Cisco Analog Telephone Adaptor Overview

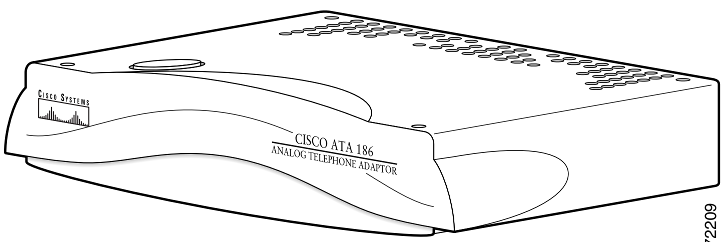
This section describes the hardware and software features of the Cisco Analog Telephone Adaptor (Cisco ATA) and includes a brief overview of the Session Initiation Protocol (SIP).

The Cisco ATA analog telephone adaptors are handset-to-Ethernet adaptors that allow regular analog telephones to operate on IP-based telephony networks. Cisco ATAs support two voice ports, each with an independent telephone number. The Cisco ATA 188 also has an RJ-45 10/100BASE-T data port.

This section covers the following topics:

- Session Initiation Protocol (SIP) Overview, page 1-2
- Hardware Overview, page 1-5
- Software Features, page 1-7
- Installation and Configuration Overview, page 1-10

Figure 1-1 Cisco ATA Analog Telephone Adaptor



The Cisco ATA, which operates with Cisco voice-packet gateways, makes use of broadband pipes that are deployed through a digital subscriber line (DSL), fixed wireless-cable modem, and other Ethernet connections.

Note The term *Cisco ATA* refers to both the Cisco ATA 186 and the Cisco ATA 188, unless otherwise stated.

Note This guide provides information about the SIP image for the Cisco ATA. The features and functionality described in this guide do not necessarily pertain to the features and functionality provided by the other protocol loads available for the Cisco ATA. Each protocol load has its own administrator's guide. If you are looking for information about the behavior of the Cisco ATA for a protocol other than SIP, please refer to the administration guide specific to that protocol.

Figure 1-2 Cisco ATA 186 as Endpoint in SIP Network

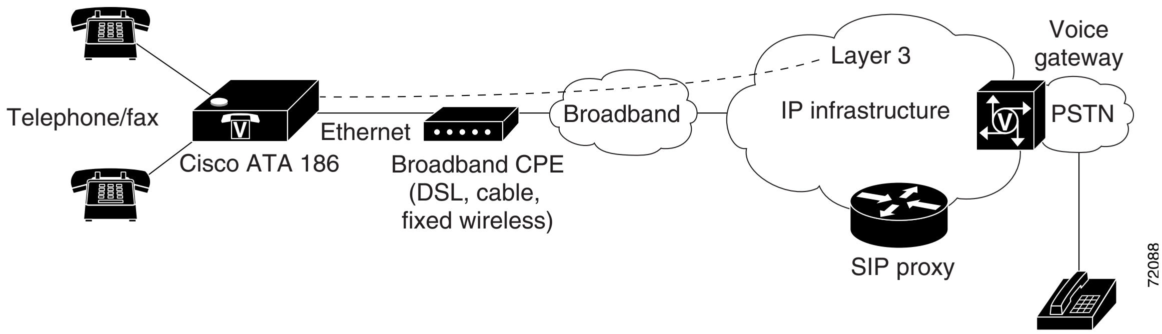
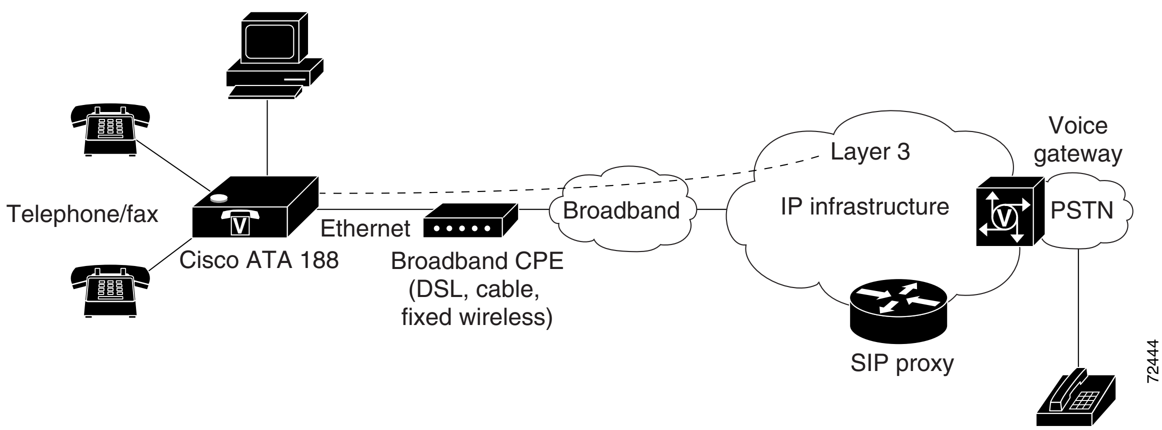


Figure 1-3 Cisco ATA 188 as Endpoint in SIP Network



Session Initiation Protocol (SIP) 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 Cisco ATA is compliant with RFC2543.

This section contains the following topics:

- SIP Capabilities, page 1-3
- Components of SIP, page 1-3

SIP Capabilities

SIP provides the following 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 are supported by all endpoints.
- 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 mid-call 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) and 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 the following roles:

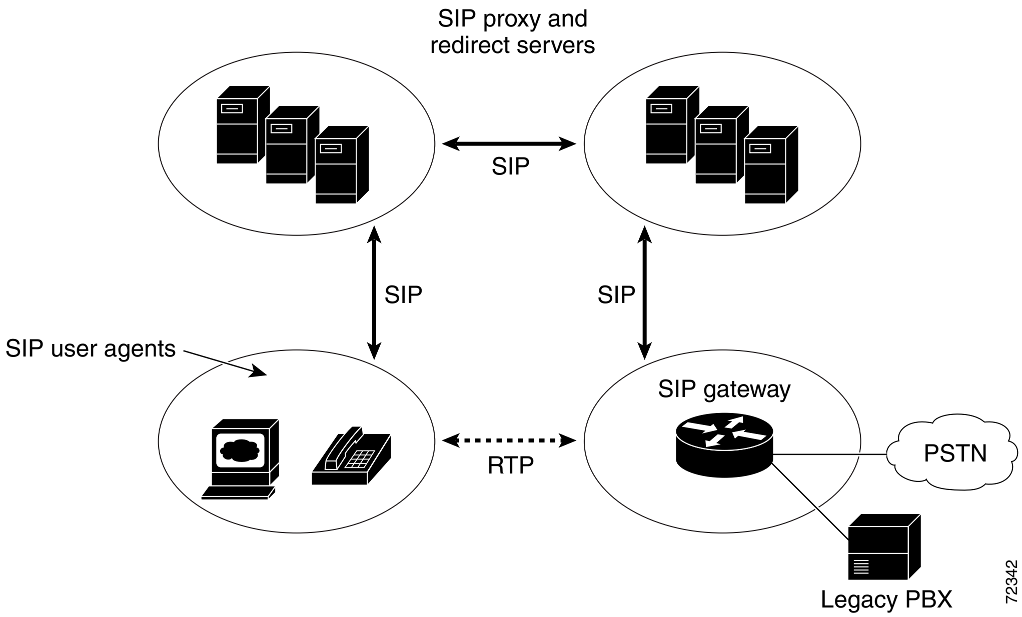
- User agent client (UAC)—A client application that initiates the SIP request.
- User agent server (UAS)—A server application that contacts the user when a SIP request is received and returns a response on behalf of the user.

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 1-4 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 1-4 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.
- Telephones—Can act as either a UAS or UAC. The Cisco ATA 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 co-located with a redirect or proxy server.

Hardware Overview

The Cisco ATA 186 and Cisco ATA 188 are compact, easy to install devices. Figure 1-5 shows the rear panel of the Cisco ATA 186. Figure 1-6 shows the rear panel of the Cisco ATA 188.

Figure 1-5 Cisco ATA 186—Rear View

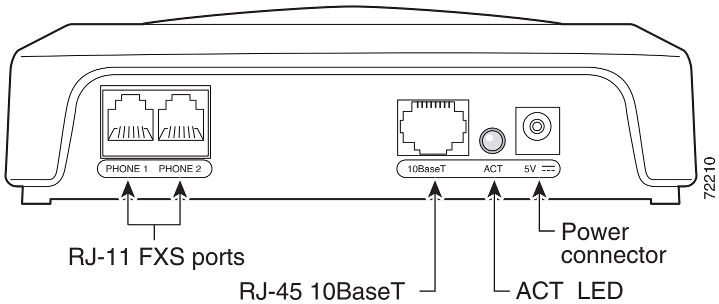
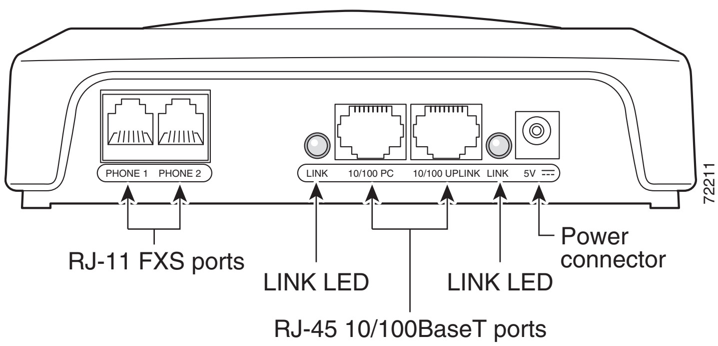


Figure 1-6 Cisco ATA 188—Rear View



The unit provides the following connectors and indicators:

- 5V power connector.
- Two RJ-11 FXS (Foreign Exchange Station) ports—The Cisco ATA supports two independent RJ-11 telephone ports that can connect to any standard analog telephone device. Each port supports either voice calls or fax sessions, and both ports can be used simultaneously.

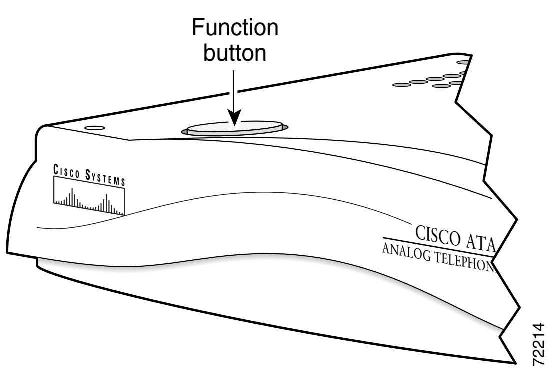
Note The Cisco ATA186-I1 and Cisco ATA188-I1 provide 600-ohm resistive impedance. The Cisco ATA186-I2 and Cisco ATA188-I2 provide 270 ohm + 750 ohm // 150-nF complex impedance. The impedance option is requested when you place your order and should match your specific application. If you are not sure of the applicable configuration, check your country or regional telephone impedance requirements.

- Ethernet ports
  - The Cisco ATA 186 has one RJ-45 10BASE-T uplink Ethernet port to connect the Cisco ATA 186 to a 10/100BASE-T hub or another Ethernet device.
  - The Cisco ATA 188 has two Ethernet ports: an RJ-45 10/100BASE-T uplink port to connect the Cisco ATA 188 to a 10/100BASE-T hub or another Ethernet device and an RJ-45 10/100BASE-T data port to connect an Ethernet-capable device, such as a computer, to the network.

Note The Cisco ATA 188 performs auto-negotiation for duplexity and speed and is capable of 10/100 Mbps, full-duplex operation. The Cisco ATA 186 is fixed at 10 Mbps, half-duplex operation.

- The Cisco ATA 188 RJ-45 LED shows network link and activity. The LED blinks twice when the Cisco ATA is first powered on, then turns off if there is no link or activity. The LED blinks to show network activity and is solid when there is a link.
- The Cisco ATA 186 RJ-45 LED is solid when the Cisco ATA is powered on and blinks to show network activity.
- Function button—The function button is located on the top panel of the unit (see Figure 1-7).

Figure 1-7 Function Button



The function button lights when you pick up the handset of a telephone attached to the Cisco ATA. The button blinks quickly when the Cisco ATA is upgrading its configuration.

Note If the function button blinks slowly, the Cisco ATA cannot find the DHCP server. Check your Ethernet connections and make sure the DHCP server is available.

Pressing the function button allows you to access to the voice configuration menu. For additional information about the voice configuration menu, see the "Voice Configuration Menu" section on page 3-20.

Caution Never press the function button during an upgrade process. Doing so may interfere with the process.

Software Features

The Cisco ATA supports the following protocols, services and methods:

Voice Codecs Supported

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

- G.711µ-law
- G.711A-law
- G.723.1
- G.726
- G.729
- G.729A
- G.729B
- G.729AB

Additional Supported Signaling Protocols

In addition to SIP, the Cisco ATA supports the following signaling protocols:

- H.323
- Skinny Client Control Protocol (SCCP)
- Media Gateway Control Protocol (MGCP)

If you wish to perform a cross-protocol upgrade from SIP to another signaling image, see the "Upgrading the Signaling Image from a TFTP Server" section on page 8-1.

Other Supported Protocols

Other protocols that the Cisco ATA supports include the following:

- 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)
- Real-Time Transport Protocol (RTP)
- Transmission Control Protocol (TCP)
- Trivial File Transfer Protocol (TFTP)
- User Datagram Protocol (UDP)

Cisco ATA SIP Services

For a list of required SIP parameters as well as descriptions of all supported Cisco ATA SIP services and cross references to the parameters for configuring these services, see Chapter 4, "Basic and Additional SIP Services."

These services include the following features:

- IP address assignment—DHCP-provided or statically configured
- Cisco ATA configuration by means of a TFTP server, web browser, or voice configuration menu
- VLAN configuration
- Cisco Discovery Protocol (CDP)
- Low-bit-rate codec selection
- User authentication
- Configurable tones (dial tone, busy tone, alert tone, reorder tone, call waiting tone)
- Dial plans
- Network Address Translation (NAT) Gateway
- NAT/Port Address Translation (PAT) translation

- SIP proxy server redundancy
- Outbound-proxy support
- SIP session-timer support
- Privacy features
- DNS SRV support
- User-configurable, call-waiting, permanent default setting
- Comfort noise during silence period when using G.711
- Advanced audio
- Billable features
- Caller ID format
- Ring cadence format
- Silence suppression
- Hook-flash detection timing configuration
- Configurable on-hook delay
- Type of Service (ToS) configuration for audio and signaling ethernet packets
- Debugging and diagnostic tools

Fax Services

The Cisco ATA supports two modes of fax services, in which fax signals are transmitted using the G.711 codec:

- Fax pass-through mode—Receiver-side Called Station Identification (CED) tone detection with automatic G.711A-law or G.711µ-law switching.
- Fax mode—The Cisco ATA is configured as a G.711-only device.

How you set Cisco ATA fax parameters depends on what network gateways are being used. You may need to modify the default fax parameter values (see Chapter 7, "Configuring and Debugging Fax Services").

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 Cisco ATA supports the methods listed below. For more information, refer to RFC3261 (*SIP: Session Initiation Protocol*).

- REGISTER
- REFER
- INVITE
- BYE
- CANCEL

- NOTIFY
- OPTIONS
- ACK

Supplementary Services

SIP supplementary services are services that you can use to enhance your telephone service. For information on how to enable and subscribe to these services, see the "CallFeatures" section on page 5-35 and the "PaidFeatures" section on page 5-36.

For information on how to use these services, see Appendix A, "Using SIP Supplementary Services."

The following list contains the SIP supplementary services that the Cisco ATA supports:

- Caller ID
- Call-waiting caller ID
- Voice mail indication
- Making a conference call
- Call waiting
- Call forwarding
- Call return
- Calling-line identification
- Unattended transfer
- Attended transfer

Installation and Configuration Overview

Table 1-1 provides the basic steps required to install and configure the Cisco ATA to make it operational in a typical SIP environment where a large number of Cisco ATAs must be deployed.

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

Action	Reference
1. Plan the network and Cisco ATA configuration.	
2. Install the Ethernet connection.	
3. Install and configure the other network devices.	
4. Install the Cisco ATA but do not power up the Cisco ATA yet.	What the Cisco ATA Package Includes, page 2-2
5. Download the desired Cisco ATA release software zip file from the Cisco web site, then configure the Cisco ATA.	Chapter 3, "Configuring the Cisco ATA for SIP"
6. Power up the Cisco ATA.	
7. Periodically, you can upgrade the Cisco ATA to a new signaling image by using the TFTP server-upgrade method or the manual-upgrade method.	Chapter 8, "Upgrading the Cisco ATA Signaling Image"