with different names.)
- Each tone can specify at most four c/i pairs (about frequency and gain) and four cadence segments (each is an on/off pair). Any additional data gets discarded.



Troubleshooting and Maintenance

This chapter provides information that can assist you in troubleshooting problems with your ATA 190 or with your IP telephony network. It also explains how to clean and maintain your phone.

Resolving Startup Problems

After installing an ATA 190 into your network and adding it to Cisco Unified Communications Manager, the phone starts up as described in the [“Install the ATA 190” section on page 3-3](#). If the phone does not start up properly, see the following sections for troubleshooting information:

- [Symptom: The ATA 190 Does Not Go Through its Normal Startup Process, page D-1](#)
- [Symptom: The ATA 190 Does Not Register with Cisco Unified Communications Manager, page D-2](#)
- [Symptom: ATA 190 Unable to Obtain IP Address, page D-4](#)

Symptom: The ATA 190 Does Not Go Through its Normal Startup Process

When you connect a phone in the network port, the phone goes through its normal startup process as described in the [“Verify the ATA 190 Startup Process” section on page 3-3](#). If the phone does not go through the startup process, the cause may be faulty cables, bad connections, network outages, lack of power, and so on. Or, the phone may not be functional.

To determine whether the phone is functional, follow these suggestions to systematically eliminate these other potential problems:

1. Verify that the network port is functional:
 - Exchange the Ethernet cables with cables that you know are functional.
 - Disconnect a functioning phone from another port and connect it to this network port to verify that the port is active.
 - Connect the phone that does not start up to a different network port that is known to be good.
 - Connect the phone that will not start up directly to the port on the switch, eliminating the patch panel connection in the office.
2. Verify that the phone is receiving power:
 - If you are using external power, verify that the electrical outlet is functional.
 - If you are using the external power supply, switch with a unit that you know to be functional.

3. If the phone still does not start up properly, perform a factory reset of the phone.

Symptom: The ATA 190 Does Not Register with Cisco Unified Communications Manager

If the phone proceeds past the first stage of the startup process, with the LED buttons flashing on and off, but continues to cycle through the messages, the phone is not starting up properly. The phone cannot successfully start up unless it is connected to the Ethernet network and it has registered with a Cisco Unified Communications Manager server.

These sections can assist you in determining the reason the phone is unable to start up properly:

- [Registering the Phone with Cisco Unified Communications Manager, page D-4](#)
- [Checking Network Connectivity, page D-2](#)
- [Verifying TFTP Server Settings, page D-2](#)
- [Verifying DNS Settings, page D-2](#)
- [Verifying Cisco Unified Communications Manager Settings, page D-3](#)
- [Cisco Unified Communications Manager and TFTP Services Are Not Running, page D-3](#)
- [Creating a New Configuration File, page D-3](#)
- [Registering the Phone with Cisco Unified Communications Manager, page D-4](#)

Checking Network Connectivity

If the network is down between the phone and the TFTP server or Cisco Unified Communications Manager, the phone cannot start up properly. Ensure that the network is currently running.

Verifying TFTP Server Settings

You can determine the IP address of the TFTP server used by the ATA 190 by entering **http://x.x.x.x** where x.x.x.x is the IP address of the ATA 190.

If you have assigned a static IP address to the phone, manually enter a setting for the TFTP Server 1 option. See the [“Accessing the IVR and Configuring Your Phone Setting”](#) section on page B-1.

If you are using DHCP, the phone obtains the address for the TFTP server from the DHCP server. Check the IP address that is configured in Option 150 or Option 66.

You can also enable the phone to use an alternate TFTP server. Such a setting is very useful if the phone was recently moved from one location to another. See the [“Accessing the IVR and Configuring Your Phone Setting”](#) section on page B-1 for instructions.

Verifying DNS Settings

If you are using DNS to refer to the TFTP server or to Cisco Unified Communications Manager, ensure that you have specified a DNS server. Verify this setting by entering **http://x.x.x.x** where x.x.x.x is the IP address of the ATA 190. Also, verify that there is an A entry in the DNS server for the TFTP server and for the Cisco Unified Communications Manager system.

Also, ensure that DNS is configured to do reverse look-ups.

Verifying Cisco Unified Communications Manager Settings

Enter `http://x.x.x.x` where `x.x.x.x` is the IP address of the ATA 190 to find the active Cisco Unified Communications Manager settings.

Cisco Unified Communications Manager and TFTP Services Are Not Running

If the Cisco Unified Communications Manager or TFTP services are not running, phones may not be able to start up properly. In such a situation, it is likely that you are experiencing a system-wide failure, and that other phones and devices are unable to start up properly.

If the Cisco Unified Communications Manager service is not running, all devices on the network that rely on it to make phone calls are affected. If the TFTP service is not running, many devices cannot start up successfully.

To start a service, follow these steps:

Procedure

-
- Step 1** From Cisco Unified Communications Manager Administration, choose **Cisco Unified Serviceability** from the Navigation drop-down list.
 - Step 2** Choose **Tools > Control Center - Network Services**.
 - Step 3** Choose the primary Cisco Unified Communications Manager server from the Server drop-down list.
The window displays the service names for the server that you chose, the status of the services, and a service control panel to stop or start a service.
 - Step 4** If a service has stopped, click its radio button and then click the **Start** button.
The Service Status symbol changes from a square to an arrow.
-



Note

A service must be activated before it can be started or stopped. To activate a service, choose **Tools > Service Activation**.

Creating a New Configuration File

If you continue to have problems with a particular phone that other suggestions in this chapter do not resolve, the configuration file may be corrupted.

To create a new configuration file, follow these steps:

Procedure

-
- Step 1** From Cisco Unified Communications Manager, choose **Device > Phone > Find** to locate the phone experiencing problems.
 - Step 2** Choose **Delete** to remove the phone from the Cisco Unified Communications Manager database.
 - Step 3** Add the phone back to the Cisco Unified Communications Manager database. See the [“Attach a Phone to the ATA 190”](#) section on page 3-3 for details.

Step 4 Power cycle the phone.

**Note**

When you remove a phone from the Cisco Unified Communications Manager database, its configuration file is deleted from the Cisco Unified Communications Manager TFTP server. The phone's directory number or numbers remain in the Cisco Unified Communications Manager database. They are called "unassigned DNs" and can be used for other devices. If unassigned DNs are not used by other devices, delete them from the Cisco Unified Communications Manager database. You can use the Route Plan Report to view and delete unassigned reference numbers. See the Cisco Unified Communications Manager Administration Guide for more information.

**Note**

Changing the buttons on a phone button template, or assigning a different phone button template to a phone, may result in directory numbers that are no longer accessible from the phone. The directory numbers are still assigned to the phone in the Cisco Unified Communications Manager database, but there is no button on the phone with which calls can be answered. These directory numbers should be removed from the phone and deleted if necessary.

Registering the Phone with Cisco Unified Communications Manager

A phone can register with a Cisco Unified Communications Manager server only if the phone has been added to the server or if autoregistration is enabled. Review the information and procedures in the ["Attach a Phone to the ATA 190" section on page 3-3](#) to ensure that the phone has been added to the Cisco Unified Communications Manager database.

To verify that the phone is in the Cisco Unified Communications Manager database, choose **Device > Phone > Find** from Cisco Unified Communications Manager Administration to search for the phone based on its MAC address. For information about determining a MAC address, see the ["Determining the MAC Address of an ATA 190" section on page 2-6](#).

If the phone is already in the Cisco Unified Communications Manager database, its configuration file may be damaged. See the ["Adding Users to Cisco Unified Communications Manager" section on page 4-6](#) for assistance.

Symptom: ATA 190 Unable to Obtain IP Address

If a phone is unable to obtain an IP address when it starts up, the phone may be not be on the same network or VLAN as the DHCP server, or the switch port to which the phone is connected may be disabled.

Ensure that the network or VLAN to which the phone is connected has access to the DHCP server, and make sure that the switch port is enabled.

ATA 190 Resets Unexpectedly

If users report that their phones are resetting during calls or while idle on their desk, investigate the cause. If the network connection and Cisco Unified Communications Manager connection are stable, an ATA 190 should not reset on its own.

Typically, a phone resets if it has problems connecting to the Ethernet network or to Cisco Unified Communications Manager. These sections can help you identify the cause of a phone resetting in your network:

- [Verifying Physical Connection, page D-5](#)
- [Identifying Intermittent Network Outages, page D-5](#)
- [Verifying DHCP Settings, page D-5](#)
- [Checking Static IP Address Settings, page D-6](#)
- [Verifying Voice VLAN Configuration, page D-6](#)
- [Eliminating DNS or Other Connectivity Errors, page D-6](#)

Verifying Physical Connection

Verify that the Ethernet connection to which the ATA 190 is connected is up. For example, check whether the particular port or switch to which the phone is connected is down and that the switch is not rebooting. Also make sure that there are no cable breaks.

Identifying Intermittent Network Outages

Intermittent network outages affect data and voice traffic differently. Your network might have been experiencing intermittent outages without detection. If so, data traffic can resend lost packets and verify that packets are received and transmitted. However, voice traffic cannot recapture lost packets. Rather than retransmitting a lost network connection, the phone resets and attempts to reconnect its network connection.

If you are experiencing problems with the voice network, investigate whether an existing problem is simply being exposed.

Verifying DHCP Settings

Follow this process to help determine if the phone has been properly configured to use DHCP:

Procedure

-
- | | |
|---------------|--|
| Step 1 | Verify that you have properly configured the phone to use DHCP. See the “Configure Startup Network Settings” section on page 3-4 for more information. |
| Step 2 | Verify that the DHCP server has been set up properly. |
| Step 3 | Verify the DHCP lease duration. Cisco recommends that you set it to 8 days. |
-

The ATA 190 sends the DHCP request message to update the IP Address at half of the lease time. If no response is received from the server, the ATA 190 starts the DHCP Discover process to get the new IP address.

Checking Static IP Address Settings

If the phone has been assigned a static IP address, verify that you have entered the correct settings. See the [“Understanding Phone Configuration Files” section on page 2-3](#) for more information.

Verifying Voice VLAN Configuration

If the ATA 190 appears to reset during heavy network usage (for example, following extensive web surfing on a computer connected to same switch as phone), it is likely that you do not have a voice VLAN configured.

Isolating the phones on a separate auxiliary VLAN increases the quality of the voice traffic.

Eliminating DNS or Other Connectivity Errors

If the phone continues to reset, follow these steps to eliminate DNS or other connectivity errors:

Procedure

-
- | | |
|---------------|--|
| Step 1 | Use the IVR to reset phone settings to their default values. See the “Accessing the IVR and Configuring Your Phone Setting” section on page B-1 for details. |
| Step 2 | Modify DHCP and IP settings: <ul style="list-style-type: none">a. Disable DHCP.b. Assign static IP values to the phone. See the “Understanding Phone Configuration Files” section on page 2-3 for instructions. Use the same default router setting used for other functioning ATA 190.c. Assign TFTP server. Use the same TFTP server used for other functioning ATA 190. |
| Step 3 | On the Cisco Unified Communications Manager server, verify that the local host files have the correct Cisco Unified Communications Manager server name mapped to the correct IP address. |
| Step 4 | From Cisco Unified Communications Manager, choose System > Server and verify that the server is referred to by its IP address and not by its DNS name. |
| Step 5 | From Cisco Unified Communications Manager, choose Device > Phone and verify that you have assigned the correct MAC address to this ATA 190. For information about determining a MAC address, see the “Determining the MAC Address of an ATA 190” section on page 2-6 . |
| Step 6 | Power cycle the phone. |
-

Troubleshooting ATA 190 Security

[Table D-1](#) provides troubleshooting information for the security features on the ATA 190. For information relating to the solutions for any of these issues, and for more troubleshooting information about security, see the *Cisco Unified Communications Manager Security Guide*.

Table D-1 ATA 190 Security Troubleshooting

Problem	Possible Cause
CTL File Problems	
Device authentication error.	CTL file does not have a Cisco Unified Communications Manager certificate or has an incorrect certificate.
Phone cannot authenticate CTL file.	The security token that signed the updated CTL file does not exist in the CTL file on the phone.
Phone cannot authenticate any of the configuration files other than CTL file.	The configuration file may not be signed by the corresponding certificate in the phone's Trust List.
Phone does not register with Cisco Unified Communications Manager.	The CTL file does not contain the correct information for the Cisco Unified Communications Manager server.
Phone does not request signed configuration files.	The CTL file does not contain any TFTP entries with certificates.

General Troubleshooting Tips

Table D-2 provides general troubleshooting information for the ATA 190.

Table D-2 ATA 190 Troubleshooting

Summary	Explanation
Poor quality when calling mobile phones using the G.729 protocol	In Cisco Unified Communications Manager, you can configure the network to use the G.729 protocol (the default is G.711). When using G.729, calls between a phone and a mobile phone have poor voice quality. Use G.729 only when absolutely necessary.
Prolonged broadcast storms cause phones to reset, or be unable to make or answer a call	A prolonged Layer 2 broadcast storm (lasting several minutes) on the voice VLAN may cause phones to reset, lose an active call, or be unable to initiate or answer a call. Phones may not come up until a broadcast storm ends.
Dual-Tone Multi-Frequency (DTMF) delay	When you are on a call that requires keypad input, if you press the keys too quickly, some of them may not be recognized.
Codec mismatch between the phone and another device	The RxType and the TxType statistics show the codec that is being used for a conversation between this ATA 190 and the other device. These values should match. If they do not, verify that the other device can handle the codec conversation or that a transcoder is in place to handle the service.
Sound sample mismatch between the phone and another device	The RxSize and the TxSize statistics show the size of the voice packets that are being used in a conversation between this ATA 190 and the other device. The values of these statistics should match.
Gaps in voice calls	Check the AvgJtr and the MaxJtr statistics. A large variance between these statistics may indicate a problem with jitter on the network or periodic high rates of network activity.

Table D-2 ATA 190 Troubleshooting (continued)

Summary	Explanation
One-way audio	When at least one person in a call does not receive audio, IP connectivity between phones is not established. Check the configurations in routers and switches to ensure that IP connectivity is properly configured.
Phone call cannot be established	<p>The phone does not have a DHCP IP address and is unable to register with Cisco Unified Communications Manager.</p> <p>Verify the following:</p> <ol style="list-style-type: none"> 1. The Ethernet cable is attached. 2. The Cisco CallManager service is running on the Cisco Unified Communications Manager server. 3. Both phones are registered to the same Cisco Unified Communications Manager.

Where to Go for More Troubleshooting Information

If you have more questions about troubleshooting the ATA 190, several Cisco.com websites can provide you with more tips. Choose from the sites available for your access level.

- ATA 190 Troubleshooting Resources:
<http://www.cisco.com/c/en/us/support/unified-communications/ata-190-series-analog-telephone-adapters/tsd-products-support-series-home.html>
- Cisco Products and Services (Technical Support and Documentation):
<http://www.cisco.com/cisco/web/support/index.html>

Cleaning the ATA 190

To clean your ATA 190, use a soft, dry cloth to wipe the surface. Do not apply liquids or powders directly on the device. As with all non-weather-proof electronics, liquids and powders can damage the components and cause failures.

Troubleshooting With Syslog

Before version 1.2.1 the ATA 190 could only send the voice syslog package to the network. To capture syslog, customer had to use Wireshark tool or Syslog tool.

Starting with version 1.2.1, Syslog-ng daemon or syslog tool is introduced. With syslog-ng, the user can capture both system log and voice log. Syslog-ng can direct the log to a local file or to the remote syslog server.

The local log file can be downloaded through a web page. The file makes troubleshooting at the customer side convenient.

Debug Options

Voice > System page

Debug Level

There are 7 levels: 0, 1, 2, 3, 3+Router, 3+Coma, 1+Coma.

Two modes are generally used: 3, 3+coma. Default value is 3.

- 3 – Only output voice log.
- 3+coma – Output both voice log and DSP log.

Syslog Output Method

Set syslog output mode:

- Direct – Output syslog to network directly. This method works as before and does not introduce any performance impact.
- Syslog-ng daemon – Output syslog to syslog-ng daemon that runs on the box.

Voice > Provisioning page

Debug Level

Control Cisco Unified Communications Manager related log output:

- none – No log output.
- info – Output log after network is ready.
- boot info – Before network is ready, output log to console or syslog-ng daemon. After network is ready, output log to network or syslog-ng daemon.

Administration > Log > Log Module page

- Status Enable/Disable – Enable/Disable syslog-ng daemon.
- Log Enable – Enable/Disable syslog output.
- Service
 - kernel – kernel syslog.
 - system – all other syslog except kernel.
- Priority – From high priority to low priority: Emerg, Alert, Critical, Error, Warning, Notice, Info, Debug.
- Local – Output log to local file.
- Syslog Server – Output log to remote syslog server.

Administration > Log > Log Setting page

- Local > Log Size – Set local log buffer size.
- Syslog Server – Set IP address, Port. When a new IP address is set, reboot the ATA 190 to make the IP address effective.

Administration > Log > Log Viewer page

- Download Log – Download log from box.
- Clear Log – Clear log on box.
- Display – Choose the type of message to display. After selection, message is displayed in this view.
- Filter – Set filter for displayed message.

Practice

Output Voice syslog to Network Directly

1. Set syslog server on page **Administration > Log > Log Setting**.
2. Set **Voice > Provisioning > Debug Level** as **boot info**.
3. Set **Voice > System > Debug Level** as **required**.
4. Set **Voice > System > Syslog Output Method** as **Direct**.

Capture Voice syslog Through Web Page

1. Enter page **Administration > Log > Log Module**.
Set system priority to **Debug**. Enable **Local**.
2. Set local log size on page **Administration > Log > Log Setting**.
3. Set **Voice > Provisioning > Debug Level** as **boot info**.
4. Set **Voice > System > Debug Level** as **required**.
5. Set **Voice > System > Syslog Output Method** as **Syslog-ng Daemon**.
6. Download log on page **Administration > Log > Log Viewer**.



Numerics

10BaseT 10-Mbps baseband Ethernet specification using two pairs of twisted-pair cabling (Categories 3, 4, or 5): one pair for sending data and the other for receiving data. 10BASET, which is part of the IEEE 802.3 specification, has a distance limit of approximately 328 feet (100 meters) per segment.

A

A-law ITU-T companding standard used in the conversion between analog and digital signals in PCM systems. A-law is used primarily in European phone networks and is similar to the North American μ -law standard. See also companding and μ -law.

AVT tones Out-of-bound signaling as defined in RFC 2833.

C

category-3 cable One of five grades of UTP cabling described in the EIA/TIA-586 standard. Category 3 cabling is used in 10BaseT networks and can send data at speeds up to 10 Mbps.

CED tone detection Called station identification. A three-second, 2100 Hz tone generated by a fax machine answering a call, which is used in the hand-shaking used to set the call; the response from a called fax machine to a CNG tone.

CELP code excited linear prediction compression. Compression algorithm used in low bit-rate voice encoding. Used in ITU-T Recommendations G.728, G.729, G.723.1.

CLIP Calling Line Identification Presentation. Shows your identity to callers with Caller ID.

CLIR Calling Line Identification Restriction. Hides your identity from callers with Caller ID.

CNG Comfort Noise Generation

codec coder decoder. In Voice over IP, Voice over Frame Relay, and Voice over ATM, a DSP software algorithm used to compress/decompress speech or audio signals.

companding Contraction derived from the opposite processes of compression and expansion. Part of the PCM process whereby analog signal values are rounded logically to discrete scale-step values on a nonlinear scale. The decimal step number then is coded in its binary equivalent prior to transmission. The process is reversed at the receiving terminal using the same nonlinear scale. Compare with compression and expansion. See also a-law and μ -law.

compression The running of a data set through an algorithm that reduces the space required to store or the bandwidth required to transmit the data set. Compare with companding and expansion.

CoS Class of service. An indication of how an upper-layer protocol requires a lower-layer protocol to treat its messages. In SNA subarea routing, CoS definitions are used by subarea nodes to determine the optimal route to establish a given session. A CoS definition comprises a virtual route number and a transmission priority field.

D

DHCP Dynamic Host Configuration Protocol. Provides a mechanism for allocating IP addresses dynamically so that addresses can be reused when hosts no longer need them.

dial peer An addressable call endpoint. In Voice over IP (VoIP), there are two types of dial peers: POTS and VoIP.

DNS Domain Name System. System used on the Internet for translating names of network nodes into addresses.

DSP digital signal processor. A DSP segments the voice signal into frames and stores them in voice packets.

DTMF dual tone multifrequency. Tones generated when a button is pressed on a phone, primarily used in the U.S. and Canada.

E

E.164 The international public telecommunications numbering plan. A standard set by the ITU-T which addresses phone numbers.

endpoint A SIP terminal or gateway. An endpoint can call and be called. It generates and/or terminates the information stream.

expansion The process of running a compressed data set through an algorithm that restores the data set to its original size. Compare with companding and compression.

F

firewall Router or access server, or several routers or access servers, designated as a buffer between any connected public networks and a private network. A firewall router uses access lists and other methods to ensure the security of the private network.

FoIP Fax over IP

FQDN Fully Qualified Domain (FQDN) format “mydomain.com” or “company.mydomain.com.”

FSK Frequency shift key

FXO Foreign Exchange Office. An FXO interface connects to the public switched phone network (PSTN) central office and is the interface offered on a standard phone. Cisco FXO interface is an RJ-11 connector that allows an analog connection at the PSTN central office or to a station interface on a PBX.

FXS Foreign Exchange Station. An FXS interface connects directly to a standard phone and supplies ring, voltage, and dial tone. Cisco's FXS interface is an RJ-11 connector that allows connections to basic phone service equipment, keysets, and PBXs.

G

G.711 Describes the 64-kbps PCM voice coding technique. In G.711, encoded voice is already in the correct format for digital voice delivery in the PSTN or through PBXs. Described in the ITU-T standard in its G-series recommendations.

G.723.1 Describes a compression technique that can be used for compressing speech or audio signal components at a very low bit rate as part of the H.324 family of standards. This Codec has two bit rates associated with it: 5.3 and 6.3 kbps. The higher bit rate is based on ML-MLQ technology and provides a somewhat higher quality of sound. The lower bit rate is based on CELP and provides system designers with more flexibility. Described in the ITU-T standard in its G-series recommendations.

G.729A Describes CELP compression where voice is coded into 8-kbps streams. There are two variations of this standard (G.729 and G.729 Annex A) that differ mainly in computational complexity; both provide speech quality similar to 32-kbps ADPCM. Described in the ITU-T standard in its G-series recommendations.

gateway A gateway allows SIP or H.323 terminals to communicate with terminals configured to other protocols by converting protocols. A gateway is the point where a circuit-switched call is encoded and repackaged into IP packets.

H

H.323 H.323 allows dissimilar communication devices to communicate with each other by using a standard communication protocol. H.323 defines a common set of CODECs, call setup and negotiating procedures, and basic data transport methods.

I

ICMP Internet Control Message Protocol

IP Internet Protocol. Network layer protocol in the TCP/IP stack offering a connectionless internetwork service. IP provides features for addressing, type-of-service specification, fragmentation and reassembly, and security. Defined in RFC 791.

IVR Interactive voice response. Term used to describe systems that provide information in the form of recorded messages over phone lines in response to user input in the form of spoken words or, more commonly, DTMF signaling.

L

LDAP Lightweight Directory Access Protocol

LEC local exchange carrier

Location Server A SIP redirect or proxy server uses a location server to get information about a caller's location. Location services are offered by location servers.

M

MWI message waiting indication

μ-law North American companding standard used in conversion between analog and digital signals in PCM systems. Similar to the European a-law. See also a-law and companding.

N

NAT Network Address Translation. Mechanism for reducing the need for globally unique IP addresses. NAT allows an organization with addresses that are not globally unique to connect to the Internet by translating those addresses into globally routable address spaces. Also known as Network Address Translator.

NSE packets Real-Time Transport Protocol (RTP) digit events are encoded using the Named Signaling Event (NSE) format specified in RFC 2833, Section 3.0.

NAT Server Network Address Translation. an Internet standard that enables a local-area network (LAN) to use one set of IP addresses for internal traffic and a second set of addresses for external traffic.

NTP Network Time Protocol. Protocol built on top of TCP that assures accurate local time-keeping with reference to radio and atomic clocks located on the Internet. This protocol is capable of synchronizing distributed clocks within milliseconds over long time periods.

P

POTS Plain old phone service. Basic phone service supplying standard single-line phones, phone lines, and access to the PSTN.

Proxy Server An intermediary program that acts as both a server and a client for making requests on behalf of other clients. Requests are serviced internally or by passing them on, possibly after translation, to other servers. A proxy interprets, and, if necessary, rewrites a request message before forwarding it.

PSTN Public switched phone network

Q

QoS Quality of Service. The capability of a network to provide better service to selected network traffic over various technologies, including Frame Relay, Asynchronous Transfer Mode (ATM), Ethernet and 802.1 networks, SONET, and IP-routed networks that may use any or all these underlying technologies. The primary goal of QoS is to provide priority including dedicated bandwidth, controlled jitter and latency (required by some real-time and interactive traffic), and improved loss characteristics.

R

Redirect Server A redirect server is a server that accepts a SIP request, maps the address into zero or more new addresses, and returns these addresses to the client. It does not initiate its own SIP request nor accept calls.

Registrar Server A registrar server is a server that accepts Register requests. A registrar is typically colocated with a proxy or redirect server and may offer location services.

router Network layer device that uses one or more metrics to determine the optimal path along which network traffic should be forwarded. Routers forward packets from one network to another based on network layer information. Occasionally called a gateway (although this definition of gateway is becoming increasingly outdated). Compare with gateway.

RTP Real-Time Transport Protocol. One of the IPv6 protocols. RTP is designed to provide end-to-end network transport functions for applications transmitting real-time data, such as audio, video, or simulation data, over multicast or unicast network services. RTP provides services such as payload type identification, sequence numbering, timestamping, and delivery monitoring to real-time applications.

S

SDP Session Definition Protocol. An IETF protocol for the definition of Multimedia Services. SDP messages can be part of SGCP and MGCP messages.

SIP Session Initiation Protocol. Protocol developed by the IETF MMUSIC Working Group as an alternative to H.323. SIP features are compliant with IETF RFC 2543, published in March 1999. SIP equips platforms to signal the setup of voice and multimedia calls over IP networks.

SIP endpoint A terminal or gateway that acts as a source or sink of Session Initiation Protocol (SIP) voice data. An endpoint can call or be called, and it generates or terminates the information stream.

SLIC Subscriber Line Interface Circuit. An integrated circuit (IC) providing central office-like phone interface functionality.

SOHO Small office, home office. Networking solutions and access technologies for offices that are not directly connected to large corporate networks.

T

T.38 T.38 is an ITU recommendation for allowing transmission of fax over IP networks in real time.

TCP Transmission Control Protocol. Connection-oriented transport layer protocol that provides reliable full-duplex data transmission. TCP is part of the TCP/IP protocol stack.

TFTP Trivial File Transfer Protocol. Simplified version of FTP that allows files to be transferred from one computer to another over a network, usually without the use of client authentication (for example, username and password).

TN power systems A TN power system is a power distribution system with one point connected directly to earth (ground). The exposed conductive parts of the installation are connected to that point by protective earth conductors.

TOS Type of service. See CoS.

U

UAC User agent client. A client application that initiates the SIP request.

UAS User agent server (or user agent). A server application that contacts the user when a SIP request is received, and then returns a response on behalf of the user. The response accepts, rejects, or redirects the request.

UDP User Datagram Protocol. Connectionless transport layer protocol in the TCP/IP protocol stack. UDP is a simple protocol that exchanges datagrams without acknowledgments or guaranteed delivery, requiring that error processing and retransmission be handled by other protocols. UDP is defined in RFC 768.

user agent See UAS.

V

VAD Voice activity detection. When enabled on a voice port or a dial peer, silence is not transmitted over the network, only audible speech. When VAD is enabled, the sound quality is slightly degraded but the connection monopolizes much less bandwidth.

voice packet gateway

Gateway platforms that enable Internet telephony service providers to offer residential and business-class services for Internet telephony.

VoIP

Voice over IP. The capability to carry normal telephony-style voice over an IP-based Internet with POTS-like functionality, reliability, and voice quality. VoIP enables a router to carry voice traffic (for example, phone calls and faxes) over an IP network. In VoIP, the DSP segments the voice signal into frames, which then are coupled in groups of two and stored in voice packets. VoIP is a blanket term, which generally refers to Cisco's standard-based (for example H.323) approach to IP voice traffic.

X**XML**

eXtensible Markup Language. Designed to enable the use of SGML on the World-Wide Web. XML allow you to define your own customized markup language.

