Cisco ATA 186 and Cisco 188 Installation and Configuration Guide

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Text Part Number: OL-1267-01

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

The following information is for FCC compliance of Class B devices: The equipment described in this manual generates and may radiate radio-frequency energy. If it is not installed in accordance with Cisco's installation instructions, it may cause interference with radio and television reception. This equipment has been tested and found to comply with the limits for a Class B digital device in accordance with the specifications in part 15 of the FCC rules. These specifications are designed to provide reasonable protection against such interference in a residential installation. However, there is no guarantee that interference will not occur in a particular installation.

Modifying the equipment without Cisco's written authorization may result in the equipment no longer complying with FCC requirements for Class A or Class B digital devices. In that event, your right to use the equipment may be limited by FCC regulations, and you may be required to correct any interference to radio or television communications at your own expense.

You can determine whether your equipment is causing interference by turning it off. If the interference stops, it was probably caused by the Cisco equipment or one of its peripheral devices. If the equipment causes interference to radio or television reception, try to correct the interference by using one or more of the following measures:

• Turn the television or radio antenna until the interference stops.

• Move the equipment to one side or the other of the television or radio.

• Move the equipment farther away from the television or radio.

• Plug the equipment into an outlet that is on a different circuit from the television or radio. (That is, make certain the equipment and the television or radio are on circuits controlled by different circuit breakers or fuses.)

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

Exhibit - J

User Manual Statement

FCC Requirements

1. The Federal Communications Commission (FCC) has established Rules, which permit this device to be directly connected to the telephone network. Standardized jacks are used for these connections. This equipment should not be used on party lines or coin phones.

2. If this device is malfunctioning, it may also be causing harm to the telephone network; this device should be disconnected until the source of the problem can be determined and until repair has been made. If this is not done, the Telephone Company may temporarily disconnect service.

3. The Telephone Company may make changes in its technical operations and procedures; if such changes affect the compatibility or use of this device, the Telephone Company is required to give adequate notice of the changes. You will be advised of your right to file a complaint with the FCC.

4. This equipment complies with Part 68 of the FCC rules. On the this equipment is a label that contains, among other information, the FCC certification number and ringer equivalence number (REN) for this equipment. If the telephone company requests information on what equipment is connected to their lines, inform them of the following:

a.The telephone number to which this unit is connected.

b.The ringer equivalence number. [0.3B]

c.The USOC jack required. [RJ11C]

d. The FCC Registration Number. [5B1XXX-XXXXX-M5-E]

Items (b) and (d) are indicated on the label. The Ringer Equivalence Number (REN) is used to determine how many devices can be connected to your telephone line. In most areas, the sum of the REN's of all devices on any one line should not exceed five (5.0). If too many devices are attached, they may not ring properly.

Service Requirements

In the event of equipment malfunction, all repairs should be performed by our Company or an authorized agent. It is the responsibility of users requiring service to report the need for service to our Company or to one of our authorized agents. Service can be obtained at:

Cisco Systems, Inc
170 West Tasman Drive
San Jose, CA 95134-1706
Telephone Number: 408-526-4000

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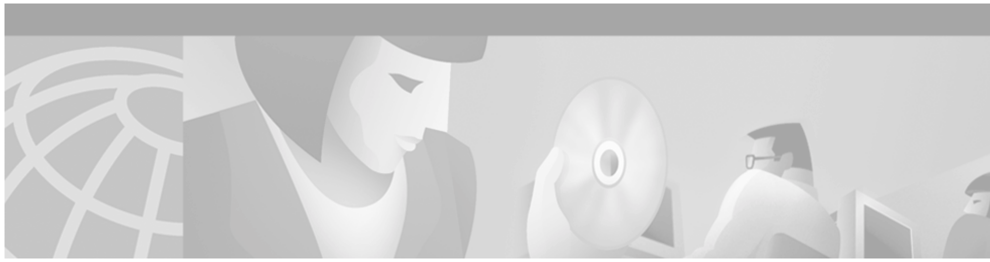


Preface

This preface describes:

- Audience
- Conventions used in this guide
- Related documentation
- How to access electronic documentation

Who Should Read This Guide

This guide is intended for service providers and network administrators who administer network connectivity of the Cisco ATA 186 and use the Cisco ATA 186 to provide voice over IP (VoIP) services to end users.

Note For additional information about the SIP protocol with Cisco ATA, also refer to the *Cisco ATA 186 and Cisco ATA 188 Analog Telephone Adaptor Administrator's Guide (SIP)*.

Related Documentation

In addition to this Cisco ATA 186 *Installation and Configuration Guide*, the Cisco ATA 186 documentation set includes the following:

- *Quick Start for the Cisco ATA 186 Analog Telephone Adaptor*
- *Quick Reference Guide for the Cisco ATA 186 Analog Telephone Adaptor*
- *Cisco ATA 186 Regulatory Compliance and Safety Information*
- Release Notes for Cisco ATA 186

Be sure to read any readme files or additional release notes for important information.

Conventions

Table 1 Conventions

Convention	Description
boldface font	Commands and keywords.
italic font	Variables for which you supply values.
[ ]	Keywords or arguments that appear within square brackets are optional.
{x | y | z}	A choice of required keywords appears in braces separated by vertical bars. You must select one.
`screen font`	Examples of information displayed on the screen.
boldface screen font	Examples of information you must enter.
< >	Nonprinting characters, for example passwords, appear in angle brackets.
[ ]	Default responses to system prompts appear in square brackets.

Table 1 Conventions (continued)

Convention	Description
Note	This symbol means *reader take note*. Notes contain helpful suggestions or references to additional information and material.
Timesaver	This symbol means *the described action saves time*. You can save time by performing the action described in the paragraph.
Caution	This symbol means *reader be careful*. In this situation, you might do something that could result in equipment damage or loss of data.
Tips	This symbol means *the following information will help you solve a problem*. The tips information might not be troubleshooting or even an action, but could be useful information, similar to a Timesaver.
Warning	**This warning symbol 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. To see translations of the warnings that appear in this publication, refer to the *Regulatory Compliance and Safety Information* document that accompanied this device.**

Table 1 Conventions (continued)

Convention	Description
Waarschuwing	**Dit waarschuwingssymbool betekent gevaar. U verkeert in een situatie die lichamelijk letsel kan veroorzaken. Voordat u aan enige apparatuur gaat werken, dient u zich bewust te zijn van de bij elektrische schakelingen betrokken risico's en dient u op de hoogte te zijn van standaard maatregelen om ongelukken te voorkomen. Voor vertalingen van de waarschuwingen die in deze publicatie verschijnen, kunt u het document *Regulatory Compliance and Safety Information* (Informatie over naleving van veiligheids- en andere voorschriften) raadplegen dat bij dit toestel is ingesloten.**
Varoitus	**Tämä varoitusmerkki merkitsee vaaraa. Olet tilanteessa, joka voi johtaa ruumiinvammaan. Ennen kuin työskentelet minkään laitteiston parissa, ota selvää sähkökytkentöihin liittyvistä vaaroista ja tavanomaisista onnettomuuksien ehkäisykeinoista. Tässä julkaisussa esiintyvien varoitusten käännökset löydät laitteen mukana olevasta *Regulatory Compliance and Safety Information* -kirjasesta (määräysten noudattaminen ja tietoa turvallisuudesta).**
Attention	**Ce symbole d'avertissement indique un danger. Vous vous trouvez dans une situation pouvant causer des blessures ou des dommages corporels. Avant de travailler sur un équipement, soyez conscient des dangers posés par les circuits électriques et familiarisez-vous avec les procédures couramment utilisées pour éviter les accidents. Pour prendre connaissance des traductions d'avertissements figurant dans cette publication, consultez le document *Regulatory Compliance and Safety Information* (Conformité aux règlements et consignes de sécurité) qui accompagne cet appareil.**

Table 1 Conventions (continued)

Convention	Description
Warnung	**Dieses Warnsymbol bedeutet Gefahr. Sie befinden sich in einer Situation, die zu einer Körperverletzung führen könnte. Bevor Sie mit der Arbeit an irgendeinem Gerät beginnen, seien Sie sich der mit elektrischen Stromkreisen verbundenen Gefahren und der Standardpraktiken zur Vermeidung von Unfällen bewußt. Übersetzungen der in dieser Veröffentlichung enthaltenen Warnhinweise finden Sie im Dokument *Regulatory Compliance and Safety Information* (Informationen zu behördlichen Vorschriften und Sicherheit), das zusammen mit diesem Gerät geliefert wurde.**
Avvertenza	**Questo simbolo di avvertenza indica un pericolo. La situazione potrebbe causare infortuni alle persone. Prima di lavorare su qualsiasi apparecchiatura, occorre conoscere i pericoli relativi ai circuiti elettrici ed essere al corrente delle pratiche standard per la prevenzione di incidenti. La traduzione delle avvertenze riportate in questa pubblicazione si trova nel documento *Regulatory Compliance and Safety Information* (Conformità alle norme e informazioni sulla sicurezza) che accompagna questo dispositivo.**
Advarsel	**Dette varselsymbolet betyr fare. Du befinner deg i en situasjon som kan føre til personskade. Før du utfører arbeid på utstyr, må du vare oppmerksom på de faremomentene som elektriske kretser innebærer, samt gjøre deg kjent med vanlig praksis når det gjelder å unngå ulykker. Hvis du vil se oversettelser av de advarslene som finnes i denne publikasjonen, kan du se i dokumentet *Regulatory Compliance and Safety Information* (Overholdelse av forskrifter og sikkerhetsinformasjon) som ble levert med denne enheten.**

Table 1 Conventions (continued)

Convention	Description
Aviso	Este símbolo de aviso indica perigo. Encontra-se numa situação que lhe poderá causar danos físicos. Antes de começar a trabalhar com qualquer equipamento, familiarize-se com os perigos relacionados com circuitos eléctricos, e com quaisquer práticas comuns que possam prevenir possíveis acidentes. Para ver as traduções dos avisos que constam desta publicação, consulte o documento *Regulatory Compliance and Safety Information* (Informação de Segurança e Disposições Reguladoras) que acompanha este dispositivo.
¡Advertencia!	Este símbolo de aviso significa peligro. Existe riesgo para su integridad física. Antes de manipular cualquier equipo, considerar los riesgos que entraña la corriente eléctrica y familiarizarse con los procedimientos estándar de prevención de accidentes. Para ver una traducción de las advertencias que aparecen en esta publicación, consultar el documento titulado *Regulatory Compliance and Safety Information* (Información sobre seguridad y conformidad con las disposiciones reglamentarias) que se acompaña con este dispositivo.
Varning!	Denna varningssymbol signalerar fara. Du befinner dig i en situation som kan leda till personskada. Innan du utför arbete på någon utrustning måste du vara medveten om farorna med elkretsar och känna till vanligt förfarande för att förebygga skador. Se förklaringar av de varningar som förkommer i denna publikation i dokumentet *Regulatory Compliance and Safety Information* (Efterrättelse av föreskrifter och säkerhetsinformation), vilket medföljer denna anordning.

Obtaining Documentation

The following sections provide sources for obtaining documentation from Cisco Systems.

World Wide Web

You can access the most current Cisco documentation on the World Wide Web at the following sites:

- http://www.cisco.com
- http://www-china.cisco.com
- http://www-europe.cisco.com

Ordering Documentation

Cisco documentation is available in the following ways:

- Registered Cisco Direct Customers can order Cisco Product documentation from the Networking Products MarketPlace:

  http://www.cisco.com/cgi-bin/order/order_root.pl
- Registered Cisco.com users can order the Documentation CD-ROM through the online Subscription Store:

  http://www.cisco.com/go/subscription
- Nonregistered Cisco.com users can order documentation through a local account representative by calling Cisco corporate headquarters (California, USA) at 408 526-7208 or, in North America, by calling 800 553-NETS(6387).

Documentation Feedback

If you are reading Cisco product documentation on the World Wide Web, you can submit technical comments electronically. Click **Feedback** in the toolbar and select **Documentation**. After you complete the form, click **Submit** to send it to Cisco.

You can e-mail your comments to bug-doc@cisco.com.

To submit your comments by mail, use the response card behind the front cover of your document, or write to the following address:

Attn Document Resource Connection
Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-9883

We appreciate your comments.

Obtaining Technical Assistance

Cisco provides Cisco.com as a starting point for all technical assistance. Customers and partners can obtain documentation, troubleshooting tips, and sample configurations from online tools. For Cisco.com registered users, additional troubleshooting tools are available from the Technical Assistance Center (TAC) website.

Cisco.com

Cisco.com is the foundation of a suite of interactive, networked services that provides immediate, open access to Cisco information and resources at anytime, from anywhere in the world. This highly integrated Internet application is a powerful, easy-to-use tool for doing business with Cisco.

Cisco.com provides a broad range of features and services to help customers and partners streamline business processes and improve productivity. Through Cisco.com, you can find information about Cisco and our networking solutions, services, and programs. In addition, you can resolve technical issues with online technical support, download and test software packages, and order Cisco learning materials and merchandise. Valuable online skill assessment, training, and certification programs are also available.

Customers and partners can self-register on Cisco.com to obtain additional personalized information and services. Registered users can order products, check on the status of an order, access technical support, and view benefits specific to their relationships with Cisco.

To access Cisco.com, go to the following website:

http://www.cisco.com

Technical Assistance Center

The Cisco TAC website is available to all customers who need technical assistance with a Cisco product or technology that is under warranty or covered by a maintenance contract.

Contacting TAC by Using the Cisco TAC Website

If you have a priority level 3 (P3) or priority level 4 (P4) problem, contact TAC by going to the TAC website:

http://www.cisco.com/tac

P3 and P4 level problems are defined as follows:

- P3—Your network performance is degraded. Network functionality is noticeably impaired, but most business operations continue.
- P4—You need information or assistance on Cisco product capabilities, product installation, or basic product configuration.

In each of the above cases, use the Cisco TAC website to quickly find answers to your questions.

To register for Cisco.com, go to the following website:

http://www.cisco.com/register/

If you cannot resolve your technical issue by using the TAC online resources, Cisco.com registered users can open a case online by using the TAC Case Open tool at the following website:

http://www.cisco.com/tac/caseopen

Contacting TAC by Telephone

If you have a priority level 1 (P1) or priority level 2 (P2) problem, contact TAC by telephone and immediately open a case. To obtain a directory of toll-free numbers for your country, go to the following website:

http://www.cisco.com/warp/public/687/Directory/DirTAC.shtml

P1 and P2 level problems are defined as follows:

- P1—Your production network is down, causing a critical impact to business operations if service is not restored quickly. No workaround is available.
- P2—Your production network is severely degraded, affecting significant aspects of your business operations. No workaround is available.

Product Disposal Warning

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

Waarschuwing **Het uiteindelijke wegruimen van dit product dient te geschieden in overeenstemming met alle nationale wetten en reglementen.**

Varoitus **Tämä tuote on hävitettävä kansallisten lakien ja määräysten mukaisesti.**

Attention **La mise au rebut ou le recyclage de ce produit sont généralement soumis à des lois et/ou directives de respect de l'environnement. Renseignez-vous auprès de l'organisme compétent.**

Warnung **Die Entsorgung dieses Produkts sollte gemäß allen Bestimmungen und Gesetzen des Landes erfolgen.**

Avvertenza **Lo smaltimento di questo prodotto deve essere eseguito secondo le leggi e regolazioni locali.**

Advarsel **Endelig kassering av dette produktet skal være i henhold til alle relevante nasjonale lover og bestemmelser.**

Aviso **Deitar fora este produto em conformidade com todas as leis e regulamentos nacionais.**

¡Advertencia! **Al deshacerse por completo de este producto debe seguir todas las leyes y reglamentos nacionales.**

Varning! **Vid deponering hanteras produkten enligt gällande lagar och bestämmelser.**

CHAPTER 1

Cisco ATA 186 Overview

This chapter provides an overview of the Cisco ATA 186 Analog Telephone Adaptor and describes the system features.

About the Cisco ATA 186

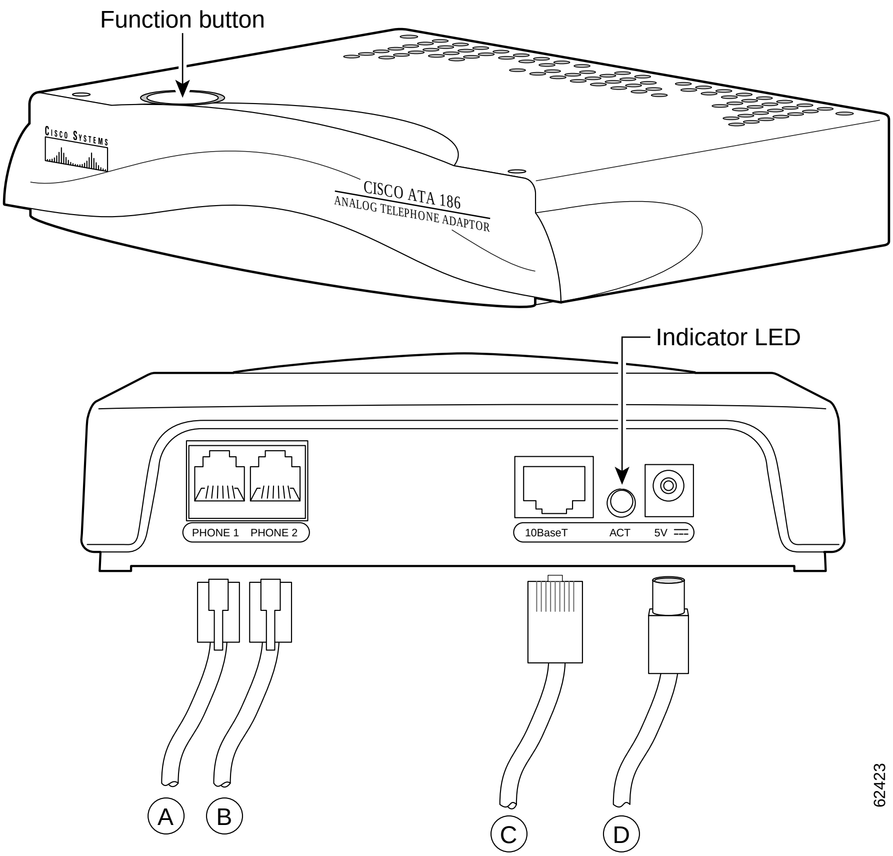
The Cisco ATA 186:

- Is an Internet telephony appliance that converts any regular analog telephone into an Internet telephone.
- Converts voice into IP data packets that are sent over a network.
- Is installed at the subscriber’s premises.
- Supports two voice ports with their own telephone numbers.
- Has one 10BaseT RJ-45 port and two RJ-11 FXS standard analog telephone ports.
- Supports both low-complexity voice codecs, such as G.711µ, G.711A-law, and high-complexity/CELP codecs, such as G.723.1 and G.729A.
- Can be configured to use either the H.323 or the SIP call signaling protocol.
- Operates with Cisco Voice Packet Gateways.
- Operates with H.323- or SIP-compliant Voice Packet Gateways.

Hardware Features

The Cisco ATA 186 features the hardware options listed in this section and shown in Figure 1-1.

Figure 1-1 Features of the Cisco ATA 186



A—Phone 1 connector (RJ-11)

B—Phone 2 connector (RJ-11)

C—10Base-T Ethernet connector

D—12V power connector

- Dual RJ-11 Ports
  - 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, so that the user can talk on one port while sending a FAX on the other.
  - There are two RJ-11 FXS port terminating impedance options, 600 Ohms resistive or 270 Ohms + 750 Ohms // 150 nF complex impedance. The impedance of the Cisco ATA 186 depends on the option ordered and must match the particular application. If you are not sure of the applicable configuration, check the country or regional telephone impedance requirements.
- Indicators
  - Function Button

    The Cisco ATA 186 function button is located in the top panel of the device. The function button lights when you pick up the handset of a telephone attached to the Cisco ATA 186. When the function button blinks, the device is in configuration search or upgrade mode. You also use the function button to access the interactive voice response (IVR) configuration menu. To configure the Cisco ATA 186 by using the IVR mode, see Chapter 3, "Configuring the Cisco ATA 186."
  - Activity LED

    The green Activity LED located on the back panel flashes to indicate network activity.
- 5 Volt power adapter connector (adapter included)
- 10Base-T Ethernet port

Additional Feature

Supports polarity reversal before and after caller-ID signal.

Software Features

Features of the Cisco ATA 186 depend on the protocol used.

General Features

Table 1-1 includes information on features that are available for all protocols.

Table 1-1 Features Available for all Signaling Protocols

Description	Details
Basic provisioning (TFTP Profiling)	API, profile generation, client provisioning, RC4 encryption, and hard key
Call forwarding always from the Cisco ATA 186	
Call forwarding on no answer from the Cisco ATA 186	
Call forwarding on busy from the Cisco ATA 186	
Call waiting	
Calling Line ID Presentation (CLIP)	
Calling Line ID Rejection (CLIR)	
Comfort noise generation (CNG)	
Configurable ring specification	
Dial Plan Support	Feature access code support
Domain Name Server (DNS) Lookup	
DTMF Caller ID	On-hook only
Dynamic Jitter Buffering	
Fax Detect/Passthrough **Note** Limited fax passthrough support is available (up to 9.6 kbps fax transmission rates for most fax machines). Extended support is planned. Please check release notes and product bulletins for updates.	G.711, Codec re-negotiation

Table 1-1 Features Available for all Signaling Protocols

Description	Details
Frequency Shift Key (FSK) Caller ID	On-hook only
Line-echo cancellation	8 ms fixed echo length setting
Local ring-back tone	
Remote diagnostics/monitor (trace of execution)	
Three-way calling (conferencing)	The Cisco ATA 186 will automatically switch to G.711 in this mode.
Type of Service (TOS) bit for Quality of Service (QOS)	
Voice activity detection (VAD)	

H.323-specific Features

Table 1-2 H.323-specific Features

Description	Details
Alternate gatekeeper	
Call proceeding	
Cisco registration- and admission-level security support	Uses MD5 hashing Uses access/clear token
Dual Tone Multiple Frequency (DTMF) relay	H.245
Empty cap set	
Fast start/tunneling/early H.245	Including H.245 messages in the Alert message

SIP Specific Features

Table 1-3 SIP Features

Description	Details
Authentication	Digest Authentication
DTMF Relay	RFC 2833
Call Forwarding	Unconditionally, on no answer, or on busy
Call Return	
Call Transfer	With or without consultation
Message Waiting Indication (MWI)	Plays an intermittent dial tone if there is a message waiting. Otherwise, plays a normal dial tone.
Third-party call control	

About Supported Standards and Protocols

The following standards are supported on the Cisco ATA 186:

- Network interface: one RJ-45 8-wire connector, IEEE 802.3 10Base-T standard
- Two RJ-11 FXS standard analog telephone voice ports, up to 5 ringer equivalency number (REN) per port, depending on loop length
- ITU G.711µ, G.711A, G.723.1 Annex A, and G.729 Annex A voice codecs
- G.723.1 Annex A, voice activity detection (VAD)/comfort noise generation (CNG): bandwidth saving algorithm
- ITU H.323 V.2 call signaling protocol
- SIP: RFC 2543bis
- LSSGR: Signaling for analog interfaces GR-506-CORE
- RTP: real-time transmission Internet protocol
- ITU-T V.42/V.42bis and MNP2-10 error correction and data compression
- AVT Tones: RFC 2833
- DHCP: RFC 2131

Electrical Specifications

Table 1-4 Electrical Specifications

Category	Specification
Voltage	+5.0 VDC at 1.5 A maximum
Power	0.25 to 7.5 Watts (idle, maximum)
Power adaptor	Universal AC/DC 3.3 x 2.0 x 1.3 in (~8.5 x 5.0 x 3.2 cm) 4.8 oz (135 gm) for the AC-input external power adaptor 4 ft (1.2 m) DC cord Class II transformer 6 ft (1.8 m) cord UL/CUL, CE agency approvals

Environmental Specifications

Table 1-5 Environmental Specifications

Category	Specification
Operating Temperature	32° to 122° F (0 to 50° C)
Storage Temperature	-22° to 149° F (-30° to 65° C)
Relative Humidity	10 to 90% non-condensing, operating and storage

Standards Compliance

Table 1-6 Standards Compliance

Category	Specification
Agency approvals	UL/C-UL FCC (Declaration of Conformity) Class B part 15 and part 68. European Union, CE mark (Declaration of Conformity) Industry Canada (Declaration of Conformity) ACA (Declaration of Conformity) VCCI (Declaration of Conformity)
Safety standards	UL60950 CAN/CSA-C22.2 No. 60950-00 IEC 60950 (Second Edition with Amendments 1, 2, 3, and 4) EN60950:1992 (with Amendments 1, 2, 3, 4, and 11) AS/NZS 3260:1993 (with Amendments 1, 2, 3, and 4) TS001:1997

Table 1-6 Standards Compliance

Category	Specification
Emissions	CFR 47 Part 15 Class B 2000 EN55024, EN50082-1 EN55022/CISPR22 Class B VCCI Class B AS/NZS 3548:1995 Class B ICES-003 (Issue 2, Class B, April 1997)
Immunity	EN50082-1 including the following EN61000-3-2, Electromagnetic Compatibility EN61000-3-3, Electromagnetic Compatibility EN61000-4-2, ESD EN61000-4-3, Radiated Immunity EN61000-4-4, Burst Transients EN61000-4-5, Surge EN61000-4-6, Injected RF EN61000-4-11, Dips and Sags

Dimensions

Table 1-7 Dimensions

Category	Specification
Length	6.5 in (16.5 cm)
Width	6 in (15.25 cm)
Height	1.5 in (3.8 cm)
Weight	15 oz (425 gm)

Installing the Cisco ATA 186

This chapter provides information about installing the Cisco ATA 186.

English is the default language. For information on the upgrade process, see Chapter 6, “Upgrading the Cisco ATA 186 Software.”

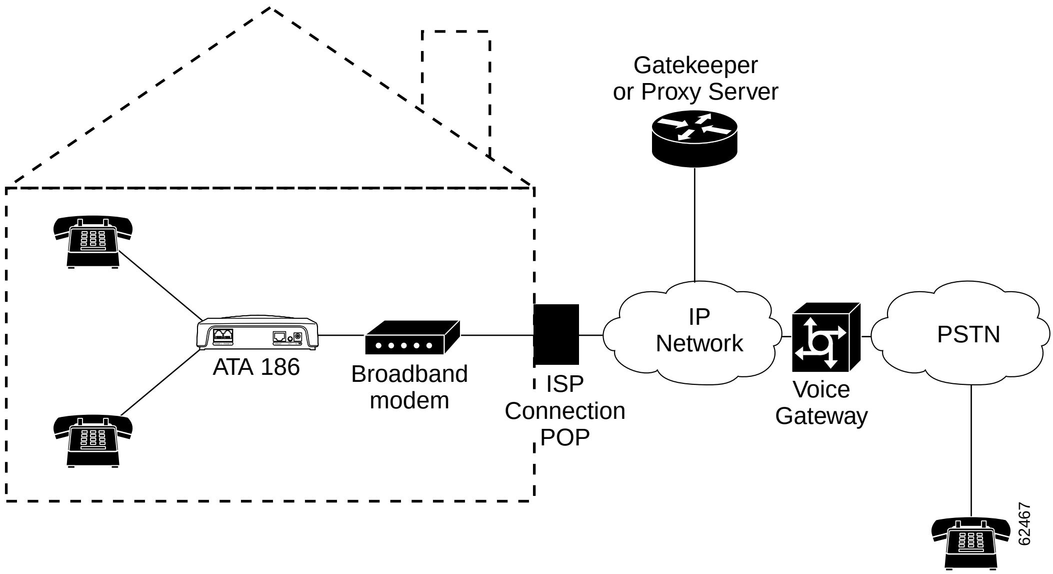
Installation Overview

The general steps necessary to install the Cisco ATA 186 are:

1. Plan the network and Cisco ATA 186 configuration.
2. Install the Ethernet connection.
3. Install and configure the other network devices; for example, gatekeeper if you are using H.323 or proxy server if you are using SIP.
4. If you will be routing calls through the Public Switched Telephone Network (PSTN), install and configure the Gateway.
5. Install the Cisco ATA 186.
6. Configure the Cisco ATA 186.
7. Perform any troubleshooting and maintenance, including upgrading the software if necessary.

Figure 2-1 shows an example of a network with a Cisco ATA 186.

Figure 2-1 Example Network Diagram



Network Requirements

The Cisco ATA 186 acts as a terminal on an IP network. You need the following equipment:

- One or two standard analog telephone handsets.
- Ethernet connection.
- Gatekeeper or proxy server—Currently, there must be a device running ITU H.323 or RFC 2543bis SIP-complaint software. The gatekeeper must be running the applicable version of software for the features and protocol you want to use.
- Voice Packet Gateway—Required if you are connecting to the Public Switched Telephone Network (PSTN).
- Fax machine (optional).
- If you are using a firewall, Cisco recommends that it be a Cisco PIX firewall, Version 5 or later versions.

Safety Recommendations

To ensure general safety, follow these guidelines:

- Do not open or disassemble this product.
- Do not get this product wet or pour liquids into this device.
- Do not perform any action that creates a potential hazard to people or makes the equipment unsafe.

System Requirements

Caution The Cisco ATA 186 is intended for use with a 5V DC power adaptor only.

The Cisco ATA 186 installation package includes:

- Cisco ATA 186
- Cisco ATA 186 documentation
- 5V Power Adaptor

You also need:

- 10BaseT category-3 cable or better
- One or two analog touch-tone telephones

Note Telephones must be set to use tone, rather than pulse dialing for the Cisco ATA 186 to operate correctly.

Installation Warnings

This section contains important safety information.

Number 26 AWG Warning

Warning **To reduce the risk of fire, use only No. 26 AWG or larger telecommunication line cord.**

Waarschuwing **Om brandgevaar te reduceren, dient slechts telecommunicatielijnsnoer nr. 26 AWG of groter gebruikt te worden.**

Varoitus **Tulipalovaaran vähentämiseksi käytä ainoastaan nro 26 AWG- tai paksumpaa tietoliikennejohdinta.**

Attention **Pour réduire le risque d’incendie, n’utiliser que des cordons de lignes de télécommunications de type AWG nº 26 ou plus larges.**

Warnung **Zur Reduzierung der Feuergefahr eine Fernmeldeleitungsschnur der Größe 26 AWG oder größer verwenden.**

Avvertenza **Per ridurre il rischio di incendio, usare solo un cavo per linea di telecomunicazioni di sezione 0,12 mm^2 (26 AWG) o maggiore.**

Advarsel **Bruk kun AWG nr. 26 eller telekommunikasjonsledninger med større dimensjon for å redusere faren for brann.**

Aviso **Para reduzir o risco de incêndio, utilize apenas terminais de fio de telecomunicações Nº. 26 AWG ou superiores.**

¡Advertencia! **Para reducir el riesgo de incendios, usar sólo líneas de telecomunicaciones de calibre No. 26 AWG o más gruesas.**

Varning! **För att minska brandrisken skall endast Nr. 26 AWG eller större telekommunikationsledning användas.**

Short-Circuit Protection Warning

Warning **This product relies on the building's installation for short-circuit (overcurrent) protection. Ensure that a fuse or circuit breaker no larger than 120VAC, 20A U.S. (240VAC, 16 to 20A international) is used on the phase conductors (all current-carrying conductors). The fuse or circuit breaker must have adequate safety approvals recognized by the country of usage.**

Waarschuwing **Dit product is afhankelijk van de installatie van het gebouw voor bescherming tegen kortsluiting (overstroom). Zorg ervoor dat de zekering of stroomonderbreker die gebruikt wordt niet groter is dan 120 V~, 20 ampère in de V.S. of 240 V~, 16-20 ampère internationaal op de fasegeleiders (alle stroomdragende geleiders). De zekering of stroomonderbreker dient de juiste veiligheidsgoedkeuringen te hebben in het land waarin het gebruikt wordt.**

Varoitus **Tämä tuote on riippuvainen rakennuksen oikosulkusuojauksesta (ylivirtasuojauksesta). Varmista, että vaihejohtimissa (kaikissa jännitteellisissä johtimissa) käytetään alle 240 V vaihtovirran, 16–20 ampeerin (kansainvälinen) tai 120 V vaihtovirran, 20 ampeerin (Yhdysvallat) sulaketta tai virtakytkintä. Sulakkeessa tai virtakytkimessä on oltava käyttömaassa tunnistetut, riittävät turvahyväksynnät.**

Attention **Pour la protection contre les courts-circuits (surtension), ce produit utilise les dispositifs intégrés au bâtiment. Assurez-vous qu'un fusible ou un disjoncteur est utilisé sur les conducteurs de phase (tous les conducteurs porteurs de courant). Le fusible ou le disjoncteur (maximum 240 V CA, 16 à 20 A [aux USA, maximum 120 V CA, 20 A]) doit être conforme aux normes de sécurité en vigueur dans votre pays.**

Warnung **Diese Produkt erfordert eine Gebäudeabsicherung gegen Kurzschluß (Überstrom). Achten Sie darauf, daß auf den Phasenleitern (allen stromführenden Leitern) eine Sicherung oder ein Schaltkreisunterbrecher verwendet wird, der nicht größer ist als 120VAC, 20A U.S. (240VAC, 16 bis 20A international). Die Sicherung oder der Schaltkreisunterbrecher muß angemessenen Sicherheitsvorschriften genügen, die den Bestimmungen des Anwendungslandes entsprechen.**

Avvertenza **La protezione di questo prodotto da cortocircuiti (sovracorrente) dipende dall'impianto elettrico dell'edificio. Assicuratevi che un fusibile o interruttore di circuito con meno di 120VAC, 20A U.S. (240VAC, da 16 a 20A internazionale) venga utilizzato sui conduttori di fase (tutti i conduttori di corrente elettrica). Il fusibile o interruttore di circuito deve rispondere alle specifiche di sicurezza invigore nel paese dove viene utilizzato.**

Advarsel **Dette produktet er avhengig av bygningens installasjoner for overstrømsbeskyttelse (kortslutning). Kontroller at det ikke brukes en sikring eller overbelastningsbryter som er større enn 120 V, 20 ampere i USA, eller 240 V, 16 til 20 ampere internasjonalt, på faselederne (alle strømførende ledere). Sikringen eller overbelastningsbryteren må være sikkerhetsgodkjent i det aktuelle landet der den skal brukes.**

Aviso **Este dispositivo depende das instalações existentes para protecção contra curto-circuitos (sobrecarga). Assegure-se de que utiliza um fusível ou um disjuntor com uma capacidade não superior a 120VAC, 20A U.S. (240VAC, 16 a 20A internacional) nos condutores de fase (todos os condutores de corrente). O fusível ou disjuntor deverá possuir as necessárias aprovações de segurança por parte das autoridades locais.**

¡Advertencia! Este producto ha sido diseñado teniendo en cuenta que la instalación del edificio contará con protección contra cortocircuitos (sobrevoltajes). Asegúrese de que se usa un fusible o cortacircuitos no superior a 120VAC, 20A en los EE.UU. (240VAC, de 16 a 20A en el resto de países) en los conductores de fase (todos los conductores de transporte corriente). El fusible o cortacircuitos debe contar con las aprobaciones de seguridad adecuadas y reconocidas por el país en el que vayan a usarse.

Varning! Denna produkt förlitar sig på att byggnadens installation är försedd med skydd mot kortslutning (överström). Se till att en säkring eller ett överspänningsskydd för högst 120 V~, 20 A USA (240 V~, 16 – 20 A internationellt) används på fasledarna (alla strömförande ledare). Säkringen eller överspänningsskyddet måste ha fullgoda säkerhetstillstånd som erkänns av användningslandet.

TN Power Systems Warning

Warning The device is designed to work with TN power systems.

Waarschuwing Het apparaat is ontworpen om te functioneren met TN energiesystemen.

Varoitus Koje on suunniteltu toimimaan TN-sähkövoimajärjestelmien yhteydessä.

Attention Ce dispositif a été conçu pour fonctionner avec des systèmes d'alimentation TN.

Warnung Das Gerät ist für die Verwendung mit TN-Stromsystemen ausgelegt.

Avvertenza Il dispositivo è stato progettato per l'uso con sistemi di alimentazione TN.

Advarsel **Utstyret er utfomet til bruk med TN-strømsystemer.**

Aviso **O dispositivo foi criado para operar com sistemas de corrente TN.**

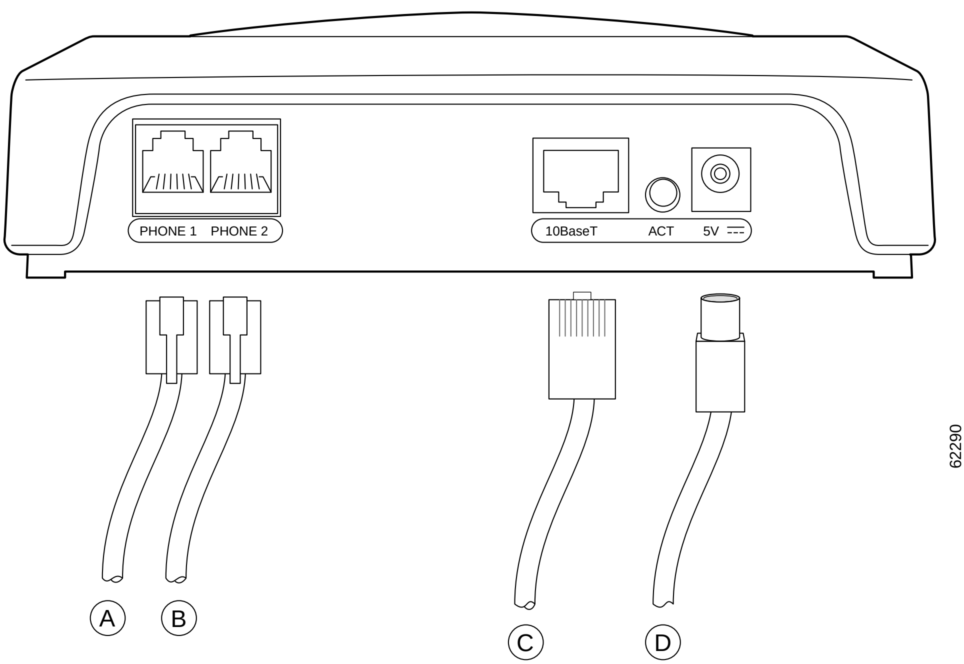
¡Advertencia! **El equipo está diseñado para trabajar con sistemas de alimentación tipo TN.**

Varning! **Enheten är konstruerad för användning tillsammans med elkraftssystem av TN-typ.**

Installing the Cisco ATA 186

Follow these steps to install the Cisco ATA 186 hardware. (See Figure 2-2.)

Figure 2-2 Installing the Cisco ATA 186



A—Phone 1 connector (RJ-11)

B—Phone 2 connector (RJ-11)

C—10Base-T Ethernet connector

D—12V power connector

Step 1 Place the Cisco ATA 186 near an electrical outlet. Connect the first telephone to the PHONE 1 input port (A) on the rear panel of the Cisco ATA 186 by using a telephone line cord with an RJ-11 connector. The PHONE 1 input port will be the primary telephone line.

Caution Do not connect the Cisco ATA 186 PHONE input ports to the telephone wall jack. To prevent damage to the device or building telephone wiring, connect each Cisco ATA 186 PHONE port to a telephone only, never to a telephone wall jack.

Step 2 You can connect a second telephone to the PHONE 2 input (B) by using a second telephone line cord. The PHONE 2 input is the secondary telephone line.

Note If you are connecting only one telephone to the Cisco ATA 186, you must use the PHONE 1 input; otherwise, the telephone cannot place calls.

Step 3 Connect one end of a 10-BaseT Ethernet cable (C) to a hub, switch, or broadband modem (DSL, cable, and so on).

Step 4 Connect the other end of the Ethernet cable to the RJ-45 input port (C) on the rear panel of the Cisco ATA 186.

Note Use a crossover Ethernet cable to connect the Cisco ATA 186 to another Ethernet device (such as a router or PC) without using a hub. Otherwise, use a straight through Ethernet cable.

Step 5 Plug the AC power adaptor into an electrical outlet. Insert the power cord into the rear panel of the unit (D). Each connected telephone should ring once, indicating that the Cisco ATA 186 is powered and ready to use. When the Cisco ATA 186 is properly connected and powered up, the green activity LED flashes. The activity LED, labeled ACT, indicates network activity.

Caution To prevent overheating during operation, do not cover or block the air vents in the top panel of the Cisco ATA 186.

If installation was successful, proceed to Chapter 3, "Configuring the Cisco ATA 186."

Verifying the Installation

If the phone does not ring and the function button does not flash after power-up, check that the power adaptor is plugged into a working electrical outlet and that the power cord is pushed securely into the connector. Additionally, verify that the Ethernet connection is secure.

CHAPTER 3

Configuring the Cisco ATA 186

This chapter provides information about configuring the Cisco ATA 186.

There are three ways to configure your Cisco ATA 186:

- Voice Configuration Menu
- Web browser
- Autoprovisioning

Configuration Requirements

The Cisco ATA 186 requires the following minimum settings for network connectivity:

- IP address
- Network route address (IP gateway)
- Subnet mask

Note To enter an IP address, press the * key to indicate a delimiter (dot). For example, `192*168*3*1`. To hear the IP address of your Cisco ATA 186, after your Cisco ATA 186 has been installed correctly, lift the telephone handset and enter **21#**.

These settings are automatically configured in a DHCP network. When the Cisco ATA 186 is downloading its DHCP configuration or software upgrade, the function button on the top panel blinks.

Caution Do not unplug the device while the function button is blinking. Doing so can cause permanent damage to the device.

Note If there is no DHCP server and the Cisco ATA 186 is programed to find one, the function button will keep blinking.

About Using the Voice Configuration Menu

Some IVR menu options will require you to enter alphanumerica characters. Alphanumeric entry differs from numeric entry in that you must enter the # key after each character selected. Using Table 3-1 as a guide, enter the appropriate number key on your telephone handset as many times as needed to select the number, letter, or symbol you want. For example, to enter 58sQ, you would enter:

```
5 # 8 # 7 7 7 7 7 # 7 7 7 7 7 7 7 # #
```

If you need to enter an alphanumeric value, the voice prompt will specifically tell you to enter an alphanumeric value; otherwise, you should enter a numeric value (0-9).

Table 3-1 lists the keys and their respective alphanumeric characters.

.

Table 3-1 Alphanumeric Characters

Key	Characters
1	1 ./_\ @@*space return +-!,?|~^ ##=$" ' ``%<>[ ] ::;{}()&
2	2 a b c A B C
3	3 d e f D E F
4	4 g h i G H I
5	5 j k l J K L
6	6 m n o M N O
7	7 p q r s P Q R S
8	8 t u v T U V
9	9 w x y z W X Y Z
0	0
*	. (delimeter)

The voice will repeat the value you entered; then, prompt you to press one of the following keys:

- 1=change your entered value
- 2=review your entered value
- 3=save your entered value
- 4=review the current saved value

Using the Voice Configuration Menu

To manually configure the Cisco ATA 186 by using the interactive voice response (IVR) system and telephone keypad, follow the steps in this section. For a list of the available configuration options, see Appendix A, "Voice Menu Options."

Step 1 Lift the handset and press the Function button located on top of the Cisco ATA 186.

Step 2 Enter the IVR menu number for the parameter that you want to configure or the command that you want to execute; then, press the pound key (#). (See Appendix A, "Voice Menu Options," and Appendix B, "Parameters and Defaults," for lists of options and their corresponding menu numbers.)

Step 3 Follow the applicable prompts.

Step 4 When you have finished, make sure you press **3** to save your changes.

Step 5 Hang up the telephone. The Cisco ATA 186 resets.he function button will fast-blink when the reset is complete.

Using the Web Configuration Page

You can configure your Cisco ATA 186 by using the web configuration page. Each configurable parameter is listed, and parameters are grouped and color-coded according to their function.

Caution Cisco recommends you do not perform the initial configuration over the Internet; the configuration page is not secure. After you have configured the parameters, password-protect the user interface to prevent it from being accessed on the Internet. The parameter for password-protecting this page is UIPasswd.

Follow these steps:

Step 1 Make sure that the PC and the Cisco ATA 186 are already networked and visible to one another.

Step 2 Open your web browser.

Step 3 Enter the URL of your configuration page. The URL of the web server is:

IP Address/**dev**

For example, the configuration page for a Cisco ATA 186 with the IP address 192.168.3.225 is:

`http://192.168.3.225/dev`

See "Using the Web Configuration Page" section on page 3-4 for information on how to find the IP address of your Cisco ATA 186.

Step 4 Select the values for the items that you want to configure. (See Appendix B, "Parameters and Defaults," for a list of options.) Scroll down to see all parameters. You can password-protect the user interface for security.

Step 5 Click **apply** to save your changes.

Step 6 Close your web browser.

About Autoprovisioning

For large-scale networks, you can use a TFTP server to host a profile for each Cisco ATA 186. The TFTP server's URL and file name can be provided (provisioned) from the DHCP server.

Set UseTFTP to 1.

TftpURL is the IP address of the TFTP server. If TftpURL is set to 0, the DHCP server will supply the IP address.

Name the file to be downloaded according to the format:

ata*xxxxxxxxxxxx*

Each ***xx*** is the 2-digit lower case hexadecimal representation of each integer in the MAC address of the Cisco ATA 186.

For example, for a Cisco ATA 186 with a MAC address of 0.1.45.2.10.20, the file name is:

```
ata00012D020A14
```

The filename has a fixed length of 15 characters, regardless of the MAC address.

In this mode of provisioning, at power-up, the Cisco ATA 186 contacts the TFTP server for a specific profile to download. The profile can be encrypted with a shared secret key. If the Cisco ATA 186 does not reach the TFTP server after 3 attempts, it continues normal operation by using its locally cached profile.

Note If the ToConfig value is set to **1**, the Cisco ATA 186 will contact the TFTP server continuously without waiting until the next CfgInterval. The value for CfgInterval is the value of CfgInterval in seconds.

At CfgInterval, the Cisco ATA 186 attempts to refresh its profile from the TFTP server.

You can configure the Cisco ATA 186 to refresh earlier than the scheduled CfgInterval by opening a refresh web page on the Cisco ATA 186. The refresh page is:

http://*ipaddress*/refresh

For example, for a Cisco ATA 186 whose IP address is 192.168.2.170, the refresh page would be:

```
http://192.168.2.170/refresh
```

If you are using TFTP when the Cisco ATA 186 is plugged in, the Cisco ATA 186 will try to contact the TFTP server to download its configuration. This method is not secure unless you are using EncryptKey. See the “Encrypt Key” section on page 3-8.

cfgfmt.exe and ptag.dat Files

Bundled with the Cisco ATA 186 software is the program cfgfmt.exe and the file ptag.dat. These should be placed the directory used to store the files for transfer using TFTP. The cfgfmt program is used to convert a text-based user profile for the Cisco ATA 186 to a binary file sent by the TFTP server to the Cisco ATA 186 to update its configuration parameters. The cfgfmt.exe program is used with the following syntax:

cfgfmt [-eRC4Password] [-tPTagFile] input output

- **eRC4Password** is the optional RC4 key to encrypt the binary TFTP file provided by the cfgfmt program
- **tPTagFile** is the optional command used to specify a ptag file other than the one provided (ptag.dat)
- **input** is the name of the text-based profile of the Cisco ATA 186 that will be converted to a TFTP binary file
- **output** is the name of the TFTP binary file produced by the cfgfmt program

Updating the Profile from the TFTP Server

To update the Cisco ATA 186 profile from the TFTP server before the CFGINTERVAL expires, open your web browser and enter:

http://*ipaddress*/refresh

where ***ipaddress*** is the IP address of the Cisco ATA 186 you want to update. The Cisco ATA 186 responds with an **ok** page if idle; otherwise, it responds with a **later** page.

If you have physical access to the Cisco ATA 186, you can power cycle the Cisco ATA 186 to update the profile from TFTP server.

About Profile and Configuration Security

This section includes information on passwords and other security methods.

Passwords

To password-protect your Cisco ATA 186:

Step 1 Set the **UIPassword** parameter to a numeric password by using the web server interface or TFTP profiling.

Step 2 You will be prompted for a password when you try to access the web server or a configurable IVR parameter.

- In web server mode, enter the password in the UIPassword field of the password challenge page.
- In IVR mode, enter the password, followed by the # key, at the **p-a-s-s-w-d** prompt.

Encrypt Key

Encrypt Key encrypts binary files being transferred over TFTP. You can change this key for each Cisco ATA 186, so that only one particular box can decode the information. You can change the encrypt key, using the IVR or web interface.

See the "cfgfmt.exe and ptag.dat Files" section for more information.

The Cisco ATA 186 polls the server at intervals set in CfgInterval to see if it needs to be upgraded. You can customize this service. For example, you can route all calls from a particular Cisco ATA 186, based on the Gatekeeper ID, to an operator.

About Using the DHCP Server

DHCP option 60, DHCP_VENDOR_CLASS_ID, is set to the value **ATA186** so that the DHCP server can identify a Cisco ATA 186.

Parameters that you can set using DHCP are:

- Client IP address
- Client Subnet mask—DHCP option 1
- Routers on the client's subnet—DHCP option 3
- Domain name servers—DHCP option 6

  (The Cisco ATA 186 takes up to two DNS servers)

- Network time protocol (NTP) servers—DHCP option 42

  (The Cisco ATA 186 takes up to two NTP servers)
- TFTP server name—DHCP option 66

DNS, TFTP, and NTP servers can be overwritten by the value of the corresponding parameters in the local box profile (for example, the DNS1IP, DNS2IP, TftpURL, and NTPIP parameters).

If you are not using DHCP, you must manually enter the IP address, network route address, and subnet mask.

Configuring Codec Options

You can configure the various Codec call options for use with the Cisco ATA 186.

Note The Cisco ATA 186 can support two simultaneous G.723 calls or one G.729A call. When using G.729A, the second line must use G.711 u-law or a-law. The default voice codec is G.723.

Step 1 To select G.723 as the preferred low-bit-rate codec (LBRCodec) for receive and transmit modes, enter **0** into the LBCodec field on the web page. To select G.729A, enter **3**.

Step 2 To select G.723 as the preferred receive codec (RxCodec) enter **0** into the RxCodec field on the web page. To select G.729A, enter **3**.

Step 3 To select G.723 as the preferred transmit codec (TxCodec), enter **0** into the TxCodec field on the web page. To select G.729A, press **3**.

CHAPTER 4

Protocol-Specific Configurations

This chapter contains information on selecting protocols and services for your system.

About Signaling Protocols

You can select either H.323 or Session Initiation Protocol (SIP) as the operating signaling protocol for the Cisco ATA 186. Both signaling protocols offer optional network control servers. With H.323, pre-call and call control services are offered by a gatekeeper, while in SIP, a proxy server can receive call transaction requests and return responses on behalf of the Cisco ATA 186.

Some parameters and supplementary services are available with SIP, some are available only with H.323, and others are available with both protocols.

About H.323-Specific Configurations

The Cisco ATA 186 uses ITU H.323, Version 2 as the default signaling protocol.

When operating in H.323 mode, the Cisco ATA 186 registers with a gatekeeper to handle call control services. A full registration request (RRQ) is performed at power-up. In order to let the gatekeeper know it is still on the network, the Cisco ATA 186 periodically refreshes this registration by sending an abbreviated RRQ. The value of the GKTimeToLive configuration parameter determines the period between refreshes, in seconds.

To use the H.323 security features, you must specify the level of authentication by means of the AutMethod configuration parameter. The settings are as follows:

- 0—no authentication
- 1—Cisco registration level
- 2—Cisco admission level

Note Make sure these levels are also enabled on the gatekeeper and gateway.

About Gatekeeper Requirements for H.323

The gatekeeper must meet these requirements:

- It must be H.323- or SIP-complaint.
- It must run the applicable version of Cisco IOS software for the features and protocol you want to use.
- It must support H.323 or SIP, but only one at a time.

Note No specific configuration is required; configure the gatekeeper as you would for any IP phone or Voice over IP (VoIP) configuration. The default configuration is IP routing off.

Enabling IP Routing

To enable IP routing so that you can run Cisco IOS software, enter:

ip routing

Connecting to a Network Time Protocol Server

If you want to use Caller ID (SIP) or security features (H.323), connect the gateway to a functioning network time protocol (NTP) server. When using Cisco IOS, enter:

ntp server ***ip_address***

clock timezone PST -8

Note For information on how to access accounting information, see your Cisco IOS documentation.

Using ISDN/EI

If you are using ISDN, the requirements for E1 are:

```
ISDN switch-type primary-5ess
voice rtp send-rcv
```

Configuration depends on your WAN connection. The following example shows a 24-channel PRI connected to a T1 VIC slot:

```
controller T1 0
framing esf
clock source linelinecode b8zs
pri-group timeslots 1-24
```

Manipulating the Dial String

You can configure the Cisco ATA 186 to add digits to the beginning of the outgoing dial string.

Note The default for dial string manipulation is off.

Follow these steps to enable dial string manipulation by using prepending:

Step 1 From the web server interface, enter **4** in the Authenticate Method field.

Step 2 Enter the digits to be prepended in the associated PIN field.

For example, if you enter the dial string 5551212 in line 0 and the Cisco ATA 186 is configured with 1234 in the PWD0 field, the outgoing dial string is 12345551212.

Configuring Security Levels

Note This is a Cisco Proprietary H.235 implementation; it uses the Cisco access/clear token structure rather than the VocalTec crypto token structure.

To configure the Cisco ATA 186 to use Cisco Registration-Level Security (or Admission-Level Security) in H.323 mode, follow these steps:

Step 1 Add **AutMethod** to match the web interface.

Step 2 Change the PIN globally to **PWD**.

Step 3 Set USELOGINID to **1**. (0 indicates LOGINID0 and LOGINID1 fields are not used; 1 indicates LOGINID0 and LOGINID1 fields are used for H.323 registration.)

Step 4 Set UID0 and UID1 to the correct E.164 IDs.

Step 5 Set LOGINID0 and LOGINID1 to the H.323 Login IDs.

Step 6 Set PWD0 and PWD1 to the correct passwords/PINs (passwords for RADIUS servers).

Step 7 Set AUTMETHOD to **1** or **2** (0 indicates no authentication; 1 indicates Cisco Registration Level Security; 2 indicates Cisco Admission Level Security).

Step 8 If the DHCP server does not provide an IP address, set NTPIP to the NTP server IP address.

About Alternate Gatekeepers and RAS

You can configure and accept acknowledgements from up to four Alternate (backup) Gatekeepers (AltGK).

The Cisco ATA 186 allows you to configure an Alternate Gatekeeper as a backup to the primary Gatekeeper. The Cisco ATA 186 can accept up to four dynamic Alternate Gatekeepers configured by the H.225 RAS messages. It accepts both temporary and permanent Alternate Gatekeepers.

When an Alternate Gatekeeper list is received with an H.225 RAS message, the secondary Gatekeeper is merged and sorted with the dynamic Alternate Gatekeepers. The secondary gatekeeper is kept and placed with the lowest priority. To allow the Cisco ATA 186 to switch back to the primary gatekeeper automatically, set a timeout value in seconds, AltGkTimeOut, to enable the feature if non-zero value is used. The Cisco ATA 186 supports the Alternate Gatekeeper list in the GCF/GRJ, ACF/ARJ, RCF/RRJ, and DRJ RAS messages. See Appendix B, “Parameters and Defaults” for more information.

About SIP-Specific Configurations

The Session Initiation Protocol (SIP) is a text-based IETF-defined protocol for establishing call sessions. To enable SIP on the Cisco ATA 186, you must set the UseSIP configuration parameter to 1.

Session Description Protocol (SDP) is the part of the SIP message that establishes parameters.

If you are using SIP, you must register with a proxy server. If you do not remember the IP address, you can use the UserID, that is, the telephone number.

About Using a Proxy Server with SIP

When using a Proxy Server, the Cisco ATA 186 sends a call to request a User ID (telephone number). The proxy server holds a table of telephone numbers and their equivalent IP addresses. You can enter this table manually (static), or the Cisco ATA 186 can dynamically send the IP address to the proxy server when it starts. This process is called Registration. To use this process, SIPRegOn must be set to 1. The default is 0 or Off.

Registration has an expiration timeout, so the Cisco ATA 186 must periodically resend its request. The interval is the number of seconds based on the value of SIPRegInterval.

If OutboundProxy is enable all messages go through the IP address or URL listed. The proxy routes to the end IP address.

About NAT Support with SIP

The following three parameters allow the Cisco ATA 186 to be used behind a firewall or a network address translation (NAT) device to access a SIP proxy server or user agent outside the firewall to register and make or receive calls. A firewall or NAT device has a WAN IP address which is reachable from outside the firewall, and a LAN IP address which is reachable from inside the firewall.

SIP Port

This is the logical UDP port where the device is listening for SIP messages. The standard port number is 5060.

Note Do not use reserved ports, such as 80 (HTTP).

To specify a port for the gatekeeper or proxy server, you can use either the IP address or the URL. If no port is specified, the device uses the default of 5060.

If you want to specify a port with the IP address, use a dot (.):

123.10.10.10.5061

With a URL, use a colon (:):

www.company.com:5061

MediaPort

The MediaPort configuration parameter specifies which base port the Cisco ATA 186 uses to receive RTP media streams. This parameter is only used for the purposes of SIP support behind a NAT.

NATIP

The NATIP configuration parameter specifies the WAN IP of the NAT. Specify the LAN IP in the StaticRoute configuration parameter.

CHAPTER 5

Configuring Supplementary Services

This chapter includes information on supplementary services for the applicable protocols. Supplementary services are services that you can use to enhance your telephone service. The features must be enabled in:

- Paid Services - the features your customers subscribe to.
- Call Features - the services you are subscribed to. You can turn these features on and off.

Changing Call Commands

To change the command for a supplementary service (for example, to change *69 to *100), change the context identifiers by changing the call command field in the web browser on the provisioning page.

Note You cannot change supplementary services by using the IVR, and you cannot change timeout values.

Cancelling a Supplementary Service

Some features, such as call waiting, are disabled automatically when you hang up. You can also deactivate them manually by pressing *70 before a call. In the case of call waiting, the second caller hears a busy signal. You can also configure your system to have the services disabled by default and enabled on a call-by-call basis. This is done in the 32-bit “Call Features” plan.

Common Supplementary Services

The supplementary services described in this section can be used with both the H.323 and SIP signaling protocols. Available services and their configuration and implementation depend on the country system you are using.

About 3-Way Calling (Conference Calling)

3-way calling works slightly differently on Swedish and U.S. systems.

Making a 3-Way Call in the U.S.

For U.S. configurations, follow these steps to make a conference call:

Step 1 Dial the first number.

Step 2 When the person you called answers, perform a "hook flash"; that is, hang up quickly and pick up the phone quickly. This will put the first person you called on hold and you will hear a dial tone.

Step 3 Dial the second person and speak normally when that person answers.

Step 4 To conference with both callers at the same time, perform a hook flash and press **3** on the telephone handset.

Step 5 To drop the second call, perform a hook flash.

Step 6 (Optional) To conference in additional callers, the last person called with a Cisco ATA 186 can call an additional person, and so on. This is known as "daisy-chaining."

Making a 3-Way Call in Sweden

For Swedish configurations, follow these steps to make a conference call:

Step 1 Dial the first number.

Step 2 When the person you called answers, perform a "hook flash"; that is, hang up quickly and pick up the phone quickly. This will put the first person you called on hold and you will hear a dial tone.

Step 3 Dial the second person and speak normally when that person answers.

Step 4 Perform a second hook flash and press **2** on the telephone handset to get back 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 and press **3** on the telephone handset. Once you conference all three callers, the only way to drop a caller is for that caller to hang up.

Step 6 (Optional) To conference in additional callers, the last person called with a Cisco ATA 186 can call an additional person, and so on. This is known as "daisy-chaining."

About Call Waiting

This section describes the Call Waiting feature.

Call Waiting in the U.S.

If someone calls you while you are speaking on the telephone, you can answer by performing a hook flash. You cannot conference in all 3 callers, but the first person you called could call someone else and "daisy-chain" them in. If there is no answer after one minute, the caller will hear 3 beeps and a busy signal.

When the Cisco ATA 186 is configured to use Call Waiting by default, press ***70** to disable Call Waiting for the duration of the next call.

Call Waiting in Sweden

If someone calls you while you are speaking on the telephone, you can answer by performing a hook flash and pressing **2** to answer or **3** to conference them with the person you are already speaking to. You can also press **3** later during the call to conference.

Performing a hook flash and pressing **1** hangs up the first caller and answers the second call. If there is no answer after one minute, the caller will hear 3 beeps and a busy signal.

To enable Call Waiting for Sweden, press ***43#**. When the Cisco ATA 186 is configured to use Call Waiting by default, press **#43#** to disable Call Waiting for the duration of the next call.

About Call Forwarding

In H.323, it is necessary to have a gatekeeper capable of handling call forwarding and call return supplementary services. In SIP, the Cisco ATA 186 handles these services.

There are 3 types of call forwarding.

- Forward Unconditional—forwards every call that comes in.
- Forward When Busy—forwards calls when the line is busy.
- Forward on No Answer—forwards calls when the telephone is not answered after the configured period of 0-63 seconds.

You can activate only one of these services at a time.

Forwarding Calls in the U.S.

Forward Unconditional

Press **#72**, the number you want to forward call to; then press **#** again.

Forward When Busy

Press **#74**, the number to forward the calls to; then press **#** again.

Forward On No Answer

Press **#75**, the number you want to forward the calls to; then press **#** again.

Cancelling Call Forwarding

To cancel call forwarding, press **#73**.

Forwarding Calls in Sweden

Forward Unconditional

Press ***21*v#**, the number you want to forward call to; then press **#** again. To cancel, press **#21#**.

Forward When Busy

Press ***67*v#**, the number to forward the calls to; then press **#** again. T0 cancel, press **#61#**.

Forward On No Answer

Press ***61*v#**, the number you want to forward the calls to; then press **#** again. To cancel, press **#67#**.

About Call Return

Call return allows you to call the last person who called you.

Returning Calls in the U.S.

Press ***69** to activate Call Return.

Returning Calls in Sweden

Press ***69#** to activate Call Return.

About Calling Line Identification Presentation

Calling Line Identification Presentation (CLIP) shows your identity to callers with Caller ID.

Calling Line Identification Presentation in the U.S.

Press ***82** to activate CLIP.

Calling Line Identification Presentation in Sweden

Calling Line Identification Presentation (CLIP) is not available for Sweden.

About Calling Line Identification Restriction

Calling Line Identification Restriction (CLIR) hides your identity from callers with Caller ID.

Calling Line Identification Restriction in the U.S.

In the U.S., press ***67** to activate CLIR. This feature is disabled when you hang up.

Calling Line Identification Restriction in Sweden

In the U.S., press ***31#** to activate CLIR. This feature is disabled when you hang up.

Upgrading the Cisco ATA 186 Software

You can upgrade the program of your Cisco ATA 186 remotely. Upgrades can be done by using either of two methods:

- executable file
- TFTP server

Note TFTP is case sensitive; if you use TFTP to upgrade your software, name your files by using all lowercase letters.

Both methods require software available from cisco.com.

About Upgrading from Release 1.*xx* to Release 2.0

Caution Release 2.0 is not backward-compatible with Release 1.*xx*.

Note When upgrading from v1.34 to v2.0, the Cisco ATA 186 does not prompt the user with the **Upgrade Successful** message when finished upgrading. Instead, observe the function button LED. The LED will flash on and off during the upgrade; then, change to a solid on state; then, turn off when the Cisco ATA 186 has finished upgrading.

About the Software Upgrade Process

Note The following information is optional detailed information about the upgrade process.

The available options when using the software upgrade are:

ata186us -h[host_ip] -p[port] -quiet ***image file***

- **-h**[*host_ip*]—if the host has more than one IP address, use this option to set a specific host IP address. The default is to use the first IP address obtained by **gethostbyname**.
- **-p**[*port*]—set the server port to specific port (default is 8000; use a different port only if you are setting up an IP directed upgrade server other than the default).
- **-quiet**—set to quiet mode; that is, send all output to a log file named *port*.log. This option is useful when the upgrade server is run as a daemon.
- **-any**—allow upgrade even if the software version is older than or the same as that of the client Cisco ATA 186.
- **-any2**—allow upgrade regardless of software type and version.
- **-d1**,**-d2**,**-d3**—set level of detailed description for debugging.

Example:

```
ata186us -h123.456.789.100 -p5786 -quiet yj2e112.kup yj2e112.el
```

Upgrading the Software by Using the Executable File

To upgrade your Cisco ATA 186 by using the ata186us.exe file and the voice menu, you need the following:

- A PC running Windows 9X/ME/NT/2000
- A network connection between the PC and the Cisco ATA 186

To upgrade, follow these steps:

Step 1 In Windows, click **Start** > **Programs** > **MSDOS Command Prompt** and locate or create a directory to store the file.

Step 2 After you unzip the files on the PC, save the executable file, the software image (the software image will have a ".zup" extension) to a directory on your PC. The images to save are:

- ata186us.exe
- ata186-v***x***.zup

where ***x*** is the version number and date of the software release; for example, ata186-v2-00-0522b.zup.

Step 3 At the DOS prompt of the directory where the files are saved, enter:

ata186us *software_file_name*. zup -d1 -any2

Step 4 Press the function button to go to the configuration menu.

Caution Step 5 will begin the upgrade process. Do not cancel the process or unplug the power cord at any time during the upgrade process; doing so can permanently damage your Cisco ATA 186. Wait until after the button stops flashing and the light has turned off.

Step 5 Using the dialpad of your telephone attached to your Cisco ATA 186, enter:

100# *ip_address_of_PC* ***8000#**

For example, if the IP address of your PC is 192.168.1.10, enter:

100#192*168*1*10*8000#

Note Press 123# to hear your code's version number.

You can later verify that you have upgraded your Cisco ATA 186.

Timesaver When upgrading many Cisco ATA 186s, you can save the software upgrade commands in your telephone's speed-dial and use them after accessing the Cisco ATA 186 voice menu.

Upgrading the Software from a TFTP Server

You can configure the Cisco ATA 186 to obtain its configuration from a TFTP server. The configuration file (profile) is unique to each Cisco ATA 186 because each profile name incorporates the unique MAC address associated with that Cisco ATA 186. The profile can be configured to instruct the Cisco ATA 186 to automatically upgrade its firmware image by using TFTP download without user input.

The profile generation tool and example is available on the Cisco ATA 186 software download page of cisco.com.

To use the TFTP server to upgrade the Cisco ATA 186's software, set the UPGRADECODE (or UPGRADELANG if you are upgrading the language) parameter in the Cisco ATA 186's user profile to the following:

3,0x301,0x0400,0x0200,*tftp_server_ip***,69,***n***,***filename_to_be_downloaded*

- *tftp_server_ip* is the TFTP server where you can download *file_name_to_be_downloaded*
- *file_name_to_be_downloaded* is the name of the TFTP upgrade method firmware image
- n is the software image release I.D. in the readme file that came with the TFTP upgrade method firmware image.

Note To help you identify the correct files to place on your TFTP server, the TFTP-method file containing the Cisco ATA 186 software has a ".kxz" extension.

Using the FAX Passthrough Feature

Both ports of the Cisco ATA 186 support FAX transmission. The Cisco ATA 186 can send FAXes by either of two methods:

- FAX passthrough—In this mode, the Cisco ATA 186 can detect a FAX tone, after which it will turn off FAX tone detection and silence suppression and switch its operating codec to G.711 u-law or G.711 A-law.
- G.711 FAX mode—In this mode, the Cisco ATA 186 simply passes the RTP packets sent between the end FAX machines without any intervention. It treats the FAX session like any normal voice call.

Note Limited fax passthrough support is available (up to 9.6 kbps fax transmission rates for most fax machines). Extended support is planned. Please check release notes and product bulletins for updates.

Both the Cisco ATA 186 and the supporting gateway must be configured appropriately to handle FAX signals.

About FAX Passthrough

To use the FAX passthrough feature, both the Cisco ATA 186 and the supporting gateway must be configured at both ends to detect the FAX tone and switch codec.

The Cisco ATA 186 supports two G.711 upspeed methods for FAX passthorough:

- The Cisco proprietary codec switch using NSE packets and RTP detection.

- The standard-based protocol level (H.323/SIP) codec switch.

To interoperate with a Cisco gateway, which does not send protocol level codec switch requests but can accept them, select the Cisco proprietary codec switch.

Note When you are using G.729A on one port, the other port automatically uses the G.711u-law or G.711A-law codec.

About Configuring the Cisco ATA 186 for FAX Passthrough

There are two possible modes of configuration, ConnectMode and AudioMode.

ConnectMode

The ConnectMode bits in the Cisco ATA 186 configuration control the functions described below. To interoperate with a Cisco gateway, Cisco recommends you set the bits as follows:

- Bit 7 (mask 0x80)—1 means enable FAX passthrough redundancy; 0 means disable.
- Bits 8-12 (mask 0x1F00)—Offset to NSE payload-type number 96. The default value is 4 (as NSE payload-type number 100), and the valid values are 2 to 23 (98 to 119).
- Bit 13 (mask 0x2000)—0 means use G.711u-law; 1 means using G.711A-law as the new codec.
- Bits 14, 15 (mask 0xC000)—Set as 00 to enable the FAX passthrough feature using the Cisco proprietary method (recommended). Set as 11 to disable FAX passthrough.

For example, a example, a *ConnectMode* setting of `0xXXXX040X` means use NSE payload-type number 100, G.711u-law codec, and no redundancy in FAX passthrough mode.

AudioMode

Set to 0xXXXX0015 for line 1, or 0x0015XXXX for line 2, (X=don't care). This setting enables FAX tone detection.

About Configuring the Cisco Gateway for FAX Passthrough

The supporting Cisco gateway can enable FAX passthrough using system level or dial-peer level commands.

System-Level Command

Enable the FAX passthrough feature using the system level commands:

voice service voip
modem passthrough NSE [payload-type number] codec {g711ulaw | g711alaw} [redundancy] [maximum-sessions value]

- The default payload-type parameter is 100. Valid values are 98 to 119. The NSE payload number must be the same on both the Cisco ATA 186 and the Cisco Gateway.
- The codec must be G.711u-law for T1 or G.711A-law for E1.
- The redundancy parameter enables RFC 2198 packet redundancy. It is disabled by default.
- The maximum sessions parameter defines the number of simultaneous FAX Passthrough calls with redundancy. The default is 16; valid values are 1 to 26.

On the Cisco ATA 186, turn off bits 14 and 15 in ConnectMode. This enables sending FAX passthrough signals and detection of incoming FAX passthrough signals using the Cisco proprietary method.

The NSE payload-type number, FAX passthrough codec (G.711u-law or G.711A-law) and redundancy must use the same settings in the Cisco ATA 186 and the supporting Cisco gateway.

Dial-Peer Level Command

The FAX passthrough feature can be enabled for communication between the Cisco gateway and the specified Cisco ATA 186 using the dial-peer level commands:

dial-peer voice *tag* voip

modem passthrough { NSE [payload-type *number*] codec {g711ulaw | g711alaw} [redundancy] | system}

The default is:

modem passthrough system

When using the default (system) configuration, the dial-peer FAX passthrough configuration is defined by the configuration in voice service voip. When system is used, no other parameter is available.

When the NSE is configured in the FAX passthrough command in the dial-peer level, the FAX passthrough definition in dial-peer takes priority over the definition in voice service voip. The payload-type number, codec, and redundancy parameters can also be used.

For example,

modem passthrough NSE codec g711ulaw

means use the NSE payload-type number 100, G.711ulaw, no redundancy in the fax passthrough mode.

Disabling the FAX Passthrough Feature

To disable FAX passthrough, turn on bits 14 and 15 of the ConnectMode in the Cisco ATA 186 configuration. On the Cisco gateway, enter:

no modem passthrough

About FAX Mode

Some users might want to set one or both lines of the Cisco ATA 186 to pure G.711 FAX mode. This mode allows the FAX machine connected to the Cisco ATA 186 to communicate directly with the FAX machine at the other end of the call with no FAX signaling events between the two FAX machines.

Configuring the Cisco ATA 186 for FAX Mode

You can configure the Cisco ATA 186 to operate in FAX mode on one or both lines.

The settings for FAX mode are:

- AudioMode—Set to 0x*XXXX*0012 for line 1, or 0x0012*XXXX* for line 2, (*X* = don't care). The value 0x0012 means we disable fax detection, disable G.711 silence suppression and use G711 only on this line.

Configuring the Cisco ATA 186 for Fax Mode on a Per-Call Basis

If you want to activate Fax Mode on a per-call basis, configure bit 15 (line 1—mask 0x8000) and Bit 31 (line 2—mask 0x8000000)—0 (1) to set the default to enable (disable) FAX mode on a per call basis.

To activate this call in the U.S., from the Fax machine enter ***99** (default) and then the telephone number to which you want to send the Fax. The next call will automatically revert to normal mode.

To activate this call in Sweden., from the Fax machine enter ***99#** (default) and then the telephone number to which you want to send the Fax. The next call will automatically revert to normal mode.

Configuring the Cisco Gateway for FAX Mode

On the Cisco Gateway, disable both FAX relay and FAX Passthrough with the following commands:

no fax relay

no modem passthrough

Testing and Troubleshooting the Cisco ATA 186

This chapter describes how to test and troubleshoot the Cisco ATA 186.

Testing the Cisco ATA 186 Configuration

To test your Cisco ATA 186 configuration:

Step 1 Lift the primary line telephone handset.

Step 2 If you are not in SIP mode, enter your user ID (UID).

Step 3 If you hear a dial tone, your Cisco ATA 186 is properly configured.

Making a Call

Note The dial plan described in these instructions is the default plan. You can configure your Cisco ATA 186 to use a different plan.

Follow these steps to make a call:

Step 1 Lift the telephone handset.

Step 2 Dial as you normally do.

Tips If you want to send the dial string out immediately, without waiting for the 2- to 4-second timeout, press # after the last digit.

If the called party is available, you will hear ringing.

Step 3 When the called party answers, speak normally.

Step 4 Hang up when finished.

You can cancel or discontinue your call at any time by hanging up the handset.

You can make a separate, simultaneous telephone call by using the second handset connected to the Cisco ATA 186. The Cisco ATA 186 can support two simultaneous G.723 calls or one G.729A call. When using G.729A, the second line must use G.711 u-law or a-law. The default voice codec is G.723.

Troubleshooting Tips

The suggestions in this section are general troubleshooting tips.

- Make sure that you are correctly registered with the gatekeeper. Check to make sure that you are using the correct gatekeeper IP address and the correct E164 address for UID0 or UID1.
- Check the green Network Activity LED to make sure that the network connection is active.
- Make sure that the DHCP server is operating correctly. Note that the Function button flashes when the Cisco ATA 186 attempts to acquire the DHCP configuration.
- If you do not see the green activity LED flashing after you connect the Ethernet cable, make sure the power cord and the Ethernet connection are secure.
- If you do not hear a dial tone, make sure that you have entered your user ID (UID) and that the telephone line cord from the telephone is plugged into the Cisco ATA 186 PHONE 1 port.

- A fast-busy tone indicates that the party you called is not available. Try your call again later.
- If you place a call to another IP telephone, hear ringing, and the called party answers but you cannot hear the speaker's voice, verify that the Cisco ATA 186 and the other IP telephone support at least one common audio codec technique: G.711A, G.711µ, G.723 and G.729A.
- If you are using a firewall, make sure it is a Cisco PIX firewall, version 5 or later versions.
- After power up, if the function button continues to blink, indicating failure to contact the DHCP server, check the Ethernet connection.
- The DHCP server should show an incoming request from the MAC address listed on the product label or given by the voice prompt.
- If your system is configured to use a gatekeeper, a dial tone is heard after the Cisco ATA 186 has been successfully registered with that gatekeeper. If you do not hear a dial tone, check that all cables are connected properly and the internet connections are operating correctly.
- Make sure you do not have duplicate user IDs.

Symptoms and Actions

Symptom Parameters with values set by using the Web Server Interface or IVR revert to their original settings.

Possible Cause You are using TFTP for provisioning (UseTFTP parameter is set to 1). The Cisco ATA 186 has a cached value of its profile stored in its flash ROM; this is what you will see or hear through the Web Server Interface or IVR. If UseTFTP is set to 1, then the cached value of its profile is synchronized with its profile located at the TFTP server. This synchronization update of the cache value happens at approximate intervals determined by the CFGInterval parameter's value or at power up reset.

Action If you are using TFTP for provisioning, do not use the Web Server Interface or IVR to modify the value of your Cisco ATA 186 profile. Only use the Web Server Interface or IVR to initially configure the Cisco ATA 186 to use TFTP for provisioning.

Symptom The Cisco ATA 186 does not appear to be provisioned by the TFTP server.

Possible Cause The TftpURL is not correctly set to the URL or IP address of the TFTP server that is hosting the profile for the Cisco ATA 186.

Action If you are using DHCP to supply the TFTP server IP address, make sure that TftpURL is set to 0. Make sure that DNS1IP and DNS2IP are properly set to resolve the TftpURL supplied by DHCP. (If the TftpURL supplied by DHCP is an IP string, the Cisco ATA 186 does not need to consult with DNS.) If you want to use DHCP server-assigned DNSes for resolving DNS requests, make sure that DNS1IP and DNS2IP are set to 0.

Symptom The Cisco ATA 186 contacts the TFTP server more often than specified in CFGINTERVAL.

Possible Cause The TOCONFIG parameter is not set to 0.

Action The TOCONFIG parameter set to 1 by default. This means that the Cisco ATA 186 does not yet have a good operating profile. Once the Cisco ATA 186 has a good operating profile, set the parameter to 0. This is best done by including this parameter in the profile downloaded from the TFTP server. If TOCONFIG is not set to 1, the Cisco ATA 186 tries to contact the TFTP server more often than necessary.

Symptom There is no dial tone.

Possible Cause The Cisco ATA 186 has not successfully registered to the gatekeeper or Proxy (GKORPROXY). If no GKORPROXY is specified, then the Cisco ATA 186 is operating in pure IP to IP mode, and the Cisco ATA 186 can be addressed only by IP address.

Action Make sure that UID0 and UID1 are set to the applicable values (telephone numbers) to get a dial tone on each line.

Symptom Unable to access the web configuration page.

Possible Cause Software versions earlier than 2.0 require the web configuration page to be enabled using option 80# on the IVR.

Action Upgrade the software.

Symptom FAX passthrough is not working or is working incorrectly.

Possible Cause By default, the VoIP dial-peers have FAX relay enabled on them; when they hear the FAX training tone, they try to load the FAX relay codec. The Cisco ATA 186 does not support this mechanism; it can only support the passthrough mechanism.

Possible Cause Disable FAX relay on the VoIP dial-peers and enable the passthrough mechanism.

Symptom IP Routing is not operating.

Possible Cause IP routing is off by default.

Action Enable IP routing on the gatekeeper by using the **ip routing** command. See your Cisco IOS documentation for information.

Symptom Time is not displayed or is displayed incorrectly.

Possible Cause The Cisco ATA 186 and gatekeeper are not pointing to the same network time protocol (NTP) server.

Action Make sure that both items are pointing to the same NTP server.

Symptom Cannot place call.

Possible Cause There may be a duplicate user ID.

Action Make sure that all user IDs are unique.

Symptom The Cisco ATA 186 registration is rejected by the gatekeeper. The gatekeeper contains more than one zone prefix command.

Possible Cause If delivering more than one zone prefix on the gatekeeper, the UserID values or E.164 values on the H.323 endpoint (that is, the Cisco ATA 186) must be within a defined zone prefix on the gatekeeper.

Action Add the zone prefixes in order for the Cisco ATA 186s to maintain their registration.

Symptom Fast busy tone.

Possible Cause There may be a duplicate user ID.

Action Make sure that all user IDs are unique.

Symptom SIP parameters are not available.

Possible Cause The SIP protocol or SIP Registration are not enabled.

Action Make sure that you have enabled the SIP protocol by setting the UseSIP parameter. If you are using proxy, enable SIP registration (for authentication).

Installation and Upgrade Issues

Note The following issues apply to the manual upgrade process only. Image and language upgrades must be performed separately.

Symptom The red LED is flashing.

Possible Cause The Cisco ATA 186 is trying to obtain the DHCP address or the software image is being upgraded

Possible Cause The Ethernet cable is unplugged.

Action Plug in the Ethernet cable.

Symptom Voice prompt returns `Upgrade not available` message.

Possible Cause You are attempting to upgrade to the existing version.

Action There is no need to upgrade.

Symptom Voice prompt returns `Upgrade failed` message.

Possible Cause You have entered an incorrect IP address.

Action Enter the correct IP address.

Symptom No dial tone.

Possible Cause No user ID was entered.

Action Enter the correct user ID.

Symptom Incorrect dial tone.

Possible Cause You installed a template for a country other than your own. Check the web interface for your Dial Plan. The default is U.S.

Action Install the correct country template.

Mass Provisioning Issues

The following cautions apply during mass provisioning.

Caution During upgrading, the function button flashes. Do not unplug the power during this procedure.

Contacting TAC

If you need to contact the Cisco Technical Assistance Center (TAC), provide the following information:

- Product codes.
- Software version number - To identify the software revision number, use the configuration menu number—**123.**
- Hardware version number - To identify the hardware revision number, use the serial number and MAC address found on the label on the bottom of the Cisco ATA 186.
- Software build information - To identify the software build information, use the voice menu option—**123123.**
- Cisco ATA 186 serial number.
- MAC address.

Debugging

A debugging tool, prserv, is available from TAC. The pserv program is used in conjunction with the NPrintf configuration parameter. Contact Cisco for more information. See Obtaining Technical Assistance, page -xviii, for instructions.

Debug commands that you can use on your Cisco IOS are

debug RAS

and

debug H225 ASN1

You should also have access to a “sniffer” or LAN analyzer.

Caution For security reasons, Cisco recommends that you do not use the web interface over the public network. Disable the web interface by using the UIPassword parameter before the Cisco ATA 186 leaves the service provider site.

Voice Menu Options

This Appendix contains a list of the Voice Menu options for the Cisco ATA 186.

Note Press # after you enter the item number.

Table A-1 Voice Menu Options in Alphabetical Order

Option	Item Number	Description
Alternate Gatekeeper (Altgk)	6	(H.323)
Alternate Gatekeeper Timeout	251	(H.323)
Alternate NTP IP Address	78	
Audio Mode	312	
Authenticate Method	92	(H.323)
Build Information	123123	Listen to the build date of the Cisco ATA 186 software
Call Features	314	
Call Wait Caller ID	317	Not currently available
Caller ID Method	316	
CfgInterval	80002	
Connect Mode	311	

Table A-1 Voice Menu Options in Alphabetical Order

Option	Item Number	Description
Dynamic Host Configuration Protocol (DHCP) 0=Disable 1=Enable (Default)	20	This command controls whether the Cisco ATA 186 can automatically obtain configuration parameters from a server over the network.
Diagnostic Information	411	
DNS 1 IP	916	
DNS 2 IP	917	
Encrypt Key	320	
Num Tx Frames Values: 1 through 12 Default=2	35	Select the number of frames per packet to be transmitted when using the audio codec G.723.1 or G.729A.
Gatekeeper ID	91	(H.323)
Gatekeeper/Proxy Server IP Address	5	Enter the gatekeeper or proxy server IP address.
Gatekeeper time to live	250	(H.323) Enter the amount of idle time before the gatekeeper times out.
Gateway IP Address	11	(H.323) Enter the IP address of the gateway. This number can be automatically assigned when DHCP is enabled.
IP Address	1	Enter the IP address of the Cisco ATA 186. This number can be automatically assigned when DHCP is enabled.
IP Dial Plan	310	
LBR Codec	300	
Login ID 0	46	
Login ID 1	47	
Media port	202	(SIP)
Network Route Address	2	Enter the network route address. This number can be automatically assigned when DHCP is enabled.

Table A-1 Voice Menu Options in Alphabetical Order

Option	Item Number	Description
NPrintf Address	81	
NTP Server Address	141	
Paid Features	315	
Polarity	304	
PWD 0	4	Enter the password associated with the gateway account.
PWD 1	14	Enter the password associated with the second gateway account.
Review IP Address	21	Listen to the IP address.
Review MAC Address	24	Listen to the media access control (MAC) address.
Review Network Route IP Address	22	Listen to the IP address of the network route.
Review Subnet Mask	23	Listen to the subnet mask.
Rx Codec 0=G.723.1 (default) 1=G.711A 2=G.711µ 3=G.729A	36	Select the audio codec type to use to decode received data. The call-receiving station automatically adjusts to the call-initiating station's audio codec type if the call-receiving station supports that audio codec.
Set Password	7387277	
SigTimer	318	
SIP Max Number of Redirects	205	(SIP)
SIP NAT IP Address	200	(SIP)
SIP Outbound Proxy	206	(SIP)
SIP Port	201	(SIP)
SIP Protocol 0=Select H.323 1=Select SIP	38	Select the signalling protocol.

Table A-1 Voice Menu Options in Alphabetical Order

Option	Item Number	Description
SIP Registration On	204	(SIP)
SIP Registration Period	203	(SIP)
Subnet Mask	10	Enter the subnet mask. This number can be automatically assigned when DHCP is enabled. The default is 255.255.255.0.
TFTP URL	905	
Timezone	302	
ToConfig 0=provisioned 1=new (default)	80001	See Appendix B, "Parameters and Defaults"
Trace Flags 0=disable (default) 1=enable	313	Enable logging of SIP messages.
Tx Codec 0=G.723.1 (default) 1=G.711A 2=G.711µ 3=G.729A	37	Select the audio codec type to use to encode data for transmission. The audio codec type for receiving does not have to be the same as the audio codec type for transmitting.
UDP TOS Bits	255	
UID 1	13	Enter the User ID (telephone number) for the Phone 2 port.
Upgrade Software	100	
Upgrade Language to English	101	Change or upgrade the voice prompt language to English. See Chapter 6, "Upgrading the Cisco ATA 186 Software" for more information.
Use Login ID	93	

Table A-1 Voice Menu Options in Alphabetical Order

Option	Item Number	Description
Use TFTP 0=disable (default) 1=enable	305	
UID 0	3	Enter the User ID (telephone number) for the Phone 1 port.
Version Number	123	Listen to the version number of the Cisco ATA 186 software.

Table A-2 Voice Menu Options in Numerical Order

Item Number	Option	Description
1	IP Address	Enter the IP address. This number can be automatically assigned when DHCP is enabled.
2	Network Route Address	Enter the network route address. This number can be automatically assigned when DHCP is enabled.
3	UID 0	Enter the User ID (telephone number) for the Phone 1 port.
4	PWD 0	Enter the password associated with the gateway account.
5	Gatekeeper/Proxy Server IP Address	Enter the gatekeeper or proxy server IP address.
6	Alternate Gatekeeper (Altgk)	(H.323)
10	Subnet Mask	Enter the subnet mask. This number can be automatically assigned when DHCP is enabled. The default is 255.255.255.0.
11	Gateway IP Address	Enter the IP address of the gateway. This number can be automatically assigned when DHCP is enabled.

Table A-2 Voice Menu Options in Numerical Order

Item Number	Option	Description
13	UID 1	Enter the User ID (telephone number) for the Phone 2 port.
14	PWD 1	Enter the password associated with the second gatekeeper account.
20	Dynamic Host Configuration Protocol (DHCP) 0=Disable 1=Enable (Default)	This command controls whether the Cisco ATA 186 can automatically obtain configuration parameters from a server over the network.
21	Review IP Address	Listen to the IP address.
22	Review Network Route IP Address	Listen to the IP address of the network route.
23	Review Subnet Mask	Listen to the subnet mask.
24	Review MAC Address	Listen to the media access control (MAC) address.
35	Num Tx Frames Values: 1 through 12. Default=2	Select the number of frames per packet to be transmitted when using the audio codec G.723.1 or G.729A.
36	Rx Codec 0=G.723.1 (default) 1=G.711A 2=G.711µ 3=G.729A	Select the audio codec type to use to decode received data. The call-receiving station automatically adjusts to the call-initiating station's audio codec type if the call-receiving station supports that audio codec.
37	Tx Codec 0=G.723.1 (default) 1=G.711A 2=G.711µ 3=G.729A	Select the audio codec type to use to encode data for transmission. The audio codec type for receiving does not have to be the same as the audio codec type for transmitting.

Table A-2 Voice Menu Options in Numerical Order

Item Number	Option	Description
38	SIP Protocol 0=Select H.323 1=Select SIP	Select the signalling protocol.
46	Login ID 0	
47	Login ID 1	
78	Alternate NTP IP Address	
81	NPrintf Address	
91	Gatekeeper ID	(H.323)
92	Authenticate Method	(H.323)
93	Use Login ID	
100	Upgrade Software	
101	Upgrade Language to English	Change or upgrade the voice prompt language to English. See Chapter 6, "Upgrading the Cisco ATA 186 Software" for more information.
123	Version Number	Listen to the version number of the Cisco ATA 186 software.
141	NTP Server Address	
200	SIP IP Proxy Address	(SIP)
201	SIP Eproxy Address	(SIP)
202	Media port	(SIP)
203	SIP Registration Period	(SIP)
204	SIP Registration On	(SIP)
205	SIP Max Number of Redirects	(SIP)
206	SIP Outbound Proxy	(SIP)
250	Gatekeeper time to live	(H.323) Enter the amount of idle time before the gatekeeper times out.
251	Alternate Gatekeeper timeout	(H.323)
255	UDP TOS Bits	

Table A-2 Voice Menu Options in Numerical Order

Item Number	Option	Description
300	LBR Codec	
302	Timezone	
304	Polarity	
305	Use TFTP 0=disable (default) 1=enable	
310	IP Dial Plan	
311	Connect Mode	
312	Audio Mode	
313	Trace Flags 0=disable (default) 1=enable	Enable logging of SIP messages
314	Call Features	
315	Paid Features	
316	Caller ID Method	
317	Call Wait Caller ID	Not currently available
318	SigTimer	
320	Encrypt Key	
411	Diagnostic Information	
905	TFTP URL	
916	DNS 1 IP	
917	DNS 2 IP	
123123	Build Information	Listen to the build date of the Cisco ATA 186 software
80001	ToConfig 0=provisioned 1=new (default)	See Appendix B, "Parameters and Defaults"

Table A-2 Voice Menu Options in Numerical Order

Item Number	Option	Description
80002	CfgInterval	
7387277	Set Password	

APPENDIX B

Parameters and Defaults

This Appendix provides information on the parameters and defaults that you can use to provision your Cisco ATA 186.

Provisioning files must begin with **#txt** for the formating tool, cfgfmt.exe, to treat it as text file.

Begin at new line indicates a comment.

All parameter/value pairs are optional but might be needed for the Cisco ATA 186 to function properly if a value has not already been programmed.

Where an IVR Access Code is available, you can use either the alphanumeric entry method or a numeric entry method to specify the value of the parameter using an IVR alternate interface. Convert the hexadecimal value to decimal when using the IVR alternate interface (numeric entry method) before you manually enter the value by using IVR.

Parameter values can be one of the following types:

- alphanumeric string
- numeric digit string
- array of short integer
- IP address (e.g. 192.168.2.170)
- Extended IP address—IP address with Port (for example, 192.168.2.170.9001)
- boolean (1 or 0)
- bitmap value—unsigned hex integer (for specifying bits in 32-bit integer)
- integer (32 bit integer).

User Interface (UI) Parameter

Table B-1 User Interface Parameter and Default

Parameter	Value Type	Description	IVR Access Code	Default
UIPassword	Alpha-numeric string	Control access to web page or IVR interface. If set to non-zero, then you must enter the value of UIPassword to access the web page and IVR. **Note** When UIPassword contains letters, you cannot enter the password from the telephone keypad.	7387277	0

Provisioning Parameters

Table B-2 TFTP Provisioning Parameters and Defaults

Parameter	Value Type	Description	IVR Access Code	Default
CfgInterval		Specifies the number of seconds (interval) between each configuration update. For example, when using TFTP for provisioning, the Cisco ATA 186 contacts TFTP each time the interval expires to get its configuration file the next time the box is idle. You can set CfgInterval to a random value to achieve random contact intervals from individual Cisco ATA 186 to the TFTP server.	80002	3600
EncryptKey	Alpha-numeric string		320	0

Table B-2 TFTP Provisioning Parameters and Defaults

Parameter	Value Type	Description	IVR Access Code	Default
TftpURL	Alpha-numeric string	IP address or URL of the TFTP server to use. This is needed if the Dynamic Host Configuration Protocol (DHCP) does not give the TFTP address. You can optionally include the path prefix to the TFTP file to download. For example, if the TFTP server IP address is 192.168.2.170 or www.cisco.com, and the path to download the TFTP file is in /ata186, then you can specify the URL as 192.168.2.170/ata186 or www.cisco.com/ata186. **Note** From the IVR, you can only enter the IP address; from the web server, you can enter the actual URL.	905	0
ToConfig	Boolean	The default is **1**. This indicates that the operating parameters of the box have not previously been set. Once the Cisco ATA 186 has been provisioned, set the parameter to **0**, or the Cisco ATA 186 will unnecessarily contact the TFTP server.	80001	0
UseTFTP	Boolean	1 means use TFTP for provisioning; 0 means do not.	305	1

There are two additional parameters that are used to control the upgrade of the firmware code and language. The values of these parameters depend on the software release.

Firmware Upgrade Parameters

The upgradecode and upgradelang parameters are special parameters that provide information on how to upgrade firmware code or language image.

The values are:

- upgrade_policy
- hardware_version
- software_type
- software_version
- tftp_ip_address
- tftp_port
- image_id
- firmware_image_file_name

All values must be present. Except for upgrade_policy field, Cisco supplies the values to be used when a new upgrade image becomes available

The values are:

- upgrade_policy—one of the following:
  – default upgrade— software_type must be the same as the current Cisco ATA 186. Internal software_type and software_version must be greater than the current internal software_version for the upgrade to take place.
  – cross software type upgrade—the software_type can be different but software_version must be greater than the current firmware internal version.
  – force upgrade—the image_id is used to determine if the upgrade has been performed (a different image ID from the Cisco ATA 186 internally stored image ID means a forced upgrade is needed).
- hardware_version— 0x301. The hardware version must match exactly for upgrade to take place.
- tftp_ip_address, tftp_port—TFTP IP and port address where the firmware can be found.

- image_id—used for force upgrade to determine if an upgrade had taken place previously.
- firmware_image_file_name—firmware image file name.
  - For example,
    upgradecode:1,0x301,0x0400,0x0200,192.168.2.170,69,0x1,
    ef2_200h.kxz
  - upgradelang:1,0x301,0x0400,0x0200,192.168.2.170,69,0x1,ef2e200h.kbx
  - The default values listed below will not trigger any upgrade.

    upgradecode:0,0x301,0x0400,0x0200,0.0.0.0,69,0,none

    upgradelang:0,0x301,0x0400,0x0200,0.0.0.0,69,0,none

Example:

1,0x031,0x0400,0x0200,192.168.2.170,69,0x1,ef2_200h.kxz

1= upgrade policy

0x031=hardware version

0x0400=software type

0x0200=software version

192.168.2.170=tftp_ip_address

69=TFTP port

0x1=image id

ef2_200h.kxz=firmware image file name

Operating Parameters

Table B-3 Common Operating Parameters and Defaults

Parameter	Value Type	Description	IVR Access Code	Default
AutMethod	Bitmap	Authentication method. Possible values are: • bit 0-1 (mask 0x3): 0 for no authentication, 1 for authentication • Cisco registration level security (H.235), 2 for Cisco Systems admission level security. • bit 2 (mask 0x4): prefix password field when registering.	92	0x00000000
CallCmd	Alpha-numeric string	Controls call commands such as turning on and off Caller ID and so on. (248 characters maximum.) U.S. Cmd Table CallCmd:Af;AH;BS;NA;CS;NA;Df;EB;Ff;EP;Kf;EFh;HQ;Jf;AFh;HQ;I*67;gA*82;fA90v;OI;H72v;bA74v;cA75v;dA73;eA*67;gA*82;fA*70;iA*69;DA*99;xA;Uh;GQ; Swedish Cmd Table CallCmd:BS;NA;CS;NA;Df;EB;Ff0;ARf1;HPf2;EPf3;AP;Kf1;HFf2;EFf3;AFf4;HQ;Jf1;HFf2;EFf3;AFf4;HQ;Af4;HQ;I*31;gA31;gA*90*v;OI;H*21*v;bA*61*v;dA*67*v;cA21;eA61;eA67;eA*31;gA31;gA*43;hA43;iA*69;DA*99;xA;Uh;GQ;	N/A	U.S. Cmd Table

Table B-3 Common Operating Parameters and Defaults (continued)

Parameter	Value Type	Description	IVR Access Code	Default

CallerIdMethod	Integer	CallerID/DTMFMethod. Possible values are:
		bit 0-1 (mask 0x3): method; 0=Bellcore, 1=DTMF, 2=ETSI, 3=FSK (0)
		bit 2 (mask 0x4): method type; 0=type 1, 1=type 2 (0)
		if(method=0) {
		bit 3-8 (mask 0x1f8): max no. of digits (12)
		bit 9-14 (mask 0x7e00): max no. of chars (15)
		bit 15-20 (mask 0x1f8000): special chars (3)
		}
		else {
		bit 3-6 (mask 0x78): start digit (12)
		bit 7-10 (mask 0x780): end digit (14)
		bit 11 (mask 0x800): polarity reversal (1)
		bit 12-16 (mask 0x1f000): max no. of digits (15)
		}
		– Sweden=0x0ff61
		– Denmark=0x0fde1
		– USA=0x19e60
			316	0x00019e60

Table B-3 Common Operating Parameters and Defaults (continued)

Parameter	Value Type	Description	IVR Access Code	Default
CallWaitCallerId	Integer	Caller ID on CallWaiting. Possible values are: • bit 0-5 (mask 0x3f): max no of digits (16) • bit 6-11 (mask 0xfc0): max no of characters (15) • bit 12-17 (mask 0x3f000): special chars (3) • bit 18-21 (mask 0x3c0000): ack digit (15)	317	0x003c33d0

Table B-3 Common Operating Parameters and Defaults (continued)

Parameter	Value Type	Description	IVR Access Code	Default
ConnectMode	Bitmap	Connection mode of the protocol used. Possible values are: • bit 0 (mask 0x1): 0 for slow start and 1 for fast start (h323). • bit 1 (mask 0x2): 1 use h245 tunneling. • bit 3 (mask 0x8): 1 means alternate gatekeeper need register. • bit 12-8 (mask 0x1f00): offset to payload 96 (0-23). • bit 13 (mask 0x2000): 0 use G.711ulaw; 1 use G.711alaw. • bit 14 (mask 0x4000): 0 use fax pass through and 1 use codec negotiation in sending fax. • bit 15 (mask 0x8000): 0 means enable/1 means disable detecting fax pass through. • bit 16 (mask 0x10000): 1 enables SIP to remove the registration before adding a new one. • bit 17 (mask 0x20000): 1 enables SIP call forwarding to be performed by the Cisco ATA 186. • bit 18 (mask 0x40000): 1 enables SIP call return performed by the Cisco ATA 186.	311	0x00060000

Table B-3 Common Operating Parameters and Defaults (continued)

Parameter	Value Type	Description	IVR Access Code	Default
DHCP	Boolean	Can be used to automatically set the IP address, network route IP, subnet mask, DNS, NTP, TFTP, and other parameters. To enable DHCP, set this value to 1; to disable DHCP, set this value to 0.	20	1
DNS1IP, DNS2IP		Primary and secondary domain name server (DNS) IP (if the DHCP server does not provide one).	DNS1I=916 DNS2IP=917	0.0.0.0
GkOrProxy	Alpha-numeric string	Gatekeeper or proxy address. Null-terminated alpha-numeric string with up to 31 characters. For SIP proxy server, this can be an IP address with or without a port parameter such as 123.123.110.45, 123.123.110.45.5060, or 123.123.110.45:5061, or a URL such as sip.cisco.com, or sip.ata.cisco.com:5061. For IP address, a '.' or ':' can be used to delimit a port parameter. For URL, a ':' must be used to indicate a port31 characters maximum.	5	0
IPDialPlan	Integer	Allows for detection of an IP-like destination address in the dial plan. Possible values are: 1: if two '.' are seen, then assume IP address is entered. 2: if three '.' are seen, then assume IP address is entered.	310	1

Table B-3 Common Operating Parameters and Defaults (continued)

Parameter	Value Type	Description	IVR Access Code	Default
LBRCodec	Integer	Low-bit-rate codec. Possible values are: • 0—select G.723 as low-bit-rate codec. • 3—select G.729A as low-bit-rate codec.	300	0
NTPIP	IP	NTP IP address (needed if DHCP server does not provide one).	141	0.0.0.0
NumTxFrames	Integer	Transmit frames per packet. Use only the recommended default.	35	2
Op Flag	Bitmap	Enables/disables various operational features. If Bit 0 (mask 0x1) is set to 1, always use the internally generated TFTP file name. If Bit 1 (mask 0x2) is set to 1, do not perform static network router probing at cold start.	323	0x2

Table B-3 Common Operating Parameters and Defaults (continued)

Parameter	Value Type	Description	IVR Access Code	Default
Polarity	Bitmap	Control connect and disconnect polarity. Possible values are: • bit 0 (mask 0x1): CALLER_CONNECT_ POLARITY • bit 1 (mask 0x2): CALLER_DISCONNECT_ POLARITY • bit 2 (mask 0x4): CALLEE_CONNECT_ POLARITY • bit 3 (mask 0x8): CALLEE_DISCONNECT_ POLARITY	304	0x00000000
PWD0, PWD1	Alpha-numeric string	Password for line 0 and 1.	PWD0=4 PWD1=14	0
RxCodec	Integer	Receiving audio codec preference. Possible values are: • 0—G.723 (can be selected only if LBRCodec is set to 0). • 1—711a. • 2—G.711u. • 3—729a (can be selected only if LBRCodec is set to 3).	36	0

Table B-3 Common Operating Parameters and Defaults (continued)

Parameter	Value Type	Description	IVR Access Code	Default
SigTimer	Bitmap	Controls the timeouts listed. A 32-bit value is divided into bit fields: Call Waiting Period (bits 0-7): the period between each burst of call-waiting tone. Range=0 to 255 in 0.1 seconds; 0 implies the factory default value of 100 (0x64=10 seconds) Reorder delay (bits 8-13): the delay in playing reorder (fast busy) tone after far end hangs up. Range=0 to 62 in seconds, 63=never play reorder. Default: 5 (seconds) Ring Timeout (bits 14-19): the amount of time after which, when there is no answer, the Cisco ATA 186 rejects the incoming call. Range=0 to 63 in 10 seconds, 0=never times out. Default: 6 (60 seconds). No Answer Timeout (bits 20-25): time to declare no answer and initiate call forwarding on no answer (used in SIP only at present). Range=0 to 63 in seconds. Default: 20 (0x14=20 seconds) Reserved (bits 26-31).		
StaticIp	IP	Statically assigned IP address. Used if the DHCP parameter is set to 0.	1	0.0.0.0
StaticRoute	IP	Statically assigned route. Used if the DHCP parameter is set to 0.	2	0.0.0.0
StaticSubNetMask	IP	Statically assigned subnetmask. Used if the DHCP parameter is set to 0.	10	0.0.0.0

Table B-3 Common Operating Parameters and Defaults (continued)

Parameter	Value Type	Description	IVR Access Code	Default
TimeZone	Integer	Timezone offset from GMT for time-stamping incoming calls with local time (for caller-id display, and so on.) Possible values are: 0-24. Local time is generated by the following formula: • Local Time=GMT + TimeZone, if TimeZone <= 12 • Local Time=GMT + TimeZone - 25, if TimeZone > 12	302	17
TxCodec	Integer	Transmitting audio codec preference. Possible values are: • 0—G.723 (can be selected only if LBRCodec is set to 0). • 1—G.711A. • 2—G.711u. • 3—G.729A (can be selected only if LBRCodec is set to 3).	37	1

Table B-3 Common Operating Parameters and Defaults (continued)

Parameter	Value Type	Description	IVR Access Code	Default
UDPTOS	Bitmap	The UDPTOS parameter specifies the default IP precedence of UDP packets as follows: Bits 0-1: Unused Bit 2: Reliability bit—1=request high reliability Bit 3: Throughput bit—1=request high throughput Bit 4: Delay bit—1=request low delay Bits 5-7: Specify datagram precedence. Values range from 0 (normal precedence) to 7 (network control).	255	0x000000a0
UID0, UID1	Alpha-numeric string	User ID for line 0 and 1. 31 characters or digits maximum.	UID0, 3 UID1, 13	0
UseLoginID	Boolean	1 means use Login id specified in LoginId0 and LoginId1 and 0 means not. For H323, this field is needed if autm is set to 1. For SIP, UID0 and UID1 will be used for authentication if this is 0.	93	0

Table B-4 H.323 Operating Parameters and Defaults

Parameter	Value Type	Description	IVR Access Code	Default
AltGk	Alpha-numeric string	Alternate gatekeeper **Note** ConnectMode bit 3 (mask 0x8) controls whether alternate gatekeeper needs to register.	6	0
AltGkTimeOut	Integer	Alternate gatekeeper timeout value in seconds.	251	0
GateWay	Alpha-numeric string	Gateway IP, if gatekeeper is not used for routing calls. 31 characters maximum.	11	0
GkId	Alpha-numeric string	Gatekeeper zone ID. 31 characters maximum.	91	.
GkTimeToLive	Integer	Gatekeeper time to live value.	250	300
LoginId0, LoginId1	Alpha-numeric string	H.323 or SIP Login ID for line 0 and 1. For H323, this is needed if AuthMethod is set to 1. For SIP, UID0 and UID1 are used for authentication if UseLoginID is 0. 19 characters maximum.	LoginId0, 46 LoginId1, 47	0

Table B-5 SIP Parameters

Parameter	Value Type	Description	IVR Access Code	Default
MAXRedirect	Integer	Maximum number of times to try redirection.	205	5
MediaPort	Integer	Base port to receive RTP media; currently only used to support SIP behind a NAT.	202	10000
NATIP	IP	WAN address of the attached router/NAT; currently only used to support SIP behind a NAT.	200	0.0.0.0

Table B-5 SIP Parameters

Parameter	Value Type	Description	IVR Access Code	Default
OutBoundProxy	Null-terminated alpha-numeric string	SIP Proxy Server for all outbound SIP requests. Web-Tag: OutBoundProxy. Maximum 31 characters. Can be an IP address with or without a port parameter, such as 123.123.110.45, 123.123.110.45.5060, or 123.123.110.45:5061, or a URL such as sip.cisco.com, sip.ata.cisco.com:5061. For IP addresses, a '.' or ':' can be used to delimit a port parameter. For URL, a ':' must be used to indicate a port.	206	0
SIPPort	Integer	Port to listen for incoming SIP requests.	201	5060
SIPRegInterval	Integer	Seconds between registration renewal.	203	3600
SIPRegOn	Integer	Enable SIP registration.	204	0
UseSIP	Boolean	Use SIP mode (default H.323).	38	0

APPENDIX C

Audio Mode Parameters and Defaults

This Appendix provides information on the audio mode parameters and defaults that you can use to provision your Cisco ATA 186.

The Audio Mode format is bitmap.

The IVR access code is 312.

The default format is 0x00150015.

This parameter represents the audio operating mode. The lower 16 bits apply to channel 0; the upper 16 bits apply to channel 1.

- bit 0 (mask 0x1): G.711SilenceSuppress: enable G.711 silent suppression (1)
- bit 1 (mask 0x2): G.711Only: use G.711 only; do not use the low-bit-rate codec (0)
- bit 2 (mask 0x4): FaxDetCED: enable fax CED tone detection (1)
- bit 3 (mask 0x8): FaxDetCNG: enable fax CNG tone detection (0)
- bits 4-5 (mask 0x30): DtmfMethod:
  - 0=always inband (send and receive; don't send SDP info)
  - 1=by negotiation (send SDP info, enable rcv, decode others SDP info, send depends on other SDP info)
  - 2=always out-of-band (send SDP info, enable rcv, decode other SDP info, always send) (1)

- bits 6-7 mask 0xc0: HookflashMethod:
  - 0=disable sending hookflash
  - 1=by negotiation
  - 2=always send hookflash
  - 3=use Q931 to send user keypad input (DTMF or hookflash) information (0)

APPENDIX D

Dial Plan Parameters and Defaults

This appendix provides information on the dial plan parameters and defaults that you can use to provision your Cisco ATA 186. You cannot configure the dial plan by using voice input; you must use the web browser or autoprovisioning.

About Programmable Dial Plans

The programmable dial plan allows the service provider to customize the way the Cisco ATA 186 reads and sends the sequence of digits input by the user. In general, the dial plan defines which sequences of digits will be recognized by the Cisco ATA 186 as valid dial strings, suitable to send through the network (e.g., 555-1212, 1-408-555-1212, 011 46 8 1234 1234). Additionally, the dial plan specifies:

- An optional send character (for example, # or *) which, when entered, will automatically send the dial string).
- The number of digits to look for before the dial string is automatically sent.
- The timeout (in seconds) between when the last digit is entered and when the Cisco ATA 186 sends the dial string. The timeout can be programmed to change value in mid-dial string.

About Dial Plan Commands

- **.**—Match any digit entered
- **-**—Additional digits can be entered. This command can be used only at the end of a dial plan rule (e.g., 1408t5- is legal usage of the - command, but 1408t5-3... is illegal).
- >#—Defines the # character as a termination character. When the termination character is entered, the dial string will automatically be sent. The termination character can be entered only after at least one digit entered by the user has been matched by a dial plan rule. Alternatively, the command >* can be used to define * as the termination character.
- tn—Defines timeout as n seconds. Valid values are 0-9 and a-z, where a-z indicates a range of 10 to 36.
- rn—Repeat the last pattern n times, where n is 0-9 or a-z. The values a-z indicate a range of 10 to 36. Use the repeat modifier to specify more rules in less space.

Note The commands **>#** and **tn** are modifiers, not patterns, and will be overlooked by the rn command.

- |—Used to separate multiple dial plan rules.
- ^—Logical not. Match any character except the character immediately following the ^ command.
- S—Seize rule matching. If a dial plan rule matches the sequence of digits entered by the user to this point, and the modifier S is the next command in the dial plan rule, all other rules are negated for the remainder of the call (e.g., a dial plan beginning with *S will be the only one in effect if the user first enters the * key).

Note All rules apply in the order listed (whichever rule is completely matched first will immediately send the dial string).

Note No syntax check is performed by the actual implementation. It is the responsibility of the provisioner to make sure that the dial plan is syntactically valid.

Example 1

The dial plan:

```
.t7>#......t4-|911|1t7>#..........t1-|0t4>#.t7-
```

or

```
.t7>#r6t4-|911|1t7>#.r9t1-|0t4>#.t7-
```

consists of the following rules:

- **.t7>#r6t4-**—You must enter at least one digit. After the first digit is entered and matched by the dial plan, the timeout before automatic send is 7 seconds, and the terminating character # can be entered at any time to manually send the dial string. After 7 digits are entered, the timeout before automatic send changes to 2 seconds. The - at the end of the rule allows further digits to be entered until the dial string is sent by the timeout or the user entering the # character.
- **911**—If the dial string 911 is entered, send it immediately.
- **1t7>#.r9t1**—If the first digit entered is 1, the timeout before automatic send is 7 seconds, and the terminating character # can be entered at any time to manually send the dial string. After the 11th digit is entered, the timeout before automatic send changes to 1 second. The user can enter more digits until the dial string is sent by the timeout or by the user entering the # character.
- **0t4>#.t7**—If the first digit entered is 0, the timeout before automatic send is 4 seconds, and the terminating character # can be entered at any time to manually send the dial string. After the 2nd digit is entered, the timeout before automatic send changes to 7 seconds. The user can enter more digits until the dial string is sent by the timeout or by the user entering the # character.

Example 2 (Default Dial Plan)

The dial plan:

```
*St4-|#St4-|911|1>#t8.r9t2-|0>#t811.rat4-|^1t4>#.-
```

consists of the following rules:

*St4-—If the first digit entered is *, all other dial plan rules are voided. Additional digits can be entered after the initial * digit, and the timeout before automatic dial string send is 4 seconds.

- #St4—Same as above, except with # as the initial digit entered.
- 911—If the dial string 911 is entered, send it immediately.
- 1>#t8.r9t2—If the first digit entered is 1, the timeout before automatic send is 8 seconds. The terminating character # can be entered at any time to manually send the dial string. After the 11th digit is entered, the timeout before automatic send changes to 2 seconds. The user can enter more digits until the dial string is sent by the timeout or by the user entering the # character.
- 0>#t811.rat4—If the first digit entered is 0, the timeout before automatic send is 8 seconds, and the terminating character # can be entered at any time to manually send the dial string. If the first 3 digits entered are 011, then after an additional 11 digits are entered, the timeout before automatic send changes to 4 seconds. The user can enter more digits until the dial string is sent by the timeout or by the user entering the # character.
- ^1t4>#.—If the first digit entered is anything other than 1, the timeout before automatic send is 4 seconds. The terminating character # can be entered at any time to manually send the dial string. The user can enter more digits until the dial string is sent by the timeout or by the user entering the # character.

Dial Plan Blocking

Dial Plan blocking prevents most invalid dialed digits from being sent. You can change the default interdigit timeout of 9 seconds by adding the following rule to your dial plan string:

I*n*

where *n* is 1-9 or a-z (for 10-35) seconds.

For example, enter an interdigit timeout of 12 seconds as:

`Ic|…[`*`the rest of your dial plan rules`*`]`

Specifying your own interdigit timeout also changes the behavior of the dial plan so that, rather than the entire dial string being sent at timeout, it is sent only as a result of a matching rule or time intended by a matching rule.

APPENDIX E

Paid Services and Call Features Parameters and Defaults

This appendix provides information on the paid services and call features parameters and defaults that you can use to provision your Cisco ATA 186. The paid services parameter is in bitmap format.

The IVR access code for paid services is 314.

The IVR access code for call features is 315.

The default for paid services is 0xffffffff.

The default for call features is 0xffffffff.

Call features is a 32-bit bitmap value. The lower 16-bits are used for channel 0; the upper 16-bits are used for channel 1. The call features and paid services parameters use the same bit masks. Paid services indicates which service the user has subscribed to, while call features indicates which feature is statically enabled by the user. Not all supplementary services can be disabled by the user.

The valid flags in call features are CLIP_CLIR, CALL_WAITING, and FAXMODE. A subscribed service enable/disabled by the user can be disabled/enabled dynamically on a per call basis.

Table E-1 Possible Values for PaidServices and CallFeatures

Bit	Mask	Value	Explanation
bit 0	0x1	FORWARD_ALL	Forward unconditional
bit 1	0x2	FORWARD_BUSY	Forward on busy
bit 2	0x4	FORWARD_NOANS	Forward on no answer
bit 3	0x8	CLIP_CLIR	CLIP
bit 4	0x10	CALL_WAITING	Call waiting
bit 5	0x20	3WAY_CALLING	3-way calling
bit 6	0x40	XFER_BLIND	Blind transfer
bit 7	0x80	XFER_CONSULT	Consult transfer
bit 8	0x100	CALLER ID	Caller ID
bit 9	0x200	CALL RETURN	Call return
bit 10	0x400	MWI	Message waiting indication
bit 15	0x8000	FAXMODE	FAX mode

APPENDIX F

Call Progress Tone Parameters and Defaults

This Appendix provides detailed information on the call progress tone parameters and defaults you can use to provision your Cisco ATA 186.

Call Progress Tone Parameters

The call progress tone parameter is used to set the tone and frequency of the call progress tone. This value is expressed as an integer. You cannot configure the call progress tone by using the IVR mode. You must use the web interface or autoprovisioning.

The default is the recommended U.S. value of 2,4,25.

The recommended value for Sweden is 1,5,25.

The tone parameter controls the characteristics of the following call progress tones:

- Dial tone
- Busy tone
- Reorder tone
- Ringback tone
- Call waiting tone

Call Waiting and Alert Tone

Each parameter is an array of 9 short integers.

Playback Tone

The playback tone is specified in terms of frequency. The values provided below are precomputed internal values.

If a frequency entry format is needed, add **Freq** to the end of each tone name; for example, RingBackToneFreq. The format of RingBackToneFreq is:

ntone, freq0, freq1, level0, level1, steady, on-time, off-time, total time

where:

- **ntone** is the number of frequency components of the tone (0, 1, or 2).
- **freq0** is the transformed frequency of the first frequency component (-32768 to 32767).

Note Only positive values can be provisioned to the Cisco ATA 186. For negative values, use the complement value of the 16-bit 2. For example, enter -1 as 65535 or 0xffff.

- **freq1** is the transformed frequency of the second frequency component (-32768 to 32767).
- **level0** is the transformed amplitude of the first frequency component (-32768 to 32767).
- **level1**is the transformed amplitude of the second frequency component (-32768 to 32767).
- **steady** controls whether the tone is constant or intermittent. A value of 1 indicates a steady tone and causes the Cisco ATA 186 to ignore the **on-time** and **off-time** parameters. A value of 0 indicates an on/off tone pattern and causes the Cisco ATA 186 to use the **on-time** and **off-time** parameters.
- **on-time** controls the length of time the tone is heard in milliseconds (ms) expressed as an integer from 1 to 0x7FFFFFFF.

- **off-time** controls the length of time between audible tones in milliseconds (ms) expressed as an integer from 1 to 0x7FFFFFFF.
- **total time** controls the length of time the tone is audible in multiples of 10 milliseconds (ms) expressed as an integer from 1 to 0x7FFFFFFF. A value of 0 causes the tone to remain audible indefinitely.

Notes

Frequency ranges from 0 to 4000 (Hz)

Transformed Frequency = 32767 · cos (2pi·Frequency/8000)

Transformed Amplitude = *A* · 32768 · sin (2pi·Frequency/8000)

The scaling factor *A* selects the volume level of the tone. To customize the playback tone parameters, select a value for *A* based on the desired volume and the number of frequency components in the relevant tone.

Table F-1 shows several values of *A* and the approximate volume level for each value of *A* for tones that consist of one and two components.

Example -- Calculating The Volume Levels For Dial Tone

A U.S. dial tone consists of two frequency components, 350 Hz, shown in this example as level0, and 440 Hz, shown as level1.

To set the dial tone to a volume of -10dBm for a two-component tone at -10dBm, use a multiplier of 0.35.

The formula for level0 uses this value of *A* in the Transformed Amplitude formula as follows:

0.35 · 32768 · sin (2pi · 350 / 8000) = 3194

The formula for level1 is:

0.35 · 32768 · sin (2pi · 440 / 8000) = 4047

The approximate values to enter for level0 and level1 are 3194 and 4047, respectively.

Table F-1 Volume Levels for a Standard U.S. 600-Ohm Impedance Board

Scale Factor (A)	Volume Level for One Tone Component (dBm)	Volume Level for Two Tones Components (dBm)
0.10	-24	-21
0.175	-19	-16
0.35	-13	-10
0.50	-10	-7

Example Call Progress Tone Parameters

U.S. Values are:

- RingOnOffTime: 2,4,25
- DialTone: 2,31538,30831,3100,3885,1,0,0,1000
- BusyTone: 2,30467,28959,1191,1513,0,4000,4000,0
- ReorderTone: 2,30467,28959,1191,1513,0,2000,2000,0
- RingBackTone: 2,30831,30467,1943,2111,0,16000,32000,0
- CallWaitTone: 1,30831,0,5493,0,0,2400,2400,4800
- AlertTone: 1,30467,0,5970,0,0,480,480,1920

Swedish values are:

- RingOnOffTime: 1,5,25
- DialTone: 1,30959,0,4253,0,1,0,0,1500
- BusyTone: 1,30959,0,2392,0,0,2000,2000,0
- ReorderTone: 1,30959,0,2392,0,0,2000,6000,0
- RingBackTone: 1,30959,0,2392,0,0,8000,40000,0
- CallWaitTone: 1,30959,0,2392,0,0,1600,4000,11200
- AlertTone: 1,30959,0,2392,0,0,2000,6000,0

Call Commands

This Appendix provides detailed information on call commands for the Cisco ATA 186.

There are several supplementary services that can be offered by a Service Provider. Additionally, supplementary services can be activated, configured, or deactivated in more than one way. The Cisco ATA 186 allows the behavior of supplementary services to be defined using its "CallCmd" field.

Context Command Lists

The entry in the "CallCmd" field is a character string composed of a sequence of instructions, called Context-Command-Lists. These Context-Command-Lists each consist of a combination of three elements:

- Context -- The Cisco ATA 186's supplementary service operation is dependent upon a state- transition process. For example, the most common state is IDLE, in which the Cisco ATA 186 is on-hook, waiting for an incoming call. Picking up the telephone handset causes the Cisco ATA 186 to transition to the PREDIAL state, in which the user hears a dial tone and the Cisco ATA 186 is waiting for DTMF digits to be entered. The Context portion of the Context-Command-List specifies the state for which the following commands are defined. The complete default state-transition sequence is shown in.
- Input-Sequence -- The input sequence is simply the input from the user, a combination of hook-flash and DTMF digits.

- Action -- This specifies the action(s) taken by the Cisco ATA 186. An action depends on the Input Sequence entered by the user and the Context in which it was entered.

Syntax

The Call Command string has the following structure:

CallCmd: Context-Command-List; Context-Command-List; ... Context-Command-List;0

Context-Command-List: Context-Identifier (1 character) Command Command ... Command;

Command: Input-Sequence; Action-Identifier 1 (1 character), Action-Identifier 2 (1 character)

Input-Sequence: One or more characters from the following set

· 0-9,#*: DTMF digits

· f: hook flash

· o: off-hook

· @f: hook flash at any time

· h: on-hook

· S: #|*

· N: 0|1|2|3|4|5|6|7|8|9

· D: N|S

· v is a variable number (1 or more) of characters from the above list. It must be followed by a character which acts as the terminator of this variable part.

As can be seen from the above description, each Context-Command-List has one Context-Identifier followed by one or more Commands. This allows a variable number of actions to be triggered by the relevant user input commands for any state. Each command is composed of an Input-Sequence (which the user will enter when the Cisco ATA 186 is in a given state) and two Action-Identifier characters which define the action(s) taken by the Cisco ATA 186 in response to the Context and Input-Sequence. If the Cisco ATA 186 only takes one action, one of the two Action-Identifier characters will be a null-action.

Context-Identifiers

Table G-1 Context Identifiers

Identifier	Context
A	CONFERENCE
B	PREDIAL
C	PREDIAL_HOLDING
D	CONNECTED
E	CONNECTED_HOLDING
F	CONNECTED_ALERTING
G	HOLDING
H	CONFIGURING
I	CONFIGURING_HOLDING
J	3WAYCALLING
K	CALLWAITING
L	IDLE
M	RINGING
N	DIALING
O	CALLING
P	Reserved (ANSWERING)
Q	Reserved (CANCELING)
R	Reserved (DISCONNECTING)
S	WAITHOOK
T	DIALING_HOLDING
U	CALLING_HOLDING
V	Reserved (ANSWERING_HOLDING)
W	Reserved (HOLDING_HOLDING)
X	Reserved (CANCELING_HOLDING)

Table G-1 Context Identifiers (continued)

Identifier	Context
Y	Reserved (DISCONNECTING_HOLDING)
Z	Reserved (HOLDING_ALERTING)
a	WAITHOOK_ALERTING
b	WAITHOOK_HOLDING

Action Identifiers

Table G-2 Action Identifiers

Identifier	Action
A	NONE
B	Seizure (User Intendes To Dial Or Configure
C	Continue to dial
D	Call Return
E	Hold the active call
F	Retrieve the waiting call
G	Cancel the call attempt
H	Disconnect the call
I	Blind transfer the call to the number
N	Go to configuration mode
O	Release the call
P	Answer the incoming call
Q	Transfer with consultation
R	Say busy to the caller
a	None
b	Forward all calls to the given number

Table G-2 Action Identifiers (continued)

c	Forward on busy to the given number
d	Forward on no answer to the given number
e	Cancel call forward
f	CLIP for the next call
g	CLIR for the next call
h	Enable Call Waiting for the next call
i	Disable Call Waiting for the next call
x	Enable Fax Mode for the next call

Call Command Example

In addition to the provisioned call commands, the Cisco ATA 186 has a default list of call commands to handle common call scenarios. The default call commands can be overwritten by the provisioned call commands. If any Context/Input-Sequence appears in both the default Call Command string and the manually entered string, the manually entered value takes precedence.

The default Call Command string is as follows:

```
Bf;BAN;CA;CN;CAf;OF;Df;EB;I@f;OF;H@f;OA;Lo;BAf;BA;Mo;PA;ND;CAf;OA;Of;G
A;Pf;HA;Qf;OA;Rf;OA;Sf;OA;TD;CAf;OF;Uf;GF;Vf;HF;Wf;FF;Xf;AF;Yf;AF;Zf;A
P;bf;OF;af;OP;
```

In this section, the Call Command string is broken down into its component Context-Command-Lists as follows:

```
Call Command Fragment;
    Context-Identifier
        Input-Sequence1 Action1 Action2;
        (optional) Input-Sequence2 Action1 Action2;
```

Note When reading a Call Command string, you can identify the end of one Context-Command-List and the beginning of the next by noting whether there is a terminating ';' immediately after the second Action-Identifier, or whether it is followed by an input sequence.

```
Bf;BAN;CA;
    Predial
         hook-flash; Seizure NONE
         0|1|...|9; Continue-to-dial NONE;
CN;CAf;OF;
    Predial_Holding
         0|1|...|9; Continue-to-dial NONE
         hook-flash; Release-the-call Retrieve-the-waiting-call;
Df;EB;
    Connected
         hook-flash; Hold-the-active-call Seizure;
I@f;OF;
    Configuring_Holding
         hook-flash (at any time); Release-the-call
        Retrieve-the-waiting-call;
H@f;OA;
    Configuring
         hook-flash (at any time); Release-the-call NONE;
Lo;BAf;BA;
    Idle
         off-hook; Seizure NONE;
         hook-flash; Seizure NONE;
Mo;PA;
    Ringing
         off-hook; Answer-the-incoming-call NONE;
ND;CAf;OA
    Dialing
         0|1|...|9|#|*; Continue-to-dial NONE
         hook-flash; Release-the-call NONE;
Of;GA;
    Calling
         hook-flash; Cancel-the-call-attempt NONE;
Pf;HA;
    Answering
         hook-flash; Disconnect-the-call NONE;
Qf;OA;
    Canceling
         hook-flash; Release-the-call NONE;
Rf;OA;
    Disconnecting
         hook-flash; Release-the-call NONE;
Sf;OA;
    Waithook
         hook-flash; Release-the-call NONE;
TD;CAf;OF;
    Dialing_Holding
         0|1|...|9|#|*; Continue-to-dial NONE;
         hook-flash; Release-the-call NONE;
```

```
Uf;GF;
    Calling_Holding
         hook-flash; Cancel-the-call-attempt
        Retrieve-the-waiting-call;
Vf;HF;
    Answering_Holding
         hook-flash; Disconnect-the-call Retrieve-the-waiting-call;
Wf;FF;
    Holding_Holding
         hook-flash; Retrieve-the-waiting-call
        Retrieve-the-waiting-call;
Xf;AF;
    Canceling_Holding
         hook-flash; NONE Retrieve-the-waiting-call;
Yf;AF;
    Disconnecting_Holding
         hook-flash; NONE Retrieve-the-waiting-call;
Zf;AP;
    Holding_Alerting
         hook-flash; NONE Answering;
bf;OF;
    Waithook_Holding
         hook-flash; Release-the-call Retrieve-the-waiting-call;
af;OP;
    Waithook_Holding
         hook-flash; Release-the-call Answer-the-incoming-call;
```

Call Command Behaviors

Table G-3 summarizes the differing Call Command behaviors.

Note This information is based on the U.S. and Sweden default call commands.

The notations used in the table are:

- FE: Far end
- AFE: Active Far End, a connected far end which is not placed on hold
- WFE: Waiting Far End, a connected far end which is being placed on hold, or an incoming caller waiting to be answered
- R: Hook Flash
- ONH: On Hook
- OFH: Off Hook
- 0-9,*,#: DTMF digits
- v: a variable length string, usually a phone number, and does not include '#'
- CWT: call waiting tone

Table G-3 Call Command Behaviors

Context	Commands
IDLE: Phone is on-hook; Cisco ATA 186 is waiting for incoming call	OFH: Start dial-tone and goto PREDIAL;New incoming call, or a waiting call (started before it enters IDLE): Start ringing the phone and goto RINGING
PREDIAL: Phone just went off-hook but no DTMF entered yet; Cisco ATA 186 plays dial-tone	US and Sweden # , * : Stop dial-tone, goto CONFIG, and prepare to accept a complete configuration sequence; 0-9: Stop dial-tone, start invoking dial-plan rules, and goto DIALING to accept a complete phone number
DIALING: User entering phone number, which is parsed with the given dial-plan rules	R: Abort dialing, restarts dial-tone, and revert to PREDIAL;Invalid phone number: Abort dialing, plays fast-busy, and goto WAITHOOK;

Table G-3 Call Command Behaviors (continued)

Context	Commands
CONFIG: User configuring a supplementary service	US *69: Call Return #72v#: Forward unconditiona to number specified in 'v'l; (PacBell use 72#)#73:Cancel any call forwarding; (PacBell use 73#)#74v#: Forward on busy to number specified in 'v'; (PacBell don't enable this service from the phone)#75v#: Forward on no answer to number specified in 'v'; (Pac Bell don't enable this service from the phone)*67: CLIR in the next call (if global profile is CLIP);*82: CLIP for the next call (if global user profile is CLIR);*70: Disable call waiting in the next call;*99: Enable Fax Mode in the next call; (non-standard) Sweden *21*v#: Forward unconditionally to number specified in 'v';*67*v#: Forward on busy to number specified in 'v';*61*v#: Forward on no answer to number specified in 'v';#21#: Cancel any call forwarding;#67#: Cancel any call forwarding;#61#: Cancel any call forwarding;#31#: CLIR in the next call;*31#: CLIR in the next call; (Note: no CLIP, maybe all calls by default is CLIP in Sweden?)*43#: Enable call waiting in the next call (Sweden allows globally disable call waiting?);#43#: Disable call waiting in the next call;*69#: Call Return; (non-standard)*99#: Enable Fax Mode in the next call; (non-standard)All RegionsR or any unrecognized sequence: Abort configuration, restart dial-tone, and revert to PREDIAL;Any complete configuration sequence: Carry out the configuration command, restart dial-tone, and revert to PREDIAL
CALLING: Phone number is sent; Cisco ATA 186 is waiting for response from the far end	R: Cancel the outgoing call, restarts dial-tone, and revert to PREDIAL

Table G-3 Call Command Behaviors (continued)

Context	Commands
RINGING: Cisco ATA 186 is ringing the phone to alert user of an incoming call	OFH: Stop ringing, answer the call, and goto CONNECTED
CONNECTED: The Cisco ATA 186 is connected with one far end party; Cisco ATA 186 may be the caller or the callee	US and Sweden: R: Hold current call, play dial-tone to dial 2nd number, and goto PREDIAL_HOLDING
WAITHOOK: Far end hangs up while in CONNECTED state; Cisco ATA 186 plays fast-busy after 5 seconds in this state	R: Stop fast-busy, start dial-tone, and goto PREDIAL
CONNECTED_ALERTING: Cisco ATA 186 receives another call while in CONNECTED state; Cisco ATA 186 plays Call Waiting tone periodically (every 10 seconds for US; l second for Sweden)	US: R: Place current call on-hold, answer the waiting call, and goto CALLWAITING; Sweden: R0: Continue current call, reject (say busy to) the waiting call, and revert to CONNECTED; R1: Disconnect current call, answer the waiting call, and goto CONNECTED; R2: Place current call on-hold, answer waiting call, and goto CALLWAITING; R3: Continue with current call, answer the waiting call and goto CONFERENCE; All Regions ONH: Disconnect current call and goto IDLE (in which Cisco ATA 186 automatically starts ringing the phone, and goto RINGING); AFE hangs up: Goto WAITHOOK_ALERTING, continue to play CWT; WFE cancels the call: Stop CWT and revert to CONNECTED

Table G-3 Call Command Behaviors (continued)

Context	Commands
CALL WAITING: Cisco ATA 186 is connected to two far end users on the same line; one of them is in active conversation (the active far end or AFE) while the other is on-hold (the waiting far end or WFE). This state is initially entered when the Cisco ATA 186 is connected to one of the far end while the other far end call into the Cisco ATA 186	US: R: Place the AFE on-hold and retrieve the WFE;(ONH: Transfer the WFE to the AFE, drop out of the call, and goto PREDIAL;)SwedenR1: Disconnect current call, answer the waiting call, and goto CONNECTED;R2: Place the AFE on-hold and retrieve the WFE;R3: Retrieve the WFE, and goto CONFERENCE;(R4: Transfer the WFE to the AFE, drop out of the call, and goto PREDIAL;)
3WAYCALLING: Cisco ATA 186 is connected to two far end users on the same line; one of them is in active conversation (the active far end or AFE) while the other is on-hold (the waiting far end or WFE). This state is initially entered when the Cisco ATA 186 is connected to one of the far end, then places this far end on hold and call the second far end	US R: Retrieve the WFE and goto CONFERENCE;(ONH: Transfer the WFE to the AFE, drop out of the call, and goto PREDIAL;) Sweden Same as in CALLWAITING state
CONFERENCE: Cisco ATA 186 is connected to two active far end simultaneously; Cisco ATA 186 performs audio mixing such that evey party can hear the other two parties but not themselves	USR: Disconnect the last callee and stay connected with the first party, and revert to CONNECTEDSweden(R4: Transfer one FE to the other, drop out of the call, and goto PREDIAL;)
PREDIAL_HOLDING: Cisco ATA 186 user places a connected call on-hold and prepares to dial a second number; Cisco ATA 186 plays dial-tone	US *,#: Stop dial-tone, goto CONFIG_HOLDING, and prepare to collect a configuration command;0-9: Stop dial-tone, goto DIALING_HOLDING, and prepare to complete dialing a second phone number; SwedenSame as USAll RegionsR: Stop dial-tone, retrieve the WFE, and revert to CONNECTED

Table G-3 Call Command Behaviors (continued)

Context	Commands
CONFIG_HOLDING: A connected FE is placed on hold, while the Cisco ATA 186 is entering a configuration command	US*67: CLIR for the next call;*82: CLIP for the next call;#90v#: Blind transfer to the number specified in 'v'; disconnect the call and goto PREDIALSweden#31# or *31#: CLIR in the next call*90*v#: Blind transfer to the number specified in 'v'; disconnect the call and goto PREDIAL (non-standard)All RegionsR or any unrecognized sequence: Abort configuration, restarts dial-tone, and goto to PREDIAL_HOLDING A complete configuration sequence: Carries out the command, and goto PREDIAL_HOLDING
DIALING_HOLDING: Cisco ATA 186 user entering a second phone number to call while placing a connected call on-hold	Collected digits match a dial-plan rule: Call the new number, and goto CALLING_HOLDING;R: Aborts dialing and revert to PREDIAL_HOLDING
CALLING_HOLDING: Cisco ATA 186 waiting for a second far end to response while placing a connected call on-hold	R: Cancel the call and revert to PREDIAL_HOLDING;(ONH: Cancel the call and transfer the waiting party to the callee, and revert back to PREDIAL)
WAITHOOK_HOLDING: The AFE hangs-up to disconnect the current call while there is a WFE being put on-hold	R: Retrieve the WFE and goto CONNECTED
AITHOOK_ALERTING: The AFE hangs-up while a waiting call is alerting	W R: Stop CWT, answer the waiting call, and goto CONNECTED;WFE cancels the call: Stop CWT, goto WAITHOOK;ONH: Goto IDLE (in which Cisco ATA 186 automatically starts ringing the phone, and goto RINGING)

U.S. Call Command

The default U.S. Call command is:

Af;AH;BS;NA;CS;NA;Df;EB;Ff;EP;Kf;EFh;HQ;Jf;AFh;HQ;I*67;gA*82;fA#90v#;OI;H#72v#;bA#74v#;cA#75v#;dA#73;eA*67;gA*82;fA*70;iA*69;DA*99;xA;Uh;GQ;

Af;AH;

Conference

hook-flash; NONE Disconnect-the-call;

BS;NA;

Predial

#|*; Go-to-configuration-mode NONE;

CS;NA;

Predial_Holding

#|*; Go-to-configuration-mode NONE;

Df;EB;

Connected

hook-flash; Hold-the-active-call Seizure;

Ff;EP;

Connected_Alerting

hook-flash; Hold-the-active-call Answer-the-incoming-call;

Kf;EFh;HQ;

CallWaiting

hook-flash; Hold-the-active-call Retrieve-the-waiting-call

on-hook; Disconnect-the-call Transfer-with-consultation;

Jf;AFh;HQ;

3WayCalling

hook-flash; NONE Retrieve-the-waiting-call

on-hook; Disconnect-the-call Transfer-with-consultation;

I*67;gA*82;fA#90v#;OI;

Configuring_Holding

*67; CLIR-for-the-next-call NONE

*82; CLIP-for-the-next-call NONE

#90v#; Release-the-call Blind-transfer-the-call-to-the-given-number;

H#72v#;bA#74v#;cA#75v#;dA#73;eA*67;gA*82;fA*70;iA*69;DA*99;xA;

Configuring

#72v#; Forward-all-calls-to-the-given-number NONE

#74v#; Forward-on-busy-to-the-given-number NONE

#75v#; Forward-on-no-answer-to-the-given-number NONE

#73; Cancel-call-forward NONE

*67; CLIR-for-the-next-call NONE

*82; CLIP-for-the-next-call NONE

*70; Disable-call-waiting-for-the-next-call NONE

*99; Enable-fax-mode-for-the-next-call NONE;

Uh;GQ;

Calling_Holding

on-hook; Cancel-the-call-attempt Transfer-with-consultation;;

Sweden Call Command

BS;NA;CS;NA;Df;EB;Ff0;ARf1;HPf2;EPf3;AP;Kf1;HFf2;EFf3;AFf4;HQ;Jf1;
HFf2;EFf3;AFf4;HQ;Af4;HQ;I*31#;gA#31#;gA*90*v#;OI;H*21*v#;bA*61*v#
;dA*67*v#;cA#21#;eA#61#;eA#67#;eA*31#;gA#31#;gA*43#;hA#43#;iA*69#;
DA*99#;xA;Uh;GQ;

BS;NA;

Predial

#|*; Go-to-configuration-mode NONE;

```
CS;NA;
Predial_Holding
   #|*;   Go-to-configuration-mode NONE;
Df;EB;
Connected
   hook-flash;  Hold-the-active-call Seizure;
Ff0;ARf1;HPf2;EPf3;AP;
Connected_Alerting
   hook-flash0;  NONE Say-busy-to-caller;
   hook-flash1;  Disconnect-the-call Answer-the-incoming-call;
   hook-flash2;  Hold-the-active-call Answer-the-incoming-call;
   hook-flash3;  NONE Answer-the-incoming-call;
Kf1;HFf2;EFf3;AFf4;HQ;
CallWaiting
   hook-flash1;  Disconnect-the-call Retrieve-the-waiting-call;
   hook-flash2;  Hold-the-active-call Retrieve-the-waiting-call;
   hook-flash3;  NONE Retrieve-the-waiting-call;
   hook-flash4; Disconnect-the-call Transfer-with-consultation;
Jf1;HFf2;EFf3;AFf4;HQ;
3WayCalling
   hook-flash1;  Disconnect-the-call Retrieve-the-waiting-call;
   hook-flash2;  Hold-the-active-call Retrieve-the-waiting-call;
   hook-flash3;  NONE Retrieve-the-waiting-call;
   hook-flash4;  Disconnect-the-call Transfer-with-consultation;
Af4;HQ;
Conference
   hook-flash4;  Disconnect-the-call Transfer-with-consultation;
I*31#;gA#31#;gA*90*v#;OI;
Configuring_Holding
```

*31#; CLIR-for-the-next-call NONE;

#31#; CLIR-for-the-next-call NONE;

*90*v#; Release-the-call Blind-transfer-the-call-to-the-given-number;

H*21*v#;bA*61*v#;dA*67*v#;cA#21#;eA#61#;eA#67#;eA*31#;gA#31#;gA*43#;hA#43#;iA*69#;DA*99#;xA;

Configuring

*21*v#; Forward-all-calls-to-the-given-number NONE;

*61*v#; Forward-on-no-answer-to-the-given-number NONE;

*67*v#; Forward-on-busy-to-the-given-number NONE;

#21#; Cancel-call-forward NONE;

#61#; Cancel-call-forward NONE;

#67#; Cancel-call-forward NONE;

*31#; CLIR-for-the-next-call NONE;

#31#; CLIR-for-the-next-call NONE;

*43#; Enable-call-waiting-for-the-next-call NONE;

#43#; Disable-call-waiting-for-the-next-call NONE;

*69#; Call-return NONE;

*99#; Enable-fax-mode-for-the-next-call NONE

Uh;GQ;

Calling_Holding

on-hook; Cancel-the-call-attempt Transfer-with-consultation

Terms and Acronyms

This Appendix contains definitions of some of the terms and acronyms used in the Cisco ATA 186 Installation and Configuration Guide. For additional definitions, see the Cisco Systems *Internetworking Terms and Acronyms Guide*.

C

CNG Comfort Noise Generation.

D

DSP Digital Signal Processor.

DTMF Dual Tone Multi-Frequency.

F

FoIP FAX over IP

FXO Foreign Exchange Office. An FXO connects to a central office – this is the interface a standard phone offers.

FXS Foreign Exchange Station

H

H.323 An ITU standard that provide a foundation for audio, video and data communications across IP-based network.

I

IVR Interactive Voice Response

S

SDP Session Description Protocol.

SIP Session Initiation Protocol. A signaling protocol for internet telephony.

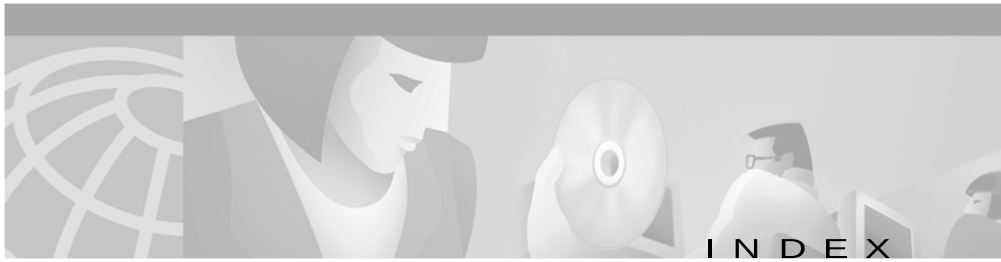
SLIC Subscriber Line Interface Circuit. An IC providing central office-like telephone interface functionality.

SOHO Small Office / Home Office.

V

VAD Voice Activity Detection.

VoIP Voice over Internet Protocol. A packet based network protocol.



Symbols

A

C

D

E

T

U

V

W