CHAPTER 1

Cisco ATA 187 Analog Telephone Adaptor Overview

This section describes the hardware and software features of the Cisco ATA 187 Analog Telephone Adaptor (ATA 187) and includes a brief overview of the Session Initiation Protocol (SIP).

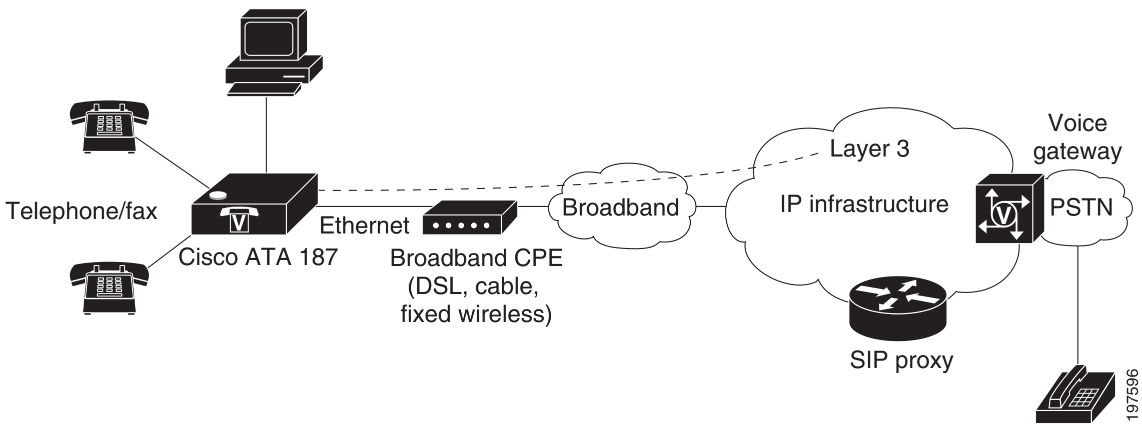
The ATA 187 analog telephone adaptors are handset-to-Ethernet adaptors that allow regular analog phones to operate on IP-based telephony networks. The ATA 187 support two voice ports, each with an independent phone number. The ATA 187 also has an RJ-45 10/100BASE-T data port.

This section covers these topics:

Figure 1-1 Cisco Analog Telephone Adaptor

The ATA187, 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.

Figure 1-2 ATA 187 as Endpoint in SIP Network



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 187 is compliant with RFC2543.

This section contains these topics:

- SIP Capabilities, page 1-2
- Components of SIP, page 1-3

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 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 these roles:

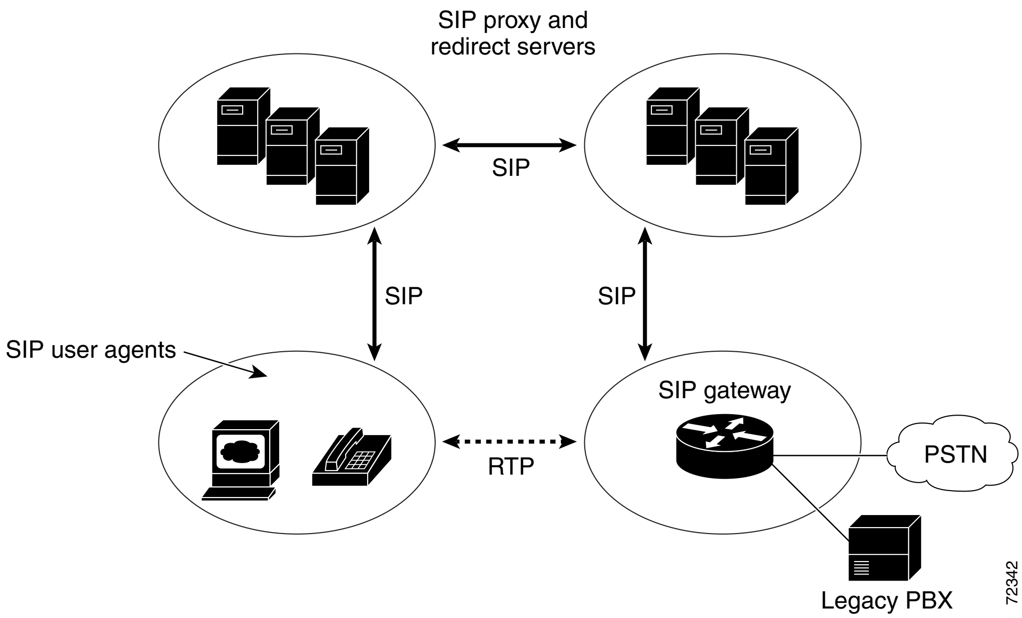
- User agent client (UAC)—A client application that initiates the SIP request.
- User agent server (UAS)—A server application that contacts the user when a SIP request is received and returns a response on behalf of the user.

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 1-3 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 1-3 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 187 can initiate SIP requests and respond to requests.

SIP Servers

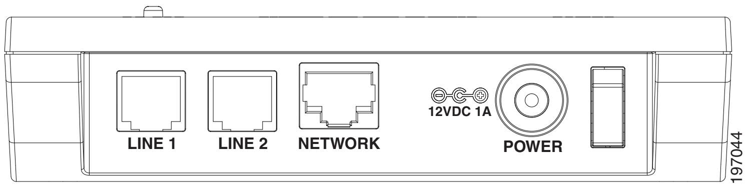
SIP servers include:

- 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 ATA 187 is a compact, easy to install device. Figure 1-4 shows the rear panel of the ATA 187.

Figure 1-4 ATA 187—Rear View



The unit provides these connectors and indicators:

- 12Vb power connector.
- Two RJ-11 FXS (Foreign Exchange Station) ports—The ATA 187 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 187 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 187 performs auto-negotiation for duplexity and speed and is capable of 10/100 Mbps, full-duplex operation.

Software Features

The ATA 187 supports these protocols, services and methods:

- Secure Real-Time Transport Protocol, page 1-6
- Name Signaling Event based passthrough, page 1-6
- Transport Layer Security Protocol, page 1-6
- T.38 Fax Relay, page 1-6
- Voice Codecs Supported, page 1-6
- Other Supported Protocols, page 1-6
- ATA 187 SIP Services, page 1-7
- Modem Standards, page 1-7
- Fax Services, page 1-8
- Methods Supported, page 1-8
- Supplementary Services, page 1-9

Secure Real-Time Transport Protocol

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

Name Signaling Event based passthrough

Name Signaling Event (NSE)-based passthrough is simply thet transport of fax or modem communications using the G.711 codec.

The ATA 187 does not support NSE-based modem passthrough.

Transport Layer Security Protocol

Transport Layer Security (TLS) is a cryptographic protocol that secures data communications such as e-mail 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, sent to the T.38 fax device, and converted 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 Security Description (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 187 supports these voice codecs (check your other network devices for the codecs they support):

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

Other Supported Protocols

The ATA 187 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)
- Real-Time Transport Protocol (RTP)
- Transmission Control Protocol (TCP)
- Trivial File Transfer Protocol (TFTP)
- User Datagram Protocol (UDP)

ATA 187 SIP Services

These services include these features:

- IP address assignment—DHCP-provided or statically configured
- ATA 187 configuration by Cisco Unified Communications Manager configuration interface
- 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
- SIP proxy server redundancy
- Privacy features
- User-configurable, call waiting, permanent default setting
- Comfort noise during silent period when using G.711
- Advanced audio mode
- Caller ID format
- Ring cadence format
- Silence suppression
- Hookflash detection timing configuration
- Configurable onhook delay
- Type of Service (ToS) configuration for audio and signaling ethernet packets
- Debugging and diagnostic tools

Modem Standards

The ATA 187 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 187 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.
- 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, sent to the T.38 fax device, and converted 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 187 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 7, "Using SIP Supplementary Services".

The ATA 187 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
- MeetMe
- Pick Up
- Redial

Installation and Configuration Overview

Table 1-1 provides the basic steps required to install and configure the ATA 187 to make it operational in a typical SIP environment where a large number of ATA 187s must be deployed.

Table 1-1 Overview of the Steps Required to Install and Configure the ATA 187 and Make it Operational

Action	Reference
1. Plan the network and ATA 187 configuration.	
2. Install the Ethernet connection.	
3. Install and configure the other network devices.	
4. Install the ATA 187 but do not power up the ATA 187 yet.	Installing the ATA 187, page 3-3
5. Power up the ATA 187.	