Cisco ATA 191 Analog Telephone Adapter Overview

Your Analog Telephone Adapter

The ATA 191 analog telephone adapter is a telephony-device-to-Ethernet adapter that allows regular analog phones to operate on IP-based telephony networks. The ATA 191 supports two voice ports, each with an independent phone number. The ATA 191 also has an RJ-45 10/100BASE-T data port.

Figure 1: Cisco Analog Telephone Adapter

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Session Initiation Protocol

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). It is 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 is used to control the attributes of an end-to-end call.

Note SIP for the ATA 191 is compliant with RFC2543.

SIP Capabilities

Session Initiation Protocol (SIP) provides these capabilities:

- Determines the availability of the target endpoint. If the target endpoint is unavailable, SIP determines whether the called party is already on the phone or didn't 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.

SIP Components

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:

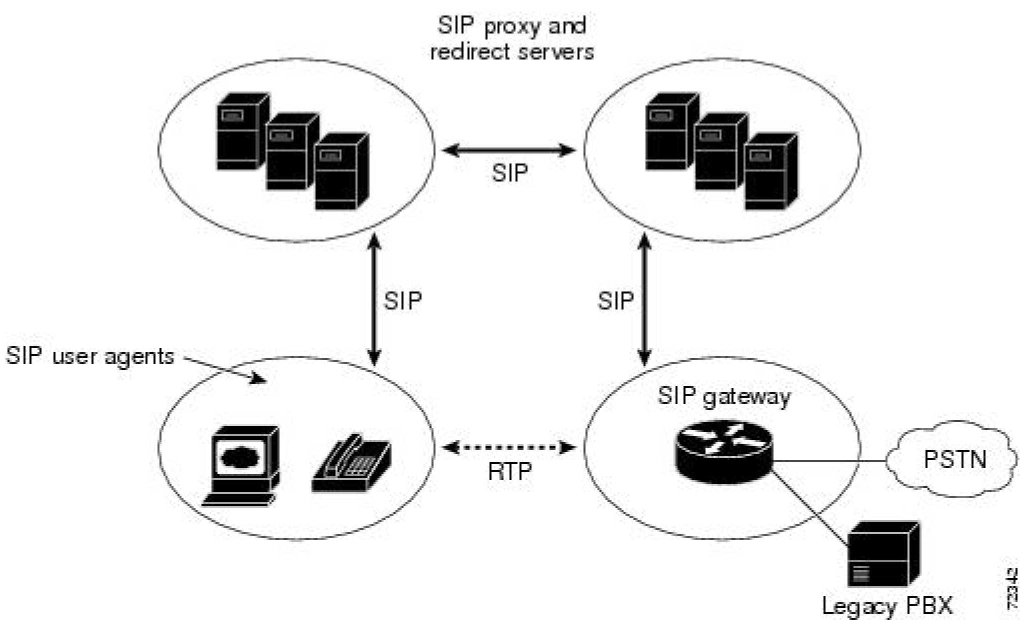
- User agent client (UAC)—A client application that initiates the SIP request.
- User agent server (UAS)—A server application that contacts the user when a SIP request is received and returns a response on behalf of the user.

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. The following figure shows 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 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 191 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.

Cisco ATA 191 Hardware

The ATA 191 is a compact, easy to install device. The following figures show the top and rear panels of the ATA 191.

The unit provides these connectors:

- 5V DC power connector.
- Two RJ-11 FXS (Foreign Exchange Station) ports—The ATA 191 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 191 has one WAN network port—An RJ-45 10/100BASE-T data port to connect an Ethernet-capable device, such as a computer, to the network. You connect to the network switch or a computer running Cisco Unified Communications Manager using this port.

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

ATA 191 Top Panel

Figure 3: ATA 191 Top Panel

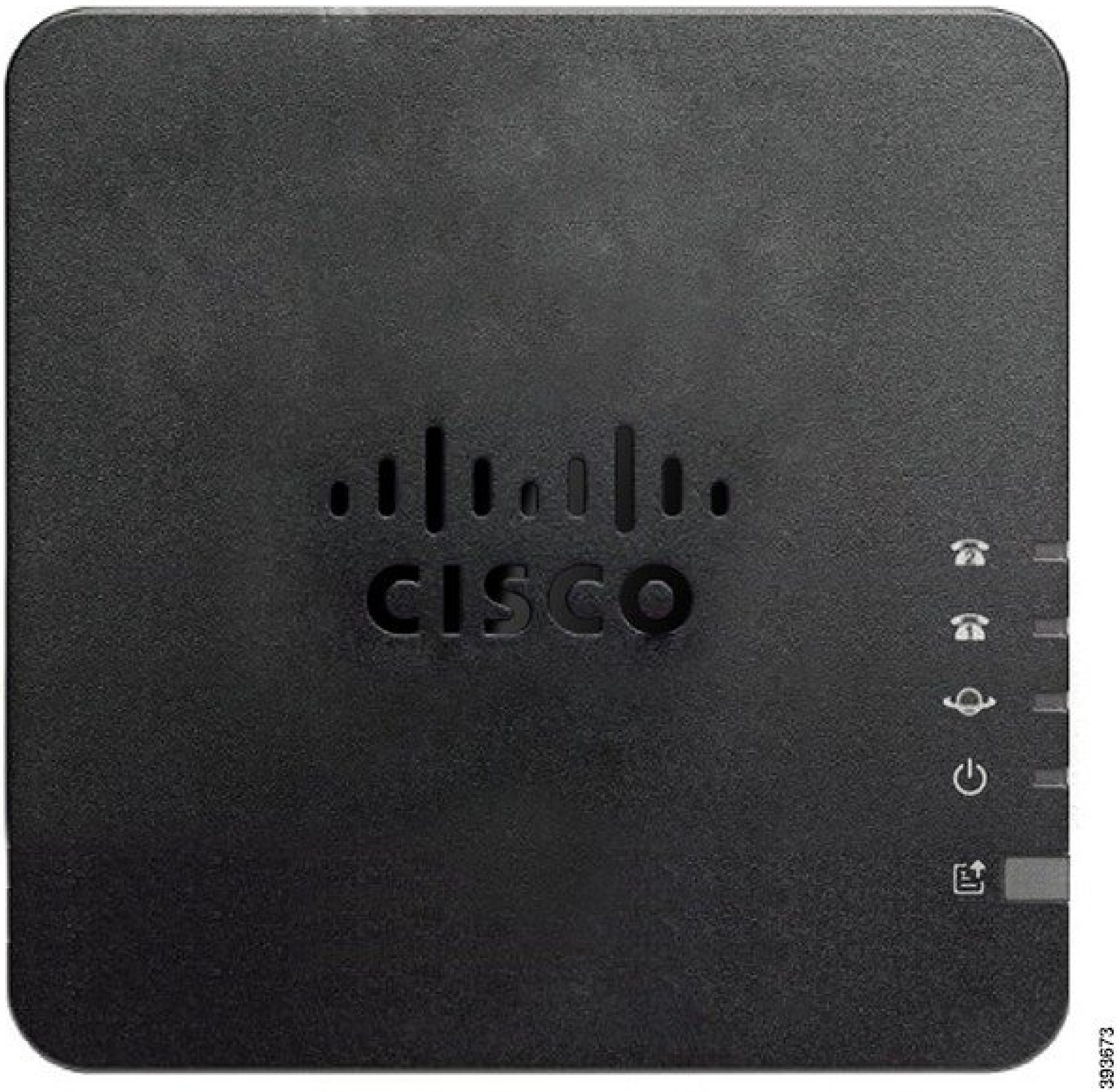


Table 1: ATA 191 Top Panel Items

Item	Description
Power LED	**Steady green:** System booted up successfully and is ready for use. **Slow flashing green:** System is booting up. **Fast flashing green three times, then repeats:** System failed to boot up. **Off:** Power is off.
Network LED	**Flashing green:** Data transmission or reception is in progress through the WAN port. **Off:** No link.
Phone 1 LED Phone 2 LED	**Steady green:** On hook. **Slow flashing green:** Off hook. **Fast flashing green three times, then repeats:** The analog device failed to register. **Off:** The port is not configured.
Problem Report Tool (PRT) Button	Press this button to create a problem report using the Problem Report Tool. **Note** This is not a power button. When you press this button, a problem report is generated and uploaded to a server for the system administrator.
Problem Report Tool (PRT) LED	**Flashing amber:** The PRT is preparing the data for the problem report. **Fast Flashing amber:** The PRT is sending the problem report log to the PRT server. **Solid green for five seconds, then off:** The PRT report was sent successfully. **Flashing red:** The PRT report failed. Press the PRT button to turn the LED off. Once it is off, another press triggers a new PRT report.

Problem Report Tool Button

The ATA top panel includes the Problem Report Tool (PRT) button. When the button is pressed, a log file is prepared and uploaded to either an HTTP server or to Cisco Unified Communications Manager depending on your configuration.

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

To upload the PRT log file from the ATA, one of the following must be completed:

- The HTTP server must be set up to upload the PRT log file from the ATA.

- You configure the customer support upload URL in CUCM and apply it to the ATA.

Related Topics

Problem Report Tool

ATA 191 Back Panel

Figure 4: ATA 191 Back Panel

Table 2: ATA 191 Back Panel Items

Item	Description
RESET	To restart the ATA, use a paper clip or similar object to press this button briefly. To restore the factory default settings, press and hold for 10 seconds.
PHONE 1	Use an RJ-11 phone cable to connect an analog phone or fax machine.
PHONE 2	Use an RJ-11 phone cable to connect a second analog phone or fax machine.
NETWORK	Use an Ethernet cable to connect to the network.
DC 5V POWER	Use the power adapter that was provided to connect to a power source.

Software Features

The ATA 191 supports these protocols, services, and methods:

- Secure Real-Time Transport Protocol, on page 8
- Fax Passthrough, on page 8

Secure Real-Time Transport Protocol

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

Fax Passthrough

Name Signaling Event (NSE) -based and re-INVITE-based passthrough provide 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 Session Description Protocol (SDP) entries in the initial SIP INVITE message indicate that T.38 fax relay is present. 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.

The ATA 191 only supports T38 Fax Relay Version 0 (G3).

Supported Voice Codecs

The ATA 191 supports these voice codecs:

- G.711 mu-law
- G.711 A-law

- G.729a
- G.729ab

Check your other network devices for the codecs they support.

Other Supported Protocols

The ATA 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) v4 and IPv6
- Link Layer Discovery Protocol (LLDP)
- Secure Real-Time Transport Protocol (SRTP)
- Transmission Control Protocol (TCP)
- Trivial File Transfer Protocol (TFTP)
- User Datagram Protocol (UDP)
- Transport Layer Security (TLS)
- Secure Socket Shell (SSH)
- Network Time Protocol (NTP)
- HyperText Transfer Protocol (HTTP)

Supported SIP Services

The following SIP services are supported on the ATA:

- IP address assignment—DHCP-provided or statically configured
- ATA 191 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

Supported Call Services

The following call services are supported on the ATA:

- IP address assignment—DHCP-provided or statically configured
- ATA 191 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
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Modem Standards

The ATA supports these 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 191 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.

Related Topics

Configure Fax Services

Supported Methods

The ATA 191 supports these methods:

- REGISTER
- REFER
- INVITE
- BYE
- CANCEL

- NOTIFY
- OPTIONS
- ACK
- SUBSCRIBE

For more information, see RFC3261, SIP: Session Initiation Protocol.

Supported ATA Call Features

SIP supplementary services are services that you can use to enhance your phone service.

The ATA 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

The following basic steps are required to install and configure the ATA. The steps also make the ATA operational in a typical SIP environment where many ATAs are deployed.

1. Plan the network and the ATA configuration.
2. Install the Ethernet connection.
3. Install and configure the other network devices.
4. Install the ATA but do not power it up yet.

5 Power up the ATA.

Related Topics

Prepare to Install the ATA 191 on Your Network
Install the ATA 191