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- 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.)

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Cisco ATA 186 and Cisco ATA 188 Analog Telephone Adaptor Administrator’s Guide for SIP (version 3.0)
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CONTENTS

Preface 13
  Overview 13
  Audience 13
  Organization 14
  Conventions 14
  Related Documentation 18
  Obtaining Documentation 18
    World Wide Web 18
    Documentation CD-ROM 19
    Ordering Documentation 19
    Documentation Feedback 19
  Obtaining Technical Assistance 19
    Cisco.com 20
    Technical Assistance Center 20
      Cisco TAC Web Site 20
      Cisco TAC Escalation Center 21

Cisco Analog Telephone Adaptor Overview 1
  Session Initiation Protocol (SIP) Overview 2
    SIP Capabilities 3
    Components of SIP 3
      SIP Clients 4
      SIP Servers 4
  Hardware Overview 5
  Software Features 7
    Voice Codecs Supported 7
    Additional Supported Signaling Protocols 8
    Other Supported Protocols 8
    Cisco ATA SIP Services 8
    Fax Services 9
    Methods Supported 9
    Supplementary Services 10
  Installation and Configuration Overview 10
Installing the Cisco ATA

Network Requirements
Safety Recommendations
What the Cisco ATA Package Includes
What You Need
Installation Procedure
Power-Down Procedure

Configuring the Cisco ATA for SIP

Default Boot Load Behavior
Specifying a Preconfigured VLAN ID or Disabling VLAN IP Encapsulation
Steps Needed to Configure the Cisco ATA
   Basic Configuration Steps in a TFTP Server Environment
   Basic Configuration Steps in a Non-TFTP Server Environment
Configuring the Cisco ATA Using a TFTP Server
   Setting Up the TFTP Server with Cisco ATA Software
   Configurable Features and Related Parameters
Creating Unique and Common Cisco ATA Configuration Files
   Using atapname.exe Tool to Obtain MAC Address
   Using Encryption With the cfgfmt Tool
   Examples of Upgrading to Stronger Encryption Key
atadefault.cfg Configuration File
Configuring the Cisco ATA to Obtain its Configuration File from the TFTP Server
   Using a DHCP Server
   Without Using a DHCP Server
Voice Configuration Menu
   Using the Voice Configuration Menu
   Entering Alphanumeric Values
   Resetting the Cisco ATA to Factory Default Values
Cisco ATA Web Configuration Page
Refreshing or Resetting the Cisco ATA
   Procedure to Refresh the Cisco ATA
   Procedure to Reset the Cisco ATA
Obtaining Cisco ATA Configuration File After Failed Attempt
Upgrading the SIP Signaling Image
### Basic and Additional SIP Services

<table>
<thead>
<tr>
<th>Section</th>
<th>Page</th>
</tr>
</thead>
<tbody>
<tr>
<td>Important Basic SIP Services</td>
<td>1</td>
</tr>
<tr>
<td>Required Parameters</td>
<td>1</td>
</tr>
<tr>
<td>Establishing Authentication</td>
<td>2</td>
</tr>
<tr>
<td>Setting the Codec</td>
<td>3</td>
</tr>
<tr>
<td>Configuring Refresh Interval</td>
<td>3</td>
</tr>
<tr>
<td><strong>Additional SIP Services</strong></td>
<td>3</td>
</tr>
<tr>
<td>Advanced Audio Configuration</td>
<td>4</td>
</tr>
<tr>
<td>Billable Features</td>
<td>4</td>
</tr>
<tr>
<td>Call Forwarding Setting Removal Using HTTP</td>
<td>5</td>
</tr>
<tr>
<td>Call-Waiting Hang-Up Alert</td>
<td>5</td>
</tr>
<tr>
<td>Enabling the Call-Waiting Hang-Up Alert Feature</td>
<td>6</td>
</tr>
<tr>
<td>Default Behavior of Call-Waiting Calls</td>
<td>6</td>
</tr>
<tr>
<td>Comfort Noise During Silence Period When Using G.711</td>
<td>6</td>
</tr>
<tr>
<td>Configurable Hook Flash Timing</td>
<td>7</td>
</tr>
<tr>
<td>Configurable Mixing of Call Waiting Tone and Audio</td>
<td>7</td>
</tr>
<tr>
<td>Configurable On-hook delay</td>
<td>7</td>
</tr>
<tr>
<td>Configurable Reboot of Cisco ATA</td>
<td>7</td>
</tr>
<tr>
<td>Diagnostics for Debugging</td>
<td>7</td>
</tr>
<tr>
<td>Dial Plan</td>
<td>7</td>
</tr>
<tr>
<td>Disabling Access To The Web Interface</td>
<td>8</td>
</tr>
<tr>
<td>Display-Name Support for Caller ID</td>
<td>8</td>
</tr>
<tr>
<td>Distinctive Ringing</td>
<td>8</td>
</tr>
<tr>
<td>DNS SRV Support</td>
<td>9</td>
</tr>
<tr>
<td>Hardware Information Display</td>
<td>9</td>
</tr>
<tr>
<td>NAT Gateway</td>
<td>9</td>
</tr>
<tr>
<td>NAT/PAT Translation</td>
<td>10</td>
</tr>
<tr>
<td>Network Timing</td>
<td>10</td>
</tr>
<tr>
<td>Obtaining Network Status Before and After Getting IP Connectivity</td>
<td>10</td>
</tr>
<tr>
<td>Privacy Options</td>
<td>10</td>
</tr>
<tr>
<td>Network Infrastructure Requirements</td>
<td>11</td>
</tr>
<tr>
<td>Anonymity for Called Party</td>
<td>11</td>
</tr>
<tr>
<td>Anonymous User Name Support for SIP INVITE Requests</td>
<td>11</td>
</tr>
<tr>
<td>Privacy Token Support for SIP Diversion Header</td>
<td>12</td>
</tr>
<tr>
<td>Progress Tones</td>
<td>13</td>
</tr>
<tr>
<td>Real-Time Transport Protocol (RTP) Statistics Reporting</td>
<td>13</td>
</tr>
<tr>
<td>Receiver-tagged VIA header</td>
<td>13</td>
</tr>
<tr>
<td>Redundant Proxy Support for BYE/CANCEL Request</td>
<td>13</td>
</tr>
</tbody>
</table>
Contents

Repeat Dialing on Busy Signal  14
Retransmitting SIP requests and SIP Responses  15
Setting Up and Placing a Call Without Using a SIP Proxy  15
  Configuration  15
  Placing an IP Call  16
SipOutBoundProxy Support  16
SIP Proxy Server Redundancy  16
SIP Session-Timer Support  17
Status of Phone Service Using HTTP  17
STUN Support  18
  Types of NATs  18
  NAT Traversal  18
  STUN Configuration Parameters  19
Stuttering Dial Tone on Unconditional Call Forward  19
Toll Restrictions for Call Forwarding and Outgoing Calls  19
User Configurable Call Waiting Permanent Default Setting  20
User Configurable Timeout On No Answer for Call Forwarding  20
Voice Prompt Confirmation for Call Waiting and Call Forwarding  20
  Base Number  21
  Relevant Bit of OpFlags Parameter  21
XML Pages of Cisco ATA Information  22
  Current configuration  22
  Current statistics  22
  Current service values  22
Complete Reference Table of all Cisco ATA SIP Services  23

Parameters and Defaults  1
  Configuration Text File Template  2
  User Interface (UI) Security Parameter  4
    UIPassword  4
  Parameters for Configuration Method and Encryption  4
    UseTFTP  5
    TftpURL  5
    CfgInterval  6
    EncryptKey  6
    EncryptKeyEx  7
  Network Configuration Parameters  8
    DHCP  8
    StaticIp  9
<table>
<thead>
<tr>
<th>Contents</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>StaticRoute</td>
<td>9</td>
</tr>
<tr>
<td>StaticNetMask</td>
<td>10</td>
</tr>
<tr>
<td>DNS1IP</td>
<td>10</td>
</tr>
<tr>
<td>DNS2IP</td>
<td>11</td>
</tr>
<tr>
<td>NTPIP</td>
<td>11</td>
</tr>
<tr>
<td>AltNTPIP</td>
<td>12</td>
</tr>
<tr>
<td>VLANSetting</td>
<td>12</td>
</tr>
<tr>
<td>SIP Configuration Parameters</td>
<td>13</td>
</tr>
<tr>
<td>GkOrProxy</td>
<td>13</td>
</tr>
<tr>
<td>AltGk</td>
<td>14</td>
</tr>
<tr>
<td>AltGkTimeout</td>
<td>15</td>
</tr>
<tr>
<td>UID0</td>
<td>15</td>
</tr>
<tr>
<td>PWD0</td>
<td>16</td>
</tr>
<tr>
<td>UID1</td>
<td>16</td>
</tr>
<tr>
<td>PWD1</td>
<td>17</td>
</tr>
<tr>
<td>LoginID0</td>
<td>17</td>
</tr>
<tr>
<td>LoginID1</td>
<td>18</td>
</tr>
<tr>
<td>UseLoginID</td>
<td>18</td>
</tr>
<tr>
<td>SIPPort</td>
<td>19</td>
</tr>
<tr>
<td>SIPRegInterval</td>
<td>19</td>
</tr>
<tr>
<td>SIPRegOn</td>
<td>20</td>
</tr>
<tr>
<td>MAXRedirect</td>
<td>20</td>
</tr>
<tr>
<td>SipOutBoundProxy</td>
<td>21</td>
</tr>
<tr>
<td>NATIP</td>
<td>21</td>
</tr>
<tr>
<td>NatServer</td>
<td>22</td>
</tr>
<tr>
<td>NatTimer</td>
<td>22</td>
</tr>
<tr>
<td>MsgRetryLimits</td>
<td>24</td>
</tr>
<tr>
<td>SessionTimer</td>
<td>26</td>
</tr>
<tr>
<td>SessionInterval</td>
<td>28</td>
</tr>
<tr>
<td>MinSessionInterval</td>
<td>28</td>
</tr>
<tr>
<td>DisplayName0</td>
<td>29</td>
</tr>
<tr>
<td>DisplayName1</td>
<td>29</td>
</tr>
<tr>
<td>Audio Configuration Parameters</td>
<td>30</td>
</tr>
<tr>
<td>MediaPort</td>
<td>30</td>
</tr>
<tr>
<td>RxCodec</td>
<td>31</td>
</tr>
<tr>
<td>TxCodec</td>
<td>31</td>
</tr>
<tr>
<td>LBRCodec</td>
<td>32</td>
</tr>
<tr>
<td>AudioMode</td>
<td>32</td>
</tr>
<tr>
<td>NumTxFrames</td>
<td>34</td>
</tr>
<tr>
<td>TOS</td>
<td>34</td>
</tr>
</tbody>
</table>
Operational Parameters  35
  CallFeatures  35
  PaidFeatures  36
  CallCmd  37
  FeatureTimer  38
  FeatureTimer2  39
  SigTimer  40
  ConnectMode  41
  OpFlags  45
  TimeZone  48

Telephone Configuration Parameters  49
  CallerIdMethod  49
  Polarity  51
  FXSInputLevel  52
  FXSOutputLevel  52

Tone Configuration Parameters  53
  Tone Parameter Syntax—Basic Format  53
  Tone Parameter Syntax—Extended Formats  54
    Extended Format A  55
    Extended Format B  56
  Recommended Values  59
  Specific Tone Parameter Information  60
    DialTone  60
    BusyTone  61
    ReorderTone  61
    RingbackTone  62
    CallWaitTone  62
    AlertTone  63
    SITone  63
    RingOnOffTime  64

Dial Plan Parameters  64
  DialPlan  64
    Dial Plan Commands  65
    Dial Plan Rules  66
    Dial Plan Examples  70
  DialPlanEx  72
  IPDialPlan  72

Diagnostic Parameters  73
<table>
<thead>
<tr>
<th>Section</th>
<th>Page</th>
</tr>
</thead>
<tbody>
<tr>
<td>NPrintf</td>
<td>73</td>
</tr>
<tr>
<td>TraceFlags</td>
<td>73</td>
</tr>
<tr>
<td>SyslogIP</td>
<td>74</td>
</tr>
<tr>
<td>SyslogCtrl</td>
<td>75</td>
</tr>
<tr>
<td>CFGID—Version Parameter for Cisco ATA Configuration File</td>
<td>76</td>
</tr>
<tr>
<td><strong>Call Commands</strong></td>
<td>1</td>
</tr>
<tr>
<td>Call Command Structure</td>
<td>1</td>
</tr>
<tr>
<td>Syntax</td>
<td>2</td>
</tr>
<tr>
<td>Context-Identifiers</td>
<td>3</td>
</tr>
<tr>
<td>Input Sequence Identifiers</td>
<td>4</td>
</tr>
<tr>
<td>Action Identifiers</td>
<td>4</td>
</tr>
<tr>
<td>Call Command Example</td>
<td>5</td>
</tr>
<tr>
<td>Call Command Behavior</td>
<td>7</td>
</tr>
<tr>
<td><strong>Configuring and Debugging Fax Services</strong></td>
<td>1</td>
</tr>
<tr>
<td>Using Fax Pass-through Mode</td>
<td>1</td>
</tr>
<tr>
<td>Configuring the Cisco ATA for Fax Pass-through mode</td>
<td>2</td>
</tr>
<tr>
<td>AudioMode</td>
<td>2</td>
</tr>
<tr>
<td>ConnectMode</td>
<td>3</td>
</tr>
<tr>
<td>Configuring Cisco IOS Gateways to Enable Fax Pass-through</td>
<td>3</td>
</tr>
<tr>
<td>Enable Fax Pass-through Mode</td>
<td>4</td>
</tr>
<tr>
<td>Disable Fax Relay Feature</td>
<td>5</td>
</tr>
<tr>
<td>Using FAX Mode</td>
<td>6</td>
</tr>
<tr>
<td>Configuring the Cisco ATA for Fax Mode</td>
<td>6</td>
</tr>
<tr>
<td>Configuring the Cisco ATA for Fax Mode on a Per-Call Basis</td>
<td>7</td>
</tr>
<tr>
<td>Configuring the Cisco IOS Gateway for Fax Mode</td>
<td>7</td>
</tr>
<tr>
<td>Debugging the Cisco ATA 186/188 Fax Services</td>
<td>7</td>
</tr>
<tr>
<td>Common Problems When Using IOS Gateways</td>
<td>7</td>
</tr>
<tr>
<td>Using prserv for Diagnosing Fax Problems</td>
<td>9</td>
</tr>
<tr>
<td>prserv Overview</td>
<td>9</td>
</tr>
<tr>
<td>Analyzing prserv Output for Fax Sessions</td>
<td>10</td>
</tr>
<tr>
<td>Using rtpcatch for Diagnosing Fax Problems</td>
<td>12</td>
</tr>
<tr>
<td>rtpcatch Overview</td>
<td>12</td>
</tr>
<tr>
<td>Example of rtpcatch</td>
<td>14</td>
</tr>
<tr>
<td>Analyzing rtpcatch Output for Fax Sessions</td>
<td>16</td>
</tr>
<tr>
<td>Using rtpcatch to Analyze Common Causes of Failure</td>
<td>18</td>
</tr>
</tbody>
</table>
Contents

rtpcatch Limitations 20

Upgrading the Cisco ATA Signaling Image 1
Upgrading the Signaling Image from a TFTP Server 1
Upgrading the Signaling Image Manually 2
  Preliminary Steps 3
  Running the Executable File 3
  Upgrade Requirements 3
  Syntax 3
  Upgrade Procedure 4
Confirming a Successful Signaling Image Upgrade 5
  Using a Web Browser 5
  Using the Voice Configuration Menu 5

Troubleshooting 1
  General Troubleshooting Tips 1
  Symptoms and Actions 2
  Installation and Upgrade Issues 3
  Debugging 4
  Using System Diagnostics 6
  Local Tone Playout Reporting 10
  Obtaining Network Status Prior to Getting IP Connectivity 11
  Obtaining Network Status After Getting IP Connectivity 12
  DHCP Status HTML Page 13
  Real-Time Transport Protocol (RTP) Statistics Reporting 13
  Frequently Asked Questions 14
  Contacting TAC 15

Using SIP Supplementary Services 1
  Changing Call Commands 1
  Cancelling a Supplementary Service 1
  Common Supplementary Services 1
    Caller ID 2
    Call-Waiting Caller ID 2
    Voice Mail Indication 2
    Unattended Transfer 3
Contents

Semi-unattended Transfer 3
Fully Unattended Transfer 3
Attended Transfer 4
Making a Conference Call in the United States 4
Making a Conference Call in Sweden 4
Call Waiting in the United States 5
Call Waiting in Sweden 5
About Call Forwarding 5
Call Forwarding in the United States 5
Call Forwarding in Sweden 6
Call Return in the United States 6
Call Return in Sweden 6
Calling Line Identification Presentation 6
About Calling Line Identification Restriction 6
Calling Line Identification Restriction in the United States 7
Calling Line Identification Restriction in Sweden 7

Voice Menu Codes 1

Cisco ATA Specifications 1
  Physical Specifications 1
  Electrical Specifications 2
  Environmental Specifications 2
  Physical Interfaces 2
  Ringing Characteristics 3
  Software Specifications 3
  SIP Compliance Reference Information 5

SIP Call Flows 1
  Supported SIP Request Methods 1
  Call Flow Scenarios for Successful Calls 2
    Cisco ATA-to-SIP Server—Registration without Authentication 2
    Cisco ATA-to-SIP Server—Registration with Authentication 3
    Cisco ATA-to-Cisco ATA—Basic SIP to SIP Call without Authentication 6
    Cisco ATA-to-Cisco ATA—Basic SIP to SIP Call with Authentication 12

Recommended Cisco ATA Tone Parameter Values by Country 1

Glossary
Index
Preface

This preface includes the following sections:

- Overview, page 13
- Audience, page 13
- Organization, page 14
- Conventions, page 14
- Related Documentation, page 18
- Obtaining Documentation, page 18
- Obtaining Technical Assistance, page 19

Overview

The *Cisco ATA 186 and Cisco ATA 188 Analog Telephone Adaptor Administrator’s Guide (SIP)* provides the information you need to install, configure and manage the Cisco ATA 186 and Cisco ATA 188 on a Session Initiation Protocol (SIP) network.

**Note**

The term *Cisco ATA* is used throughout this manual to refer to both the Cisco ATA 186 and the Cisco ATA 188, unless differences between the Cisco ATA 186 and Cisco ATA 188 are explicitly stated.

Audience

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

Table 1 provides an overview of the organization of this guide.

<table>
<thead>
<tr>
<th>Table 1</th>
<th>Cisco ATA 186 and Cisco ATA 188 Analog Telephone Adaptor Administrator’s Guide (SIP) Organization</th>
</tr>
</thead>
<tbody>
<tr>
<td>Chapter</td>
<td>Description</td>
</tr>
<tr>
<td>Chapter 1, “Cisco Analog Telephone Adaptor Overview”</td>
<td>Provides descriptions of hardware and software features of the Cisco ATA Analog Telephone Adaptor along with a brief overview of the Session Initiation Protocol (SIP).</td>
</tr>
<tr>
<td>Chapter 2, “Installing the Cisco ATA”</td>
<td>Provides information about installing the Cisco ATA.</td>
</tr>
<tr>
<td>Chapter 3, “Configuring the Cisco ATA for SIP”</td>
<td>Provides information about configuring the Cisco ATA and the various methods for configuration.</td>
</tr>
<tr>
<td>Chapter 4, “Basic and Additional SIP Services”</td>
<td>Provides information about SIP services that the Cisco ATA supports.</td>
</tr>
<tr>
<td>Chapter 5, “Parameters and Defaults,”</td>
<td>Provides information on all parameters and defaults that you can use to configure the Cisco ATA.</td>
</tr>
<tr>
<td>Chapter 6, “Call Commands”</td>
<td>Provides the Cisco ATA call commands for SIP.</td>
</tr>
<tr>
<td>Chapter 7, “Configuring and Debugging Fax Services”</td>
<td>Provides instructions for configuring both ports of the Cisco ATA to support fax transmission.</td>
</tr>
<tr>
<td>Chapter 8, “Upgrading the Cisco ATA Signaling Image”</td>
<td>Provides instructions for remotely upgrading Cisco ATA software.</td>
</tr>
<tr>
<td>Chapter 9, “Troubleshooting”</td>
<td>Provides basic testing and troubleshooting procedures for the Cisco ATA.</td>
</tr>
<tr>
<td>Appendix A, “Using SIP Supplementary Services”</td>
<td>Provides end-user information about pre-call and mid-call services.</td>
</tr>
<tr>
<td>Appendix B, “Voice Menu Codes”</td>
<td>Provides a quick-reference list of the voice configuration menu options for the Cisco ATA.</td>
</tr>
<tr>
<td>Appendix C, “Cisco ATA Specifications”</td>
<td>Provides physical specifications for the Cisco ATA.</td>
</tr>
<tr>
<td>Appendix D, “SIP Call Flows”</td>
<td>Provides Cisco ATA call flows for SIP scenarios.</td>
</tr>
<tr>
<td>Appendix E, “Recommended Cisco ATA Tone Parameter Values by Country”</td>
<td>Provides tone parameters for various countries.</td>
</tr>
<tr>
<td>Glossary</td>
<td>Provides definitions of commonly used terms.</td>
</tr>
<tr>
<td>Index</td>
<td>Provides reference information.</td>
</tr>
</tbody>
</table>

Conventions

This document uses the following conventions:

- Alternative keywords are grouped in braces and separated by vertical bars (for example, \{x | y | z\}).
- Arguments for which you supply values are in italic font.
- Commands and keywords are in boldface font.
- Elements in square brackets ([ ]) are optional.
- Information you must enter is in boldface screen font.
• Optional alternative keywords are grouped in brackets and separated by vertical bars (for example, \[x \mid y \mid z\]).

• Terminal sessions and information the system displays are in screen font.

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

**Timesaver**
Means *the described action saves time*. You can save time by performing the action described in the paragraph.

**Tip**
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.

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

**Warning**
**IMPORTANT SAFETY INSTRUCTIONS**

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. Use the statement number provided at the end of each warning to locate its translation in the translated safety warnings that accompanied this device.

Statement 1071

**SAVE THESE INSTRUCTIONS**

**Waarschuwing**
**BELANGRIJKE VEILIGHEIDSINSTRUCTIES**

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 de standaard praktijken om ongelukken te voorkomen. Gebruik het nummer van de verklaring onderaan de waarschuwing als u een vertaling van de waarschuwing die bij het apparaat wordt geleverd, wilt raadplegen.

**BEWAAR DEZE INSTRUCTIES**
TÄRKEITÄ TURVALLISUUSOHJEITA


SÄILYTÄ NÄMÄ OHJEET
INSTRUÇÕES IMPORTANTES DE SEGURANÇA

Este símbolo de aviso significa perigo. Você está em uma situação que poderá ser causadora de lesões corporais. Antes de iniciar a utilização de qualquer equipamento, tenha conhecimento dos perigos envolvidos no manuseio de circuitos elétricos e familiarize-se com as práticas habituais de prevenção de acidentes. Utilize o número da instrução fornecido ao final de cada aviso para localizar sua tradução nos avisos de segurança traduzidos que acompanham este dispositivo.

GUARDE ESTAS INSTRUÇÕES

¡Advertencia! INSTRUCCIONES IMPORTANTES DE SEGURIDAD

Este símbolo de aviso indica peligro. Existe riesgo para su integridad física. Antes de manipular cualquier equipo, considere los riesgos de la corriente eléctrica y familiarícese con los procedimientos estándar de prevención de accidentes. Al final de cada advertencia encontrará el número que le ayudará a encontrar el texto traducido en el apartado de traducciones que acompaña a este dispositivo.

GUARDE ESTAS INSTRUCCIONES

Varning! VIKTIGA SÄKERHETSANVISNINGAR


SPARA DESSA ANVISNINGAR

Fontos biztonsági előírások

Ez a figyelmezetel vett veszély utal. Sérülésveszélyt jelent, amikor valamely kockázatot fenyeget az elektromos áramkörök okozta kockázatokra, és ismerkedjen meg a szokásos balesetvédelmi eljárásokkal. A kiadóban szereplő figyelmeztetések fordítása a készülékekhöz mellékeltek biztonsági figyelmeztetések között található; a fordítás az egyes figyelmeztetések végén látható szám alapján kereshető meg.

ORIZZE MEG EZEKET AZ UTASÍTÁSOKAT!

ВАЖНЫЕ ИНСТРУКЦИИ ПО СОБЛЮДЕНИЮ ТЕХНИКИ БЕЗОПАСНОСТИ

Этот символ предупреждения обозначает опасность. То есть имеет место ситуация, в которой следует опасаться телесных повреждений. Перед эксплуатацией оборудования выясните, каким опасностям может подвергаться пользователь при использовании электрических цепей, и ознакомьтесь с правилами техники безопасности для предотвращения возможных несчастных случаев. Воспользуйтесь номером заявления, приведенным в конце каждого предупреждения, чтобы найти его переведенный вариант в переводе предупреждений по безопасности, прилагаемом к данному устройству.

СОХРАНИТЕ ЭТИ ИНСТРУКЦИИ
Related Documentation

- RFC3261 (SIP: Session Initiation Protocol)
- RFC2543 (SIP: Session Initiation Protocol)
- Cisco ATA SIP Compliance Reference Information
- RFC768 (User Datagram Protocol)
- RFC2198 (RTP Payload for Redundant Audio Data)
- RFC2833 (RTP Payload for DTMF Digits, Telephony Phones and Telephony Signals)
- RFC2327 (SDP: Session Description Protocol)
- RFC3266 (Support for IPv6 in Session Description Protocol (SDP))
- Read Me First - ATA Boot Load Information
- Cisco ATA 186 and Cisco 188 Analog Telephone Adaptor At a Glance
- Regulatory Compliance and Safety Information for the Cisco ATA 186 and Cisco 188
- Cisco ATA Release Notes

Obtaining Documentation

These sections explain how to obtain documentation from Cisco Systems.

World Wide Web

You can access the most current Cisco documentation on the World Wide Web at this URL:

http://www.cisco.com
Translated documentation is available at this URL:

Documentation CD-ROM

Cisco documentation and additional literature are available in a Cisco Documentation CD-ROM package, which is shipped with your product. The Documentation CD-ROM is updated monthly and may be more current than printed documentation. The CD-ROM package is available as a single unit or through an annual subscription.

Ordering Documentation

You can order Cisco documentation in these ways:

- Registered Cisco.com users (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 Systems Corporate Headquarters (California, U.S.A.) at 408 526-7208 or, elsewhere in North America, by calling 800 553-NETS (6387).

Documentation Feedback

You can submit comments electronically on Cisco.com. In the Cisco Documentation home page, click the Fax or Email option in the “Leave Feedback” section at the bottom of the page.

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

You can submit your comments by mail by using the response card behind the front cover of your document or by writing to the following address:

Cisco Systems
Attn: Document Resource Connection
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 online documentation, troubleshooting tips, and sample configurations from online tools by using the Cisco Technical Assistance Center (TAC) Web Site. Cisco.com registered users have complete access to the technical support resources on the Cisco TAC Web Site.
Cisco.com

Cisco.com is the foundation of a suite of interactive, networked services that provides immediate, open access to Cisco information, networking solutions, services, programs, and resources at any time, from anywhere in the world.

Cisco.com is a highly integrated Internet application and a powerful, easy-to-use tool that provides a broad range of features and services to help you with these tasks:

- Streamline business processes and improve productivity
- Resolve technical issues with online support
- Download and test software packages
- Order Cisco learning materials and merchandise
- Register for online skill assessment, training, and certification programs

If you want to obtain customized information and service, you can self-register on Cisco.com. To access Cisco.com, go to this URL:

http://www.cisco.com

Technical Assistance Center

The Cisco Technical Assistance Center (TAC) is available to all customers who need technical assistance with a Cisco product, technology, or solution. Two levels of support are available: the Cisco TAC Web Site and the Cisco TAC Escalation Center.

Cisco TAC inquiries are categorized according to the urgency of the issue:

- Priority level 4 (P4)—You need information or assistance concerning Cisco product capabilities, product installation, or basic product configuration.
- Priority level 3 (P3)—Your network performance is degraded. Network functionality is noticeably impaired, but most business operations continue.
- Priority level 2 (P2)—Your production network is severely degraded, affecting significant aspects of business operations. No workaround is available.
- Priority level 1 (P1)—Your production network is down, and a critical impact to business operations will occur if service is not restored quickly. No workaround is available.

The Cisco TAC resource that you choose is based on the priority of the problem and the conditions of service contracts, when applicable.

Cisco TAC Web Site

You can use the Cisco TAC Web Site to resolve P3 and P4 issues yourself, saving both cost and time. The site provides around-the-clock access to online tools, knowledge bases, and software. To access the Cisco TAC Web Site, go to this URL:

http://www.cisco.com/tac

All customers, partners, and resellers who have a valid Cisco service contract have complete access to the technical support resources on the Cisco TAC Web Site. The Cisco TAC Web Site requires a Cisco.com login ID and password. If you have a valid service contract but do not have a login ID or password, go to this URL to register:

http://www.cisco.com/register/
If you are a Cisco.com registered user, and you cannot resolve your technical issues by using the Cisco TAC Web Site, you can open a case online by using the TAC Case Open tool at this URL:

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

If you have Internet access, we recommend that you open P3 and P4 cases through the Cisco TAC Web Site.

**Cisco TAC Escalation Center**

The Cisco TAC Escalation Center addresses priority level 1 or priority level 2 issues. These classifications are assigned when severe network degradation significantly impacts business operations. When you contact the TAC Escalation Center with a P1 or P2 problem, a Cisco TAC engineer automatically opens a case.

To obtain a directory of toll-free Cisco TAC telephone numbers for your country, go to this URL:


Before calling, please check with your network operations center to determine the level of Cisco support services to which your company is entitled: for example, SMARTnet, SMARTnet Onsite, or Network Supported Accounts (NSA). When you call the center, please have available your service agreement number and your product serial number.
Cisco Analog Telephone Adaptor Overview

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

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

This section covers the following topics:

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

![Cisco ATA Analog Telephone Adaptor](image)

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

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**Note**

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

**Note**

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

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

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

Note: SIP for the Cisco ATA is compliant with RFC2543.

This section contains the following topics:
- SIP Capabilities, page 1-3
- Components of SIP, page 1-3
SIP Capabilities

SIP provides the following capabilities:

- Determines the availability of the target endpoint. If a call cannot be completed because the target endpoint is unavailable, SIP determines whether the called party is already on the phone or did not answer in the allotted number of rings. SIP then returns a message indicating why the target endpoint was unavailable.

- Determines the location of the target endpoint. SIP supports address resolution, name mapping, and call redirection.

- Determines the media capabilities of the target endpoint. Using the Session Description Protocol (SDP), SIP determines the lowest level of common services between endpoints. Conferences are established using only the media capabilities that are supported by all endpoints.

- Establishes a session between the originating and target endpoint. If the call can be completed, SIP establishes a session between the endpoints. SIP also supports mid-call changes, such as adding another endpoint to the conference or changing the media characteristic or codec.

- Handles the transfer and termination of calls. SIP supports the transfer of calls from one endpoint to another. During a call transfer, SIP establishes a session between the transferee and a new endpoint (specified by the transferring party) and terminates the session between the transferee and the transferring party. At the end of a call, SIP terminates the sessions between all parties. Conferences can consist of two or more users and can be established using multicast or multiple unicast sessions.

Components of SIP

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

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

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

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

From an architectural standpoint, the physical components of a SIP network can also be grouped into two categories—Clients and servers. Figure 1-4 illustrates the architecture of a SIP network.

Note

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

SIP clients include:

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

SIP Servers

SIP servers include:

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

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

Figure 1-5  Cisco ATA 186—Rear View

Figure 1-6  Cisco ATA 188—Rear View
The unit provides the following connectors and indicators:

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

**Note**

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

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

**Note**

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

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

---

**Figure 1-7 Function Button**

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

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

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

Caution

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

Software Features

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

- Voice Codecs Supported, page 1-7
- Additional Supported Signaling Protocols, page 1-8
- Other Supported Protocols, page 1-8
- Cisco ATA SIP Services, page 1-8
- Fax Services, page 1-9
- Methods Supported, page 1-9
- Supplementary Services, page 1-10

Voice Codecs Supported

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

- G.711μ-law
- G.711A-law
- G.723.1
- G.726
- G.729
- G.729A
- G.729B
- G.729AB
Additional Supported Signaling Protocols

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

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

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

Other Supported Protocols

Other protocols that the Cisco ATA supports include the following:

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

Cisco ATA SIP Services

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

These services include the following features:

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

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

Fax Services

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

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

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

Note

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

Methods Supported

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

- REGISTER
- REFER
- INVITE
- BYE
- CANCEL
Supplementary Services

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

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

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

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

Installation and Configuration Overview

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

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

This section provides instructions for installing the Cisco ATA 186 and Cisco ATA 188. Before you perform the installation, be sure you have met the following prerequisites:

- Planned the network and Cisco ATA configuration.
- Installed the Ethernet connection.
- Installed and configured the other network devices.

This section contains the following topics:

- Network Requirements, page 2-2
- Safety Recommendations, page 2-2
- What the Cisco ATA Package Includes, page 2-2
- What You Need, page 2-3
- Installation Procedure, page 2-3
- Power-Down Procedure, page 2-6

Note: The term *Cisco ATA* is used throughout this manual to refer to both the Cisco ATA 186 and the Cisco ATA 188, unless differences between the Cisco ATA 186 and Cisco ATA 188 are explicitly stated.
Network Requirements

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

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

Safety Recommendations

To ensure general safety, follow these guidelines:

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

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

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

Warning: The plug/socket combination must be accessible at all times because it serves as the main disconnecting device.

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

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

For translated warnings, see the Regulatory Compliance and Safety Information for the Cisco ATA 186 and Cisco ATA 188 manual.

What the Cisco ATA Package Includes

The Cisco ATA package contains the following items:

- Cisco ATA 186 or Cisco ATA 188 Analog Telephone Adaptor
- Cisco ATA 186 and Cisco ATA 188 Analog Telephone Adaptor at a Glance
What You Need

You also need the following items:

- Category-3 10BASE-T or 100BASE-T or better Ethernet cable. One cable is needed for each Ethernet connection.

  A Category-3 Ethernet cable supports 10BASE-T for up to 100 meters without quality degradation, and a Category-3 Ethernet cable supports 100BASE-T for up to 10 meters without quality degradation.

  For uplink connections, use a crossover Ethernet cable to connect the Cisco ATA to another Ethernet device (such as a router or PC) without using a hub. Otherwise, use straight-through Ethernet cables for both uplink and data port connections.

- Access to an IP network
- One or two analog Touch-Tone telephones or fax machines, or one of each

Installation Procedure

After the equipment is in place, see Figure 2-1 (for Cisco ATA 186) or Figure 2-2 (for Cisco ATA 188) and follow the next procedure to install the Cisco ATA.
**Procedure**

**Step 1**  Place the Cisco ATA near an electrical power outlet.

**Step 2**  Connect one end of a telephone line cord to the **Phone 1** input on the rear panel of the Cisco ATA. Connect the other end to an analog telephone set.
Chapter 2 Installing the Cisco ATA

Installation Procedure

If you are connecting a telephone set that was previously connected to an active telephone line, unplug the telephone line cord from the wall jack and plug it into the **Phone 1** input.

<table>
<thead>
<tr>
<th>Warning</th>
<th>To reduce the risk of fire, use only No. 26 AWG or larger telecommunication line cord.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Caution</td>
<td>Do not connect the Phone input ports to a telephone wall jack. To avoid damaging the Cisco ATA or telephone wiring in the building, do not connect the Cisco ATA to the telecommunications network. Connect the Phone port to a telephone only, never to a telephone wall jack.</td>
</tr>
<tr>
<td>Note</td>
<td>The telephone must be switched to tone setting (not pulse) for the Cisco ATA to operate properly.</td>
</tr>
</tbody>
</table>

**Step 3** (Optional) Connect the telephone line cord of a second telephone to the **Phone 2** input port.

**Note** If you are connecting only one telephone to the Cisco ATA, you must use the **Phone 1** input port.

**Step 4** Connect an Ethernet cable to the uplink RJ-45 connector on the Cisco ATA. For the Cisco ATA 186, this is the 10BASE-T connector; for the Cisco ATA 188, this is the 10/100UPLINK connector.

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

**Step 5** (Cisco ATA 188 only—optional) Connect a straight-through Ethernet cable from your PC to the 10/100 PC RJ-45 connector on the Cisco ATA.

**Step 6** Connect the socket end of the power cord to the Cisco-supplied 5V DC power adaptor.

**Step 7** Insert the power adaptor cable into the power connector on the Cisco ATA.

<table>
<thead>
<tr>
<th>Caution</th>
<th>Use only the Cisco-supplied power adaptor.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Warning</td>
<td>This product relies on the building’s installation for short-circuit (overcurrent) protection. Ensure that a fuse or circuit breaker no larger than 120 VAC, 15A U.S. (240VAC, 10A international) is used on the phase conductors (all current-carrying conductors).</td>
</tr>
</tbody>
</table>
Step 8  Connect the plug end of the 5V DC power adaptor cord into an electrical power outlet.

When the Cisco ATA is properly connected and powered up, the green activity LED flashes to indicate network activity. This LED is labeled ACT on the rear panel of the Cisco ATA 186 and is labeled LINK on the rear panel of the Cisco ATA 188.

⚠️ **Caution**

Do not cover or block the air vents on either the top or the bottom surface of the Cisco ATA. Overheating can cause permanent damage to the unit.

For more information about LEDs and the function button, see the “Hardware Overview” section on page 1-5.

---

### Power-Down Procedure

⚠️ **Caution**

If you need to power down Cisco ATA 186 or Cisco 188 at any time, use the following power-down procedure to prevent damage to the unit.

**Procedure**

1. **Step 1** Unplug the RJ45 Ethernet cable
2. **Step 2** Wait for 20 seconds.
3. **Step 3** Unplug the power cable.

⚠️ **Warning**

This equipment contains a ring signal generator (ringer), which is a source of hazardous voltage. Do not touch the RJ-11 (phone) port wires (conductors), the conductors of a cable connected to the RJ-11 port, or the associated circuit-board when the ringer is active. The ringer is activated by an incoming call.
Configuring the Cisco ATA for SIP

This section describes how to configure the Cisco ATA to operate with the Session Initiation Protocol (SIP) signaling image and how the Cisco ATA obtains the latest signaling image.

You can configure the Cisco ATA for use with SIP with any of the following methods:

- **By using a TFTP server**—This is the Cisco-recommended method for deploying a large number of Cisco ATAs. This method allows you to set up a unique Cisco ATA configuration file or a configuration file that is common to all Cisco ATAs. The Cisco ATA can automatically download its latest configuration file from the TFTP server when the Cisco ATA powers up, is refreshed or reset, or when the specified TFTP query interval expires.

- **By using manual configuration**:
  - **Voice configuration menu**—This is the method you must use if the process of establishing IP connectivity for the Cisco ATA requires changing the default network configuration settings. These settings are CDP, VLAN, and DHCP. You also can use the voice configuration menu to review all IP connectivity settings. The voice configuration menu can also be used when Web access is not available.
  - **Web-based configuration**—This method is convenient if you plan to deploy a small number of Cisco ATAs in your network. To use this method, the Cisco ATA must first obtain IP connectivity, either through the use of a DHCP server or by using the voice configuration menu to statically configure IP addresses.

This section contains the following topics:

- **Default Boot Load Behavior, page 3-2**—This section describes the process that the Cisco ATA follows by default when it boots up. It is very important to understand this process because, if your network environment is not set up to follow this default behavior, you need to make the applicable configuration changes. For example, by default, the Cisco ATA attempts to contact a DHCP server for the necessary IP addresses to achieve network connectivity. However, if your network does not use a DHCP server, you must manually configure various IP settings as described in this section.

- **Specifying a Preconfigured VLAN ID or Disabling VLAN IP Encapsulation, page 3-3**—This section includes a table of the parameters you can configure for VLAN and CDP settings.

- **Steps Needed to Configure the Cisco ATA, page 3-5**—This section provides tables that summarize the general configuration steps you must follow to configure the Cisco ATA.

- **Configuring the Cisco ATA Using a TFTP Server, page 3-8**—This section describes procedures for configuring the Cisco ATA by using a TFTP server, which is the recommended configuration method for the deployment of a large number of Cisco ATAs.

- **Voice Configuration Menu, page 3-20**—This section includes information on how to obtain basic network connectivity for the Cisco ATA and how to perform a factory reset if necessary.
Default Boot Load Behavior

Before configuring the Cisco ATA, you need to know how the default Cisco ATA boot load process works. Once you understand this process, you will be able to configure the Cisco ATA by following the instructions provided in this section and in the sections that follow.

All Cisco ATAs are shipped with a bootload signaling-protocol image. However, because this image is not a fully functional signaling image, the image must be upgraded. The image is designed to be automatically upgraded by a properly configured TFTP server. To configure the Cisco ATA to automatically upgrade to the latest signaling image, see the “Upgrading the Signaling Image from a TFTP Server” section on page 8-1.

In addition, the Cisco ATA obtains its configuration file during the bootload process.

The following list summarizes the default Cisco ATA behavior during its boot-up process:

1. The Cisco ATA uses the Cisco Discovery Protocol (CDP) to discover which VLAN to enter. If the Cisco ATA receives a VLAN ID response from the network switch, the Cisco ATA enters that VLAN and adds 802.1Q VLAN tags to its IP packets. If the Cisco ATA does not receive a response with a VLAN ID from the network switch, then the Cisco ATA assumes it is not operating in a VLAN environment and does not perform VLAN tagging on its packets.

   Note If your network environment is not set up to handle this default behavior, make the necessary configuration changes by referring to the “Specifying a Preconfigured VLAN ID or Disabling VLAN IP Encapsulation” section on page 3-3.

2. The Cisco ATA contacts the DHCP server to request its own IP address.

   Note If your network environment does not contain a DHCP server, you need to statically configure various IP addresses so that the Cisco ATA can obtain network connectivity. For a list of parameters that you must configure to obtain network connectivity, see Table 3-6 on page 3-21. For instructions on how to use the voice configuration menu, which you must use to perform this configuration, see the “Voice Configuration Menu” section on page 3-20.
3. Also from the DHCP server, the Cisco ATA requests the IP address of the TFTP server.

4. The Cisco ATA contacts the TFTP server and downloads the Cisco ATA release software that contains the correct signaling image for the Cisco ATA to function properly.

Note  If you are not using a TFTP server, you need to manually upgrade the Cisco ATA to the correct signaling image. For information on this procedure, see the “Upgrading the Signaling Image Manually” section on page 8-2.

5. The Cisco ATA looks for a Cisco ATA-specific configuration file (designated by the MAC address of the Cisco ATA and named ata<macaddress> with a possible file extension) on the TFTP server and downloads this file if it exists. For information about configuration file names, see the “Configuration Files that the cfgfmt Tool Creates” section on page 3-13.

6. If the Cisco ATA does not find an ata<macaddress> configuration file, it looks for an atadefault.cfg configuration file and downloads this file if it exists. This file can contain default values for the Cisco ATA to use.

Note  When the Cisco ATA is downloading its DHCP configuration, the function button on the top panel blinks.

Specifying a Preconfigured VLAN ID or Disabling VLAN IP Encapsulation

If you want the Cisco ATA to use a preconfigured VLAN ID instead of using the Cisco Discovery Protocol to locate a VLAN, or if you want to disable VLAN IP encapsulation, refer to Table 3-1 for a reference to the parameters and bits you may need to configure. Use the voice configuration menu to configure these parameters. (See the “Voice Configuration Menu” section on page 3-20 for instructions on using this menu.) Also, refer to Table 3-2 for a matrix that indicates which VLAN-related parameters and bits to configure depending on your network environment.

Note  Bits are numbered from right to left, starting with bit 0.
Example

The following procedure shows you how to configure the OpFlags and VLANSetting parameters to allow the Cisco ATA to use a user-specified VLAN ID. In this example, the voice VLAN ID is 115 (in decimal format).

Step 1
Set bits 4-6 of the OpFlags parameter to 1, 0, and 1, respectively. This setting translates to the following bitmap:

```
xxxx xxxx xxxx xxxx xxxx xxxx x101 xxxx
```

The remaining bits of the OpFlags parameter, using all default values, make up the following bitmap representation:

```
0000 0000 0000 0000 0000 0000 0xxx 0010
```

Therefore, the resulting value of the OpFlags parameter becomes the following bitmap representation:

```
0000 0000 0000 0000 0000 0000 0101 0010
```
Steps Needed to Configure the Cisco ATA

This section contains the following topics:

- Basic Configuration Steps in a TFTP Server Environment, page 3-5
- Basic Configuration Steps in a Non-TFTP Server Environment, page 3-7

Basic Configuration Steps in a TFTP Server Environment

Table 3-3 shows the basic steps for configuring the Cisco ATA and making it operational in a typical SIP environment, which includes a TFTP server.
### Table 3-3  Basic Steps to Configure the Cisco ATA in a TFTP Environment

<table>
<thead>
<tr>
<th>Action</th>
<th>Reference</th>
</tr>
</thead>
<tbody>
<tr>
<td>1. Download the desired Cisco ATA release software zip file from the Cisco web site and store it on the TFTP server.</td>
<td>“Setting Up the TFTP Server with Cisco ATA Software” section on page 3-8</td>
</tr>
<tr>
<td>2. Follow these basic steps to create a unique Cisco ATA configuration file, which actually entails creating two files:</td>
<td>“Creating Unique and Common Cisco ATA Configuration Files” section on page 3-9</td>
</tr>
<tr>
<td>a. Create a Cisco ATA configuration text file that contains parameters that are common to all Cisco ATAs in your network.</td>
<td></td>
</tr>
<tr>
<td>b. Create a unique Cisco ATA configuration text file that contains parameters that are specific to a Cisco ATA. Make sure to use an include command in the unique configuration file to pull in values from the common configuration file.</td>
<td></td>
</tr>
<tr>
<td>c. Convert the unique configuration file to binary format.</td>
<td></td>
</tr>
<tr>
<td>d. Place the unique binary configuration file on the TFTP server.</td>
<td></td>
</tr>
<tr>
<td>3. Optionally, create a default configuration file called atadefault.cfg, which the Cisco ATA will download from the TFTP server only if the unique Cisco ATA file called ata&lt;macaddress&gt; (with a possible file extension) does not exist on the TFTP server. For information about possible configuration file names, see the “Configuration Files that the cfgfmt Tool Creates” section on page 3-13.</td>
<td>“atadefault.cfg Configuration File” section on page 3-17</td>
</tr>
<tr>
<td>4. Configure the upgradecode parameter so that the Cisco ATA will obtain the correct signaling image from the TFTP server when the Cisco ATA powers up.</td>
<td>“Upgrading the Signaling Image from a TFTP Server” section on page 8-1</td>
</tr>
<tr>
<td>5. Configure the desired interval for the Cisco ATA to contact the TFTP server to check for a configuration-file update or an upgrade of the signaling image file.</td>
<td>“Configuring Refresh Interval” section on page 4-3</td>
</tr>
<tr>
<td>6. Configure the method with which the Cisco ATA will locate the TFTP server at boot up time.</td>
<td>“Configuring the Cisco ATA to Obtain its Configuration File from the TFTP Server” section on page 3-18</td>
</tr>
<tr>
<td>7. Power up the Cisco ATA.</td>
<td></td>
</tr>
<tr>
<td>8. If you make configuration changes to the Cisco ATA or upgrade the signaling image on the TFTP server, you can refresh the Cisco ATA so that these changes take effect immediately. Otherwise, these changes will take effect when the specified interval (CfgInterval parameter value) for the TFTP query expires.</td>
<td>“Refreshing or Resetting the Cisco ATA” section on page 3-26</td>
</tr>
</tbody>
</table>
Basic Configuration Steps in a Non-TFTP Server Environment

Table 3-4 shows the basic steps for configuring the Cisco ATA without using the TFTP server method.

Table 3-4  Basic Steps to Configure the Cisco ATA Without Using the TFTP Server Method

<table>
<thead>
<tr>
<th>Action</th>
<th>Reference</th>
</tr>
</thead>
<tbody>
<tr>
<td>1. Download the desired Cisco ATA release software zip file from the Cisco web site:</td>
<td></td>
</tr>
<tr>
<td>a. If you are a registered CCO user, go to the following URL:</td>
<td></td>
</tr>
<tr>
<td><a href="http://www.cisco.com/cgi-bin/tablebuild.pl/ata186">http://www.cisco.com/cgi-bin/tablebuild.pl/ata186</a></td>
<td></td>
</tr>
<tr>
<td>b. Download the zip file that contains the software for the applicable release and signaling image you are using. The contents of each file are described next to the file name.</td>
<td></td>
</tr>
<tr>
<td>c. Extract the files to the desired location on your PC.</td>
<td></td>
</tr>
<tr>
<td><strong>Note</strong> The file that contains the protocol signaling image has an extension of .zup.</td>
<td></td>
</tr>
<tr>
<td>2. Manually upgrade the Cisco ATA to the correct signaling image.</td>
<td>Upgrading the Signaling Image Manually, page 8-2</td>
</tr>
<tr>
<td>3. Configure the Cisco ATA by using either one of the manual-configuration methods.</td>
<td>• Voice Configuration Menu, page 3-20</td>
</tr>
<tr>
<td></td>
<td>• Cisco ATA Web Configuration Page, page 3-23</td>
</tr>
<tr>
<td>4. Power up the Cisco ATA.</td>
<td></td>
</tr>
</tbody>
</table>
Configuring the Cisco ATA Using a TFTP Server

The TFTP method of configuration is useful when you have many Cisco ATA because you can use a TFTP server for remote, batch configuration of Cisco ATAs. A TFTP server can host one unique configuration file for each Cisco ATA.

This section contains the following topics:

- Setting Up the TFTP Server with Cisco ATA Software, page 3-8
- Configurable Features and Related Parameters, page 3-8
- Creating Unique and Common Cisco ATA Configuration Files, page 3-9
- atadefault.cfg Configuration File, page 3-17
- Configuring the Cisco ATA to Obtain its Configuration File from the TFTP Server, page 3-18

Setting Up the TFTP Server with Cisco ATA Software

This section provides the procedure for the Cisco ATA administrator to obtain the correct Cisco ATA software and set up the TFTP server with this software.

Procedure

Step 1
If you are a registered CCO user, go to the following URL:
http://www.cisco.com/cgi-bin/tablebuild.pl/ata186

Step 2
Download the zip file that contains the software for the applicable release and signaling image you are using. The contents of each file are described next to the file name. Save the zip file onto a floppy disc.

Note
The file that contains the protocol signaling image has an extension of .zup.

Step 3
Extract the signaling files onto the TFTP server. This should be the same TFTP server that will contain the binary Cisco ATA configuration file that you create (either ata<macaddress> with a possible file extension or atadefault.cfg). For information about possible configuration file names, see the “Configuration Files that the cfgfmt Tool Creates” section on page 3-13.

Configurable Features and Related Parameters

Table 4-1 on page 4-2 contains a list of all required SIP parameters. These parameters must be properly configured for the Cisco ATA to work.

For descriptions of important Cisco ATA SIP services that you can configure, and references to their configuration parameters, see the “Important Basic SIP Services” section on page 4-1 and the “Additional SIP Services” section on page 4-3.

Table 4-4 on page 4-23 lists, in alphabetical order, various features that you can configure for the Cisco ATA. Table 4-4 on page 4-23 also includes links to the related parameter that allows you to configure each of these features. Each link takes you to a detailed description of the parameter that includes its default values.
For an example of how to configure parameters for the TFTP Server configuration method, see the “Creating Unique and Common Cisco ATA Configuration Files” section on page 3-9.

Creating Unique and Common Cisco ATA Configuration Files

If you have many Cisco ATAs to configure, a good approach is to create two configuration files:

- One file that will contain only parameter values unique to a specific Cisco ATA.
- One file for parameters that will be configured with values common to a group of Cisco ATAs. If this file is updated, all Cisco ATA devices in this common group can obtain the new configuration data in a batch-mode environment.

The following procedure demonstrates the steps needed to create these configuration files.

Procedure

**Step 1**
Use the sip_example.txt file as a template for creating a text file of values that are common to one group of Cisco ATAs. The sip_example.txt file is included in the software-release zip file and contains all default values. This file is shown without its annotations in the “Configuration Text File Template” section on page 5-2.

Copy the sip_example.txt file and save it with a meaningful name, such as common.txt.

**Step 2**
Configure all common parameters by editing the text file as desired. For example, you might configure the following parameters:

```
UseTftp:1
DHCP:1
TFtpURL:10.10.10.1
```

The settings in this example indicate that a group of Cisco ATAs is using the TFTP server with an IP address of 10.10.10.1 to obtain their configuration files. These Cisco ATAs will use a DHCP server to obtain their own IP addresses but not to obtain the TFTP server IP address (because the TftpURL parameter has a configured value).

**Step 3**
Save your changes.
**Step 4** Use the sip_example.txt file again, this time as a template for creating a text file of values that are specific to one Cisco ATA. For example, you might configure the following parameters:

```
UserID:8530709
GkorProxy:192.168.1.1
```

Save this file of Cisco ATA-specific parameters as:

```
ata<macaddress>.txt
```

where `macaddress` is the non-dotted hexadecimal version of the MAC address of the Cisco ATA you are configuring. This non-dotted hexadecimal MAC address is labeled on the bottom of most Cisco ATAs next to the word “MAC.” The file name must be exactly 15 characters long. (However, if this filename is supplied by the DHCP server, the name can be as long as 31 characters and can be any name with printable ASCII characters.)

If necessary, you can obtain the non-dotted hexadecimal MAC address by using the atapname.exe command. For information on using the atapname.exe command, see the “Using atapname.exe Tool to Obtain MAC Address” section on page 3-11. That section includes an example of a dotted decimal MAC address and its corresponding non-dotted hexadecimal address.

**Note** The `ata<macaddress>.txt` file should contain only those parameters whose values are different from the file of common parameters. Parameter values in the `ata<macaddress>` configuration file will overwrite any manually configured values (values configured through the web or voice configuration menu) when the Cisco ATA powers up or refreshes.

**Step 5** On the top line of the `ata<macaddress>.txt` file, add an `include` command to include the name of the common-parameters file, and save the file.

```
include:common.txt
UserID:8530709
GkorProxy:192.168.1.1
```

**Step 6** Run the cfgfmt.exe tool, which is bundled with the Cisco ATA software, on the `ata<macaddress>.txt` text file to generate the binary configuration file. If you wish to encrypt the binary file, see the “Using Encryption With the cfgfmt Tool” section on page 3-12.

The syntax of the cfgfmt program follows:

```
Syntax
```

```
cfgfmt [Encryption options] -sip -tptag.dat input-text-file output-binary-file
```

- Encryption options are described in the “Using Encryption With the cfgfmt Tool” section on page 3-12.
- `-sip` is the protocol you are using, which you must specify so that the cfgfmt tool will include only the applicable protocol in the converted output binary file.
- The `tptag.dat` file, provided with the Cisco ATA software version you are running, is used by `cfgfmt.exe` to format a text input representation of the parameter/value pairs to its output binary representation. Be sure this file resides in the same directory from which you are running the `cfgfmt` program.
- `input-text-file` is the input text file representation of the Cisco ATA configuration file.
- `output-binary-file` is the final output binary file that Cisco ATA uses as the TFTP configuration file.
Example

cfgfmt -sip -tptag.dat ata0a141e28323c.txt ata0a141e28323c

This example is based on a Cisco ATA MAC address of 10.20.30.40.50.60, which converts to the two-digit, lower-case hexadecimal representation of each integer as 0a141e28323c.

When you convert the ata<macaddress>.txt file to a binary file, the binary file will merge the two text files to form one Cisco ATA-specific binary configuration file for your Cisco ATA.

If the same parameter is configured with different values in these two files, the value in the ata<macaddress>.txt file takes precedence over the value in the common.txt file.

**Step 7** Store all binary configuration file(s) in the TFTP server root directory. For information about possible configuration file names, see the “Configuration Files that the cfgfmt Tool Creates” section on page 3-13.

When the Cisco ATA powers up, it will retrieve its configuration file(s) from the TFTP server.

**Step 8** If you want to make configuration changes after boot up, repeat the process of creating or editing the text files containing the desired parameters, then converting the ata<macaddress>.txt text file to the binary file(s) and storing the binary file(s) on the TFTP server. For the configuration changes to take effect immediately, refresh the Cisco ATA. (See the “Refreshing or Resetting the Cisco ATA” section on page 3-26.)

After being refreshed, the Cisco ATA will download the updated ata<macaddress> configuration file(s).

---

**Note** If you do not perform a refresh procedure, the Cisco ATA will update its configuration the next time it contacts the TFTP server, which is based on the configured value of the CfgInterval parameter.

---

**Using atapname.exe Tool to Obtain MAC Address**

This bundled tool is useful for converting the dotted decimal version of the Cisco ATA MAC address (available on the Cisco ATA Web configuration page or from the voice configuration menu code 24#) to its default Cisco ATA profile name. This name has the following format:

ataxxxxxxxxxxxx

where each xx is the two-digit, lower-case hexadecimal representation of each integer in the dotted, decimal version of the Cisco ATA MAC address. This is the name you use for the unique Cisco ATA binary configuration file.
The following command and output show an example of this command.

**Command Example**

```plaintext
atapname.exe 10.20.30.40.50.60
```

**Command Output**

```plaintext
ata0a141e28323c
```

**Note**
The same functionality is available from the voice configuration menu (voice menu code 84#), which will announce the Cisco ATA profile name.

### Using Encryption With the `cfgfmt` Tool

The EncryptKey or EncryptKeyEx parameter can be used to encrypt binary files that are transferred over TFTP. You can change encryption keys for each Cisco ATA so that only one specific Cisco ATA can decode the information.

Cisco strongly recommends using the EncryptKeyEx parameter for encryption because this parameter provides a stronger encryption than the EncryptKey parameter that was used in Cisco ATA software releases prior to release 2.16.

You must use version 2.3 of the `cfgfmt` configuration-file generation tool to use the new EncryptKeyEx parameter. This tool comes bundled with Cisco ATA software version 3.0. To verify that you have version 2.3 of the `cfgfmt` tool type the following command:

```plaintext
cfgfmt
```

The version number of the `cfgfmt` tool will be returned.

You can configure the EncryptKeyEx parameter by using the Cisco ATA Web configuration page or by using the TFTP configuration method. (For more information, see the “EncryptKeyEx” section on page 5-7.)

You can configure the EncryptKey parameter by using the Cisco ATA Web configuration page, the voice configuration menu, or by using the TFTP configuration method. (For more information, see the “EncryptKey” section on page 5-6.)

By default, the Cisco ATA-specific `ata<macaddress>` configuration file(s) are not encrypted. If encryption is required, however, you must manually configure the EncryptKeyEx or EncryptKey parameter before you boot up the Cisco ATA so that the TFTP method is secure. The Cisco ATA uses the RC4 cipher algorithm for encryption.

**Note**
Because the factory-fresh ATA cannot accept encrypted configuration files, the first unencrypted file, if intercepted, can easily be read. (You would still have to know the data structure format in order to decode the binary information from the unencrypted file.) Therefore, the new encryption key in the unencrypted file can be compromised.

**Note**
For security reasons, Cisco recommends that you set the UIPassword parameter (if desired) in the configuration file and not by using one of the manual configuration methods.
This section contains the following topics:

- Configuration Files that the cfgfmt Tool Creates, page 3-13
- cfgfmt Tool Syntax and Examples, page 3-14

### Configuration Files that the cfgfmt Tool Creates

The number of output binary configuration files that the Cisco ATA produces is dependent on two factors:

- Which encryption key parameter is used—EncryptKey or EncryptKeyEx
- The total size of the binary output

Table 3-5 shows the names of the binary files that can be generated. One, two or four files can be generated.

(Note) <macaddress> in Table 3-5 is the MAC address of the Cisco ATA.

(Note) If you are creating an atadefault configuration file, the generated binary file name will be atadefault.cfg.x if you encrypt the text file with the EncryptKeyEx parameter; the binary file name will be atadefault.cfg if you do not use the EncryptKeyEx parameter to encrypt the text file. For information on creating an atadefault configuration file, see the “atadefault.cfg Configuration File” section on page 3-17.

<table>
<thead>
<tr>
<th>Value of EncryptKeyEx Parameter</th>
<th>Total Binary Output Size Less Than or Equal to 2,000 Bytes</th>
<th>Total Binary Output Size Greater Than 2,000 Bytes</th>
</tr>
</thead>
<tbody>
<tr>
<td>0</td>
<td>ata&lt;macaddress&gt;</td>
<td>ata&lt;macaddress&gt;</td>
</tr>
<tr>
<td></td>
<td></td>
<td>ata&lt;macaddress&gt;.ex</td>
</tr>
<tr>
<td>Non-zero</td>
<td>ata&lt;macaddress&gt;</td>
<td>ata&lt;macaddress&gt;</td>
</tr>
<tr>
<td></td>
<td>ata&lt;macaddress&gt;.x</td>
<td>ata&lt;macaddress&gt;.ex</td>
</tr>
<tr>
<td></td>
<td></td>
<td>ata&lt;macaddress&gt;.xex</td>
</tr>
<tr>
<td></td>
<td></td>
<td>ata&lt;macaddress&gt;.xex</td>
</tr>
</tbody>
</table>

(Note) Place all generated binary configuration files onto the TFTP server.
cfgfmt Tool Syntax and Examples

The syntax of the cfgfmt tool follows:

**Syntax**

```
cfgfmt [options] input output
```

**Syntax Definitions—Options**

- `-eRc4Passwd`—This option directs the Cisco ATA to use `Rc4Passwd` as the key (up to eight hexadecimal characters) to encrypt or decrypt the input text file. However, if the Cisco ATA EncryptKey parameter in the input text file is not 0, then the value of that parameter is used to encrypt the output binary file, and `Rc4Passwd` is ignored. The `-e` portion of this option means that the Cisco ATA will use the weaker encryption method.

- `-E`—This option directs the Cisco ATA to not use the value of the EncryptKey parameter, as set in the input text file, to encrypt the output binary configuration file.

- `-xRc4Passwd`—This option directs the Cisco ATA to use `Rc4Passwd`, which must be a hexadecimal string of as many as 64 characters, as the key to encrypt or decrypt the input text file. However, if the Cisco ATA EncryptKeyEx parameter in the input text file is not 0, then the value of that parameter is used to encrypt the output binary file, and `Rc4Passwd` is ignored. The `-x` portion of this option means that the Cisco ATA will use the stronger encryption method.

- `-X`—This option directs the Cisco ATA to not use the value of the EncryptKeyEx parameter, as set in the input text file, to encrypt the output binary configuration file.

- `-tPtag.dat`—This file, provided with the Cisco ATA software version you are running, is used by the cfgfmt tool to format a text input representation of the parameter/value pairs to its output binary representation. Be sure this file resides in the same directory from which you are running the cfgfmt program.

- `-sip`—Specify this tag if you are using the SIP protocol so that the cfgfmt tool will include only the SIP protocol parameters in the converted output binary file.

- `-h323`—Specify this tag if you are using the H.323 protocol so that the cfgfmt tool will include only the H.323 protocol parameters in the converted output binary file.

- `-mgcp`—Specify this tag if you are using the MGCP protocol so that the cfgfmt tool will include only the MGCP protocol parameters in the converted output binary file.

- `-sccp`—Specify this tag if you are using the SCCP protocol so that the cfgfmt tool will include only the SCCP protocol parameters in the converted output binary file.

- `-g`—This tag omits sensitive parameters in an ata<macaddress> file that was created with a version of the cfgfmt tool prior to version 2.3.

Some parameters, specified in the ptag.dat file used by the `cfgfmt` tool, are marked as sensitive information (these parameters could include UIPassword, UID, PWD0). These parameters are not included in the output binary file if the `-g` switch is specified in the cfgfmt syntax.

**Syntax Definitions—Required Parameters**

- Input—This is the input text file representation of the Cisco ATA configuration file.

- Output—This is the final output binary file that Cisco ATA uses as the TFTP configuration file.
Syntax examples

The `cfgfmt.exe` syntax affects how the `EncryptKeyEx` or `EncryptKey` parameters are used, as shown in the following examples. In these examples, `input-text-file` is the `ata<macaddress>.txt` file that you will convert to binary to create the `ata<macaddress>` configuration file(s) for the Cisco ATA; `output-binary-file` is that binary `ata<macaddress>` file, and `Secret` is the encryption key.

- `cfgfmt -sip -tptag.dat input-text-file output-binary-file`
  If `input-text-file` sets the Cisco ATA `EncryptKey` parameter to 0, then `output-binary-file` is not encrypted. If the `input-text-file` sets `EncryptKey` to a non-zero value, then `output-binary-file` is encrypted with that value.

- `cfgfmt -X -sip -tptag.dat input-text-file output-binary-file`
  This is an example of how you might perform encryption on a first-time Cisco ATA.
  The `-X` (uppercase) option means that any value specified for the Cisco ATA `EncryptKeyEx` parameter in `input-text-file` is ignored. However, because `Secret` is not specified in this example, `output-binary-file` is not encrypted. Nevertheless, the `EncryptKeyEx` parameter and its value, if specified in `input-file-text`, will be included in `output-binary-file` for possible encryption at a later time. The next time the Cisco ATA fetches the configuration file from the TFTP server, the file will be encrypted with `Secret`.

- `cfgfmt -X -xSecret -sip -tptag.dat input-text-file output-binary-file`
  This is an example of changing the encryption key from one key to another key.
  The `-X` (uppercase) option means that any value specified for the Cisco ATA `EncryptKeyEx` parameter in `input-text-file` is ignored and the `output-binary-file` is encrypted with the `Secret` key. However, the `EncryptKeyEx` parameter and its value, if specified in `input-text-file`, will be included in `output-binary-file`.

Examples of Upgrading to Stronger Encryption Key

This section contains two examples of how you would upgrade your Cisco ATA configuration to use the stronger encryption method if the current Cisco ATA firmware version was a version earlier than version 2.16.2. Versions earlier than 2.16.2 do not support the stronger `EncryptKeyEx` parameter.

Example 1

In this example, the Cisco ATA has not yet been deployed, but its firmware version is earlier than 2.16.2. Therefore, the Cisco ATA will upgrade to firmware version 3.0 to use the `EncryptKeyEx` parameter as its encryption key.

The Cisco ATA in this example has a MAC address of 102030405060.

Perform the following steps:

**Procedure**

- **Step 1**
  Create a file called `ata102030405060.txt` by using the applicable `example.txt` file provided with the Cisco ATA software. (For example, for SIP, the `example.txt` file is called `sip_example.txt`.)

- **Step 2**
  Modify the `ata102030405060.txt` file with desired parameter values. The value of the `EncryptKey` parameter should be 0.
Step 3 Set the value of the EncryptKeyEx parameter to the chosen encryption key with which you want the output binary file to be encrypted. In the EncryptKeyEx parameter specified in the configuration file, you can also restrict the EncryptKeyEx value to apply only to the Cisco ATA with a particular MAC address. For example, if the chosen key value is 231e2a7f10bd7fe, you can specify EncryptKeyEx as:

```
EncryptKeyEx:231e2a7f10bd7fe/102030405060
```

This means that only the Cisco ATA with MAC address 102030405060 will be allowed to apply this EncryptKeyEx value to its internal configuration.

Step 4 Update the `upgradecode` parameter to instruct the Cisco ATA to upgrade to firmware version 3.0 by means of TFTP configuration. The `upgradecode` parameter is described in Chapter 8, “Upgrading the Cisco ATA Signaling Image.”

Step 5 Run the `cfgfmt` tool as follows:

```
cfgfmt -g ata102030405060.txt ata102030405060
```

This will generate the following two binary configuration files:

- `ata102030405060`
- `ata102030405060.x`

`ata102030405060` is unencrypted. `ata102030405060.x` is encrypted with EncryptKeyEx value.

Step 6 Place these two files on the TFTP server that the Cisco ATA will contact for its configuration files.

When the Cisco ATA powers up, it will obtain its IP address from the DHCP server. If the DHCP server specifies the TFTP server address, the Cisco ATA will contact the TFTP server obtained from DHCP because the Cisco ATA is not preconfigured with a TFTP server address. The boot process is as follows:

- a. The Cisco ATA downloads the configuration file `ata102030405060` from the TFTP server.
- b. The Cisco ATA applies parameter values in the file `ata102030405060` to its internal configuration while ignoring the EncryptKeyEx parameter (because the older version of the Cisco ATA does not yet recognize the EncryptKeyEx parameter).
- c. The Cisco ATA upgrades to the 3.0 firmware load.
- d. The Cisco ATA reboots.
- e. The Cisco ATA again downloads the configuration file `ata102030405060`.
- f. The Cisco ATA applies the value of the EncryptKeyEx parameter to its internal configuration.
- g. The Cisco ATA reboots.
- h. The Cisco ATA EncryptKeyEx value is in effect, so from this point forward the Cisco ATA will download the `ata102030405060.x` file at each reboot and each time the value configured in the `CfgInterval` parameter expires.

Note Although `EncryptKeyEx` is encrypted in the `ata<macaddress>` file, and the `ata<macaddress>` file does not contain other sensitive information, Cisco recommends that for absolute security you pre-configure the Cisco ATA as described in this example for a private network. Alternatively, you should remove `ata<macaddress>` once `EncryptKeyEx` takes effect.
Example 2

In this example, a new Cisco ATA has already been deployed (with the EncryptKey value set) with a firmware version earlier than 2.16.2. The Cisco ATA needs to be upgraded to version 2.16.2 firmware or greater to use EncryptKeyEx parameter to encrypt its configuration file.

In this scenario, you would follow the same procedure as in Example 1, except that you would need to set the EncryptKey value to the previously configured EncryptKey value. The difference is that the ata<macaddress> file is now encrypted with EncryptKey because the Cisco ATA expects the ata<macaddress> file to be encrypted with EncryptKey. The Cisco ATA can then begin using the ata<macaddress>.x file that is encrypted with the EncryptKeyEx parameter.

atadefault.cfg Configuration File

You can create a configuration file, called atadefault.cfg, that is common to all Cisco ATAs. This configuration file is applied to a Cisco ATA only if a unique configuration file (such as ata<macaddress>) does not exist for the Cisco ATA on the TFTP server during the Cisco ATA power-up procedure.

You can use the atadefault.cfg file to provide limited functionality for when you first install the Cisco ATA. For example, if your service provider provides the ethernet connection and VoIP telephony service, you may need to call customer service to activate the service. If the atadefault.cfg file is configured to provide a direct connection to the customer service center, you can simply pick up the telephone and wait to be connected without using your regular phone.

The following procedure illustrates how to create the Cisco ATA default configuration file, convert it to the required binary format that the Cisco ATA can read, and store it on the TFTP server so that the Cisco ATA will download it during the boot-up process:

Procedure

Step 1 Make a copy of the sip_example.txt file and rename it atadefault.txt.

Step 2 Make the desired configuration changes by editing the atadefault.txt file, then save the file.

Step 3 Convert the atadefault.txt file to a binary file by running the cfgfmt.exe tool, which is bundled with the Cisco ATA software.

Note If you wish to encrypt the binary file for security reasons, see the “Using Encryption With the cfgfmt Tool” section on page 3-12. If you encrypt the file using the EncryptKeyEx parameter, the resulting binary file will be called atadefault.cfg.x; if not encrypted with the EncryptKeyEx parameter the resulting binary file name will be atadefault.cfg.

Step 4 Store the binary atadefault.cfg (or atadefault.cfg.x) configuration file in the TFTP server root directory. During the boot-up process, the Cisco ATA will download this file as its configuration file unless it first finds a Cisco ATA-specific configuration file named for the MAC address of the Cisco ATA.
Configuring the Cisco ATA to Obtain its Configuration File from the TFTP Server

This section describes three methods for how the Cisco ATA contacts the TFTP server to obtain its configuration file:

- Using a DHCP Server, page 3-18
  - The Cisco ATA contacts the DHCP server, which provides the IP address of the TFTP server
  - The Cisco ATA uses the DHCP server but the DHCP server does not know about the TFTP server
- Without Using a DHCP Server, page 3-20

Using a DHCP Server

When using a DHCP server, configuration settings vary depending on whether or not the DHCP server is under the control of the Cisco ATA system administrator or the service provider. The simplest configuration is when the DHCP server is under the control of the Cisco ATA administrator, in which case the DHCP server provides the IP address of the TFTP server. Depending on who controls the DHCP server, follow the applicable configuration procedure:

- Procedure if DHCP Server is Under Control of Cisco ATA Administrator, page 3-18
- Procedure if DHCP Server is not Under Control of Cisco ATA Administrator, page 3-19

This section also includes the topic:
- Other DHCP Options You Can Set, page 3-19

Note

If no DHCP server is found and the Cisco ATA is programmed to find one, the function button continues to blink.

Procedure if DHCP Server is Under Control of Cisco ATA Administrator

Procedure

Step 1

On the DHCP server, set one of the following two options:

- DHCP option 150 (TFTP server IP address)
- Standard DHCP option 66 (TFTP server name)

If you use DHCP option 150, the Cisco ATA will ignore DHCP option 66. However, if you use DHCP option 66, you must turn off DHCP option 150 or set its value to 0.

Note

You can turn off the DHCP option 150 request by using the Cisco ATA OpFlags parameter (see the “OpFlags” section on page 5-45).

Step 2

Make sure to use default values for the following Cisco ATA parameters:

- TftpURL=0
- UseTftp=1
- DHCP=1
This completes the parameter settings and DHCP options you need to configure for this procedure. The Cisco ATA will contact the DHCP server for the IP address of the TFTP server that contains the Cisco ATA configuration file.

**Procedure if DHCP Server is not Under Control of Cisco ATA Administrator**

This is the procedure to use if the DHCP server is not under the control of the Cisco ATA administrator, which means that the URL of the TFTP server must be manually configured.

**Procedure**

**Step 1** Using the voice configuration menu, set the parameter TftpURL to the IP address or URL of the TFTP server. For more information on setting the TftpURL parameter, see the “TftpURL” section on page 5-5. For information about using the Cisco ATA voice configuration menu, see the “Voice Configuration Menu” section on page 3-20.

**Note** If you are not using a DHCP server to provide the TFTP server location, you must manually configure the TftpURL. You can do this by using the voice configuration menu without first obtaining network connectivity for the Cisco ATA. If you want to configure this value using the Web configuration page, you first must obtain network connectivity by using the voice configuration menu to statically configure IP address information (see the “Voice Configuration Menu” section on page 3-20).

**Step 2** Use the default value of 1 for the Cisco ATA parameter DHCP.

**Step 3** Use the default value of 1 for the Cisco ATA parameter UseTftp.

This completes the parameter settings you need to configure for this procedure. The Cisco ATA will contact the manually configured TFTP server that contains the Cisco ATA configuration file.

**Other DHCP Options You Can Set**

The following parameters can also be configured with DHCP:

- Boot file name of DHCP header—The ata<macaddress> binary Cisco ATA configuration file, which can have a maximum of 31 characters and can be any name with printable ASCII characters
- Client PC address
- DHCP option 1—Client Subnet Mask
- DHCP option 3—Routers on the client’s subnet
- DHCP option 6—One or two Domain Name servers
- DHCP option 42—One or two Network Time Protocol servers

**Note** DHCP options 43 and 60 are set by the Cisco ATA. Option 43 specifies the protocol and option 60 identifies the vendor class of the Cisco ATA box.
Without Using a DHCP Server

Use the following procedure if you are not using a DHCP server in your environment but are still using a TFTP server to obtain the Cisco ATA configuration file:

**Procedure**

**Step 1** Set the DHCP parameter to 0.

**Step 2** Set the UseTFTP parameter to 1.

**Step 3** Set the Cisco ATA parameter TftpURL to the IP address or URL of the TFTP server. For more information on setting the TftpURL parameter, see the “TftpURL” section on page 5-5.

**Note** If you are not using a DHCP server to provide the TFTP server location, you must manually enter the TftpUrl using either the voice configuration menu or the Web configuration page.

**Step 4** If you have done already done so, statically configure the following parameters using the voice configuration menu (see the “Voice Configuration Menu” section on page 3-20). These are the parameters you need to configure for the Cisco ATA to obtain network connectivity:

- StaticIP
- StaticRoute
- StaticNetMask

Other parameters that are normally supplied by DHCP may be provided statically by configuring their values. These parameters are:

- DNS1IP
- DNS2IP
- NTPIP
- AltNTPIP
- Domain

This completes the parameter settings you need to configure in order for the Cisco ATA to contact the TFTP server (without using DHCP) that will contain the configuration file for the Cisco ATA.

**Voice Configuration Menu**

The main reasons to use the voice configuration menu are to establish IP connectivity for the Cisco ATA if a DHCP server is not being used in your network environment, and to reset the Cisco ATA to its factory values if necessary. You can also use the voice configuration menu if you need to configure a small number of parameters or if the web interface and TFTP configuration are not available.
Note

Do not use the voice configuration menu to attempt to change any values that you configured by means of the TFTP configuration file method. Whenever the Cisco ATA refreshes, it downloads its ata<macaddress> configuration file or atadefault.cfg default configuration file from the TFTP server, and the values in either of these files will overwrite the values of any corresponding parameters configured with the voice configuration menu.

See Chapter 5, “Parameters and Defaults,” for a complete list of parameters and their definitions. Also see Table 4-4 on page 4-23 for an alphabetical listing of configurable features and references to their corresponding parameters.

This section contains the following topics:

- Using the Voice Configuration Menu, page 3-21
- Entering Alphanumeric Values, page 3-22
- Resetting the Cisco ATA to Factory Default Values, page 3-23

Using the Voice Configuration Menu

To manually configure the Cisco ATA by using the voice configuration menu and the telephone keypad, perform the following steps:

Procedure

Step 1 Connect an analog touch-tone phone to the port labeled Phone 1 on the back of the Cisco ATA.

Step 2 Lift the handset and press the function button located on the top of the Cisco ATA. You should receive the initial voice configuration menu voice prompt.

Step 3 Using the telephone keypad, enter the voice menu code for the parameter that you want to configure or the command that you want to execute, then press #. For a list of voice menu codes, see Appendix B, “Voice Menu Codes.”

Table 3-6 lists the menu options that you need to configure basic IP connectivity for the Cisco ATA, after which you can use the Cisco ATA web configuration page to configure additional parameters.

Note If you are using the voice configuration menu to statically configure the Cisco ATA IP address, you must disable DHCP by setting its value to 0.

Table 3-6 Parameters that Provide Basic IP Connectivity for the Cisco ATA

<table>
<thead>
<tr>
<th>Voice Menu Number</th>
<th>Features</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>StaticIP—IP address of the Cisco ATA.</td>
</tr>
<tr>
<td>2</td>
<td>StaticRoute—Default gateway for the Cisco ATA to use.</td>
</tr>
<tr>
<td>10</td>
<td>StaticNetMask—Subnet mask of the Cisco ATA.</td>
</tr>
<tr>
<td>20</td>
<td>DHCP—Set value to 0 to disable the use of a DHCP server; set value to 1 to enable DHCP.</td>
</tr>
<tr>
<td>21</td>
<td>Review the IP address of the Cisco ATA.</td>
</tr>
</tbody>
</table>
Table 3-6 Parameters that Provide Basic IP Connectivity for the Cisco ATA (continued)

<table>
<thead>
<tr>
<th>Voice Menu Number</th>
<th>Features</th>
</tr>
</thead>
<tbody>
<tr>
<td>22</td>
<td>Review the default router for the Cisco ATA to use.</td>
</tr>
<tr>
<td>23</td>
<td>Review subnet mask of the Cisco ATA.</td>
</tr>
</tbody>
</table>

**Step 4** Follow the voice prompts and enter the appropriate values, then press the # key.

**Note** Use the * key to indicate a delimiter (dot). For example, to enter an IP address of 192.168.3.1, you would enter 192*168*3*1 on your telephone keypad.

**Note** When entering values for a field that contains a hexadecimal value, you must convert the hexadecimal value to a decimal value in order to enter it into the voice configuration menu system. For example, to enter the hexadecimal value 0x6A, you would enter the number 106 on the telephone keypad.

The voice configuration menu repeats the value you entered, then prompts 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

**Step 5** Cisco strongly recommends that you set a password. Use the voice menu code 7387277 (SETPASS) to configure a password through the voice configuration menu, after which you are prompted for the password whenever you attempt to change a parameter value.

**Step 6** After completing the configuration through the voice configuration menu, press the # key to exit.

**Step 7** Hang up the telephone. The Cisco ATA configuration refreshes. The function button fast-blinks when the refresh completes.

### Entering Alphanumeric Values

Some voice configuration menu options require you to enter alphanumeric characters. Alphanumeric entry differs from numeric entry because you must press # after each character selected.

If you need to enter an alphanumeric value, the voice prompt tells you to enter an alphanumeric value; otherwise, enter a numeric value (0 to 9).

**Table 3-7** lists the keys on a telephone keypad and their respective alphanumeric characters.

Using Table 3-7 as a guide, enter the appropriate number key on the telephone keypad as many times as needed to select the number, letter, or symbol required. For example, to enter 58sQ, you would enter:

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

**Table 3-7 Alphanumeric Characters**

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

**Resetting the Cisco ATA to Factory Default Values**

It is possible that you may, under some circumstances, want to reset the Cisco ATA to its factory default values. For example, this is the only way to recover a forgotten password without contacting your Cisco representative.

To perform a factory reset, you must use the voice configuration menu and follow these steps:

**Procedure**

**Step 1** Press the function button on the Cisco ATA.

**Step 2** Press the digits 322873738 (FACTRESET) then press # on your telephone keypad.

**Step 3** Press * on your telephone keypad to confirm that you want to reset the Cisco ATA, then hang up the phone.

**Cisco ATA Web Configuration Page**

You can use the Cisco ATA web configuration page in a non-TFTP configuration environment, or in a TFTP configuration environment as a read-only record of individual customer parameters.

You can display the most recent Cisco ATA configuration file from the TFTP server by opening your web browser and typing the following:

http://<ipaddress>/refresh

where ipaddress is the IP address of the Cisco ATA.

Figure 3-1 shows and example of the Cisco ATA web configuration page, which displays all configurable parameters.
**Note**

Do not use the web configuration page to attempt to change any values that you configured by means of the TFTP configuration file method. Whenever the Cisco ATA refreshes, it downloads its ata<macaddress> configuration file(s) or atadefault.cfg default configuration file from the TFTP server, and the values in either of these files will overwrite the values of any corresponding parameters configured with the web configuration method.
### Figure 3-1  Cisco ATA Web Configuration Page

| UTPassword:         | *                           | UseTftp:     | 0                           |
|                     |                             | TftpURL:     |                             |
| EncryptKey:         | *                           | CfgInterval: | 3600                        |
| Dhep:               |                             | EncryptKeyEx:| 00000000000000000000000000000000 |
| StaticRoute:        | 0.0.0.0                     | StaticNetMask:| 255.255.255.0             |
| UID0:               | 9300                        | PWD0:        |                             |
| UID1:               | 0                           | PWD1:        |                             |
| GkOrProxy:          | 192.166.2.81                | UseLoginID:  | 0                           |
| LoginID:            | 0                           | LoginID1:    |                             |
| AltGk:              |                             | AltGkTimeout:| 60                          |
| SIPRegInterval:     | 3600                        | MaxRedirect: | 5                           |
| SIPRegOn:           | 1                           | NATIP:       | 0.0.0.0                     |
| SIPPort:            | 5060                        | MediaPort:   | 16384                       |
| OutBoundProxy:      | 0                           | NatServer:   |                             |
| NatTimer:           | 0:00000000                  | MsgRetryLimits:| 0:00000000             |
| SessionTimer:       | 0:00000000                  | SessionInterval:| 7000             |
| MinSessionInterval: | 1800                        | DisplayName0:| 0                           |
| DisplayName1:       | 0                           | LBRCodec:    |                             |
| AudioMode:          | 0:03150015                  | RxCodec:     | 2                           |
| TxCodec:            | 2                           | NumTxFrames: | 2                           |
| CallFeatures:       |                             | PaidFeatures:| 0:00000000                  |
| CallerIdMethod:     | 0:00019e60                  | FeatureTimer:| 0:00000000                  |
| FeatureTimer2:      | 0:0000000e                  | Polarity:    | 0:00000000                  |
| ConnectMode:        | 0:00600400                  | TimeZone:    | 17                          |
| NTPIP:              | 0.0.0.0                     | AllNTPIP:    | 0.0.0.0                     |
| DNS1IP:             | 0.0.0.0                     | DNS2IP:      | 0.0.0.0                     |
| TOS:                | 0x00000888                  | SigTimer:    | 0x01418564                  |
| OpFlags:            | 0x00000002                  | VLANSetting:| 0x00000002b                 |
| FXSInputLevel:      | -1                          | FXSOuputLevel:| -1                      |
| NPrintf:            | 192.166.3.105.9300          | TraceFlags:  | 0:00000000                  |
| SyslogIP:           | 0.0.0.0.514                 | SyslogCrtl:  | 0:00000000                  |
| RingOnOffTime:      | 2.425                       | IPDialPlan:  |                             |
| DialPlan:           |                             | DialPlanEx:  |                             |
| DialTone:           | 2.31536,308321,1360,1740    | BusyTone:    | 2.30467,20959,1191,1513     |
| reorderTone:        | 2.30467,20959,1191,1513     | RingBackTone:|                             |
| CallWaitTone:       | 1.30831,0.5493,0,0.2400,0,2 | AlertTone:   | 1.30467,0.5970,0,0.480,0,48 |
| STone:              |                             | CallCmd:     |                             |
| CFGID:              | 0x00000000                  |               |                             |
You can access the web configuration page from any graphics-capable browser, such as Microsoft Internet Explorer or Netscape. This provides easy initial access to the Cisco ATA configuration within the administrator's private network.

Follow these steps to set parameters using the web configuration page:

Procedure

Step 1  Make sure that your PC and the Cisco ATA are already networked and visible to each other.

Step 2  Open your web browser.

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

\[\text{http://IP Address/dev}\]

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

\[\text{http://192.168.3.225/dev}\]

Step 4  Select the values for the items that you want to configure. See Chapter 5, “Parameters and Defaults,” for a complete list of parameters and their definitions. Also see Table 4-4 on page 4-23 for an alphabetical listing of configurable features and references to their corresponding parameters.

Note  Cisco strongly recommends that you set a password. Use the UIPassword parameter to configure a password, after which you are prompted for the password whenever you attempt to change a parameter value. Configuration parameters cannot be accessed through the voice configuration menu if the password contains one or more letters and can be changed only by using the web interface or the TFTP configuration method.

Step 5  Click apply to save your changes.

The Cisco ATA automatically refreshes its configuration.

Step 6  Close your web browser.

**Refreshing or Resetting the Cisco ATA**

Whenever you make configuration changes to your Cisco ATA configuration file, you can refresh or reset the Cisco ATA for these configuration changes to immediately take effect. If you do not refresh or reset the Cisco ATA, the configuration changes will take effect the next time the Cisco ATA contacts the TFTP server, which occurs based on the configured value of the CfgInterval parameter.

Note  A refresh procedure will update the Cisco ATA configuration file. A reset procedure will also update the Cisco ATA configuration file, and will additionally power-down and power-up the Cisco ATA. A reset should not be necessary if your only goal is to update the configuration file.
Procedure to Refresh the Cisco ATA

To refresh the Cisco ATA, enter the following command from your web browser:

http://<ipaddress>/refresh

where ipaddress is the IP address of the Cisco ATA that you are refreshing.

Procedure to Reset the Cisco ATA

To reset the Cisco ATA, enter the following command from your web browser:

http://<ipaddress>/reset

where ipaddress is the IP address of the Cisco ATA that you are resetting.

Obtaining Cisco ATA Configuration File After Failed Attempt

The Cisco ATA uses the following formula for determining how soon to contact the TFTP server for the Cisco ATA configuration file after a failed attempt at getting the file. The result of the formula is called the random back-off amount.

\[
\text{random back-off amount} = \text{CfgInterval} + \text{random}(	ext{min}(1800, \text{CfgInterval}))
\]

where

- CfgInterval is the value of the CfgInterval configuration parameter (in seconds). For more information about this parameter, see the “CfgInterval” section on page 5-6.
- random(x) function yields a value between 0 and x-1.
- min(x,y) function yields the smaller value of x and y.

Upgrading the SIP Signaling Image

For instructions on how to upgrade the Cisco ATA to the most recent SIP signaling image, refer to the following list:

- To use the recommended TFTP method of upgrading the Cisco ATA, see the “Upgrading the Signaling Image from a TFTP Server” section on page 8-1.
- In the rare instance that you are not using the TFTP server to configure the Cisco ATA and to obtain software upgrades, you must manually upgrade to the latest signaling image immediately after the Cisco ATA boots up. In this case, see the “Upgrading the Signaling Image Manually” section on page 8-2.
Basic and Additional SIP Services

This section provides information about key basic and additional SIP services that the Cisco ATA supports:

- **Important Basic SIP Services, page 4-1**—This section includes a list of parameters that you must configure in order for the Cisco ATA to function in a SIP environment.
- **Additional SIP Services, page 4-3**—This section contains information about additional, commonly used SIP features, with references to the parameters for configuring these services.
- **Complete Reference Table of all Cisco ATA SIP Services, page 4-23**—This section contains a complete listing of Cisco ATA services supported for SIP, and includes cross references to the parameters for configuring these services. This section includes services not described in the sections about the key basic SIP services and the commonly used additional SIP services.

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

### Important Basic SIP Services

This section provides descriptions and cross references for configuring required SIP parameters and also for configuring other important basic SIP services:

- **Required Parameters, page 4-1**
- **Establishing Authentication, page 4-2**
- **Setting the Codec, page 4-3**
- **Configuring Refresh Interval, page 4-3**

### Required Parameters

If you are using the SIP protocol, you need to supply values for the required SIP parameters shown in **Table 4-1**. The Parameter column provides the name of the parameter and a cross reference which provides a more-detailed description of the parameter.

**Note**
See Chapter 5, “Parameters and Defaults,” for information about additional Cisco ATA parameters.
### Important Basic SIP Services

#### Establishing Authentication

The Cisco ATA supports two levels of authentication, depending on the setting of the UseLoginID parameter:

- If UseLoginID is set to 0, the user ID (UID0 or UID1) is used with a user-supplied password (PWD0 or PWD1) for authentication.
- If UseLoginID is set to 1, you must supply a login ID (LoginID0 or LoginID1) and a password (PWD0 or PWD1) for authentication.

#### Related Configuration Parameters

- UseLoginID, page 5-18
- UID0, page 5-15
- UID1, page 5-16
- LoginID0, page 5-17
- LoginID1, page 5-18
- PWD0, page 5-16
- PWD1, page 5-17

---

### Table 4-1 Required SIP Parameters and Defaults

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value Type</th>
<th>Description</th>
<th>Voice</th>
<th>Minimum Value</th>
<th>Maximum Value</th>
<th>Default</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIPRegInterval, page 5-19</td>
<td>Integer</td>
<td>Seconds between registration renewal</td>
<td>203</td>
<td>1</td>
<td>86400</td>
<td>3600</td>
</tr>
<tr>
<td>MAXRedirect, page 5-20</td>
<td>Integer</td>
<td>Maximum number of times to try redirection</td>
<td>202</td>
<td>0</td>
<td>10</td>
<td>5</td>
</tr>
<tr>
<td>SIPRegOn, page 5-20</td>
<td>Integer</td>
<td>Enable SIP registration</td>
<td>204</td>
<td>0</td>
<td>1</td>
<td>0</td>
</tr>
<tr>
<td>NATIP, page 5-21</td>
<td>IP address</td>
<td>WAN address of the attached router/NAT; currently only used to support SIP behind a NAT.</td>
<td>200</td>
<td>0</td>
<td>255</td>
<td>0.0.0.0</td>
</tr>
<tr>
<td>SIPPort, page 5-19</td>
<td>Integer</td>
<td>Port to listen for incoming SIP requests</td>
<td>201</td>
<td>1</td>
<td>65535</td>
<td>5060</td>
</tr>
<tr>
<td>MediaPort, page 5-30</td>
<td>Integer</td>
<td>Base port to receive RTP media; only used to support SIP behind a NAT.</td>
<td>202</td>
<td>1</td>
<td>65535</td>
<td>16384</td>
</tr>
<tr>
<td>SipOutBoundProxy, page 5-21</td>
<td>Alphanumeric string</td>
<td>Proxy server for all outbound SIP requests. All SIP requests are sent to SipOutBoundProxy, when configured, instead of to the configured GkOrProxy.</td>
<td>206</td>
<td>—</td>
<td>—</td>
<td>0</td>
</tr>
<tr>
<td>GkOrProxy, page 5-13</td>
<td>Alphanumeric string</td>
<td>SIP proxy server address or registrar address.</td>
<td>5</td>
<td>—</td>
<td>—</td>
<td>0</td>
</tr>
</tbody>
</table>

---

4-2
Chapter 4      Basic and Additional SIP Services

Setting the Codec

The LBRCodec (low-bit-rate codec) parameter determines whether the G.723, G.726 or G.729A codec, in addition to G.711A-law and G.711µ-law, can be used for receiving and transmitting. For configuration information, see the “LBRCodec” section on page 5-32.

Configuring Refresh Interval

When the value specified in the CfgInterval parameter is reached, the Cisco ATA attempts to refresh its configuration file from the TFTP server. By opening a web page for the Cisco ATA, you can perform a refresh before the scheduled refresh. Set the CfgInterval parameter to an interval value (in seconds) for refreshing the Cisco ATA configuration file. Cisco recommends that the interval be semi-random to prevent many simultaneous contacts with the TFTP server. For more information, see the “CfgInterval” section on page 5-6.

When the Cisco ATA contacts the TFTP server, it also checks to see if an upgrade signaling image has been placed on the TFTP server. If such an image exists, the Cisco ATA will download this image.

Additional SIP Services

This section describes additional SIP services and, where applicable, provides configuration information and cross references to the parameters for configuring these services. These services are listed alphabetically.

- Advanced Audio Configuration, page 4-4
- Billable Features, page 4-4
- Call Forwarding Setting Removal Using HTTP, page 4-5
- Call-Waiting Hang-Up Alert, page 4-5
- Comfort Noise During Silence Period When Using G.711, page 4-6
- Configurable Hook Flash Timing, page 4-7
- Configurable Mixing of Call Waiting Tone and Audio, page 4-7
- Configurable On-hook delay, page 4-7
- Configurable Reboot of Cisco ATA, page 4-7
- Diagnostics for Debugging, page 4-7
- Dial Plan, page 4-7
- Disabling Access To The Web Interface, page 4-8
- Display-Name Support for Caller ID, page 4-8
- Distinctive Ringing, page 4-8
- DNS SRV Support, page 4-9
- Hardware Information Display, page 4-9
- NAT Gateway, page 4-9
- NAT/PAT Translation, page 4-10
- Network Timing, page 4-10
Advanced Audio Configuration

The TOS (specifies the precedence and delay of audio and signaling IP packets) and AudioMode (audio operating mode) parameters allow you to tune audio configuration.

Related Parameters

TOS, page 5-34
AudioMode, page 5-32

Billable Features

You can customize specific features on a subscription basis by changing the values of specific bits in several different parameters. Table 4-2 contains a list of billable features and their related parameters:

<table>
<thead>
<tr>
<th>Feature</th>
<th>Related Parameters</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Conferencing</td>
<td>PaidFeatures, page 5-36, CallFeatures, page 5-35</td>
</tr>
<tr>
<td>Call Forwarding</td>
<td>PaidFeatures, page 5-36, CallFeatures, page 5-35, ConnectMode, page 5-41, SigTimer, page 5-40</td>
</tr>
<tr>
<td>Call Transfer</td>
<td>PaidFeatures, page 5-36, CallFeatures, page 5-35</td>
</tr>
</tbody>
</table>
Call Forwarding Setting Removal Using HTTP

The service provider can remotely reset a call forwarding setting for which a subscriber configured an incorrect phone number to receive forwarded calls.

The service provider issues the following command, which removes call forwarding settings for both Cisco ATA phone lines:

```
http://ipaddress/resetcfwd/
```

where \textit{ipaddress} is the IP address of the Cisco ATA whose call forwarding numbers are being removed. This Web page is password protected. Once the service provider issues this command, this Web page shows that the current call-waiting and call-forwarding settings are \textit{N/A}.

Call-Waiting Hang-Up Alert

This feature provides an audible alert (ringtone) whenever the user inadvertently hangs up from a call-waiting call while an active call is still on hold.

This section contains the following topics:

- Enabling the Call-Waiting Hang-Up Alert Feature, page 4-6
- Default Behavior of Call-Waiting Calls, page 4-6

\begin{table}[h]
\centering
\begin{tabular}{|c|c|}
\hline
Feature & Related Parameters \\
\hline
Call Waiting & PaidFeatures, page 5-36, CallFeatures, page 5-35, SigTimer, page 5-40 \\
\hline
Caller ID & PaidFeatures, page 5-36, CallFeatures, page 5-35, CallerIdMethod, page 5-49 \\
\hline
Call Return & ConnectMode, page 5-41, PaidFeatures, page 5-36, CallFeatures, page 5-35 \\
\hline
Polarity & Polarity, page 5-51 \\
\hline
Voice Mail Indicator & PaidFeatures, page 5-36, CallFeatures, page 5-35 \\
\hline
\end{tabular}
\caption{Billable Features and Related Parameters (continued)}
\end{table}
Enabling the Call-Waiting Hang-Up Alert Feature

To enable the call-waiting hang-up alert feature, perform the following steps:

**Procedure**

1. **Step 1** Enable bit 25 of the Cisco ATA ConnectMode parameter. (For more information, see the “ConnectMode” section on page 5-41.

2. **Step 2** Make sure the call-waiting call command is set to one of the following values:
   - Kf;EFh;HF; (for U.S. users)
   - Kh;HFf;EF; (for U.S. users)
   - Kf1;HFf2;EFF3;AFf4;HQh;HF; (for Sweden users)

**Note** The \( F \) Action-Identifier specifies the retrieval of the held call after the active call is disconnected when the user hangs up. For more information about call commands, see Chapter 6, “Call Commands.”

Default Behavior of Call-Waiting Calls

Without the call-waiting hang-up alert feature enabled, both the call-waiting call and active call are disconnected as soon as the user hangs up the phone.

To check whether this default behavior is in effect, search for the appearance of the string \( h;HH \) within the sequence of call-waiting call commands. The context-identifier \( K \) denotes the beginning of the call-waiting call commands.

With this default behavior, the call-waiting call command string could be one of the following examples:
   - Kf;EFh;HH; (for U.S. users)
   - Kh;HFf;EF; (for U.S. users)
   - Kf1;HFf2;EFF3;AFf4;HQh;HH; (for Swedish users)
   - Kf1;HFf2;EFF3;AFf4;HQ; (for Swedish users with no specific on-hook treatment defined)

For more information about call commands, see Chapter 6, “Call Commands.”

Comfort Noise During Silence Period When Using G.711

When silence suppression is turned on in ITU G.711, the Cisco ATA calculates and transmits its noise level to the far end to enable the remote endpoint to generate the appropriate amount of comfort noise. This provides the remote user with a similar experience to that of a PSTN call and prevents silent gaps when neither party is talking.

**Related Parameter**

AudioMode, page 5-32—Bit 0 disables/enables silence suppression.
Configurable Hook Flash Timing

This feature provides the ability to adjust the hook-flash timing to meet local requirements.

**Related Parameter**

SigTimer, page 5-40—Bits 26 and 27 are for configuring the minimum on-hook time required for a hook flash event, and bits 28 through 31 are for configuring maximum on-hook time.

Configurable Mixing of Call Waiting Tone and Audio

This feature allows the call-waiting tone to be mixed with the audio in an active call. Therefore, the call-waiting tone will sound without a pause in the audio.

**Related Parameter**

ConnectMode, page 5-41—Bit 24

Configurable On-hook delay

This feature is available only for the recipient (callee) of a call. If the callee picks up the phone and then later hangs up to retrieve another call, the hang-up is not considered on-hook until the specified delay expires.

**Related Parameter**

FeatureTimer, page 5-38—Bits 8 to 12

Configurable Reboot of Cisco ATA

The Cisco ATA continuously monitors its Ethernet connection to the switch or hub. If this connection is broken, the Cisco ATA starts an internal timer that runs until a configurable timeout period expires. Once the timeout value is reached, the Cisco ATA automatically reboots. This timeout value is configured by using the FeatureTimer2 configuration parameter. For more information, see the “FeatureTimer2” section on page 5-39.

Diagnostics for Debugging

You can use the following parameters to troubleshoot operation issues:

- **NPrintf, page 5-73**—Specify the IP address and port where debug information is sent.
- **TraceFlags, page 5-73**—Use to turn on specific trace features.

Dial Plan

You can set specific dial plan rules and timeout values. Many of these values are determined on a country-by-country basis.
Related Parameters

- DialPlan, page 5-64
- DialPlanEx, page 5-72

Disabling Access To The Web Interface

To prevent tampering and unauthorized access to the Cisco ATA configuration, the Cisco ATA built-in web server can be disabled.

Related Parameter

OpFlags, page 5-45—Bit 7

Display-Name Support for Caller ID

For caller ID purposes, you can configure a name to correspond to the phone number of the Cisco ATA input ports. This name will be displayed at the remote endpoint when a call originates from this Cisco ATA.

Related Parameters

- DisplayName0, page 5-29—for the Phone 1 port
- DisplayName1, page 5-29—for the Phone 2 port

Distinctive Ringing

This feature allows a user to identify a caller based on the ringing pattern the user selects for the incoming number.

This feature is dependent on the proxy or remote UA, including the Alert-Info header with the appropriate value in the INVITE message. The Cisco ATA supports standard distinctive ringing pattern 1 to 5 as defined in the standard GR-506-CORE.

The following Alert-Info header values are allowed:

- Bellcore-dr1
- Bellcore-dr2
- Bellcore-dr3
- Bellcore-dr4
- Bellcore-dr5

If the Alert-Info header value is not recognized, the Cisco ATA plays the regular ring tone, Bellcore-dr1.

Note

The Bellcore-dr5 ringing pattern is the same as the Bellcore-dr1 ringing pattern.
DNS SRV Support

The Cisco ATA supports DNS SRV lookup for the SIP proxy server. If the GkOrProxy parameter value begins with _sip._udp. or sip.udp., the Cisco ATA performs a DNS SRV lookup for the SIP proxy server. A DNS SRV lookup results in one of the following conditions:

- Zero host is returned or DNS SRV lookup failed. The Cisco ATA then performs a regular DNS A-record lookup for the given name.
- One host is returned. The single host is used as the primary proxy and AltGk is the backup proxy, if specified.
- Two or more hosts are returned. The two hosts with the highest priorities are used as the primary and backup proxy servers (AltGk is ignored in this case).

Related Parameters
- GkOrProxy, page 5-13
- AltGk, page 5-14

Hardware Information Display

Cisco ATA hardware information is displayed in the lower-left corner of the Cisco ATA Web configuration page.

NAT Gateway

Network Address Translation (NAT) supports port mapping and forwarding to standard default SIP signaling port 5060 and media base port 16384, or other ports as configured in the Cisco ATA. Media ports are evenly numbered from the base port. NAT must support multiple port mappings. The Cisco ATA can use up to four media ports to handle conference calls on both lines. For example, if media base port 16384 is used for one call, the next call uses port 16386 and other calls will use ports 16388 and 16390.

Note
Routers such as D-Link, WinRoute, and WinProxy may not route correctly if both caller and callee are behind the same NAT.

To configure the Cisco ATA to work in a NAT environment, modify the following parameters:
- StaticRoute, page 5-9—Enter the LAN IP address of the NAT through which the Cisco ATA will communicate.
- NATIP, page 5-21—Enter the WAN IP address of the NAT through which all external SIP user agents will communicate.
- SIPPort, page 5-19—Enter a new port for SIP messages (optional).
- MediaPort, page 5-30—Enter a new base port for RTP media (optional).
NAT/PAT Translation

To maintain Network Address Translation/Port Address Translation (NAT/PAT) for a session, the Cisco ATA can be configured to periodically send a dummy UDP packet to a server (the Cisco ATA does not expect any response from the server).

Related Parameters

- **NatTimer**, page 5-22—Bits 0 to 11 are for specifying the retransmission period.
- **NatServer**, page 5-22—Specify the server to which the dummy packet is sent.

Network Timing

You can fine tune your network timing with the following parameters:

- **TimeZone**, page 5-48—Use for time-stamping incoming calls (offset from Greenwich Mean Time) with local time.
- **NTPIP**, page 5-11—Use for configuring the IP address of the Network Time Protocol server. NTP is a protocol built on top of TCP that ensures accurate local time-keeping with reference to radio and atomic clocks located on the Internet.
- **AltNTPIP**, page 5-12—Use to configure an alternate NTP server IP address.
- **ConnectMode**, page 5-41—Used to control the connection mode of the SIP protocol.

Obtaining Network Status Before and After Getting IP Connectivity

Using voice configuration menu code 3123#, you can obtain basic network status to use for diagnostic purposes prior to getting IP connectivity. For detailed information, see the “Obtaining Network Status Prior to Getting IP Connectivity” section on page 9-11.

Use the Cisco ATA Stats Web page (http://<Cisco ATA IP address>/stats) to display network status information after obtaining IP connectivity. For detailed information, see the “Obtaining Network Status After Getting IP Connectivity” section on page 9-12.

Privacy Options

The privacy options described in this section provide users with stricter control over the appearance of their caller line identification at the SIP message level. These options not only protect the user’s anonymity at the caller site but also prevent network-level sniffer-type applications from gaining access to restricted information that is in transit.

This section contains the following topics:

- **Network Infrastructure Requirements**, page 4-11
- **Anonymity for Called Party**, page 4-11
- **Anonymous User Name Support for SIP INVITE Requests**, page 4-11
- **Privacy Token Support for SIP Diversion Header**, page 4-12
Network Infrastructure Requirements

For user privacy to work effectively on the Cisco ATA, the proxy or proxies deployed in the IP network must be capable of the following:

- Privacy token support.
- Determining whether a third party involved in a diversion case is trusted so that the proxy can forward private Diversion headers to that site.
- Determining that a site is not trusted and being able to change the user names to Anonymous in Diversion headers before including these headers in resulting INVITE requests.

Gateways deployed in the IP Network must be capable of the following:

- Privacy token support
- Redirecting a number by correctly setting the presentation bits in the Redirecting Number field of the Initial Address Message (IAM) message, based on the level of privacy requested in the Diversion header.

Anonymity for Called Party

The Cisco ATA provides an option to use the Anonymous user name in the TO header in all outgoing SIP INVITE requests when the Cisco ATA is the calling party.

To enable this option, set bit 30 of the Cisco ATA ConnectMode parameter to 1. For more information, see the “ConnectMode” section on page 5-41.

This feature guarantees the anonymity of the called party (in the To header) on every call that the Cisco ATA initiates. However, the main intent of this feature is to hide the identity of the called party in the event that the call is diverted to a not-trusted address and caller-ID-restricted is configured on the diverting Cisco ATA. In this case, the proxy has the responsibility of removing user-sensitive data before the call is sent to the not-trusted address.

The Cisco ATA handles the TO header because proxies are not allowed to change the TO header.

Anonymous User Name Support for SIP INVITE Requests

The Cisco ATA provides an option to use the Anonymous user name in the FROM and CONTACT headers and in the o= line (also called the origin line) of the Session Description Protocol (SDP) header in SIP INVITE requests to the far end. The Anonymous user name is used when the following three conditions are met:

- The Cisco ATA is acting as the caller.
- The Cisco ATA Caller Line Identification Restriction (CLIR) feature is enabled, which can be performed in one of two ways:
  - By setting bit 3 of both the CallFeatures and PaidFeatures parameters to 0. (For the Phone2 port of the Cisco ATA, you would set bit 19 to 0 for each parameter.) For more information on these parameters, see the “CallFeatures” section on page 5-35 and the “PaidFeatures” section on page 5-36.
  - By enabling the CLIR on a per-call basis by using the call command dial string. Enabling the CLIR on a per-call basis requires that the dial string sequence (typically *69) that users enter on their dialpad prior to dialing the phone number match the specified CLIR string defined in the call command. For more information, see Chapter 6, “Call Commands.”
Example Cisco ATA INVITE Messages
The following example SIP INVITE messages show how the Anonymous user name would appear in the context of these messages:

```
INVITE sip:9401@192.168.2.146:5060;user=phone SIP/2.0
Via: SIP/2.0/UDP 192.168.2.96:5060;branch=7cca152b-232af34f-3f0b0c31-3cea3a52-1
Via: SIP/2.0/UDP 192.168.3.117:5060;received=192.168.3.117
Supported: timer
From: "Anonymous" <sip:Anonymous@192.168.2.96;user=phone>;tag=173234376
To: <sip:9401@192.168.2.96;user=phone>
Call-ID: 1157628352@192.168.3.117
CSeq: 1 INVITE
Contact: "Anonymous" <sip:Anonymous@192.168.3.117:5060;user=phone;transport=udp>
User-Agent: Cisco ATA 186 v3.0.0 atasip (030619A)
Allow: ACK, BYE, CANCEL, INVITE, NOTIFY, OPTIONS, REFER, REGISTER
Expires: 300
Content-Length: 252
Content-Type: application/sdp
v=0
o=Anonymous 6269 6269 IN IP4 192.168.3.117
```

Privacy Token Support for SIP Diversion Header
Proxies typically use the privacy token value contained in the Diversion header of SIP INVITE messages and 3xx Redirection responses to determine whether any of the diverting party's user names should be changed before forwarding the message to untrusted addresses.

This feature applies only when the following two conditions are met:
- The Cisco ATA is the callee and is forwarding or diverting a call.
- Bit 27 of the ConnectMode parameter is set to 1. For more information, see the “ConnectMode” section on page 5-41.

Before forwarding the call, the Cisco ATA appends a privacy=[full|off] field to the end of the Diversion header in a 302 Moved Temporarily message.

The value of the privacy=[full|off] field depends on the setting of bit 3 of the CallFeatures and PaidFeatures parameters. (Bit 19 is the applicable bit for the Phone2 port of the Cisco ATA.) This bit is for configuring either the Caller Line Identification Restriction (CLIR) or Caller Line Identification Presentation (CLIP) feature.

- If CLIR is the configured feature, then privacy=full is appended to the Diversion header.
- If CLIP is the configured feature, then privacy=off is appended to the Diversion header.

For more information on CLIR and CLIP, see the “CallFeatures” section on page 5-35 and the “PaidFeatures” section on page 5-36.
Progress Tones

Values for the following parameters (all defined in the “Tone Configuration Parameters” section on page 5-53) must be determined based on the country in which the Cisco ATA is located:

- DialTone
- BusyTone
- ReorderTone
- RingBackTone
- CallWaitTone
- AltertTone

Real-Time Transport Protocol (RTP) Statistics Reporting

To monitor the quality of service for the media stream, you can access RTP packet statistics of the two voice ports and their channels by opening the following page on the Cisco ATA Web server:

<Cisco ATA IP address>/rtps

For detailed information about RTP statistics reporting, see the “Real-Time Transport Protocol (RTP) Statistics Reporting” section on page 9-13.

Receiver-tagged VIA header

You can disable or enable the processing the received = parameter in the Via header. This feature is disabled by default.

Related Parameter
ConnectMode, page 5-41—Bit 22

Redundant Proxy Support for BYE/CANCEL Request

The Cisco ATA retries a BYE or CANCEL request using an alternate SIP proxy if the GkOrProxy parameter value is configured with a domain name. The BYE request requires special consideration because the destination can be either the SIP endpoint client or proxy server.

For a SIP user agent client, if a SIP proxy server does not include a Record-Route header in its 200 OK response to an INVITE request, the destination of a BYE request is the SIP URL specified in the Contact header of the response. This URL is usually an IP address; therefore, redundancy is not possible.

For a SIP user agent server, if a SIP proxy server does not include a Record-Route header in the original INVITE request, the destination of a BYE request is the SIP URL specified in the Contact header of the request. This URL is also usually an IP address; therefore, redundancy is not possible.
If a Record-Route header is present in the SIP proxy server 200 OK response to an INVITE request or in an original INVITE request, the BYE request is sent to the first SIP URL specified in the Record-Route header. If the SIP URL is a proxy domain name, then proxy redundancy is possible. Therefore, for SIP proxy redundancy to work for a BYE request, the RequestURL must be a domain name.

**Related Parameter**

GkOrProxy, page 5-13

---

### Repeat Dialing on Busy Signal

This feature allows the Cisco ATA to repeatedly call a busy number at a periodic interval for a specific length of time. Both the interval and total time can be specified by the user.

To use this feature, configure FeatureTimer bits 0-7 and add the new command/action values "#37#;kA" to the existing “H” context and “5;jA” to the existing “S” context in the CallCmd parameter.

This feature is invoked by pressing 5 after the busy tone sounds. The caller then gets a beep confirmation followed by silence. When the subscriber hangs up, the Cisco ATA starts to redial at the interval specified in FeatureTimer bits 4-7. When the called party rings, the caller is notified with a special ring. If the called party picks up the call first, the called party receives a ringback. If the caller picks up the call first, the caller receives the ringback. This feature is automatically cancelled when the called party rings.

**Note**

For this feature to work properly, the remote user agent server must return a 486 (Busy Here) response to an INVITE request if it detects that the remote party (IP or PSTN) is busy. If the server returns a 183 (Session Progress) response with an SDP before a 486, the Cisco ATA considers the call successful and automatically cancels repeat dialing.

**Related Parameters**

- **FeatureTimer, page 5-38**—Bits 0 to 3 control the maximum time the Cisco ATA redials a number.
- **FeatureTimer, page 5-38**—Bits 4 to 7 control the interval between each redial that the Cisco ATA performs. A value of zero (0) sets the default redial interval to 15 seconds.
- **CallCmd, page 5-37**—The following context commands are used as follows:

  **Parameter:** CallCmd  
  **Context:** S (may also include ‘a’ or ‘b’)  
  **Command/action:** 5;jA  
  **Description:** This context command adds the service activation code to enable repeat dialing.

  **Parameter:** CallCmd  
  **Context:** H  
  **Command/action:** #37#;kA  
  **Description:** This context command adds the service deactivation code to disable repeat dialing

**Note**

For complete information about call commands, see Chapter 6, “Call Commands.”
Retransmitting SIP requests and SIP Responses

You can configure the number of Cisco ATA transmission attempts for some SIP requests and responses to requests from the SIP user agent. For details on which requests and responses are configurable, see the “MsgRetryLimits” section on page 5-24.

Setting Up and Placing a Call Without Using a SIP Proxy

The Cisco ATA supports direct IP-to-IP calls without using a SIP proxy. When a call is placed, the Cisco ATA sends the INVITE request directly to the remote user agent and expects the usual 100/180/200 responses from the user agent.

This section contains the following topics:
- Configuration, page 4-15
- Placing an IP Call, page 4-16

Configuration

To perform the necessary configuration of the Cisco ATA, follow this procedure:

**Procedure**

**Step 1** Open your Web browser.

**Step 2** Enter the URL: `http://<Cisco_ATA_IP_address>/dev`

where Cisco_ATA_IP_address is the IP address of your Cisco ATA. This takes you to the Cisco ATA Web configuration page.

**Step 3** Configure the following parameters as shown:
- **GkOrProxy**, page 5-13—Set to the value of 0 (zero).
- **UID0**, page 5-15—Set to the unique telephone number of the **Phone 1** port of the Cisco ATA.
- **UID1**, page 5-16—Set to the unique telephone number of the **Phone 2** port of the Cisco ATA.
- **SIPRegOn**, page 5-20—Set to 0 to disable SIP registration with a SIP proxy server.

**Step 4** Click the **Apply** button to save these changes.
Placing an IP Call

To place an IP call, dial the telephone number and the IP address of the remote user agent. The dial format is shown below:

**Dial Format**

\[ <\text{phone number}>**<\text{ipaddress}># \]

Use the star (*) key on the telephone keypad to represent the dot (.) in an IP address. Use the pound (#) key on the telephone keypad to terminate the dial string and place the call.

**Note**

URL dialing is not supported.

**Example**

To place a call to a user agent with an ID of 408-555-1212 at IP address 192.168.1.100, you would enter the following string on your telephone keypad:

4085551212**192*168*1*100#

SipOutBoundProxy Support

If the SipOutBoundProxy parameter is a fully qualified domain name (FQDN), and DNS returns multiple IP addresses, the first IP address is used as the primary outbound proxy and the second IP address as the secondary outbound proxy. If SipOutBoundProxy is an IP address or if DNS returns only one IP address, then a backup outbound proxy is not available. The AltGkTimeOut parameter determines the backup proxy timeout value for the outbound proxy.

If the backup proxy fails, the Cisco ATA automatically switches back to the primary proxy if the unit has been using the backup proxy for at least 30 seconds. This effectively prevents the Cisco ATA from switching indefinitely between failing primary and failing backup proxies for the same transactions.

Switching between primary and secondary proxies can occur only for initial INVITE and REGISTER requests. Other requests, such as CANCEL, BYE, ACK, and re-INVITE, do not retry the backup proxy but give up if the current proxy fails.

When SipOutBoundProxy is enabled, the Cisco ATA determines whether to retry to connect with the backup SipOutBoundProxy or backup SIP proxy if the INVITE or REGISTER requests fail. If the reason for failure is an ICMP error (such as an unreachable host), the Cisco ATA retries with the backup outbound proxy. If failure is due to timeout while waiting for a response or a 5xx response, the Cisco ATA retries the backup SIP proxy.

**Related Parameter**

- SipOutBoundProxy, page 5-21
- AltGkTimeOut, page 5-15

SIP Proxy Server Redundancy

SIP proxy server redundancy can be enabled by entering a fully qualified domain name (FQDN) or IP address (and optional port number) in the GkOrProxy and AltGk parameters, and by configuring the AltGkTimeOut parameter. If you provide hostnames for GkOrProxy or AltGk, the names are resolved by the configured DNS. DNS results are hard-coded in cache memory for 10 minutes.
If DNS returns multiple IP addresses, the Cisco ATA uses only the first IP address. If AltGk is set to 0 (disabled) and DNS returns two or more IP addresses for GkOrProxy, then the Cisco ATA uses the first IP address as the primary proxy and the second IP address as the secondary proxy. If GkOrProxy is an IP address or DNS returns one IP address, then the backup SIP proxy is not available. A special case exists if GkOrProxy and AltGk are the same values and are not IP addresses. In this case, the AltGk parameter is assumed to have the value 0.

Related parameters
- GkOrProxy, page 5-13
- AltGk, page 5-14
- AltGkTimeOut, page 5-15

**SIP Session-Timer Support**

The SIP Session Timer is a keepalive mechanism for a SIP session, and is used to determine whether a call is still active when one of the following conditions occurs:

- The user agent fails to send a BYE message to the Cisco ATA at the end of a SIP session.
- The BYE message that the user agent sends to the Cisco ATA is lost because of network problems.

To avoid a situation where a user agent waits indefinitely for a BYE message, the user agent sends periodic re-INVITE requests (or session refresh requests) to the Cisco ATA to keep the session alive. The interval between these session-refresh requests is negotiated with the use of Session-Expires/Min-SE headers and 422 (Session Interval Too Small) messages. If the Cisco ATA does not receive a session-refresh request before the negotiated interval expires, the session is considered terminated. Both user agents then send BYE messages and disconnect the session.

The Cisco ATA supports session timing only when both the caller and callee support this feature. Also, the Cisco ATA does not support the UPDATE method; therefore, all session-refresh requests are performed by means of the re-INVITE method.

> Note

The Cisco ATA implementation of the SIP Session Timer is based on the document *draft-ietf-sip-session-timer-11.txt*, which can be found on the Internet.

**Related Parameters**

The following three parameters are used to configure SIP session timing:

- SessionTimer, page 5-26—Example settings are included and described.
- SessionInterval, page 5-28
- MinSessionInterval, page 5-28

**Status of Phone Service Using HTTP**

You can use the following command to provide the status of various call features:

http://ipaddress/service/

where *ipaddress* is the IP address of the Cisco ATA whose status you are checking.

The Web page invoked from this command provides information about the following features:
• CWaIt (call waiting)—The status is shown as either on or off.
• CFwdu (call forwarding unconditional)
• CFwdNA (call forwarding no answer)
• CFwdB (call forward busy)
• CRtn (call return number)

For all features except call waiting, the information on the Web page either provides the applicable phone number, which also means that the feature has been activated, or N/A is shown if the feature has not been activated.

STUN Support

The Cisco ATA supports a Simple Traversal of UDP through NAT (STUN) client, as described in RFC 3489. The Cisco ATA obtains the IP address and port mappings of the NAT and uses them accordingly in a SIP message.

This section contains the following topics:

• Types of NATs, page 4-18
• NAT Traversal, page 4-18
• STUN Configuration Parameters, page 4-19

Types of NATs

Four types of NATs can be used:

• Full Cone NAT—This type of NAT maps all requests from the same internal IP address and port to the same external IP address and port. Any external host can send a packet to the internal host only if the internal host had previously sent a packet through the NAT.
• Restricted Cone NAT—This type of NAT maps all requests from the same internal IP address and port to the same external IP address and port. An external host (with IP address X) can send a packet to the internal host only if the internal host had previously sent a packet to IP address X.
• Port Restricted Cone NAT—This type of NAT maps all requests from the same internal IP address and port to the same external IP address and port. An external host (with IP address X and source port P) can send a packet to the internal host only if the internal host had previously sent a packet to IP address X and port P.
• Symmetric NAT—In a symmetric NAT, all requests from the same internal IP address and port to a specific destination IP address and port are mapped to the same external IP address and port. If the same host sends a packet with the same source address and port, but to a different destination, a different mapping is used. Furthermore, only the external host that receives a packet can send a UDP packet back to the internal host.

A STUN server can help facilitate traversing through most NATs, except for symmetric NATs.

NAT Traversal

To help facilitate traversal through a NAT, the Cisco ATA uses the same signaling port for transmitting and receiving SIP messages and the same media port for transmitting and receiving media. Before an external host can send a packet to the Cisco ATA behind a NAT, the Cisco ATA must already have sent a packet through the NAT.
An external host can communicate with a Cisco ATA behind a full cone NAT by sending packets to the mapped port. If the Cisco ATA is behind a restricted cone NAT, an external host would have to send packets from the same IP address that it used to receive packets from the Cisco ATA.

If the Cisco ATA is behind a port-restricted cone NAT, an external host would have to send packets from the same IP address and port it used to receive packets from the Cisco ATA. Because a symmetric NAT creates different mapping for every packet with a different destination IP address and/or port, the Cisco ATA cannot traverse through this type of NAT.

If properly configured, the Cisco ATA on power up contacts the specified STUN server to obtain the IP address and port mappings of the NAT before registering with the registration server. The Cisco ATA substitutes this mapping information into the Via and Contact headers. Each time the Cisco ATA sends an INVITE message, the Cisco ATA obtains IP address and port mappings from the STUN server and substitutes this mapping information into the Via, Contact, and SDP c= and m= headers. This allows the SIP proxy server and remote user agents to communicate with the Cisco ATA through most NATs.

**STUN Configuration Parameters**

Two parameters control the operation of the Cisco ATA with a STUN server:

- **NatTimer**—This parameter allows the following configuration:
  - Interval of keep-alive packets
  - STUN mode to use
  - Destination of keep-alive packets

  For more information, see the “NatTimer” section on page 5-22.

- **NatServer**—This parameter is used to specify a server to which keep-alive packets are sent. If the NatServer is a STUN server and STUN mode is selected, the Cisco ATA obtains IP address and port mapping information from this server.

  For more information, see the “NatServer” section on page 5-22.

**Stuttering Dial Tone on Unconditional Call Forward**

If unconditional call forwarding is enabled, the Cisco ATA plays a continuous stuttering dial tone when the telephone handset is picked up. This reminds the user that all incoming calls are forwarded to another number. For more information, see the “Call Forwarding in the United States” section on page A-5 and the “Call Forwarding in Sweden” section on page A-6.

**Toll Restrictions for Call Forwarding and Outgoing Calls**

You can configure the Cisco ATA to block certain numbers from call forwarding and to display certain numbers for caller ID.

**Related Dial Plan Rules**

- ‘F’ Rule for Call Forwarding Blocking, page 5-69
- ‘D’ Rule for Displaying Caller ID, page 5-70
User Configurable Call Waiting Permanent Default Setting

This feature allows you to specify the default call-waiting setting for every call on a permanent basis by means of the service activation and deactivation codes.

Related Parameter
ConnectMode, page 5-41—Bit 23

User Configurable Timeout On No Answer for Call Forwarding

This feature allows you to specify the timeout before a call is forwarded to another number on no answer. This feature is activated by entering the service activation code followed by the phone number and delay. The entry sequence is as follows:

\(<\text{Service Activation Code}\> \ <\text{Phone Number}\> \ * \ <\text{Delay}\> \ #\)

Delay can be from 1 to 255 seconds. If the delay is zero (0) or not provided by the user, the delay specified in the SigTimer parameter (bits 20-25), which has a default value of 20 seconds, is in effect.

Example
Using the U.S. Call Command parameter string, the U.S. service activation code is #75 and the deactivation code is #73.

To forward calls to the number 555-1212 after a no-answer for 15 seconds, enter the following:

#755551212*15#

To deactivate this feature, enter the following:

#73

Related Parameter
SigTimer, page 5-40—Bits 20 to 25

Voice Prompt Confirmation for Call Waiting and Call Forwarding

You can configure the Cisco ATA to automatically call a voice announcement server whenever the status of call-waiting or call-forwarding services changes. The telephone number of the server to which the Cisco ATA will send an INVITE request is specified by a configurable base number and several pre-assigned extension numbers. The extension numbers correspond to the service and the state change of the service.

You must configure the following information:

- Base Number, page 4-21
- Relevant Bit of OpFlags Parameter, page 4-21
**Base Number**

The base number is the first part of a number that the Cisco ATA calls, as specified in the dial plan using rule ‘B.’ To set this base number using rule plan ‘B’, you would use ‘B’ followed by the desired base number. If, for example, the desired base number is 1234, you would add the rule ‘B1234’ to your dial plan.

**Note**

Each dial plan rule must be partitioned from other rules with a vertical bar ( | ).

The telephone number that the Cisco ATA will call will always consist of the base number followed by a two-digit extension. The extensions corresponding to the service type and service transition, as shown below. These extensions are not configurable. If the administrator has configured a base number of 1234 and call forward on busy is enabled, the called number is 123403.

**Cisco ATA Service/Transition Extensions**

- Call Waiting Enable—Extension 00
- Call Waiting Disabled—Extension 01
- Call Forward All Enabled—Extension 02
- Call Forward All Disabled—Extension 05
- Call Forward Busy Enabled—Extension 03
- Call Forward Busy Disabled—Extension 05
- Call Forward No Answer Enabled—Extension 04
- Call Forward No Answer Disabled—Extension 05

**Relevant Bit of OpFlags Parameter**

The relevant OpFlags parameter bit is determined by the service that is enabled or disabled, and the identity of the transition. The service types that will prompt a call to the announcement server are call waiting and call forward. Both of these services can undergo an enable transition or a disable transition. For the Cisco ATA to call the server, the applicable OpFlags bit must be set. **Table 4-3** provides a mapping of each relevant OpFlags bit to its corresponding service/transition state or states.

**Table 4-3 Service/Transition and Corresponding OpFlags Bit**

<table>
<thead>
<tr>
<th>Service/Transition</th>
<th>OpFlags Bit</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Waiting Enabled</td>
<td>Bit 18 (mask = 0x40000)</td>
</tr>
<tr>
<td>Call Waiting Disabled</td>
<td>Bit 19 (mask = 0x80000)</td>
</tr>
<tr>
<td>Call Forward All Enabled</td>
<td>Bit 16 (mask = 0x10000)</td>
</tr>
<tr>
<td>Call Forward All Disabled</td>
<td>Bit 17 (mask = 0x20000)</td>
</tr>
<tr>
<td>Call Forward Busy Enabled</td>
<td>Bit 16 (mask = 0x10000)</td>
</tr>
<tr>
<td>Call Forward Busy Disabled</td>
<td>Bit 17 (mask = 0x20000)</td>
</tr>
<tr>
<td>Call Forward No Answer Enabled</td>
<td>Bit 16 (mask = 0x10000)</td>
</tr>
<tr>
<td>Call Forward No Answer Disabled</td>
<td>Bit 17 (mask = 0x20000)</td>
</tr>
</tbody>
</table>
XML Pages of Cisco ATA Information

The Cisco ATA provides XML pages that contain the following Cisco ATA information:

- Current configuration, page 4-22
- Current statistics, page 4-22
- Current service values, page 4-22

**Current configuration**

To obtain the XML configuration page, issue the following command:

```
http://ipaddress/dev.xml
```

where `ipaddress` is the IP address of the Cisco ATA whose configuration you wish to access.

This XML page is only for retrieving the configuration of a Cisco ATA; you cannot change configuration values on this page.

This XML page is password protected. You can enter the password by means of your Web browser, or you can issue the following command:

```
curl -d "ChangeUIPasswd=<passwd>&ChangeUIPasswd=&ChangeUIPasswd=&apply=apply" <ip addr>/dev.xml
```

where `<passwd>` is the Cisco ATA password and `<ip addr>` is the IP address of the Cisco ATA.

**Current statistics**

To obtain this statistics page, issue the following command:

```
http://ipaddress/stats.xml
```

where `ipaddress` is the IP address of the Cisco ATA whose statistics you wish to access.

This XML page is password protected. You can enter the password by means of your Web browser, or you can issue the following command:

```
curl -d "ChangeUIPasswd=<passwd>&ChangeUIPasswd=&ChangeUIPasswd=&apply=apply" <ip addr>/stats.xml
```

where `<passwd>` is the Cisco ATA password and `<ip addr>` is the IP address of the Cisco ATA.

**Current service values**

To obtain this service page, issue the following command:

```
http://ipaddress/service.xml
```

where `ipaddress` is the IP address of the Cisco ATA whose service values you wish to access.

This XML page is password protected. You can enter the password by means of your Web browser, or you can issue the following command:

```
curl -d "ChangeUIPasswd=<passwd>&ChangeUIPasswd=&ChangeUIPasswd=&apply=apply" <ip addr>/service.xml
```

where `<passwd>` is the Cisco ATA password and `<ip addr>` is the IP address of the Cisco ATA.
## Complete Reference Table of all Cisco ATA SIP Services

Table 4-4 is a reference table that lists all configurable features for the Cisco ATA (using SIP), and includes links to the detailed descriptions of the parameters used for configuring these features.

### Table 4-4  Configurable Features and Related Parameters

<table>
<thead>
<tr>
<th>Configurable Feature</th>
<th>Related Parameter</th>
</tr>
</thead>
<tbody>
<tr>
<td>802.1Q packet tagging</td>
<td>VLANSetting, page 5-12</td>
</tr>
<tr>
<td>Anonymity for called third party</td>
<td>ConnectMode, page 5-41—Bit 30</td>
</tr>
<tr>
<td>Anonymous user name support</td>
<td>ConnectMode, page 5-41—Bit 27</td>
</tr>
<tr>
<td>Audio compression and decompression</td>
<td>LBRCodec, page 5-32</td>
</tr>
<tr>
<td>Audio level of FXS ports</td>
<td>FXSInputLevel, page 5-52, FXSOutputLevel, page 5-52</td>
</tr>
<tr>
<td>Backup proxy configuration</td>
<td>AltGk, page 5-14</td>
</tr>
<tr>
<td>Backup proxy timeout</td>
<td>AltGkTimeOut, page 5-15</td>
</tr>
<tr>
<td>Call forward enable/disable</td>
<td>ConnectMode, page 5-41—Bit 17</td>
</tr>
<tr>
<td>Call forwarding—Maximum times allowed</td>
<td>MAXRedirect, page 5-20</td>
</tr>
<tr>
<td>Call commands</td>
<td>CallCmd, page 5-37, Chapter 6, “Call Commands”</td>
</tr>
<tr>
<td>Call features</td>
<td>CallFeatures, page 5-35</td>
</tr>
<tr>
<td>Caller ID format</td>
<td>CallerIdMethod, page 5-49</td>
</tr>
<tr>
<td>Call waiting</td>
<td>SigTimer, page 5-40</td>
</tr>
<tr>
<td>Call-waiting call ring timeout</td>
<td>FeatureTimer, page 5-38</td>
</tr>
<tr>
<td>Call-waiting hang-up alert</td>
<td>ConnectMode, page 5-41—Bit 25</td>
</tr>
<tr>
<td>Call-waiting state specified</td>
<td>ConnectMode, page 5-41</td>
</tr>
<tr>
<td>Cisco Discovery Protocol</td>
<td>OpFlags, page 5-45</td>
</tr>
<tr>
<td>CNG tone detection</td>
<td>AudioMode, page 5-32</td>
</tr>
<tr>
<td>Configuration update interval</td>
<td>CfgInterval, page 5-6</td>
</tr>
<tr>
<td>Debugging and diagnostics</td>
<td>NPrintf, page 5-73, TraceFlags, page 5-73, SyslogIP, page 5-74, SyslogCtrl, page 5-75</td>
</tr>
<tr>
<td>Dial plan commands</td>
<td>DialPlan, page 5-64</td>
</tr>
<tr>
<td>Domain name server</td>
<td>DNS1IP, page 5-10</td>
</tr>
<tr>
<td>DNS hostname lookup</td>
<td>ConnectMode, page 5-41</td>
</tr>
<tr>
<td>DTMF method</td>
<td>AudioMode, page 5-32</td>
</tr>
<tr>
<td>Encryption</td>
<td>EncryptKey, page 5-6, EncryptKeyEx, page 5-7</td>
</tr>
<tr>
<td>Fax CED tone</td>
<td>AudioMode, page 5-32</td>
</tr>
<tr>
<td>Fax mode on a per-call basis</td>
<td>CallFeatures, page 5-35, PaidFeatures, page 5-36</td>
</tr>
<tr>
<td>Fax pass-through</td>
<td>AudioMode, page 5-32, ConnectMode, page 5-41</td>
</tr>
<tr>
<td>Configurable Feature</td>
<td>Related Parameter</td>
</tr>
<tr>
<td>---------------------------------------------------------</td>
<td>-------------------------------</td>
</tr>
<tr>
<td>G.711 codec</td>
<td>AudioMode, page 5-32</td>
</tr>
<tr>
<td>Hook flash</td>
<td>AudioMode, page 5-32, SigTimer, page 5-40</td>
</tr>
<tr>
<td>IDs for phone lines</td>
<td>UID0, page 5-15, UID1, page 5-16</td>
</tr>
<tr>
<td>IP audio and signaling packets—precedence and delay</td>
<td>TOS, page 5-34</td>
</tr>
<tr>
<td>IP-like address in dial plan</td>
<td>IPDialPlan, page 5-72</td>
</tr>
<tr>
<td>Login ID</td>
<td>LoginID0, page 5-17, LoginID1, page 5-18</td>
</tr>
<tr>
<td>Low bit-rate codec</td>
<td>LBRCCodec, page 5-32</td>
</tr>
<tr>
<td>Mixing of tones</td>
<td>ConnectMode, page 5-41</td>
</tr>
<tr>
<td>Network Address Translation (NAT) server—Maintain during session</td>
<td>NatServer, page 5-22</td>
</tr>
<tr>
<td>NSE payload number</td>
<td>ConnectMode, page 5-41</td>
</tr>
<tr>
<td>NTP IP address</td>
<td>NATIP, page 5-21</td>
</tr>
<tr>
<td>On-hook delay</td>
<td>FeatureTimer, page 5-38</td>
</tr>
<tr>
<td>Outbound proxy</td>
<td>SipOutBoundProxy, page 5-21</td>
</tr>
<tr>
<td>Paid features</td>
<td>PaidFeatures, page 5-36</td>
</tr>
<tr>
<td>Passwords for phone lines</td>
<td>PWD0, page 5-16, PWD1, page 5-17</td>
</tr>
<tr>
<td>Polarity</td>
<td>Polarity, page 5-51</td>
</tr>
<tr>
<td>Polarity reversal before and after caller ID signal</td>
<td>CallerIdMethod, page 5-49</td>
</tr>
<tr>
<td>Privacy token support for SIP diversion header</td>
<td>ConnectMode, page 5-41—Bit 27</td>
</tr>
<tr>
<td>Received = tag enable/disable</td>
<td>ConnectMode, page 5-41</td>
</tr>
<tr>
<td>Receiving-audio codec preference</td>
<td>RxCodec, page 5-31</td>
</tr>
<tr>
<td>Redial time if line is busy</td>
<td>FeatureTimer, page 5-38</td>
</tr>
<tr>
<td>Refresh Cisco ATA using Web server</td>
<td>OpFlags, page 5-45</td>
</tr>
<tr>
<td>REGISTER messages</td>
<td>ConnectMode, page 5-41</td>
</tr>
<tr>
<td>Registration removal</td>
<td>ConnectMode, page 5-41</td>
</tr>
<tr>
<td>Reset Cisco ATA using Web server</td>
<td>OpFlags, page 5-45</td>
</tr>
<tr>
<td>Retransmission interval for NAT server</td>
<td>NatTimer, page 5-22</td>
</tr>
<tr>
<td>Retry interval if line is busy</td>
<td>FeatureTimer, page 5-38</td>
</tr>
<tr>
<td>Ringback tone—send to caller</td>
<td>ConnectMode, page 5-41</td>
</tr>
<tr>
<td>Ring-cadence pattern</td>
<td>RingOnOffTime, page 5-64</td>
</tr>
<tr>
<td>RTP media port</td>
<td>MediaPort, page 5-30</td>
</tr>
<tr>
<td>RTP packet size</td>
<td>NumTxFrames, page 5-34</td>
</tr>
<tr>
<td>RTP statistics</td>
<td>TraceFlags, page 5-73</td>
</tr>
</tbody>
</table>
## Table 4-4 Configurable Features and Related Parameters (continued)

<table>
<thead>
<tr>
<th>Configurable Feature</th>
<th>Related Parameter</th>
</tr>
</thead>
<tbody>
<tr>
<td>Secondary domain name server</td>
<td>DNS2IP, page 5-11</td>
</tr>
<tr>
<td>Silence suppression setting</td>
<td>AudioMode, page 5-32</td>
</tr>
<tr>
<td>SIP call return</td>
<td>ConnectMode, page 5-41</td>
</tr>
<tr>
<td>SIP proxy registrar address</td>
<td>GkOrProxy, page 5-13</td>
</tr>
<tr>
<td>SIP retransmission attempts for requests or responses</td>
<td>MsgRetryLimits, page 5-24</td>
</tr>
<tr>
<td>SIP proxy registration renewal</td>
<td>SIPRegInterval, page 5-19</td>
</tr>
<tr>
<td>SIP registration enable/disable</td>
<td>SIPRegOn, page 5-20</td>
</tr>
<tr>
<td>SIP-request listening port</td>
<td>SIPPort, page 5-19</td>
</tr>
<tr>
<td>SIP session-timer settings</td>
<td>SessionTimer, page 5-26, SessionInterval, page 5-28, MinSessionInterval, page 5-28</td>
</tr>
<tr>
<td>Static network router probe</td>
<td>OpFlags, page 5-45</td>
</tr>
<tr>
<td>STUN support</td>
<td>NatTimer, page 5-22, NatServer, page 5-22</td>
</tr>
<tr>
<td>TFTP file—not using internally generated name</td>
<td>OpFlags, page 5-45</td>
</tr>
<tr>
<td>Timing values</td>
<td>SigTimer, page 5-40, FeatureTimer, page 5-38, FeatureTimer2, page 5-39</td>
</tr>
<tr>
<td>Time zone offset</td>
<td>TimeZone, page 5-48</td>
</tr>
<tr>
<td>Tones: BusyTone, CallWaitTone, AlertTone, DialTone, ReorderTone, and RingBackTone parameters</td>
<td>Tone Configuration Parameters, page 5-53</td>
</tr>
<tr>
<td>Tracing</td>
<td>TraceFlags, page 5-73</td>
</tr>
<tr>
<td>Transmitting-audio codec preference</td>
<td>TxCodec, page 5-31</td>
</tr>
<tr>
<td>VLAN encapsulation</td>
<td>OpFlags, page 5-45</td>
</tr>
<tr>
<td>VLAN mode</td>
<td>OpFlags, page 5-45</td>
</tr>
<tr>
<td>WAN address of NAT</td>
<td>NATIP, page 5-21</td>
</tr>
<tr>
<td>Web configuration—disallowing</td>
<td>OpFlags, page 5-45</td>
</tr>
</tbody>
</table>
Parameters and Defaults

This section provides information on the parameters and defaults that you can use to create your own Cisco ATA configuration file. This section also includes the voice configuration menu code for each parameter that has such a code.

Parameters are divided into categories based on their functionality. The following categories of parameters are covered in this section:

- User Interface (UI) Security Parameter, page 5-4
- Parameters for Configuration Method and Encryption, page 5-4
- Network Configuration Parameters, page 5-8
- SIP Configuration Parameters, page 5-13
- Audio Configuration Parameters, page 5-30
- Operational Parameters, page 5-35
- Telephone Configuration Parameters, page 5-49
- Tone Configuration Parameters, page 5-53
- Dial Plan Parameters, page 5-64
- Diagnostic Parameters, page 5-73
- CFGID—Version Parameter for Cisco ATA Configuration File, page 5-76

The following list contains general configuration information:

- Your configuration file must begin with #txt.
- The Cisco ATA uses the following parameter types:
  - Alphanumeric string
  - Array of short integers
  - Boolean (1 or 0)
Configuration Text File Template

This is a listing of the sip_example.txt text file, without its annotations, that comes bundled with the Cisco ATA software.

You can make a copy of this file and use it as a template for creating your own default configuration file or Cisco ATA-specific configuration file. For instructions on how to create these configuration files, see the “Creating Unique and Common Cisco ATA Configuration Files” section on page 3-9.

The sip_example.txt file contains all the Cisco ATA default values. The sections that follow this listing describe all the parameters in this file.

#txt
UIPassword:0
UseTftp:1
TftpURL:0
cfgInterval:3600
EncryptKey:0
EncryptKeyEx:0
upgradecode:0,0x301,0x0400,0x0200,0.0.0.0,69,0,none
upgradelang:0,0x301,0x0400,0x0200,0.0.0.0,69,0,none
Dhcp:1
StaticIp:0
StaticRoute:0
StaticNetMask:0
DNS1IP:0.0.0.0
DNS2IP:0.0.0.0
NTPIP:0.0.0.0
The sections that follow describe these parameters.
Chapter 5  Parameters and Defaults

User Interface (UI) Security Parameter

This parameter type contains one parameter—UIPassword.

UIPassword

**Description**
This parameter controls access to web page or voice configuration menu interface. To set a password, enter a value other than zero. To prompt the user for this password when the user attempts to perform a factory reset or upgrade using the voice configuration menu, see the “OpFlags” section on page 5-45.

To clear a password, change the value to 0.

You cannot recover a forgotten password unless you reset the entire configuration of the Cisco ATA (see the “Resetting the Cisco ATA to Factory Default Values” section on page 3-23).

**Note**
When UIPassword contains letters, you cannot enter the password from the telephone keypad.

**Value Type**
Alphanumeric string

**Range**
Maximum nine characters

**Default**
0

**Voice Configuration Menu Access Code**
7387277

**Related Parameter**
OpFlags, page 5-45

Parameters for Configuration Method and Encryption

This section describes parameters for instructing the Cisco ATA how to locate its TFTP server and how to encrypt its configuration file. These parameters are:

- **UseTFTP**, page 5-5
- **TftpURL**, page 5-5
- **CfgInterval**, page 5-6
- **EncryptKey**, page 5-6
- **EncryptKeyEx**, page 5-7
UseTFTP

Settings
1 — Use the TFTP server for Cisco ATA configuration.
0 — Do not use the TFTP server for Cisco ATA configuration.

Value Type
Boolean

Range
0 or 1

Default
1

Voice Configuration Menu Access Code
305

Related Parameters
- TftpURL, page 5-5
- EncryptKey, page 5-6
- OpFlags, page 5-45 (bits 0 and 3)

TftpURL

Description
Use this parameter to specify the IP address or URL of the TFTP server. This string is needed if the DHCP server does not provide the TFTP server IP address. When the TftpURL parameter is set to a non-zero value, this parameter has priority over the TFTP server IP address supplied by the DHCP server.

Optionally, you can 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, you can specify the URL as 192.168.2.170/ata186 or www.cisco.com/ata186.

Note
From the voice configuration menu, you can only enter the IP address; from the web server, you can enter the actual URL.

Value Type
Alphanumeric string

Range
Maximum number of characters: 31

Default
0
Voice Configuration Menu Access Code
905

Related Parameters
- UseTFTP, page 5-5
- CfgInterval, page 5-6

CfgInterval

Description
Use this parameter to specify the number of seconds between each configuration update. The Cisco ATA will also upgrade its signaling image if it detects that the TFTP server contains an upgraded image.
For example, when using TFTP for configuration, the Cisco ATA contacts TFTP each time the interval expires to get its configuration file.
You can set CfgInterval to a random value to achieve random contact intervals from the Cisco ATA to the TFTP server.

Value Type
Decimal

Range
60 to 4294967295

Default
3600

EncryptKey

Description
This parameter specifies the encryption key that is used to encrypt the Cisco ATA configuration file on the TFTP server.
The cfgfmt tool, which is used to create a Cisco ATA binary configuration file (see the “Using Encryption With the cfgfmt Tool” section on page 3-12), automatically encrypts the binary file when the EncryptKey parameter has a value other than 0. The cfgfmt tool uses the rc4 encryption algorithm.
If this parameter value is set to 0, the Cisco ATA configuration file on the TFTP server is not encrypted.

Note
Cisco recommends using the stronger Cisco ATA encryption method, which requires the use of the EncryptKeyEx parameter. For more information, see the “EncryptKeyEx” section on page 5-7.

For examples on how to upgrade from the EncryptKey parameter to the stronger encryption method that uses the EncryptKeyEx parameter, see the “Examples of Upgrading to Stronger Encryption Key” section on page 3-15.
**Value Type**
Hexadecimal string

**Range**
Maximum number of characters: 8

**Default**
0

**Voice Configuration Menu Access Code**
320

**Related Parameters**
- UseTFTP, page 5-5
- TftpURL, page 5-5
- EncryptKeyEx, page 5-7

---

**EncryptKeyEx**

**Description**
This parameter specifies an encryption key that is stronger than the key specified with the EncryptKey parameter. This stronger key is used to encrypt the Cisco ATA configuration file on the TFTP server.

**Note**
Cisco recommends using the EncryptKeyEx parameter instead of the EncryptKey parameter for the strongest possible encryption of the Cisco ATA configuration file.

When the EncryptKeyEx parameter is set to a non-zero value, the Cisco ATA uses this value as the encryption key and ignores any value that has been set for the EncryptKey parameter. The cfgfmt tool, which is used to create a Cisco ATA binary configuration file (see the “Using Encryption With the cfgfmt Tool” section on page 3-12), automatically encrypts the binary file using the stronger rc4 encryption algorithm.

When EncryptKeyEx is used for encryption, the Cisco ATA searches for the configuration file with the format ata<macaddress>.x on the TFTP server.

If the value of the EncryptKeyEx parameter is 0, then the Cisco ATA uses the value of the EncryptKey parameter for encryption.

**Note**
The cfgfmt tool (version 2.3) program generate an ata<macaddress>.x file in addition to an ata<macaddress> file if the EncryptKeyEx parameter is specified. You should place both such configuration files on the TFTP server.

For examples on how to upgrade from the EncryptKey parameter to the stronger encryption method that uses the EncryptKeyEx parameter, see the “Examples of Upgrading to Stronger Encryption Key” section on page 3-15.
Value Type
Hexadecimal string of the form:
Rc4PasswdInHex/macinHex_12
- rc4KeyInHex_n is a hexadecimal string of one to 64 characters.
- /macInHex_12 is the optional extension consisting of a forward slash (/) followed by the six-byte MAC address of the Cisco ATA to which the configuration file will be downloaded.

Range
Maximum number of characters: 64

Default
0

Voice Configuration Menu Access Code
Not applicable for this parameter.

Related Parameters
- UseTFTP, page 5-5
- TftpURL, page 5-5
- EncryptKey, page 5-6

Network Configuration Parameters
This section includes the parameters for enabling or disabling the use of a DHCP server to obtain IP address information, and parameters that you need to statically configure if you disable DHCP:

- DHCP, page 5-8
- StaticIp, page 5-9
- StaticRoute, page 5-9
- StaticNetMask, page 5-10
- DNS1IP, page 5-10
- DNS2IP, page 5-11
- NTPIP, page 5-11
- AltNTPIP, page 5-12
- VLANSetting, page 5-12

DHCP
Description
A DHCP server can be used to automatically set the Cisco ATA IP address, the network route IP address, the subnet mask, DNS, NTP, TFTP, and other parameters.

- 1—Enable DHCP
- 0—Disable DHCP
Network Configuration Parameters

Value Type
Boolean

Range
0 or 1

Default
1

Voice Configuration Menu Access Code
20

Related Parameters
- StaticIp, page 5-9
- StaticRoute, page 5-9
- StaticNetMask, page 5-10
- OpFlags, page 5-45 (bits 3 and 11)

**StaticIp**

Description
Use this parameter to statically assign the Cisco ATA IP address if the DHCP parameter is set to 0.

Value Type
IP address

Default
0.0.0.0

Voice Configuration Menu Access Code
1

Related Parameters
- DHCP, page 5-8
- StaticRoute, page 5-9
- StaticNetMask, page 5-10

**StaticRoute**

Description
Use this parameter to statically assign the Cisco ATA route if the DHCP parameter is set to 0.

Value Type
IP address
**Network Configuration Parameters**

**Default**

0.0.0.0

**Voice Configuration Menu Access Code**

2

**Related Parameters**

- DHCP, page 5-8
- StaticIp, page 5-9
- StaticNetMask, page 5-10

---

**StaticNetMask**

**Description**

Use this parameter to statically assign the Cisco ATA subnet mask if the DHCP parameter is set to 0

**Value Type**

IP address

**Default**

255.255.255.0

**Voice Configuration Menu Access Code**

10

**Related Parameters**

- DHCP, page 5-8
- StaticIp, page 5-9
- StaticRoute, page 5-9

---

**DNS1IP**

**Description**

This parameter is for setting the primary domain name server (DNS) IP address, if the DHCP server does not provide one. If DHCP provides DNS1IP (and if it is non-zero), this parameter overwrites the DHCP-supplied value. You *cannot* specify a port parameter. The Cisco ATA uses the default DNS port only.

**Value Type**

IP address

**Default**

0.0.0.0
**DNS2IP**

**Description**
This parameter is for setting the secondary domain name server (DNS) IP address, if the DHCP server does not provide one. If DHCP provides DNS2IP (if it is non-zero), this parameter overwrites the DHCP-supplied value. You cannot specify a port parameter. The Cisco ATA uses the default DNS port only.

**Value Type**
IP address

**Default**
0.0.0.0

**Voice Configuration Menu Access Code**
916

**Related Parameter**
DHCP, page 5-8

**NTPIP**

**Description**
This parameter is the NTP IP address, required if DHCP server does not provide one.
The Cisco ATA requires an NTP Server from which to obtain Coordinated Universal Time (UTC) to time-stamp incoming calls (H.323 and SIP) to drive an external Caller-ID device.
DHCP may also supply a NTP server. If NTPIP is specified, it overwrites the value supplied by DHCP. NTPIP is ignored if its value is 0 or 0.0.0.0.
The user must not specify a port parameter. The Cisco ATA uses the default NTP port only.

**Value Type**
IP address

**Default**
0.0.0.0

**Voice Configuration Menu Access Code**
141
AltNTPIP

Description
This parameter is the alternate NTP IP address, if you want redundancy. You can set this parameter to 0 or point to the same NTP IP if only one NTP server exists.

Value Type
IP address

Default
0.0.0.0

Voice Configuration Menu Access Code
142

VLANSetting

Description
This parameter is for specifying VLAN-related settings.

Bitmap definitions are as follows for the VLANSetting parameter:
- Bits 0-2—Specify VLAN Class of Service (CoS) bit value (802.1P priority) for signaling IP packets.
- Bits 3-5—Specify VLAN CoS bit value (802.1P priority) for voice IP packets.
- Bits 6-17—Reserved.
- Bits 18-29—User-specified 802.1Q VLAN ID.
- Bits 30-31—Reserved.

Value Type
Bitmap

Default
0x0000002b

Voice Configuration Menu Access Code
324

Related Parameter
OpFlags, page 5-45


SIP Configuration Parameters

This section describes the following parameters, which include SIP port and SIP proxy configuration parameters:

- GkOrProxy, page 5-13
- AltGk, page 5-14
- AltGkTimeOut, page 5-15
- UID0, page 5-15
- PWD0, page 5-16
- UID1, page 5-16
- PWD1, page 5-17
- LoginID0, page 5-17
- LoginID1, page 5-18
- UseLoginID, page 5-18
- SIPPort, page 5-19
- SIPRegInterval, page 5-19
- SIPRegOn, page 5-20
- MAXRedirect, page 5-20
- SipOutBoundProxy, page 5-21
- NATIP, page 5-21
- NatServer, page 5-22
- NatTimer, page 5-22
- MsgRetryLimits, page 5-24
- SessionTimer, page 5-26
- SessionInterval, page 5-28
- MinSessionInterval, page 5-28
- DisplayName0, page 5-29
- DisplayName1, page 5-29

GkOrProxy

Description
This parameter is the proxy address or registrar address.

For a 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 an IP address, a ‘.’ or ‘:’ can be used to delimit a port parameter. For a URL, a ‘:’ must be used to indicate a port.

The default port is port number 5060.
Note
If the SIP proxy server and registration server reside on separate hardware, enter the SIP registration server address in this field.

If the hostname specified in GkOrProxy is not an IP address (but is, for example, a domain name), the Cisco ATA performs a DNS A-record query on the domain name. To force the Cisco ATA to perform a DNS SRV query first, you must set ConnectMode parameter bit 29 to a value of 1. (See the “ConnectMode” section on page 5-41.) If the query results in a zero DNS SRV entry, then the Cisco ATA performs a DNS A-record query on the hostname.

If the SRV lookup returns two hosts, they become primary and backup proxies according to their priority (as specified in the DNS SRV RFC), and the hostname specified in the AltGk parameter is ignored.

If the SRV lookup returns only one host, this host is the primary proxy, and the hostname specified in the AltGk parameter is the backup proxy.

Value Type
Alphanumeric string

Range
Maximum number of characters: 31

Default
0—Disables proxy registration and proxy-routed calls.

In this case, you can make direct IP calls by dialing the user-id@IP:port of the callee, where user-id must be a numeric value, '@' is dialed as "**", and '.' and ':' are dialed as a '*'..

The following list shows some examples of direct SIP IP dialing:
- 1234**192*168*1*10*5060
- 102*210**9*101*5061
- 4084281002**100*123*89*10

Voice Configuration Menu Access Code
5

AltGk

Description
You have the option of using this parameter to specify a backup proxy. However, if a DNS SRV performed on the GkOrProxy parameter returns more than one host, the AltGk parameter is ignored. For more information, refer to the description and syntax examples in the “GkOrProxy” section on page 5-13.

Value Type
Alphanumeric string

Range
Maximum number of characters: 31
AltGkTimeOut

**Description**
You can use this parameter to specify the timeout in seconds before the Cisco ATA fails back to the primary proxy server from the backup proxy server. Re-registration does not occur until the current registration period expires.

**Value Type**
Integer

**Default**
0—The Cisco ATA continues to use the backup proxy server until it fails before attempting to fail back to the primary proxy server.

**Range**
30 to 4294967295 seconds

UID0

**Description**
This parameter is the User ID (for example, the phone number) for the Phone 1 port. If the value is set to zero, the port will be disabled and no dial tone will sound.

Use this parameter for registration and authentication. If a proxy server requires separate registration and authentication IDs, use this parameter to specify the registration ID only. You can use the LoginID0 parameter specify the authentication ID.

**Value Type**
Alphanumeric string

**Range**
Maximum number of characters: 31

**Default**
0
Voice Configuration Menu Access Code
3

Related Parameter
LoginID0, page 5-17

PWD0

Description
This parameter is the password for the Phone 1 port.

Value Type
Alphanumeric string

Range
Maximum number of characters: 31

Default
0

Voice Configuration Menu Access Code
4

UID1

Description
This parameter is the User ID (for example, the phone number) for the Phone 2 port. If the value is set to zero, the port will be disabled and no dial tone will sound.

Use this parameter for registration and authentication. If a proxy server requires separate registration and authentication IDs, use this parameter to specify the registration ID only. You can use the LoginID1 parameter specify the authentication ID.

Value Type
Alphanumeric string

Range
Maximum number of characters: 31

Default
0

Voice Configuration Menu Access Code
13

Related Parameter
LoginID1, page 5-18
PWD1

Description
This parameter is the password for the Phone 2 port.

Value Type
Alphanumeric string

Range
Maximum number of characters: 31

Default
0

Voice Configuration Menu Access Code
14

LoginID0

Description
If a proxy server requires separate registration and authentication IDs, use this parameter to specify the authentication ID for the Phone1 port of the Cisco ATA.

Note
UID0 is used for authentication if UseLoginID is 0.

Value Type
Alphanumeric string

Range
Maximum number of characters: 51

Default
0

Voice Configuration Menu Access Code
46

Related Parameters
- UID0, page 5-15
- UseLoginID, page 5-18
LoginID

Description
If a proxy server requires separate registration and authentication IDs, use this parameter to specify the authentication ID for the Phone2 port of the Cisco ATA.

Note
UID1 is used for authentication if UseLoginID is 0.

Value Type
Alphanumeric string

Range
Maximum number of characters: 51

Default
0

Voice Configuration Menu Access Code
47

Related Parameters
- UID1, page 5-16
- UseLoginID, page 5-18

UseLoginID

Description
0—Use UID0 and UID1 as the authentication ID.
1—Use LoginID0 and LoginID1 as the authentication ID.

Value Type
Boolean

Range
0 or 1

Default
0

Voice Configuration Menu Access Code
93
SIPPort

**Description**
This parameter is used to configure the port through which the Cisco ATA listens for incoming SIP requests and sends outgoing SIP requests.

**Value Type**
Integer

**Range**
1 to 65535

**Default**
5060

**Voice Configuration Menu Access Code**
201

SIPRegInterval

**Description**
Use this parameter to configure the number of seconds between Cisco ATA registration renewal with the SIP proxy server. The Cisco ATA renews the registration at some percentage of time earlier than the specified interval to prevent a registration from expiring.

**Value Type**
Integer

**Range**
1 to 86400

**Default**
3600

**Voice Configuration Menu Access Code**
203
### SIPRegOn

**Description**

0—Disable SIP registration.

1—Enable SIP registration. When this flag is enabled, the Cisco ATA registers with the SIP Proxy Server that is specified in the GkorProxy parameter. The Cisco ATA also registers with the interval that is specified in the SIPRegInterval parameter.

**Value Type**

Boolean

**Range**

0 or 1

**Default**

0

**Voice Configuration Menu Access Code**

204

### MAXRedirect

**Description**

This parameter specifies the maximum number of times that a called number is allowed to forward the call to another number.

**Value Type**

Integer

**Range**

0 to 10

**Default**

5

**Voice Configuration Menu Access Code**

205
SipOutBoundProxy

**Description**
The SIP Outbound Proxy Server is a SIP proxy server which can be different from the Registration Proxy Server (specified in the GkOrProxy parameter) and to which all outgoing SIP requests are sent. Outgoing SIP responses are not affected by this out-bound-proxy and are still sent according to the VIA header and source address of the incoming SIP requests.

If the outgoing SIP request has a ROUTE header, the first route in the header is removed if it resolves to the same IP address as the out-bound-proxy. This process guards against the case when the out-bound-proxy also inserts its IP address into the RECORD-ROUTE header.

The OutBoundProxy parameter 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 period (.) or colon (:) can be used to delimit a port parameter. For a URL, a colon (:) must be used to indicate a port. If no port parameter is specified, the port 5060 is assumed.

**Value Type**
Alphanumeric string

**Range**
Maximum number of characters: 31

**Default**
0

**Voice Configuration Menu Access Code**
206

NATIP

**Description**
This is the WAN address of the attached router/NAT; currently only used to support SIP behind a NAT.

**Value Type**
IP address

**Default**
0.0.0.0

**Voice Configuration Menu Access Code**
200
**NatServer**

**Description**
This parameter allows you to specify the server to which dummy, single-byte UDP packets are sent to maintain a NAT translation for a session.

The parameter value can be an IP address with or without a port parameter (such as 123.123.110.45 or 123.123.110.45:5060), or a URL(such as nat.cisco.com or dummy.cisco.com:5061).

**Syntax**

```
<IP_address>[.:<Port>] | <URL>[:<Port>]
```

The following syntax rules apply:
- For an IP address, a dot (.) or colon (:) can be used to delimit a port parameter.
- For a URL, a colon (:) must be used as a delimiter.
- If the port parameter is not specified, the default port 5060 is assumed.
- A value of 0 indicates that a server is not available.

**Value Type**
IP address or FQDN format

**Range**
Maximum number of characters: 47

**Default**
5060 is the default port if no port is specified.

**Voice Configuration Menu Access Code**
207

**NatTimer**

**Description**
The NatTimer parameter provides the following configuration:
- Interval of Keep-Alive packets
- STUN mode to use
- Destination of keep-alive packets.

See Table 5-1 for definitions of each bit.

**Value Type**
Bitmap

**Default**
0x00000000
Table 5-1 NatTimer Parameter Bit Definitions

<table>
<thead>
<tr>
<th>Bit Number</th>
<th>Definition</th>
</tr>
</thead>
</table>
| 0-11       | Use these bits to specify the interval in seconds for sending keep-alive packets from the SIP port and RTP port. Keep-alive packets are sent in the following manner:  
- The packets are sent constantly for the SIP port once the keep-alive interval is reached.  
- The packets are sent only during a call for the media ports once the keep-alive interval is reached.  
Range: 0 to 2047 seconds  
Default: 0 (no keep-alive packets are sent). |
| 12-15      | Reserved. These bits must be set to 0. |
| 16-17      | Use these bits with one of the following values to specify NAT traversal mode:  
- 0—Send keep-alive packets only  
The Cisco ATA sends keep-alive packets only and does not perform any discovery or substitution of the NAT WAN IP address or port in SIP messages.  
- 1—STUN  
The Cisco ATA obtains IP address and port mappings from the STUN server specified in the NatServer parameter and substitutes these mappings into SIP messages. The Cisco ATA automatically obtains port mappings, as needed, for the following parameters:  
  - SIPPort  
  - MediaPort  
  - MediaPort + 2  
  - MediaPort + 4  
- 2—NATIP auto-mapping  
The Cisco ATA obtains only the IP address mapping from the STUN server specified in the NatServer parameter and substitutes the mapping into SIP messages. With this mode, you must manually map the Cisco ATA port to an external port.  
- 3—Reserved. Do not use this value. |
| 18         | Use this bit to set the destination of keep-alive UDP packets that are sent from the SIP port:  
- 0—Keep-alive UDP packets are sent to the server specified in the NatServer parameter. This setting is normally used when only keep-alive packets are sent (see bits 16-17).  
- 1—Keep-alive UDP packets are sent to the server specified in the GkOrProxy parameter. This setting is normally used when STUN mode is selected (see bits 16-17). |
| Related Parameters |  
- NatServer, page 5-22  
- GkOrProxy, page 5-13 |
| 19         | Reserved. |
Chapter 5  Parameters and Defaults

SIP Configuration Parameters

Table 5-1  NatTimer Parameter Bit Definitions

<table>
<thead>
<tr>
<th>Bit Number</th>
<th>Definition</th>
</tr>
</thead>
</table>
| 20         | 0—This setting keeps the Cisco ATA signaling port (SIPPort parameter value) bound to the NAT-assigned external port during a call. This prevents the binding from expiring during a period of inactivity when no UDP packets pass through the mapped port.  
            | 1—If this setting is selected, a NAT router may remove the signaling port binding after a period of inactivity.  
            | Related Parameter  
            | SIPPort, page 5-19                                                                                                                                  |
| 21         | 0—This setting keeps the first Cisco ATA media port (MediaPort parameter value) bound to the NAT-assigned external port during a call. This prevents the binding from expiring during a period of silence when no UDP packets pass through the mapped port.  
            | 1—If this setting is selected, a NAT router may remove the media port binding after a period of inactivity.  
            | Related Parameter  
            | MediaPort, page 5-30                                                                                                                                 |
| 22         | 0—This setting keeps the second Cisco ATA media port (MediaPort parameter value + 4) bound to the NAT-assigned external port during a call. This prevents the binding from expiring during a period of silence when no UDP packets pass through the mapped port.  
            | 1—If this setting is selected, a NAT router may remove the media port binding after a period of inactivity.  
            | Related Parameter  
            | MediaPort, page 5-30                                                                                                                                 |
| 23-31      | Reserved; must be set to 0.                                                                                                                          |

MsgRetryLimits

Description
This parameter is a bitmap that allows you to specify the number of times that the Cisco ATA retransmits various SIP requests to the current proxy as well as the number of times that the Cisco ATA sends responses to specific requests from the SIP user agent.

When the Cisco ATA sends a SIP message to the remote SIP user agent, the message does not always reach its destination for various reasons. When this occurs, the Cisco ATA re-sends the same message a specified number of times before timing out.

The number of retries is configurable for the following Cisco ATA SIP requests:

- REGISTER
- INVITE
- BYE
The number of retries is also configurable for the following Cisco ATA responses to requests from the SIP user agent:

- Cisco ATA responses to an INVITE request
- Cisco ATA final responses (200-to-600 class responses) to an OPTIONS request

See Table 5-2 for definitions of each bit.

**Value Type**
Bitmap

**Default**
0x00000000

**Voice Configuration Menu Access Code**
360

### Table 5-2  *MsgRetryLimits Parameter Bit Definitions*

<table>
<thead>
<tr>
<th>Bit Number</th>
<th>Definition</th>
</tr>
</thead>
</table>
| 0-3        | Number of times to retransmit the following Cisco ATA SIP requests or responses to the SIP user agent:  
             • NOTIFY request  
             • Response to an OPTIONS request  
             • Final response to an INVITE request  
             Range: 0-15  
             Default: 0. This means the following numbers of retransmission attempts are used:  
             • NOTIFY request: 6  
             • OPTIONS response: 10  
             • INVITE final response: 7  
| 4-7        | Number of times to retransmit REGISTER request.  
             Default: 0 (Number of attempts is 10.)  
| 8-11       | Number of times to retransmit INVITE request.  
             Default: 0 (Number of attempts is five.)  
| 12-15      | Number of times to retransmit BYE request.  
             Default: 0 (Number of attempts is four.)  
| 16-19      | Number of times to retransmit CANCEL request.  
             Default: 0 (Number of attempts is four.)  
| 20-23      | Number of times to retransmit REFER request.  
             Default: 0 (Number of attempts is five.)  
| 24-31      | Reserved. The value of these bits must be 0.  

SessionTimer

Description
Use this parameter to configure various options of the SIP session timer. See Table 5-3 for bit definitions.

Value Type
Bitmap

Default
0x00000000

Voice Configuration Menu Access Code
362

Examples
- SessionTimer: 0x00000001
  With this setting, the Cisco ATA accepts requests from the far end for a session timer but never initiates requests for a session timer.
  By default, the Cisco ATA will not be the session refresher except if one of the following conditions is true:
    - If the caller has already specified the Cisco ATA as the refresher in the initial call setup request.
    - If the party that the Cisco ATA called has specified that the Cisco ATA be the refresher.

- SessionTimer: 0x00000003
  With this setting, the Cisco ATA is always the requester of a session timer, providing the following condition is true:
    - Session timing is supported by the far end regardless of whether the far end or the Cisco ATA is the call initiator.
  By default, the Cisco ATA will not be the session refresher except if one of the following conditions is true:
    - If the caller has already specified the Cisco ATA as the refresher in the initial call setup request.
    - If the party that the Cisco ATA called has specified that the Cisco ATA be the refresher.

- SessionTimer: 0x00000005
  With this setting, the Cisco ATA only accept requests from the far end for a session timer but never initiates requests for a session timer.
  By default, the Cisco ATA will not be the session refresher except if one of the following conditions is true:
    - The caller has already specified itself to as the session refresher in the initial call setup request.
    - The party that the Cisco ATA called has specified itself as the session refresher.
• SessionTimer: 0x00000007

With this setting, the Cisco ATA is always the requester of a session timer, providing the following condition is true:

− Session timing is supported by the far end regardless of whether the far end or the Cisco ATA is the call initiator.

In this mode, the Cisco ATA is designated as the session refresher except if one of the following conditions is true:

− The caller has already specified itself as the session refresher in the initial call-setup request.
− The party that the Cisco ATA called has specified itself as the session refresher.

Related Parameters

- SessionInterval, page 5-28
- MinSessionInterval, page 5-28

Table 5-3  SessionTimer Parameter Bit Definitions

<table>
<thead>
<tr>
<th>Bit Number</th>
<th>Definition</th>
</tr>
</thead>
<tbody>
<tr>
<td>0</td>
<td>Set this bit to 1 to enable the SIP session-timer feature.</td>
</tr>
<tr>
<td>1</td>
<td>Set this bit to 1 to request that a SIP session timer be in effect on every call if the far endpoint also supports this feature.</td>
</tr>
</tbody>
</table>
| 2          | Use this bit to select the party responsible for initiating session refreshes if the caller has not specified the party. This field is applicable only when the Cisco ATA is the callee. Set the bit value as follows:  
  • 0—The caller will perform session refreshes.  
  • 1—The callee will perform session refreshes. |
| 3          | Reserved.                                                                   |
| 4          | Set this bit to 1 to enable the collection of session debug messages.       |
| 5-31       | Reserved.                                                                   |

Note

If the caller specifies which party will perform session refreshes (by means of the SIP Session-Expires field of the initial INVITE request), that selection takes precedence over the configured value of this bit.
SessionInterval

Description
Use this parameter to configure the interval, in number of seconds, between SIP session refreshes.

Conditions for Usage
One of the following conditions must exist for this parameter to be used:

- The Cisco ATA is the caller requesting the use of a session timer.
  In this case, bits 0 and 1 of the SessionTimer parameter must be enabled.
- The Cisco ATA is the callee requesting the use of a session timer when none has been requested by the calling side.
  - The initial SIP INVITE request did not contain a Session-Expires header or a Min-SE header (for minimum session interval); however the INVITE request did include the Supported:timer line to indicate that this feature is supported.
  - Bits 0 and 1 of the SessionTimer parameter must be enabled.

Value Type
Integer

Range
60 to 86400 seconds

Default
1800 seconds (one half hour)

Voice Configuration Menu Access Code
363

Related Parameters
- SessionTimer, page 5-26
- MinSessionInterval, page 5-28

MinSessionInterval

Description
This parameter is used only if the Cisco ATA is the callee and if the caller has requested the use of a session timer. The Cisco ATA then uses this parameter to validate the requested refresh interval.

Use this parameter to configure the minimum required interval, in number of seconds, between session refreshes.

If the Session-Expires value in the INVITE request sent to the Cisco ATA is smaller than the value of the MinSessionInterval parameter, the Cisco ATA will reject the INVITE request and return a 422 Session Interval Too Small message to the caller.
If the Session-Expires value in the received INVITE request is greater than or equal to the Cisco ATA MinSessionInterval, the Cisco ATA will accept this interval by sending a 200 OK message to the caller with the refresh interval listed in the Session-Expires field and the Required:timer line included in the Cisco ATA response.

**Value Type**

Integer

**Range**

60 to 86400 seconds

**Default**

1800

**DisplayName0**

**Description**

For caller ID purposes, you can configure a name to correspond to the phone number of the Cisco ATA Phone 1 input port. This name will be displayed at the remote endpoint when a call originates from this Cisco ATA.

**Value Type**

Alphanumeric string

**Range**

Maximum 31 characters

**Default**

0. This means that a display name will not be included in any SIP messages.

**DisplayName1**

**Description**

For caller ID purposes, you can configure a name to correspond to the phone number of the Cisco ATA Phone 2 input port. This name will be displayed at the remote endpoint when a call originates from this Cisco ATA.
### Audio Configuration Parameters

This section describes the following audio parameters, which allow you to configure such items as codecs and silence suppression:

- MediaPort, page 5-30
- RxCodec, page 5-31
- TxCodec, page 5-31
- LBRCCodec, page 5-32
- AudioMode, page 5-32
- NumTxFrames, page 5-34
- TOS, page 5-34

#### MediaPort

**Description**

Use this parameter to specify the base port where the Cisco ATA transmits and receives RTP media. This parameter **must** be an even number. Each connection uses the next available even-numbered port for RTP.

**Value Type**

Integer

**Range**

1 to 65535

**Default**

16384
Voice Configuration Menu Access Code
202

Related Parameters
• TOS, page 5-34
• VLANSetting, page 5-12

RxCodec

Description
Use this parameter to specify receiving-audio codec preference. The following values are valid:
• 0—G.723 (can be selected only if LBRCODEC is set to 0)
• 1—G.711A-law
• 2—G.711µ-law
• 3—G.729a (can be selected only if LBRCODEC is set to 3)
• 6—G.726-32kbps

Value Type
Integer

Range
0-3, 6

Default
2

Voice Configuration Menu Access Code
36

TxCodec

Description
Use this parameter to specify the transmitting-audio codec preference. The following values are valid:
• 0—G.723 (can be selected only if LBRCODEC is set to 0)
• 1—G.711A-law
• 2—G.711µ-law
• 3—G.729A (can be selected only if LBRCODEC is set to 3)
• 6—G.726-32kbps

Value Type
Integer
Audio Configuration Parameters

Chapter 5  Parameters and Defaults

Range
0-3, 6

Default
2

Voice Configuration Menu Access Code
37

LBRCodec

Description
This parameter allows you to specify which low-bit-rate codecs are available. The Cisco ATA is capable of supporting two G.723.1 connections, two G.726 connections, or one G.729 connection.

When G.723.1 is selected as the low-bit-rate codec, each FXS port is allocated with one G.723.1 connection.

When G.729 is selected, only one FXS port is capable of operating with the G.729 codec. The allocation of the G.729 resource to the FXS port is dynamic. The G.729 resource, if available, is allocated to an FXS port when a call is initiated or received; the resource is released when a call is completed.

When G.726 is selected, the Cisco ATA can support as many as two active simultaneous G.726 connections. The allocation of the G.726 codec is dynamic. Two G.726 connections can be used by one Cisco ATA voice port for conferencing or shared by two Cisco ATA voice ports.

The following values are valid:

- 0—Select G.723.1 as the low-bit-rate codec.
- 3—Select either G.729 as the low-bit-rate codec.
- 6—Select G.726-32kbps as the low-bit-rate codec.

Value Type
Integer

Range
0, 3 or 6

Default
0

Voice Configuration Menu Access Code
300

AudioMode

Description
This parameter represents the audio operating mode. The lower 16 bits are for the Phone 1 port, and the upper 16 bits are for the Phone 2 port. Table 5-4 on page 5-33 provides definitions for each bit.
**Value Type**
Bitmap

**Default**
0x00150015

**Voice Configuration Menu Access Code**
312

**Related Parameter**
ConnectMode, page 5-41

### Table 5-4  AudioMode Parameter Bit Definitions

<table>
<thead>
<tr>
<th>Bit Number</th>
<th>Definition</th>
</tr>
</thead>
</table>
| 0 and 16   | 0/1—Disable/enable G.711 silence suppression.  
|            | Default: 1 |
| 1 and 17   | 0—Enable selected low-bit-rate codec in addition to G.711.  
|            | 1—Enable G.711 only.  
|            | Default: 0 |
| 2 and 18   | 0/1—Disable/enable fax CED tone detection.  
|            | Default: 1 |
| 3 and 19   | Reserved. |
| 4-5 and 20-21: DtmfMethod | 0—Always in-band (send and receive, do not send SDP info)  
|            | 1—By negotiation (send SDP info, enable receive, decode others’ SDP information, send depends on others’ SDP information)  
|            | 2—Always out-of-band (send SDP info, enable receive, decode others’ SDP information, always send).  
|            | 3—Reserved. |
| 6-15 and 22-31 | Reserved. |
NumTxFrames

**Description**
Use this parameter to select the default RTP packet side in number of frames per packet. The Cisco ATA default frame sizes are as follows:
- G.711 and G.729—10 ms
- G.723.1—30 ms
For example, to receive 20 ms of G.729 packets, set the parameter to 2.

**Value Type**
Integer

**Range**
1-6

**Default**
2

**Voice Configuration Menu Access Code**
35

TOS

**Description**
This parameter allows you to configure Type of Service (ToS) bits by specifying the precedence and delay of audio and signaling IP packets, as follows:
- Bits 0-7—These bits are for the ToS value for voice data packets.
  - Range: 0-255
  - Default: 184
- Bits 8-15—These bits are for the ToS value for signaling-data packets
  - Range: 0-255
  - Default: 168
- Bits 16-31—Reserved.

**Value Type**
Bitmap

**Default**
0x0000A8B8

**Voice Configuration Menu Access Code**
255
Operational Parameters

This section describes the following parameters, which allow you to configure such items as call features and various timeout values:

- CallFeatures, page 5-35
- PaidFeatures, page 5-36
- CallCmd, page 5-37
- FeatureTimer, page 5-38
- FeatureTimer2, page 5-39
- SigTimer, page 5-40
- ConnectMode, page 5-41
- OpFlags, page 5-45
- TimeZone, page 5-48

CallFeatures

**Description**

Disable/enable CallFeatures by setting each corresponding bit to 0 or 1.

The lower 16 bits are for the **Phone 1** port, and the upper 16 bits are for the **Phone 2** port. Table 5-5 provides definitions of each bit.

**Note**

The subscribed features that can be permanently disabled by the user are Caller Line Identification Restriction_Caller Line Identification Presentation (CLIP_CLIR), call waiting and Fax mode. A subscribed service enable/disabled by the user can be disabled/enabled dynamically on a per-call basis.

**Value Type**

Bitmap

**Default**

0xffffffff

**Voice Configuration Menu Access Code**

314
Table 5-5  CallFeatures Parameter Bit Definitions

<table>
<thead>
<tr>
<th>Bit Number</th>
<th>Definition</th>
</tr>
</thead>
<tbody>
<tr>
<td>0 and 16</td>
<td>Forward unconditionally</td>
</tr>
<tr>
<td>1 and 17</td>
<td>Forward on busy</td>
</tr>
<tr>
<td>2 and 18</td>
<td>Forward on no answer</td>
</tr>
<tr>
<td>3 and 19</td>
<td>Caller Line Identification Restriction (CLIR)—0; Caller Line Identification Presentation (CLIP)—1</td>
</tr>
<tr>
<td>4 and 20</td>
<td>Call waiting</td>
</tr>
<tr>
<td>5 and 21</td>
<td>three-way calling</td>
</tr>
<tr>
<td>6 and 22</td>
<td>Blind transfer</td>
</tr>
<tr>
<td>7 and 23</td>
<td>Transfer with consultation. This service allows the user to transfer the remote party to a different number by first calling that number and consulting with the callee.</td>
</tr>
<tr>
<td>8 and 24</td>
<td>Caller ID. This service enables the Cisco ATA 186 to generate a Caller ID signal to drive a Caller ID display device attached to the FXS line.</td>
</tr>
<tr>
<td>9 and 25</td>
<td>Call return</td>
</tr>
<tr>
<td>10 and 26</td>
<td>Message waiting indication</td>
</tr>
<tr>
<td>11 and 27</td>
<td>Call Waiting Caller ID. This is available only if the Method bit in CallerIdMethod is set to Bellcore (FSK).</td>
</tr>
<tr>
<td>12-14 and 28-30</td>
<td>Reserved.</td>
</tr>
<tr>
<td>15 and 31</td>
<td>Fax mode. This service allows the user to set the Cisco ATA to Fax mode on a per-call basis. For Fax mode, use the following settings:</td>
</tr>
<tr>
<td></td>
<td>• G711 codec only</td>
</tr>
<tr>
<td></td>
<td>• No silence suppression</td>
</tr>
<tr>
<td></td>
<td>• No FAX tone detection</td>
</tr>
</tbody>
</table>

PaidFeatures

Description
Unsubscribe/subscribe to CallFeatures by setting each corresponding bit to either 0 or 1. The lower 16 bits are for the Phone 1 port, and the upper 16 bits are for the Phone 2 port. Table 5-6 provides definitions of each bit.

Value Type
Bitmap

Default
0xffffffff

Voice Configuration Menu Access Code
315
Table 5-6  PaidFeatures Parameter Bit Definitions

<table>
<thead>
<tr>
<th>Bit Number</th>
<th>Definition</th>
</tr>
</thead>
<tbody>
<tr>
<td>0 and 16</td>
<td>Forward unconditionally</td>
</tr>
<tr>
<td>1 and 17</td>
<td>Forward on busy</td>
</tr>
<tr>
<td>2 and 18</td>
<td>Forward on no answer</td>
</tr>
<tr>
<td>3 and 19</td>
<td>Caller Line Identification Restriction_Caller Line Identification Presentation (CLIP_CLIR)</td>
</tr>
<tr>
<td>4 and 20</td>
<td>Call waiting</td>
</tr>
<tr>
<td>5 and 21</td>
<td>three-way calling</td>
</tr>
<tr>
<td>6 and 22</td>
<td>Blind transfer</td>
</tr>
<tr>
<td>7 and 23</td>
<td>Transfer with consultation. This service allows the user to transfer the remote party to a different number by first calling that number and consulting with the callee.</td>
</tr>
<tr>
<td>8 and 24</td>
<td>Caller ID. This service enables the Cisco ATA 186 to generate a Caller ID signal to drive a Caller ID display device attached to the FXS line.</td>
</tr>
<tr>
<td>9 and 25</td>
<td>Call return</td>
</tr>
<tr>
<td>10 and 26</td>
<td>Message waiting indication</td>
</tr>
<tr>
<td>11 and 27</td>
<td>Call Waiting Caller ID. This is available only if the Method bit in CallerIdMethod is set to Bellcore (FSK).</td>
</tr>
<tr>
<td>12-14 and 28-30</td>
<td>Reserved.</td>
</tr>
</tbody>
</table>
| 15         | Fax mode. This service allows the user to set the Cisco ATA to Fax mode on a per-call basis. For Fax mode, use the following settings:  
  • G.711 codec only  
  • No silence suppression  
  • No FAX tone detection |

CallCmd

Description
Command table that controls call commands such as turning on/off caller ID.
For detailed information on the CallCmd parameter, see Chapter 6, “Call Commands.”

Value Type
Alphanumeric string

Range
Maximum of 248 characters
Default
US command table:

CallCmd:Af;AH:BS:NA;CS:NA:Df;EB:FF;EP:KF:EFh;HH: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;

Voice Configuration Menu Access Code
930

FeatureTimer

Description
This parameter provides configurable timing values for various telephone features, as described below:

- Bits 0-3—Maximum time to spend redialing if line is busy
  - Range: 0 - 15
  - Factor: five-minute increments
  - Values: 0 - 75 minutes
  - Default: 0 (equals 30 minutes)
- Bits 4-7—Retry interval if line is busy
  - Range: 0 - 15
  - Factor: 15-second increments
  - Values: 0 - 225 seconds
  - Default: 0 (equals 15 seconds)
- Bits 8-12—On-hook delay before a call is disconnected. This feature works only when the Cisco ATA is the terminating endpoint of the call. The user can hang up the phone in one room and pick up the phone in another room without disconnecting the line.
  - Range: 0 - 31
  - Factor: five-second increments
  - Values: 0 - 155 seconds
  - Default: 0 (no delay)
- Bits 13-15—Amount of time the Cisco ATA waits for a "486 Busy Here" response from a PSTN gateway after receiving a "183 Session Progress" response.
  - Range: 0 - 7
  - Factor: one-second increments
  - Values: 0 to 7 seconds
  - Default: 0 (no waiting)
- Bits 16-18—Configurable call waiting ring timeout. When a call arrives for a Cisco ATA port that is in use and has call-waiting enabled, the Cisco ATA plays a call-waiting tone. If the incoming call is not answered within a specified period of time, the Cisco ATA can reject the call by returning a "486 Busy" response to the remote user agent.
  You can configure FeatureTimer parameter bits 16-18 to specify the ringing period for incoming call-waiting calls.
This feature can be disabled by either using the default value 0 or by setting bits 16-18 to a value greater than the standard timeout for an incoming call as specified in SigTimer parameter bits 14-19. When this feature is disabled, a "480 Temporarily Not Available" response is returned to the remote user agent when the standard ring times out.

- Range: 0 - 7
- Factor: 10-second increments
- Values: 0 to 70 seconds
- Default: 0 (never timeout)

- Bits 19-31—Reserved.

**Value Type**
Bitmap

**Default**
0x00000000

**Voice Configuration Menu Access Code**
317

### FeatureTimer2

**Description**
This parameter provides configurable timing values for various Cisco ATA features, as described below:

- Bits 0-7—Maximum time that the Ethernet connection can be disconnected before the Cisco ATA automatically reboots.
  - Range: 0 - 255
  - Factor: one-second increments
  - Values: 0 - 255 seconds
  - Default: 30 (equals 30 seconds)

**Note**
To disable this feature, set the value of bits 0-7 to 0.

**Value Type**
Bitmap
Operational Parameters

Chapter 5  Parameters and Defaults

Default
0x0000001e

Voice Configuration Menu Access Code
361

SigTimer

Description
This parameter controls various timeout values. Table 5-7 contains bit definitions of this parameter.

Value Type
Bitmap

Default
0x01418564

Voice Configuration Menu Access Code
318

Related Parameter
CallWaitTone, page 5-62

Table 5-7  SigTimer Parameter Bit Definitions

<table>
<thead>
<tr>
<th>Bit Number</th>
<th>Definition</th>
</tr>
</thead>
<tbody>
<tr>
<td>0-7</td>
<td>Call waiting period—The period between each burst of call-waiting tone. Range: 0 to 255 in 0.1 seconds Default: 100 (0x64=100 seconds)</td>
</tr>
<tr>
<td>8-13</td>
<td>Reorder delay—The delay in playing the reorder (fast busy) tone after the far-end caller hangs up. Range: 0 to 62 in seconds Default—5 (seconds) 63—Never play the reorder tone.</td>
</tr>
<tr>
<td>14-19</td>
<td>Ring timeout—When a call is not answered, this is the amount of time after which Cisco ATA rejects the incoming call. Range—0 to 63 in 10 seconds Default—6 (60 seconds) 0—Never times out</td>
</tr>
<tr>
<td>20-25</td>
<td>No-answer timeout—The time to declare no answer and initiate call forwarding on no answer (used in SIP only) Range—0 to 63 in seconds Default—20 (0x14=20 seconds)</td>
</tr>
</tbody>
</table>
Table 5-7   **SigTimer Parameter Bit Definitions (continued)**

<table>
<thead>
<tr>
<th>Bit Number</th>
<th>Definition</th>
</tr>
</thead>
</table>
| 26-27      | Minimum hook flash time—The minimum on-hook time required for hook flash event.  
Range: 0 to 3  
Default: 0 (60 ms)  
Other possible values: 1=100 ms, 2=200 ms, 3=300 ms. |
| 28-31      | Maximum hook flash time—The maximum on-hook time allowed for hook flash event.  
Range: 0 to 15  
Default: 0 (1000 ms)  
Other possible values: 1=100 ms, 2=200 ms, 3=300 ms, 4=400 ms, 5=500 ms, 6=600 ms, 7=700 ms, 8=800 ms, 9=900 ms, 10=1000 ms, 11=1100 ms, 12=1200 ms, 13=1300 ms, 14=1400 ms, 15=1500 ms. |

**ConnectMode**

**Description**
This parameter is a 32-bit bitmap used to control the connection mode of the selected call signaling protocol. Table 5-8 provides bit definitions for this parameter.

**Value Type**
Bitmap

**Default**
0x00060400

**Voice Configuration Menu Access Code**
311

Table 5-8   **ConnectMode Parameter Bit Definitions**

<table>
<thead>
<tr>
<th>Bit Number</th>
<th>Definition</th>
</tr>
</thead>
</table>
| 0—H.323 only | 0—Enable normal start.  
1—Enable fast start. |
| 1—H.323 only | 0/1—Disable/enable h245 tunneling. |
| 2 | 0—Use the dynamic payload type 126/127 as the RTP payload type (fax pass-through mode) for G.711 µ-law/G.711 A-law.  
1—Use the standard payload type 0/8 as the RTP payload type (fax pass-through mode) for G.711 µ-law/G.711 A-law.  
Default: 0 |
| 3—H.323 only | 0/1—Disable/enable the requirement for the alternate gatekeeper to register. |
| 4—H.323 only | 0—Denotes a non-Cisco CallManager environment.  
1—Enable the Cisco ATA to operate in a Cisco CallManager environment. |
| 5—H.323 only | 0/1—Enable/disable two-way cut-through of voice path before receiving CONNECT message. |
### Table 5-8 ConnectMode Parameter Bit Definitions (continued)

<table>
<thead>
<tr>
<th>Bit Number</th>
<th>Definition</th>
</tr>
</thead>
<tbody>
<tr>
<td>6—H.323 only</td>
<td>0/1—Disable/enable using the Progress Indicator to determine if ringback is supplied by the far end with RTP.</td>
</tr>
<tr>
<td>7</td>
<td>0/1—Disable/enable fax pass-through redundancy. Default: 0</td>
</tr>
<tr>
<td>8-12</td>
<td>Specifies the fax pass-through NSE payload type. The value is the offset to the NSE payload base number of 96. The valid range is 0-23; the default is 4. For example, if the offset is 4, the NSE payload type is 100.</td>
</tr>
<tr>
<td>13</td>
<td>0—Use G.711u-law for fax pass-through codec. 1—Use G.711A-law for fax pass-through codec. Default: 0</td>
</tr>
<tr>
<td>14-15</td>
<td>0—Use fax pass-through. This setting is the default. 1—Use codec negotiation in sending fax. 2—Reserved. 3—Reserved.</td>
</tr>
<tr>
<td>16</td>
<td>0/1—Disable/enable SIP to remove previous registrations before adding a new registration. Default: 0</td>
</tr>
<tr>
<td><strong>Note</strong></td>
<td>On power up, all registrations are removed. Also, after some soft reboot/resets, only the latest registration is removed.</td>
</tr>
<tr>
<td>17—SIP only</td>
<td>0/1—Disable/enable call forwarding performed by the Cisco ATA. In SIP, call forwarding can be performed locally by the Cisco ATA or it can be performed by the SIP proxy. If this bit is disabled, the Cisco ATA forwards the entire dial string, including service activation code, to the SIP proxy for processing. Default: 1</td>
</tr>
<tr>
<td>18—SIP only</td>
<td>0/1—Disable/enable SIP call return performed by the Cisco ATA. Default: 1</td>
</tr>
<tr>
<td>19</td>
<td>0/1—Disable/enable the Cisco ATA to send a ringback tone to the caller. Default: 0</td>
</tr>
<tr>
<td>20—SIP only</td>
<td>0/1—Disable/enable SIP to perform action=proxy in a REGISTER message. You cannot simultaneously set bits 20 and 21 to 1. Also, if you set both these bits 0, the action parameter is not included in the REGISTER message, forcing the proxy server to perform the next step in the call process. Default: 0</td>
</tr>
<tr>
<td>21—SIP only</td>
<td>0/1—Disable/enable SIP to perform action=redirect in a REGISTER message. Default: 0</td>
</tr>
</tbody>
</table>
Table 5-8  ConnectMode Parameter Bit Definitions  (continued)

<table>
<thead>
<tr>
<th>Bit Number</th>
<th>Definition</th>
</tr>
</thead>
<tbody>
<tr>
<td>22—SIP only</td>
<td>0/1—Disable/enable SIP to process a received= tag in the VIA header to extract the external IP addresses used by the Network Address Translation (NAT) router. &lt;br&gt;When the Cisco ATA is operating behind a NAT, the NATIP parameter must be set to the external IP address of the NAT router. This allows the correct IP address to be placed in the Contact and SDP headers. &lt;br&gt;In release 2.14 or later, you may leave the NAT IP address at the default value of “0” or “0.0.0.0” and let the ATA automatically scan the Via header for a “received=” parameter. The parameter, if present, would indicate that the Cisco ATA is operating behind a firewall. &lt;br&gt;The Cisco ATA proceeds as follows: &lt;br&gt;1. If the &quot;received=&quot; parameter is in an INVITE response, the current INVITE is canceled and a new INVITE is sent with the new IP address extracted from the &quot;received=&lt;NAT IP address&gt;&quot; parameter in the Contact and SDP headers. &lt;br&gt;In addition, the Cisco ATA will cancel all previous registrations and re-register with the new IP address in the Contact header. This step is performed only if registration is currently in an idle state. &lt;br&gt;2. If the &quot;received=&quot; parameter is in a REGISTER response as a result of a REGISTER command, the Cisco ATA will cancel all previous registrations and re-register with the new IP address extracted from the &quot;received=&lt;NAT IP address&gt;&quot; parameter in the Contact header. Default: 0 &lt;br&gt;Note For the Cisco ATA to automatically detect its presence behind a NAT, the SIP proxy server or remote user agent server must include the &quot;received=&quot; parameter in the Via header in the responses to the Cisco ATA if the proxy detects that the source address and port do not match those in the Via header.</td>
</tr>
<tr>
<td>23</td>
<td>0/1—Disable/enable user-configurable setting for call-waiting default. &lt;br&gt;If this value is 0 (default), the end user cannot configure the permanent default call-waiting setting for every call. Instead, the service provider's default call-waiting setting is used for every call. &lt;br&gt;If this value is 1, the end-user can configure the permanent default call-waiting setting for every call, thus overriding the value set by the service provider. Default: 0</td>
</tr>
<tr>
<td>24</td>
<td>0/1—Disable/enable the mixing of audio and call waiting tone during a call. Default: 0</td>
</tr>
</tbody>
</table>
Table 5-8  ConnectMode Parameter Bit Definitions (continued)

<table>
<thead>
<tr>
<th>Bit Number</th>
<th>Definition</th>
</tr>
</thead>
</table>
| 25         | 0—Both the call-waiting call and active call will be dropped the instant the call-waiting user hangs up the phone.  
1—This setting enables the call-waiting hang-up alert feature, which alerts call-waiting users that an active call is still on hold once the user has hung up from another call. The phone will continue to ring until the party on hold has hung up.  
Note For the call-waiting hang-up feature to work, the call-waiting call command sequence must be set appropriately. For more information on how to set this sequence, see the “Enabling the Call-Waiting Hang-Up Alert Feature” section on page 4-6.  
Default: 0 |
| 26         | Reserved. |
| 27         | 0—The Cisco ATA does not change the user name to Anonymous in the SIP URL of the FROM and CONTACT headers in any INVITE requests even when the display name is sent as Anonymous. Also, the Cisco ATA does not change the user name to Anonymous in the o= line (also called the origin line) of the Session Description Protocol (SDP) header in SIP INVITE requests.  
1—The Cisco ATA exhibits the following behavior, provided that the Caller Line Identification Restriction (CLIR) feature is enabled:  
• The Cisco ATA uses a user name of Anonymous in the following SIP headers and requests:  
  – FROM header  
  – CONTACT header  
  – The o= line of the SDP header in SIP INVITE requests  
  For more information and example requests showing the usage of the Anonymous user name, see the “Anonymous User Name Support for SIP INVITE Requests” section on page 4-11.  
• The Diversion header in the INVITE request contains the privacy=[full|off] field, and this field will have the value of full. For more information, see the “Privacy Token Support for SIP Diversion Header” section on page 4-12.  
Default: 0 |

Enabling the CLIR feature

CLIR is enabled in one of two ways:

• By setting bit 3 of both the CallFeatures and PaidFeatures parameters to 0. (For the Phone2 port of the Cisco ATA, you would set bit 19 to 0 for each parameter.) For more information on these parameters, see the “CallFeatures” section on page 5-35 and the “PaidFeatures” section on page 5-36.  
• By enabling CLIR on a per-call basis using the call command dial string. For more information, see Chapter 6, “Call Commands.” If CLIR is enabled in the call command string, this takes precedence over the setting of bits 3 or 19 of the CallFeatures and PaidFeatures parameters.
**OpFlags**

**Description**
This parameter enables/disables various operational features.
See Table 5-9 on page 5-46 for bit definitions of this parameter.

**Value Type**
Bitmap

**Default**
0x2

**Voice Configuration Menu Access Code**
323
### Operational Parameters

#### Related Parameters
- TftpURL, page 5-5
- DHCP, page 5-8
- VLANSetting, page 5-12

#### Table 5-9 OpFlags Parameter Operational Features to Turn On or Off

<table>
<thead>
<tr>
<th>Bit Number</th>
<th>Definition</th>
</tr>
</thead>
<tbody>
<tr>
<td>0</td>
<td>If Bit 0 = 0, the TFTP configuration filename supplied by the DHCP server overwrites the default filename for each Cisco ATA. If Bit 0 = 1, the default Cisco ATA filename is always used. Default: 0</td>
</tr>
<tr>
<td>1</td>
<td>If Bit 1 = 0, the Cisco ATA probes the static network router during the power-up process. If Bit 1 = 1, static network router probing is disabled. Default: 1</td>
</tr>
<tr>
<td>2</td>
<td>Reserved.</td>
</tr>
<tr>
<td>3</td>
<td>If Bit 3=1, the Cisco ATA does not request DHCP option 150 in the DHCP discovery message; some DHCP server do not respond if option 150 is requested. Default: 0</td>
</tr>
<tr>
<td>4</td>
<td>If Bit 4 = 1, the Cisco ATA uses the VLAN ID specified in the VLANSetting parameter for VLAN IP encapsulation (see the “VLANSetting” section on page 5-12). Default: 0</td>
</tr>
<tr>
<td>5</td>
<td>If Bit 5=1, the Cisco ATA does not use VLAN IP encapsulation. Default: 0</td>
</tr>
<tr>
<td>6</td>
<td>If Bit 6=1, the Cisco ATA does not perform CDP discovery. Default: 0</td>
</tr>
<tr>
<td>7</td>
<td>If Bit 7=1, the Cisco ATA does not allow web configuration. Once the web server is disabled, you must configure the Cisco ATA with the TFTP or voice configuration menu methods. Default: 0</td>
</tr>
</tbody>
</table>

#### Examples

1. If the existing OpFlags value is 0x2, select menu option 323 from the voice configuration menu and enter the value 130 (0x82). This disables web configuration. If you later attempt to access the Cisco ATA web configuration page, the following error messages will be displayed.
   - Netscape: The document contained no data. Try again later, or contact the server's administrator.
   - Internet Explorer: The page cannot be displayed.

2. If the existing OpFlags value is 0x82, select menu option 323 from the voice configuration menu and enter the value 2 (0x2). This disables web configuration.

8     If Bit 8=1, the Cisco ATA does not allow HTTP refresh access with the http://ip/refresh command. Default: 0
Table 5-9  OpFlags Parameter Operational Features to Turn On or Off (continued)

<table>
<thead>
<tr>
<th>Bit Number</th>
<th>Definition</th>
</tr>
</thead>
</table>
| 9          | If Bit 9=1, the Cisco ATA does not allow HTTP reset access with the http://ip/reset command.  
Default: 0 |
| 10         | Reserved.  |
| 11         | If Bit 11=0, the Cisco ATA requests the device hostname from the DHCP server.  
If Bit 11=1, the Cisco ATA uses the device hostname that is specified in DHCP option 12.  
Default: 0 |
| 12         | Reserved.  |
| 13         | DNS Servers For Name Resolution  
If Bit 13=0 (default), use statically configured DNS IP addresses, if available, for name resolution. If statically configured DNS servers are not available, use DHCP-provided DNS IP addresses for name resolution.  
If Bit 13=1, use both statically configured DNS IP addresses and as many as two DHCP-provided DNS IP addresses. Therefore, the Cisco ATA can query as many as four DNS IP addresses in one DNS query.  
Default: 0  
For more information about statically configured DNS IP addresses, see the “DNS1IP” section on page 5-10 and the “DNS2IP” section on page 5-11. |
| 14         | DNS Servers For Name Resolution 2  
If Bit 14=0 (default), use statically configured DNS IP addresses (DNS1IP and DNS2IP), if available, for name resolution; otherwise, use DHCP-provided DNS IP addresses.  
If Bit 14=1, use both statically configured (DNS1IP and DNS2IP) and DHCP-provided DNS IP addresses (maximum of two) for name resolution.  
Note: This configuration bit gives precedence to statically provided DNS IP addresses over DHCP-provided DNS IP addresses. This bit also overrides the value of OpFlags parameter bit 13. |
| 15         | Disable UDP Checksum Generation  
If Bit 13=0, generate UDP checksum in outgoing UDP packets.  
If Bit 13=1, disable generation of UDP checksum in outgoing UDP packets.  
Default: 0 |
| 16         | Set this bit to 1 for voice prompt confirmation for the following services:  
• Call Forward All Enable  
• Call Forward Busy Enable  
• Call Forward No Answer Enable  
For more information, see the “Voice Prompt Confirmation for Call Waiting and Call Forwarding” section on page 4-20. |
### Operational Parameters

#### TimeZone

**Description**

This parameter is the timezone offset (in hours) from Greenwich Mean Time (GMT) for time-stamping incoming calls with local time (to use for Caller ID display, for example). See the “Additional Description” heading later in the description of this parameter for selecting time offsets for timezones that have 30-minute-factor or 45-minute-factor offset from GMT.

Local time is generated by the following formula:

- \( \text{Local Time} = \text{GMT} + \text{TimeZone} \), if \( \text{TimeZone} \leq 12 \)
- \( \text{Local Time} = \text{GMT} + \text{TimeZone} - 25 \), if \( \text{TimeZone} > 12 \)

**Value Type**

Integer

**Range**

0-24

**Default**

17

### Table 5-9 OpFlags Parameter Operational Features to Turn On or Off (continued)

<table>
<thead>
<tr>
<th>Bit Number</th>
<th>Definition</th>
</tr>
</thead>
</table>
| 17         | Set this bit to 1 for voice prompt confirmation for the following services:  
  - Call Forward All Disable  
  - Call Forward Busy Disable  
  - Call Forward No Answer Disable  
  For more information, see the “Voice Prompt Confirmation for Call Waiting and Call Forwarding” section on page 4-20. |
| 18         | Set this bit to 1 for voice prompt confirmation for the following service:  
  - Call Waiting Enable  
  For more information, see the “Voice Prompt Confirmation for Call Waiting and Call Forwarding” section on page 4-20. |
| 19         | Set this bit to 1 for voice prompt confirmation for the following service:  
  - Call Waiting Disable  
  For more information, see the “Voice Prompt Confirmation for Call Waiting and Call Forwarding” section on page 4-20. |
| 20-27      | Reserved. |
| 28-31      | To configure the Cisco ATA to prompt the user for the UIPassword when the user attempts to perform a factory reset or upgrade using the voice configuration menu, configure bits 28 to 31 with the value of 6. Any other value for these bits means that the Cisco ATA will not prompt the user for the UIPassword in these cases.  
  The default value for these bits is 0. |
Voice Configuration Menu Access Code

302

Additional Description

Use the following list to select Timezone offset (in minutes) from GMT for the following cities and
countries that have 30-minute-factor and 45-minute-factor time zone offsets. These values are integers
and can range from -720 through -60, and from 60 through 780.
- Tehran—210 = GMT + 3:30
- Kabul— 270 = GMT + 4:30
- Calcutta, Chennai, Mumbai, and New Delhi—330 = GMT + 5:30
- Kathmandu—345 = GMT + 5:45
- Rangoon—390 = GMT + 6:30
- Darwin and Adelaide—570 = GMT + 9:30
- Newfoundland— -210 = GMT - 3:30

Note Negative timezone values must be configured through the Cisco ATA Web configuration page and
cannot be configured with the voice configuration menu.

Telephone Configuration Parameters

This section includes the following parameters, which allow you to configure items such as generating
caller ID format and controlling line polarity:
- CallerIdMethod, page 5-49
- Polarity, page 5-51
- FXSInputLevel, page 5-52
- FXSOutputLevel, page 5-52

CallerIdMethod

Description
This 32-bit parameter specifies the signal format to use for both FXS ports for generating Caller ID
format. Possible values are:
- Bits 0-1 (method)—0=Bellcore (FSK), 1=DTMF, 2=ETSI, and 3 is reserved.
If method=0 (default), set the following bits:
- Bit 2—Reserved.
- Bit 3 to 8—Maximum number of digits in phone number (valid values are 1 to 20; default is 12).
- Bit 9 to 14—Maximum number of characters in name (valid values are 1 to 20; default is 15).
- Bit 15—if this bit is enabled (it is enabled by default), send special character O (out of area) to CID
device if the phone number is unknown.
• Bit 16—If this bit is enabled (it is enabled by default), send special character P (private) to CID device if the phone number is restricted.

• Bits 17 to 27—Reserved.

---

The Cisco ATA supports the Bellcore FSK method to turn on/off the visual message waiting indicator (VMWI) on a phone when the Cisco ATA receives MWI messages from a server. The Bellcore FSK VMWI is enabled automatically if the CallerIdMethod parameter is configured to use the Bellcore method and if the PaidFeatures and CallFeatures parameters (bits 11 and 27) are configured to enable message waiting indication. (See the “PaidFeatures” section on page 5-36 and the “CallFeatures” section on page 5-35.)

If method=1, set the following bits:

• Bit 2—Reserved.

• Bits 3-6—Start digit for known numbers (valid values are 12 for “A,” 13 for “B,” 14 for “C,” and 15 for “D.”)

• Bits 7-10—End digit for known numbers (valid values are 11 for “#,” 12 for “A,” 13 for “B,” 14 for “C,” and 15 for “D.”)

• Bits 11—Polarity reversal before and after Caller ID signal (value of 0/1 disables/enables polarity reversal)

• Bits 12-16—Maximum number of digits in phone number (valid values are 1 to 20; default is 15).

• Bits 17 to 19—Start digit for unknown or restricted numbers (valid values are 4 for “A,” 5 for “B,” 6 for “C,” and 7 for “D.”)

• Bits 20 to 22—End digit for unknown or restricted numbers (valid values are 3 for “#,” 4 for “A,” 5 for “B,” 6 for “C,” and 7 for “D.”)

• Bits 23 to 24—Code to send to the CID device if the number is unknown (valid values are 0 for “00,” 1 for “0000000000,” and 2 for “3.” 3 is reserved and should not be used.

• Bits 25 to 26—Code to send to the CID device if the number is restricted (valid values are 0 for “10,” and 1 for “1.” 2 and 3 are reserved and should not be used.

• Bits 27 to 31—Reserved.

If method=2, set the following bits:

• Bit 2—Set to 0 to have the Cisco ATA transmit data prior to ringing by using the Ring-Pulse Alerting Signal (RP-AS); set to 1 to have the Cisco ATA transmit data after the first ring.

• Bits 3-8—Maximum number of digits in a phone number (valid values are 1 to 20; default is 12).

• Bits 9-14—Maximum number of characters in a name (valid values are 1 to 20; default is 15).

• Bit 15—If this bit is enabled (it is enabled by default), send special character O (out of area) to CID device if telephone number is unknown.

• Bit 16—If this bit is enabled (it is enabled by default), send special character P (private) to CID device if telephone number is restricted.

• Bits 17-27 are reserved.
Examples
The following examples are recommended values for the CallerID Method parameter:

- USA = 0x00019e60
- Sweden = 0x006aff79 or 0x006aff61
- Denmark = 0x0000fde1 or 0x033efde1
- Germany = 0x00019e62
- Austria = 0x00019e66

Value Type
Bitmap

Default
0x00019e60

Voice Configuration Menu Access Code
316

Polarity

Description
You can control line polarity of the Cisco ATA FXS ports when a call is connected or disconnected by configuring the Polarity bitmap parameter as follows:

- Bit 0: CALLER_CONNECT_POLARITY. Polarity to use when the Cisco ATA is the caller and the call is connected.
  - 0 = Use forward polarity (Default)
  - 1 = Use reverse polarity
- Bit 1: CALLER_DISCONNECT_POLARITY. Polarity to use when the Cisco ATA is the caller and the call is disconnected.
  - 0 = Use forward polarity (Default)
  - 1 = Use reverse polarity
- Bit 2: CALLEE_CONNECT_POLARITY. Polarity to use when the Cisco ATA is the callee and the call is connected.
  - 0 = Use forward polarity (Default)
  - 1 = Use reverse polarity
- Bit 3: CALLEE_DISCONNECT_POLARITY. Polarity to use when the Cisco ATA is the callee and the call is disconnected.
  - 0 = Use forward polarity (Default)
  - 1 = Use reverse polarity

Note
Bits 4-31 are reserved.
Telephone Configuration Parameters

Value Type
Bitmap

Default
0x00000000

Voice Configuration Menu Access Code
304

FXSInputLevel

Description
Use this parameter to specify the input level control (analog-to-digital path) of the Cisco ATA FXS ports.

Value Type
Integer

Range
-9 to 2 dB

Default
-1

Voice Configuration Menu Access Code
370

Related Parameter
FXSOutputLevel, page 5-52

FXSOutputLevel

Description
Use this parameter to specify the output level control (digital-to-analog path) of the Cisco ATA FXS ports.

Value Type
Integer

Range
-9 to 2 dB

Default
-4
Voice Configuration Menu Access Code
371

Related Parameter
FXSInputLevel, page 5-52

Tone Configuration Parameters

The Cisco ATA supports the following tone parameters:

- DialTone
- BusyTone
- ReorderTone
- RingBackTone
- CallWaitTone
- AlertTone
- SITone

The Cisco ATA supports two types of tone-parameter syntax—basic format and extended format. Basic format is used in most countries; use the extended format only if the country in which the Cisco ATA is used requires this format.

This section covers all the call-progress tones that the Cisco ATA supports, and contains the following topics:

- Tone Parameter Syntax—Basic Format, page 5-53
- Tone Parameter Syntax—Extended Formats, page 5-54
- Recommended Values, page 5-59
- Specific Tone Parameter Information, page 5-60

This section also covers the following parameter, which is for configuring phone-ringing characteristics:

- RingOnOffTime, page 5-64

Note
For detailed recommendations of tone-parameter values by country, see Appendix E, “Recommended Cisco ATA Tone Parameter Values by Country.”

Tone Parameter Syntax—Basic Format

Each tone is specified by nine integers, as follows:

\[
\text{parametername: NumOfFreqs, Tfreq1, Tfreq2, Tamp1, Tamp2, Steady, OnTime, OffTime, TotalToneTime}
\]

- \text{parametername} is the name of the tone.
- \text{NumOfFreqs} is the number of frequency components (0, 1 or 2).
- \text{Tfreq1} and \text{Tfreq2} are the transformed frequencies of the first and second frequencies, respectively. Their values are calculated with the following formula:

\[
32767 \times \cos\left(\frac{2\pi F}{8000}\right)
\]
where $F$ is the desired frequency in Hz. Set this value to 0 if the frequency does not exist.
The range of each value is –32768 to 32767.
For negative values, use the 16-bit 2’s complement value. For example, enter –1 as 65535 or as 0xffff.

- $Tamp_1$ and $Tamp_2$ are the transformed amplitudes of the first and second frequencies, respectively.
  Their values are calculated with the following formulas:

$$32767 \times A \times \sin(2\pi F/8000)$$

$A$ (amplitude factor) = $0.5 \times 10^{(k+10-(n-1)\times3)/20}$

where $F$ is the desired frequency in Hz, $k$ is the desired volume in $dBm$, and $n$ is the number of frequencies. The $^*$ symbol means to the order of.

- $Steady$ controls whether the tone is constant or intermittent. A value of 1 indicates a steady tone and causes the Cisco ATA to ignore the on-time and off-time parameters. A value of 0 indicates an on/off tone pattern and causes the Cisco ATA to use the on-time and off-time parameters.

- $OnTime$ controls the length of time the tone is played in milliseconds (ms).
  Specify each value as a number of samples with a sampling rate of 8 kHz. The range of each value is 0 to 0xffff. For example, for a length of 0.3 seconds, set the value to 2400.

- $OffTime$ controls the length of time between audible tones in milliseconds (ms).
  Specify each value as a number of samples with a sampling rate of 8 kHz. The range of each value is 0 to 0xffff. For example, for a length of 0.3 seconds, set the value to 2400.

- $TotalToneTime$ controls the length of time the tone is played. If this value is set to 0, the tone will play until another call event stops the tone. For DialTone, DialTone2, BusyTone, ReorderTone, and RingBackTone, the configurable value is the number of 10 ms (100 = 1 second) units.
  For the remaining tones, the configurable value is the number of samples with a sampling rate of 8 kHz.

### Note
All tones are persistent (until the Cisco ATA changes state) except for the call-waiting tone and the confirm tone. The call-waiting tone, however, repeats automatically once every 10 seconds while the call-waiting condition exists.

### Tone Parameter Syntax—Extended Formats

Two types of extended format exist for the Cisco ATA tone parameters:

- **Extended Format A, page 5-55**—This format can be used for the following tone parameters:
  - DialTone
  - BusyTone
  - RingbackTone
  - CallWaitTone
  - AlertTone

- **Extended Format B, page 5-56**—This format can be used for the following tone parameters:
  - ReorderTone
  - SITone
Extended Format A

Each tone is specified by 11 integers, as follows:

\[
\text{parametername}: \text{NumOfFreqs}, \text{Tfreq1}, \text{Tamp1}, \text{Tfreq2}, \text{Tamp2}, \text{NumOfOnOffPairs}, \text{OnTime1}, \text{OffTime1}, \text{OnTime2}, \text{OffTime2}, \text{TotalToneTime}
\]

- `parametername` is the name of the tone.
- `NumOfFreqs = 100 +` the number of frequencies in the tone. (Therefore, `NumOfFreqs = 101` for one frequency, and `102` for two frequencies.)
- `Tfreq1` and `Tfreq2` are the transformed frequencies of the first and second frequencies, respectively. Their values are calculated with the following formula:
  \[
  32767 \times \cos \left(\frac{\pi \times F}{8000}\right)
  \]
  where \( F \) is the desired frequency in Hz. Set this value to 0 if the frequency does not exist.
  The range of each value is –32768 to 32767.
  For negative values, use the 16-bit 2’s complement value. For example, enter –1 as 65535 or as 0xffff.
- `Tamp1` and `Tamp2` are the transformed amplitudes of the first and second frequencies, respectively. Their values are calculated with the following formula:
  \[
  32767 \times A \times \sin \left(\frac{\pi \times F}{8000}\right)
  \]
  \[A (\text{amplitude factor}) = 0.5 \times 10^{(k+10-(n-1)\times3)/20}\]
  where \( F \) is the desired frequency in Hz, \( k \) is the desired volume in dBm, and \( n \) is the number of frequencies. The \(^\wedge\) symbol means to the order of.
- `NumOfOnOffPairs` is the number of on-off pairs in the cadence of the tone.
  Valid values are 0, 1 and 2. Use 0 if the tone is steady.
- `OnTime1` and `OnTime2` values are the lengths of time the tone is played for the first and second on-off pairs of a cadence, respectively. (See Figure 5-1 for a graphical representation.) Specify each value as a number of samples with a sampling rate of 8 kHz. The range of each value is 0 to 0xffff. For example, for a length of 0.3 seconds, set the value to 2400.
- `OffTime1` and `OffTime2` values are the lengths of time that silence is played for the first and second on-off pairs of a cadence, respectively. (See Figure 5-1 for a graphical representation.) Specify each value as a number of samples with a sampling rate of 8 kHz. The range of each value is 0 to 0xffff. For example, for a length of 0.3 seconds, set the value to 2400.

\[
\text{Figure 5-1 Cadence With Two On-Off Pairs}
\]

- `TotalToneTime` controls the length of time the tone is played. If this value is set to 0, the tone will play until another call event stops the tone. For DialTone, DialTone2, BusyTone, ReorderTone, and RingBackTone, the configurable value is the number of 10 ms (100 = 1 second) units.
  For the remaining tones, the configurable value is the number of samples with a sampling rate of 8 kHz.
Tone Configuration Parameters

Note

All tones are persistent (until the Cisco ATA changes state) except for the call-waiting tone and the confirm tone. The call-waiting tone, however, repeats automatically once every 10 seconds while the call-waiting condition exists.

Extended Format B

The ReorderTone parameter specifies the tone that the Cisco ATA plays when the called number is not available or the external circuit is busy. The SItone parameter specifies the special-information tone that the Cisco ATA plays when the called party is not found on the network. These tones can consist of:

- Up to three frequencies played simultaneously and a cadence of up to three on-off pairs. The first on-off pair can repeat multiple times before the second on-off pair plays.
  
  For example, a 400 Hz frequency plays four times for 0.75 second followed by 0.1 second of silence after each play and then plays one time for 0.75 second followed by 0.4 second of silence. This pattern can be set to repeat until another call event stops the pattern.

- Up to three frequencies played sequentially with a cadence of up to three on-off pairs
  
  For example, the frequencies 900 Hz, 1400 Hz, and 1800 Hz play sequentially for 0.33 seconds each with no silence after the first and second frequencies but one second of silence after the third frequency.

The syntax for Extended Format B is specified by 17 integers, as follows:

```
parameter:Sequential,NumOfFreqs,TFreq1,Tamp1,TFreq2,
Tamp2,TFreq3,Tamp3,NumOfOnOffPairs,OnTime1,OffTime1,
OnTime2,OffTime2,OnTime3,OffTime3,NumOfRepeats,TotalToneTime
```

where:

- `parameter` is either ReorderTone or SITone.

- `Sequential` specifies whether multiple frequencies in a tone play simultaneously (100) or sequentially (101). Set to 100 for a tone with one frequency. If `Sequential` is 101, the number of frequencies (NumOfFreqs) has to be the same value as the number of on-off pairs in a cadence (NumOfOnOffPairs).

- `NumOfFreqs` is the number of frequencies in the tone (1, 2, or 3). The frequencies can play simultaneously or sequentially, depending on the `Sequential` setting.

- `TFreq1`, `TFreq2`, and `TFreq3` are the transformed frequencies of the first, second, and third frequencies, respectively. Calculate each value with the following formula:

  \[ 32767 \times \cos(2 \times \pi \times F/8000) \]

  where \( F \) is the desired frequency in Hz. Set this value to 0 if the frequency does not exist.

  The range of each value is \(-32768\) to \(32767\).

  For negative values, use the 16-bit 2’s complement value. For example, enter \(-1\) as 65535 or as 0xffff.

- `Tamp1`, `Tamp2` and `Tamp3` are the transformed amplitudes of the first, second and third frequencies, respectively. Their values are calculated with the following formula:

  \[ 32767 \times A \times \sin(2\pi F/8000) \]

  \( A \) (amplitude factor) \(= 0.5 \times 10^{(k+10-(n-1)*3)/20} \)

  where \( F \) is the desired frequency in Hz, \( k \) is the desired volume in dBm, and \( n \) is the number of frequencies (If `Sequential` is set to 101, \( n \) is equal to 1). The \(^\wedge\) symbol means to the order of.
### Parameters and Defaults

**Tone Configuration Parameters**

- **NumOfOnOffPairs** is the number of on-off pairs in the cadences of the tone (0, 1, 2, or 3). For a steady tone, use 0.
  
  If this value is 0, the `OnTime1`, `OnTime2`, `OnTime3`, `OffTime1`, `OffTime2`, and `OffTime3` values must also be 0.

- **OnTime1**, **OnTime2**, and **OnTime3** are the lengths of time that the first, second, and third on-off pairs of a cadence play a sound, respectively. (See Figure 5-2 for a graphical representation.)
  
  Specify each value as a number of samples with the sampling rate of 8 kHz. The range of each value is 0 to 0xffff.
  
  For example, for a length of 0.3 seconds, set a value to 2400.

- **OffTime1**, **OffTime2**, and **OffTime3** are the lengths of silence after the sound of the first, second, and third on-off pairs of a cadence, respectively.
  
  Specify each value as a number of samples with the sampling rate of 8 kHz. The range of each value is 0 to 0xffff.
  
  For example, for a length of 0.3 seconds, set a value to 2400. (See Figure 5-2 for a graphical representation.)

- **NumOfRepeats** is the number of times that the first on-off pair of the cadence (specified by `OnTime1`, `OffTime1`) repeats before the second on-off pair (specified by `OnTime2`, `OffTime2`) plays.
  
  For example, if `NumOfRepeats` is 2, the first on-off pair will play three times (it will play once and then repeat two times), then the second on-off pair will play.

- **TotalToneTime** is the total length of time that the tone plays. If this value is 0, the tone will play until another call event stops the tone.
  
  This value is in 10 ms units (100 ms = 1 second).

Two examples of Extended Format B, both using the Reorder tone, follow.

### ReorderTone Parameter Example 1

Assume that you want a reorder tone in which:

- The frequencies 900 Hz, 1400 Hz, and 1800 Hz play sequentially.
- Each frequency plays once for 0.33 seconds.
- There is no silence after the first and the second frequencies.
- There is 1 second of silence after the third frequency (before the first frequency starts again)
- The volume of each frequency is –19 dBm.
- The tone plays until another call event stops the tone.

For this reorder tone, make the following setting. See Table 5-10 for a detailed explanation.

```plaintext
ReorderTone:101,3,24917,3405,14876,4671,5126,5178,3,2640,0,2640,0,
2640,8000,0,0
```
Chapter 5      Parameters and Defaults

Tone Configuration Parameters

Table 5-10  Reorder Tone Parameter Example 1 Explanation

<table>
<thead>
<tr>
<th>Component</th>
<th>Setting</th>
<th>Explanation</th>
</tr>
</thead>
<tbody>
<tr>
<td>Sequential</td>
<td>101</td>
<td>Frequencies play sequentially</td>
</tr>
<tr>
<td>NumOfFreqs</td>
<td>3</td>
<td>Three frequencies in the tone</td>
</tr>
<tr>
<td>TFreq1</td>
<td>24917</td>
<td>First frequency is 900 Hz</td>
</tr>
<tr>
<td>TAmp1</td>
<td>3405</td>
<td>First frequency volume is –19 dBm</td>
</tr>
<tr>
<td>TFreq2</td>
<td>14876</td>
<td>Second frequency is 1400 Hz</td>
</tr>
<tr>
<td>TAmp2</td>
<td>4671</td>
<td>Second frequency volume is –19 dBm</td>
</tr>
<tr>
<td>TFreq3</td>
<td>5126</td>
<td>Third frequency is 1800 Hz</td>
</tr>
<tr>
<td>TAmp3</td>
<td>5178</td>
<td>Third frequency volume is –19 dBm</td>
</tr>
<tr>
<td>NumOfOnOffPairs</td>
<td>3</td>
<td>Three on-off pairs in the cadence of the tone</td>
</tr>
<tr>
<td>OnTime1</td>
<td>2640</td>
<td>Sound in first on-off pair plays for 0.33 seconds</td>
</tr>
<tr>
<td>OffTime</td>
<td>0</td>
<td>No silence after the first sound (the second sound plays immediately)</td>
</tr>
<tr>
<td>OnTime2</td>
<td>2640</td>
<td>Sound in second on-off pair plays for 0.33 seconds</td>
</tr>
<tr>
<td>OffTime2</td>
<td>0</td>
<td>No silence after the second sound (the third sound plays immediately)</td>
</tr>
<tr>
<td>OnTime3</td>
<td>2640</td>
<td>Sound in third on-off pair plays for 0.33 seconds</td>
</tr>
<tr>
<td>OffTime3</td>
<td>8000</td>
<td>1 second of silence after the sound in the third on-off pair (before the pattern repeats, beginning with the first on-off pair)</td>
</tr>
<tr>
<td>NumOfRepeats</td>
<td>0</td>
<td>First on-off pair of the cadence plays once (does not repeat), then the second on-off pair plays</td>
</tr>
<tr>
<td>TotalToneTime</td>
<td>0</td>
<td>Tone plays continuously (set of three on-off pairs of the cadence repeat continuously) until another call event stops the tone</td>
</tr>
</tbody>
</table>

ReorderTone Parameter Example 2

Assume that you want a reorder tone in which:

- The only frequency is 400 Hz.
- The frequency plays six times, each time for 0.1 second followed by 0.9 second of silence.
- The frequency then plays once for 0.3 second followed by 0.7 second of silence.
- The volume of the frequency is –19 dBm.
- The tone plays until another call event stops the tone.

For this reorder tone, make the following setting. See Table 5-11 for a detailed explanation.

ReorderTone:100,1,31164,1620,0,0,0,0,2,800,7200,2400,5600,0,0,5,0
Table 5-11: Reorder Tone Parameter Example 2 Explanation

<table>
<thead>
<tr>
<th>Component</th>
<th>Setting</th>
<th>Explanation</th>
</tr>
</thead>
<tbody>
<tr>
<td>Sequential</td>
<td>100</td>
<td>Required setting for a tone with one frequency</td>
</tr>
<tr>
<td>NumOfFreqs</td>
<td>1</td>
<td>One frequency in the tone</td>
</tr>
<tr>
<td>TFreq1</td>
<td>31164</td>
<td>First frequency is 400 Hz</td>
</tr>
<tr>
<td>TAmp1</td>
<td>1620</td>
<td>First frequency volume is –19 dBm</td>
</tr>
<tr>
<td>TFreq2</td>
<td>0</td>
<td>No second frequency</td>
</tr>
<tr>
<td>TAmp2</td>
<td>0</td>
<td>No second frequency</td>
</tr>
<tr>
<td>TFreq3</td>
<td>0</td>
<td>No third frequency</td>
</tr>
<tr>
<td>TAmp3</td>
<td>0</td>
<td>No third frequency</td>
</tr>
<tr>
<td>NumOfOnOffPairs</td>
<td>2</td>
<td>Two on-off pairs in the cadence of the tone</td>
</tr>
<tr>
<td>OnTime1</td>
<td>800</td>
<td>Sound in first on-off pair plays for 0.1 second</td>
</tr>
<tr>
<td>OffTime1</td>
<td>7200</td>
<td>Sound in first on-off pair is followed by 0.9 second of silence</td>
</tr>
<tr>
<td>OnTime2</td>
<td>2400</td>
<td>Sound in second on-off pair plays for 0.3 seconds</td>
</tr>
<tr>
<td>OffTime2</td>
<td>5600</td>
<td>Sound in second on-off pair is followed by 0.7 second of silence</td>
</tr>
<tr>
<td>OnTime3</td>
<td>0</td>
<td>No third on-off pair in the cadence</td>
</tr>
<tr>
<td>OffTime3</td>
<td>0</td>
<td>No third on-off pair in the cadence</td>
</tr>
<tr>
<td>NumOfRepeats</td>
<td>5</td>
<td>First on-off pair of the cadence plays six times (plays once and then repeats five times), then the second on-off pair plays</td>
</tr>
<tr>
<td>TotalToneTime</td>
<td>0</td>
<td>Tone plays continuously (set of two on-off pairs of the cadence repeat continuously) until another call event stops the tone</td>
</tr>
</tbody>
</table>

**Recommended Values**

The following settings are recommended for the US:

- **DialTone** = "2,31538,30831,1380,1740,1,0,0,1000" (approximately -17 dBm)
- **BusyTone** = "2,30467,28959,1191,1513,0,4000,4000,0" (approximately -21 dBm)
- **ReorderTone** = "2,30467,28959,1191,1513,0,2000,2000,0" (approximately -21 dBm)
- **RingBackTone** = "2,30831,30467,1943,2111,0,16000,32000,0" (approximately -16 dBm)
- **CallWaitTone** = "1,30831,0,5493,0,0,2400,2400,4800" (approximately -10 dBm)
- **AlertTone** = “1,30467,0,5970,0,0,480,480,1920”
- **SITone** = "2,30467,28959,1191,1513,0,2000,2000,0" (approximately -21 dBm)

**Note**

For detailed recommendations of tone-parameter values by country, see Appendix E, “Recommended Cisco ATA Tone Parameter Values by Country.”
Specific Tone Parameter Information

Brief descriptions, and lists of default values and the voice configuration menu code for each Cisco ATA tone parameter, appear in the following sections:

- **DialTone**, page 5-60
- **BusyTone**, page 5-61
- **ReorderTone**, page 5-61
- **RingbackTone**, page 5-62
- **CallWaitTone**, page 5-62
- **AlertTone**, page 5-63
- **SITone**, page 5-63

**DialTone**

**Description**

The Cisco ATA plays the dial tone when it is ready to accept the first digit of a remote address to make an outgoing call.

**Default values (using the Basic format)**

- NumOfFreqs — 2
- Tfreq1 — 31538
- Tfreq2 — 30831
- Tamp1 — 1380
- Tamp2 — 1740
- Steady — 1
- OnTime — 0
- OffTime — 0
- TotalToneTime — 1000

**Voice Configuration Menu Access Code**

920
BusyTone

Description
The Cisco ATA plays the busy tone when the callee is busy.

Default values (using the Basic format)
- NumOfFreqs—2
- Treq1—30467
- Tfreq2—28959
- Tamp1—1191
- Tamp2—1513
- Steady—0
- OnTime—4000
- OffTime—4000
- TotalToneTime—0

Voice Configuration Menu Access Code
921

ReorderTone

Description
The Cisco ATA plays the reorder tone (also known as congestion tone) if the outgoing call failed for reasons other than busy. This is a fast-busy tone.

Default values (using the Basic format)
- NumOfFreqs—2
- Treq1—30467
- Treq2—28959
- Tamp1—1191
- Tamp2—1513
- Steady—0
- OnTime—2000
- OffTime—2000
- TotalToneTime—0

Voice Configuration Menu Access Code
922
RingbackTone

Description
The Cisco ATA plays the ring-back tone when the callee is being alerted by the called device.

Default values (using the Basic format)
- NumOfFreqs—2
- Tfreq1—30831
- Tfreq2—30467
- Tamp1—1943
- Tamp2—2111
- Steady—0
- OnTime—16000
- OffTime—32000
- TotalToneTime—0

Voice Configuration Menu Access Code
923

CallWaitTone

Description
The Cisco ATA plays the call-waiting tone when an incoming call arrives while the user is connected to another party.

Default values (using the Basic format)
- NumOfFreqs—1
- Tfreq1—30831
- Tfreq2—0
- Tamp1—5493
- Tamp2—0
- Steady—0
- OnTime—2400
- OffTime—2400
- TotalToneTime—4800

Voice Configuration Menu Access Code
924
AlertTone

Description
The Cisco ATA plays the alert tone as a confirmation tone that a special event, such as call forwarding, is in effect.

Default values (using the Basic format)
- NumOfFreqs—1
- Tfreq1—30467
- Tfreq2—0
- Tamp1—5970
- Tamp2—0
- Steady—0
- OnTime—480
- OffTime—480
- TotalToneTime—1920

Voice Configuration Menu Access Code
925

SITone

Description
The Cisco ATA plays the SITone (special information tone) if the Cisco ATA receives a 404 (Not Found) response, which indicates that the called party cannot be located on the network.

Default values (using the Basic format)
- NumOfFreqs—2
- Tfreq1—30467
- Tfreq2—28959
- Tamp1—1191
- Tamp2—1513
- Steady—0
- OnTime—2000
- OffTime—2000
- TotalToneTime—0

Voice Configuration Menu Access Code
926
Dial Plan Parameters

This section describes the configurable parameters related to dial plans:

- DialPlan, page 5-64
- DialPlanEx, page 5-72
- IPDialPlan, page 5-72

Dial Plan

Description
The programmable dial plan is designed for the service provider to customize the behavior of the Cisco ATA for collecting and sending dialed digits. The dial plan allows the Cisco ATA user to specify the events that trigger the sending of dialed digits. These events include the following:

- The termination character has been entered.
- The specified dial string pattern has been accumulated.
- The specified number of dialed digits has been accumulated.
- The specified inter-digit timer has expired.
Chapter 5  Parameters and Defaults

Dial Plan Parameters

Value Type
Alphanumeric string

Range
Maximum number of characters is 199.

Note
If the dial plan exceeds 199 characters, use the DialPlanEx parameter instead of the DialPlan parameter. For more information, see the “DialPlanEx” section on page 5-72.

Default
*St4-|#St4-|911111>#/t8.r9t2-|0>#t811.rat4-|^1t4>#.-

Voice Configuration Menu Access Code
926

Additional DialPlan Information
The DialPlan section contains the following additional topics that describe commands and rules for creating your own dial plan, and includes many examples:

- Dial Plan Commands, page 5-65
- Dial Plan Rules, page 5-66
- Dial Plan Examples, page 5-70

Dial Plan Commands

The following list contains commands that can be used to create your own dial plans:

- . — Wildcard, match any digit entered.
- - — Additional digits can be entered. This command can be used only at the end of a dial plan rule (for example, 1408t5- is legal usage of the - command, but 1408t5-3... is illegal).
- [ ] — Range, which means to match any single digit in the list. Use an underscore (_) to indicate a range of digits. For example, [135] matches the digits 1, 3, and 5. Also, [1_5] matches the digits 1, 2, 3, 4 and 5. The pound key (#) and asterisk (*) are not allowed in the Range command. Also, the Repeat (rn) command does not apply to range, and range cannot include the Subrule matching command.
- (subrule0| subrule1| ...|subruleN) — Subrule matching. Using the ( ) and | operators allows you to specify multiple subrules within a dial plan rule so that a subrule match is reached if the entered digits fit one of the subrules. This can be used to reduce the length of the desired dial plan rule by concatenating the group of the subrules with the common rule.
  
  For example, a dial plan rule of (1900|1800|17...)555.r3 or three dial plan rules of 1900555..31800555..317...555.r3 are equivalent. A match is reached if 11 digits are entered and the first three digits are either 1900, 1800, or 17..., and the fifth, sixth, and seventh digits are all 5.
- ># — Defines the # character as a termination character. When the termination character is entered, the dial string is automatically sent. The termination character can be entered only after at least one user-entered digit matches a dial plan rule. Alternatively, the command >* can be used to define * as the termination character.
- tn — Defines the timeout value n, in the unit of seconds, for the interdigit timer. Valid values are 0-9 and a-z, where a-z indicates a range of 10 to 35.
Dial Plan Parameters

- **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 26. Use the repeat modifier to specify more rules in less space.

  | The commands ># and tn are modifiers, not patterns, and are ignored by the rn command.

- ![Note](image)

- l—Used to separate multiple dial plan rules.
- ^—Logical not. Match any character except the character immediately following the ^ command. The ^ command can also be used as a negation instruction before the range or subrule matching commands.
- 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 (for example, a dial plan beginning with *S will be the only one in effect if the user first enters the * key).

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

  | No syntax check is performed by the actual implementation. The administrator has the responsibility of making sure that the dial plan is syntactically valid.

### Dial Plan Rules

The Cisco ATA supports the following dial plan rules:

- **(In Rule) for Dial Plan Blocking**, page 5-66
- ‘H’ Rule to Support Hot/Warm Line, page 5-67
- ‘P’ Rule to Support Dial Prefix, page 5-67
- ‘B’ Rule for Base Number, page 5-68
- ‘R’ Rule for Enhanced Prefix, page 5-68
- ‘C’ Rule for Call Blocking, page 5-69
- ‘F’ Rule for Call Forwarding Blocking, page 5-69
- ‘X’ Rule for Call Blocking and Call Forwarding Blocking, page 5-70
- ‘D’ Rule for Displaying Caller ID, page 5-70

**(In Rule) for Dial Plan Blocking**

Dial plan blocking can be used to reduce the occurrences of invalid dialed digits being sent and can prevent the dialed string of a specified pattern from being sent. By adding dial plan blocking, dialed digits are discarded after the interdigit timer expires unless one of the specified matching rules is met.

In addition, the default nine-second global interdigit timeout value is also modified with the value specified in the dial plan blocking command:

**Syntax**

\[ \text{In} \]
where \( n \) specifies the global interdigit timeout and the valid values are 1-9 and a-z (10-35).

**Example**

\[ \text{Ice} \ 911 \]

This command specifies an interdigit timeout of 12 seconds, and will discard dialed digits unless 911 is entered.

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

**‘H’ Rule to Support Hot/Warm Line**

Hotline/Warmline, also known as Private Line Automatic Ringdown (PLAR), is a line used for priority telephone service. If the Hotline feature is configured, the Cisco ATA immediately dials a pre-configured number as soon as the handset goes off hook. If the Warmline feature is configured, the Cisco ATA dials a pre-configured number if no digits were entered before the specified timer value expired when the handset went offhook.

**Syntax**

\[ \text{Hdnnnn} \]

where \( d \) is a delay-in-seconds parameter 0-9,a-z (to support 0 to 35 seconds delay), and \( nnnn \) is the variable-length phone number to call when no digits are entered for \( d \) seconds after offhook.

**Example 1**

\[ \text{H05551212} \]

This is a hotline configuration; the Cisco ATA immediately dials 555-1212 when the handset goes off hook.

**Example 2**

\[ \text{H55551212} \]

This is a warmline configuration; the Cisco ATA waits for five seconds and dials 555-1212 if no digits were entered when the handset went off hook.

**‘P’ Rule to Support Dial Prefix**

This rule is for automatic pre-pending the dial string as entered by the user with a specified prefix.

**Syntax**

\[ \text{Ptnnnn} \]

where \( t \) is a single leading trigger character; if \( t \) is the first entered digit when making a new call, it triggers the prepending of a variable-length prefix (as specified by \( nnnn \)) in the dial string. The \( t \) character can take one of the following values:

0-9,*,#, 'n' (any of 1-9), 'N' (any of 'n' and 0), 'a' (any of 'n',* and #), or 'A' (any of 'a' and 0);

**Example**

\[ \text{Pn12345} \]
This rule prepends 12345 to the dial string when the first entered digit is any of 1-9. The triggered digit is not removed from the dial string.

'B' Rule for Base Number

A base number the first part of a phone number that the Cisco ATA dials when Rule ‘B’ is used in a dial plan. The telephone number that the Cisco ATA calls consists of the base number followed by a two-digit extension.

Rule ‘B’ is used for voice prompt confirmation for the call-waiting and call-forwarding features. For more information, see the “Voice Prompt Confirmation for Call Waiting and Call Forwarding” section on page 4-20.

To set a base number, use ‘B’ followed by the desired base number. For example, if the desired base number is 1234, you would add the rule ‘B1234’ to your dial plan.

Syntax

\[ B\text{nnnn} \]

where \text{nnnn} is the base number.

Example

If the administrator has configured a base number of 1234 and call forward on busy (extension 03) is enabled, the called number is 123403.

‘R’ Rule for Enhanced Prefix

This enhanced prefix rule matches entire strings, whereas the ‘P’ rules matches only a single digit. The ‘R’ rule is for automatically prepending a specified prefix to the dialed string. The string must be an exact match to trigger the rule. If more than one ‘R’ rule matches, the first matched ‘R’ rule is triggered.

The ‘R’ rule also uses negation to exclude one or more leading digits before prepending the defined prefix string.

The number of dashes (-) after the \text{R} represents the number of leading digits that will be removed preceding the prefix.

Syntax

\[ R\text{nmmn}(\text{tttt}) \]

where \text{tttt} is a trigger string. If the dialed numbers match this string, this match triggers the prepending of a variable-length prefix (as specified by \text{nmmn}) to the dial string. The triggered string is not removed from the dial string. The negation, subrule matching and range patterns can be applied to the trigger strings.

Example 1

\[ R1212(\{2_9\}-) \]

This rule prepends 1212 to dial strings that have a leading digit of 2 to 9.

Note

Note: ‘R’ rules can replace most ‘P’ rules; for example, \text{Pn12345} is the same as \text{R12345(\{1_9\}.)}.

Example 2

\[ R-0033(0\{1-9\}.r7) \]
This removes the first dialed digit, then prepends 0033 to the dialed string. For example, if the number 0148336134 is dialed, the resulting string becomes 0033148336134.

**Example 3**

\[ R----0(0033[1-9].r7) \]

This removes the first four dialed digits, then prepends 0 to the dialed string. For example, if the number 0033148336134 is dialed, the resulting string becomes 0148336134.

**Log Information**

The Call Prefix <prefix>+<num> is shown in the prserv log.

### 'C' Rule for Call Blocking

This rule is for blocking call numbers.

**Syntax**

\[ Cnnnn \]

where \( nnnn \) is the leading set of digits of the blocked call number; \( nnnn \) can be composed with subrule matching and range. The rule is triggered when the leading digits of a dialed string match the string \( nnnn \).

The ‘C’ rule does not work with negation.

**Example:**

\[ C1900|C1888 \text{ or } C(1900|1888) \]

This rule blocks call numbers beginning with 1900 or 1888.

**Log Information**

The Call Block <num> is shown in the prserv log, and a busy tone is being played.

### 'F' Rule for Call Forwarding Blocking

This rule is for blocking call forwarding numbers.

**Syntax**

\[ Fnnnn \]

where \( nnnn \) is the leading set of digits of the blocked call forwarding number; \( nnnn \) can be composed with subrule matching and range. The rule is triggered when the leading digits of a dialed forwarding number match the string \( nnnn \). The ‘F’ rule does not work with negation.

**Example:**

\[ F1900|F1888 \text{ or } F(1900|1888) \]

These rules block call forwarding numbers beginning with 1900 or 1888.

**Log Information**

The CFWD Block:<num> is shown in the prserv log, and a busy tone is being played.
‘X’ Rule for Call Blocking and Call Forwarding Blocking

This rule is for blocking call numbers and call forwarding numbers.

Syntax

\[ Xnnnn \]

where \( nnnn \) is the leading set of digits of the blocked call number and blocked call forwarding number; \( nnnn \) can be composed with subrule matching and range. The rule is triggered when the leading set of digits of a dialed call number or forwarding number match the string \( nnnn \). The ‘F’ rule does not work with negation.

Example

\[ X1900|X1888 \] 
or 
\[ X(1900|1888) \]

This rule blocks the call numbers and call forwarding numbers beginning with 1900 or 1888.

‘D’ Rule for Displaying Caller ID

This rule is for displaying caller ID at the remote site. The number must be an exact match to trigger the rule.

Syntax

\[ Dnnnn \]

where \( nnnn \) is the callee number. The caller ID will show to the callee; \( nnnn \) automatically becomes a valid calling number. Also, \( nnnn \) can be composed with negation, subrule matching and range. The ‘D’ rule is checked before the ‘R’ and ‘P’ rules.

Example

\[ D911 \]

This rule shows the caller ID at the remote side when the call number is 911.

Log Information

\( SCC \ Cmd[:CLIP] \) or \( CLIP:<num> \) are shown in the prserv log.

Dial Plan Examples

This section contains three dial plan examples that use many different rules and commands.

Dial Plan Example 1 (Default Dial Plan)

The following dial plan:

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

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 four seconds.
- \( #St4 \)—Same as above, except with \# as the initial digit entered.
- \( 911 \)—If the dial string 911 is entered, send it immediately.
Dial Plan Parameters

- **1#t8.r9t2**—If the first digit entered is 1, the timeout before automatic send is eight 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 an automatic send changes to two 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 eight seconds, and the terminating character # can be entered at any time to manually send the dial string. If the first three digits entered are 011, then, after an additional 11 digits are entered, the timeout before an automatic send changes to four seconds. The user can enter more digits until the dial string is sent by the timeout or by the user entering the # character.

- **^t14>#.**—If the first digit entered is anything other than 1, the timeout before an automatic send is four 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 Example 2**

The following dial plans:

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

or

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

consist 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 an automatic send is seven seconds, and the terminating character # can be entered at any time to manually send the dial string. After seven digits are entered, the timeout before an automatic send changes to two seconds. The - symbol 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 this string immediately.

- **1t7>#.r9t1**—If the first digit entered is 1, the timeout before an automatic send is seven 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 an automatic send changes to one 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 an automatic send is four seconds, and the terminating character # can be entered at any time to manually send the dial string. After the second digit is entered, the timeout before an automatic send changes to seven seconds. The user can enter more digits until the dial string is sent by the timeout or by the user entering the # character.

**Dial Plan Example 3**

The following dial plan:

R1408([2_9].r5|[2_9].r6)|R9^(911|.r4)|X(1900|1888)|F011

consists of the following rules:

- **R1408([2_9].r5|[2_9].r6)**—The prefix 1408 will be added to any call numbers with seven or eight digits where the leading digit is in the range of 2 to 9. For example, 5551234 will become 14085551234, but 555123 does not match this rule.

- **R9^(911.r4)**—The prefix 9 will be added to any numbers except 911 and five-digit numbers. For example, 912 will still be 911, and 51234 will still be 51234.
- **X(1900|1888)**—There will be no calls or call forwarding to numbers beginning with 1900 or 1888.
- **F011**—There will be no call forwarding to numbers beginning with 011.

In Dial Plan Example 3, there are two ‘R’ rules, so the first matched rule is triggered. Therefore, 5551234 becomes 14085551234. However, 555123 will then become 9555123 because it matches the second rule.

### DialPlanEx

If your dial plan exceeds 199 characters, then use must use the DialPlanEx parameter to configure your dial plans. The DialPlanEx parameter supports dial plans up to 499 characters in length. This range in the number of characters is the only difference between the DialPlanEx and DialPlan parameters. Therefore, all the information about the DialPlan parameter applies to the DialPlanEx parameter. For more information, see the “DialPlan” section on page 5-64.

**Note**

If you are not using this parameter for dial plan configuration, be sure this parameter is set to 0.

### IPDialPlan

**Description**

This parameter allows for detection of IP-like destination address in DialPlan. Three values are valid:

- **0**—String is dialed as is and not treated as an IP address.
- **1**—When the Cisco ATA detects two asterisks (**), IPDialPlan takes over. The user enters the pound (#) key to terminate the digit collection, and the interdigit timeout default is not used.
- **2**—When IPDialPlan is set to 2, three asterisks (***) are required for IPDialPlan to take effect.

All other values are currently undefined.

**Value Type**

Integer

**Range**

0, 1 or 2

**Default**

1

**Voice Configuration Menu Access Code**

310
Diagnostic Parameters

This section describes the following parameters, which are used for diagnostic purposes:

- NPrintf, page 5-73
- TraceFlags, page 5-73
- SyslogIP, page 5-74
- SyslogCtrl, page 5-75

NPrintf

Description
Use this parameter to specify the IP address and port of a host to which all Cisco ATA debug messages are sent. The program `prserv.exe`, which comes bundled with the Cisco ATA software, is needed to capture the debug information.

Syntax

`<HOST_IP>,<HOST_PORT>`

Example

If the program `prserv.exe` is running on a host with IP address 192.168.2.170 and listening port 9001, set NPrintf to `192.168.2.170,9001`. This causes the Cisco ATA to send all debug traces to that IP address.

Value Type
Extended IP address

Default
0

Voice Configuration Menu Access Code
81

TraceFlags

Description
Use this parameter to turn on specific trace features for diagnostic use. Bit values are as follows:

- Bits 0 to 1:
  - 0 (default) is for simple debug messages.
  - 1 is for detailed debug messages.
  - 2 and 3 are reserved.
- Bits 2-7—Reserved
- Bit 8—RTP statistics log (values 0/1 to disable/enable with default of 0) has the following format:
  
  ```
  Recv[channel number]: <call duration in seconds> <number of recv packets> <number of recv octets> <number of late packets>
  ```
Diagnostic Parameters

<number of lost packets> <average network jitter in 1/8 ms>
<count> in speeding up local clock (adjustment for 10 ms each time)
<count> in slowing down local clock (adjustment for 10 ms each time)

Note  Bit 8 is not used at this time.

- Bits 9 to 31—Reserved.

Value Type
Bitmap

Default
0x00000000

Voice Configuration Menu Access Code
313

SyslogIP

Description
Use this parameter for diagnostic purposes; specify the IP address and port number to which the Cisco ATA should send its syslog output information.

The program prserv.exe, which is included in all Cisco ATA software upgrade packages, can be used to capture syslog information if you do not have a syslog server.

Syntax
<HOST_IPaddress>.<HOST_PORT>

Example
If you want to send syslog information to the host at IP address 192.168.2.170 and port number 514, do the following:
- Configure the value of this parameter as 192.168.2.170.514
- On your PC, run the command:
  prserv 514

Value Type
Extended IP address
Default
0.0.0.514

Voice Configuration Menu Access Code
7975640

Related Parameter
SyslogCtrl, page 5-75

**SyslogCtrl**

**Description**
Use this parameter to turn on specific syslog traces. All traces are sent to the syslog server specified in the SyslogIP parameter.

See Table 5-12 for bit values and the corresponding types of messages to turn on for tracing.

**Value Type**
Bitmap

Default
0x00000000

Voice Configuration Menu Access Code
7975641

Related Parameter
SyslogIP, page 5-74

**Table 5-12  SyslogCtrl Parameter Definitions**

<table>
<thead>
<tr>
<th>Bit Number</th>
<th>Type of Messages to Trace</th>
</tr>
</thead>
<tbody>
<tr>
<td>0</td>
<td>ARP messages.</td>
</tr>
<tr>
<td>1</td>
<td>DHCP messages</td>
</tr>
<tr>
<td>2</td>
<td>TFTP messages</td>
</tr>
<tr>
<td>3</td>
<td>Cisco ATA configuration-update messages.</td>
</tr>
<tr>
<td>4</td>
<td>System reboot messages</td>
</tr>
<tr>
<td>5-7</td>
<td>Reserved.</td>
</tr>
<tr>
<td>8</td>
<td>SIP messages</td>
</tr>
<tr>
<td>9</td>
<td>Cisco ATA event messages.</td>
</tr>
<tr>
<td>10</td>
<td>FAX messages.</td>
</tr>
<tr>
<td>11-15</td>
<td>Reserved.</td>
</tr>
<tr>
<td>16</td>
<td>RTP statistics messages.</td>
</tr>
<tr>
<td>17-31</td>
<td>Reserved.</td>
</tr>
</tbody>
</table>
CFGID—Version Parameter for Cisco ATA Configuration File

Description
CFGID is a 32-bit unsigned-value parameter whose purpose is to allow the local administrator to track the version of the Cisco ATA configuration file. This parameter-value assignment is entirely the responsibility of the local administrator, and has no significance to the operation of the Cisco ATA.

Value Type
Bitmap

Default
0x00000000
Call Commands

This section provides detailed information on call commands for the Cisco ATA:

- Call Command Structure, page 6-1
- Syntax, page 6-2
- Call Command Example, page 6-5
- Call Command Behavior, page 6-7

Service providers can offer many supplementary services, which can be activated, configured, or deactivated in more than one way. The CallCmd parameter allows you to define the behavior of supplementary services that the Cisco ATA supports.

---

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

---

This section contains call command information for the United States and Sweden. For information about other countries, contact the Cisco equipment provider for a specific country.

Call Command Structure

The entry in the CallCmd field is a character string composed of a sequence of instructions, which consist of a combination of three elements:

- Context—The Cisco ATA supplementary service operation is dependent upon a state and transition process. For example, the most common state is IDLE, in which the Cisco ATA is on-hook, waiting for an incoming call. Picking up the telephone handset causes the Cisco ATA to transition to the PREDIAL state, in which the user hears a dial tone and the Cisco ATA is waiting to detect DTMF digits. The Context portion of a Call Command string specifies the state for which the commands are defined.

- 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 taken by the Cisco ATA. The action depends on the Input-Sequence that the user enters and the Context in which it is entered.
The `CallCmd` string has the following structure:

**Context-Identifier Command . . . Command; . . . Context-Identifier Command;**

- **Table 6-1** provides a list of Context-Identifiers, which show the state of the Cisco ATA.
- **Command** consists of the following items:
  - **Input-Sequence; Action-Identifier-1 Action-Identifier-2 [Input-Sequence]**
    - **Input Sequence** consists of one or more characters from the set shown in **Table 6-2**.
    - **Table 6-3** provides a list of Action Identifiers. **Action-Identifier-1** is for the first thread of a call; **Action-Identifier-2** is for the second thread of a call. Each Action Identifier is one character.

Each Context-Identifier is followed by one or more commands to allow a variable number of actions to be triggered by relevant user input commands for any state. Each command is composed of an Input-Sequence that the user enters when the Cisco ATA is in a given state and two Action-Identifier characters which define the action that the Cisco ATA performs in response to the Context-Identifier and Input-Sequence. If the Cisco ATA takes only one action, one of the two Action-Identifier characters is a null action.

**Example 6-1 Syntax Example Using One Command**

Af;AH;

In this simple example, the first “A” is the Context-Identifier, which means the Cisco ATA is in the CONFERENCE state, as shown in **Table 6-1**. The “f” is the input sequence, which is hook-flash, as shown in **Table 6-2**. Following the semicolon, the two action identifiers are “A” and “H”. These identifiers mean “NONE” and “Disconnect the call,” respectively, as shown in **Table 6-3**. Based on these action identifiers, the Cisco ATA disconnects the most recent callee, and remains connected to the first party. The state of the Cisco ATA becomes CONNECTED. **Table 6-4** explains more about the various states of the Cisco ATA.

**Example 6-2 Syntax Example Using Two Commands**

CN;CAf;OF;

In this example, the first “C” is the Context Identifier, which means the Cisco ATA is in the PREDIAL_HOLDING state, as shown in **Table 6-1**. The “N” is the first input sequence, which is any part of the set of digits 0|1|2|3|4|5|6|7|8|9, as shown in **Table 6-2**. Following the first semicolon, the two action identifiers are “C” and “A”, which mean “Continue to Dial” and “NONE,” respectively, as shown in **Table 6-3**. Following this pair of action identifiers is another input sequence, “f”, which means hook-flash, as shown in **Table 6-2**. Next is the semicolon, always required after the input sequence, followed by the corresponding action pair, “O” and “F”. These identifiers mean “Release the Call” and “Retrieve the Call,” respectively, as shown in **Table 6-3**.
## Context-Identifiers

### Table 6-1  Context-Identifiers

<table>
<thead>
<tr>
<th>Identifier</th>
<th>Context (State of Cisco ATA)</th>
</tr>
</thead>
<tbody>
<tr>
<td>A</td>
<td>CONFERENCE</td>
</tr>
<tr>
<td>B</td>
<td>PREDIAL</td>
</tr>
<tr>
<td>C</td>
<td>PREDIAL_HOLDING</td>
</tr>
<tr>
<td>D</td>
<td>CONNECTED</td>
</tr>
<tr>
<td>E</td>
<td>CONNECTED_HOLDING</td>
</tr>
<tr>
<td>F</td>
<td>CONNECTED_ALERTING</td>
</tr>
<tr>
<td>G</td>
<td>HOLDING</td>
</tr>
<tr>
<td>H</td>
<td>CONFIGURING</td>
</tr>
<tr>
<td>I</td>
<td>CONFIGURING_HOLDING</td>
</tr>
<tr>
<td>J</td>
<td>3WAYCALLING</td>
</tr>
<tr>
<td>K</td>
<td>CALLWAITING</td>
</tr>
<tr>
<td>L</td>
<td>IDLE</td>
</tr>
<tr>
<td>M</td>
<td>RINGING</td>
</tr>
<tr>
<td>N</td>
<td>DIALING</td>
</tr>
<tr>
<td>O</td>
<td>CALLING</td>
</tr>
<tr>
<td>P</td>
<td>Reserved (ANSWERING)</td>
</tr>
<tr>
<td>Q</td>
<td>Reserved (CANCELING)</td>
</tr>
<tr>
<td>R</td>
<td>Reserved (DISCONNECTING)</td>
</tr>
<tr>
<td>S</td>
<td>WAITHOOK</td>
</tr>
<tr>
<td>T</td>
<td>DIALING_HOLDING</td>
</tr>
<tr>
<td>U</td>
<td>CALLING_HOLDING</td>
</tr>
<tr>
<td>V</td>
<td>Reserved (ANSWERING_HOLDING)</td>
</tr>
<tr>
<td>W</td>
<td>Reserved (HOLDING_HOLDING)</td>
</tr>
<tr>
<td>X</td>
<td>Reserved (CANCELING_HOLDING)</td>
</tr>
<tr>
<td>Y</td>
<td>Reserved (DISCONNECTING_HOLDING)</td>
</tr>
<tr>
<td>Z</td>
<td>Reserved (HOLDING_ALERTING)</td>
</tr>
<tr>
<td>a</td>
<td>WAITHOOK_ALERTING</td>
</tr>
<tr>
<td>b</td>
<td>WAITHOOK_HOLDING</td>
</tr>
</tbody>
</table>
Input Sequence Identifiers

<table>
<thead>
<tr>
<th>Identifier</th>
<th>Input Sequence</th>
</tr>
</thead>
<tbody>
<tr>
<td>0-9,#*</td>
<td>DTMF digits</td>
</tr>
<tr>
<td>f</td>
<td>hook flash</td>
</tr>
<tr>
<td>o</td>
<td>off-hook</td>
</tr>
<tr>
<td>@</td>
<td>anytime; for example, @f means anytime hook- flash occurs</td>
</tr>
<tr>
<td>h</td>
<td>on-hook</td>
</tr>
<tr>
<td>S</td>
<td>#</td>
</tr>
<tr>
<td>N</td>
<td>0</td>
</tr>
<tr>
<td>D</td>
<td>N</td>
</tr>
<tr>
<td>v</td>
<td>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.</td>
</tr>
</tbody>
</table>

Action Identifiers

<table>
<thead>
<tr>
<th>Identifier</th>
<th>Action</th>
</tr>
</thead>
<tbody>
<tr>
<td>A</td>
<td>NONE</td>
</tr>
<tr>
<td>B</td>
<td>Seizure (User intends to dial or configure)</td>
</tr>
<tr>
<td>C</td>
<td>Continue to dial</td>
</tr>
<tr>
<td>D</td>
<td>Call Return</td>
</tr>
<tr>
<td>E</td>
<td>Hold the active call</td>
</tr>
<tr>
<td>F</td>
<td>Retrieve the waiting call</td>
</tr>
<tr>
<td>G</td>
<td>Cancel the call attempt</td>
</tr>
<tr>
<td>H</td>
<td>Disconnect the call</td>
</tr>
<tr>
<td>I</td>
<td>Blind transfer the call to the number</td>
</tr>
<tr>
<td>N</td>
<td>Go to configuration mode</td>
</tr>
<tr>
<td>O</td>
<td>Release the call</td>
</tr>
<tr>
<td>P</td>
<td>Answer the incoming call</td>
</tr>
<tr>
<td>Q</td>
<td>Transfer with consultation</td>
</tr>
<tr>
<td>R</td>
<td>Say busy to the caller</td>
</tr>
<tr>
<td>a</td>
<td>None</td>
</tr>
<tr>
<td>b</td>
<td>Forward all calls to the given number</td>
</tr>
</tbody>
</table>
In addition to call commands that you configure, the Cisco ATA has a default list of call commands to handle common call scenarios. Configured call commands overwrite default call commands. If any Context-Identifier or Input-Sequence elements appear in both the default Call Command string and the manually entered string, the manually entered value takes precedence.

The following string shows a sample Call Command:

```
Bf;BAN;CA;OF;Df;EB;I@f;OF;H@f;OA;Lo;Baf;Ba;Mo;PA;ND;Caf;Of;Ga;Pf;Ha;Qf;OA;Rf;OA;Sf;OA;TD;Caf;Of;lf;Gf;Vf;Hf;Wf;Xf;Af;Zf;Ap;Bf;Of;Af;Op;
```

In this section, the Call Command string is broken down into its components as follows:

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

**Note**

If you use a second input sequence, this sequence follows the Action Identifier pair without a separating semicolon.

Refer to the preceding tables to determine the meanings of the identifiers.

### Example 6-3 Call Command String

```
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 Behavior

Table 6-4 summarizes differing Call Command behavior based on the U.S. and Sweden default call commands.

### U.S. Call Command Default

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

### Sweden Call Command Default

BS;NA;CS;NA;DF;EB;Ff0;ARf1;HPf2;EPf3;AP;Kf1;HFf2;EFf3;AFf4;HQh;HH;Jf1;HFf2;EFf3;AFf4;HQ;AF4;HQ;I*31#;gA#31#;gA*90*v#;O1;H*21*v#;bA*61*v#;dA*67*v#;cA#21#;eA#61#;eA#67#;eA*31#;gA#31#;gA*43#;hA#43#;hA*69#;DA*99#;xA;Uh;GQ;

### Table Notations

The following notations are used in Table 6-4:

- FE—Far end
- AFE—Active Far End, which is a connected far end that is not placed on hold
- WFE—Waiting Far End, which is a connected far end 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

### Note

The notations in Table 6-4 include abbreviations for input sequence behavior. Refer to the tables and syntax examples shown earlier in this section. The Summary of Commands column in Table 6-4 is based on the actual command syntax used in the default Call Command strings for the United States and Sweden.

### Table 6-4  Call Command Behavior

<table>
<thead>
<tr>
<th>Cisco ATA State and its Definition</th>
<th>Summary of Commands (Input Sequence and Actions)</th>
</tr>
</thead>
</table>
| IDLE: Phone is on-hook; Cisco ATA is waiting for incoming call | • OFH—Start dial tone and go to PREDIAL state.  
• New incoming call or a waiting call (started before it enters IDLE)—Start ringing the phone and go to the RINGING state. |
| PREDIAL: Phone just went off-hook but no DTMF has been entered yet; Cisco ATA plays dial-tone | United States and Sweden:  
• # , *—Stop dial-tone, go to the CONFIG state, and prepare to accept a complete configuration sequence.  
• 0-9: Stop dial tone, start invoking dial-plan rules, and go to the DIALING state to accept a complete phone number. |
### Call Command Behavior

#### Cisco ATA State and its Definition

**DIALING:** User is entering phone number, which is parsed with the given dial-plan rules

- R — Abort dialing, restart dial tone, and revert to PREDIAL state.
- Invalid phone number — Abort dialing, plays fast-busy, and go to WAITHOOK state.

**CONFIG:** User configuring a supplementary service in the United States

- *69 — Call Return
- #72v# — Forward unconditional to number specified in 'v' (PacBell uses 72#).
- #73 — Cancel any call forwarding (PacBell uses 73#).
- #74v# — Forward on busy to number specified in 'v' (PacBell does not enable this service from the phone).
- #75v# — Forward on no answer to number specified in 'v' (PacBell does not 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).
- Dial-tone — Revert to PREDIAL state.
- Any complete configuration sequence — Carry out the configuration command, restart dial-tone, and revert to PREDIAL state.

**CONFIG:** User configuring a supplementary service in 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.
- #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 Regions:**

- R or any unrecognized sequence — Abort configuration, restart dial tone and revert to PREDIAL state.
- Any complete configuration sequence — Carry out the configuration command, restart dial tone, and revert to PREDIAL state.

### Table 6-4 Call Command Behavior (continued)

<table>
<thead>
<tr>
<th>Cisco ATA State and its Definition</th>
<th>Summary of Commands (Input Sequence and Actions)</th>
</tr>
</thead>
</table>
| **DIALING:** User is entering phone number, which is parsed with the given dial-plan rules | • R — Abort dialing, restart dial tone, and revert to PREDIAL state.  
• Invalid phone number — Abort dialing, plays fast-busy, and go to WAITHOOK state. |
| **CONFIG:** User configuring a supplementary service in the United States | • *69 — Call Return  
• #72v# — Forward unconditional to number specified in 'v' (PacBell use 72#).  
• #73 — Cancel any call forwarding (PacBell use 73#).  
• #74v# — Forward on busy to number specified in 'v' (PacBell does not enable this service from the phone).  
• #75v# — Forward on no answer to number specified in 'v' (PacBell does not 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).  
• Dial-tone — Revert to PREDIAL state.  
• Any complete configuration sequence — Carry out the configuration command, restart dial-tone, and revert to PREDIAL state. |
| **CONFIG:** User configuring a supplementary service in 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.  
• #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 Regions:** | • R or any unrecognized sequence — Abort configuration, restart dial tone and revert to PREDIAL state.  
• Any complete configuration sequence — Carry out the configuration command, restart dial tone, and revert to PREDIAL state. |
### Call Command Behavior (continued)

<table>
<thead>
<tr>
<th>Cisco ATA State and its Definition</th>
<th>Summary of Commands (Input Sequence and Actions)</th>
</tr>
</thead>
<tbody>
<tr>
<td>CALLING: Phone number is sent; Cisco ATA is waiting for response from the far end</td>
<td>• R—Cancel the outgoing call, restarts dial-tone, and revert to PREDIAL state.</td>
</tr>
<tr>
<td>RINGING: Cisco ATA is ringing the phone to alert user of an incoming call</td>
<td>• OFH—Stop ringing, answer the call, and go to CONNECTED state.</td>
</tr>
</tbody>
</table>
| CONNECTED: The Cisco ATA is connected with one far end party; Cisco ATA may be the caller or the callee | United States and Sweden:  
  • R—Hold current call, play dial-tone to dial second number, and go to PREDIAL_HOLDING state. |
| WAITHOOK: Far end hangs up while in CONNECTED state; Cisco ATA plays fast-busy after five seconds in this state | • R—Stop fast-busy, start dial-tone, and go to PREDIAL state. |
| CONNECTED_ALERTING: Cisco ATA receives another call while in CONNECTED state; Cisco ATA plays Call Waiting tone periodically (every 10 seconds for US; every second for Sweden) | United States:  
  • R—Place current call on-hold, answer the waiting call, and go to CALLWAITING state.  
Sweden:  
  • R0—Continue current call, reject the waiting call, and revert to CONNECTED state.  
  • R1—Disconnect current call, answer the waiting call, and go to CONNECTED state.  
  • R2—Place current call on-hold, answer waiting call, and go to CALLWAITING state.  
  • R3—Continue with current call, answer the waiting call and go to CONFERENCE state.  
All Regions:  
  • ONH—Disconnect current call and go to IDLE state (the Cisco ATA then automatically starts ringing the phone, and goes to RINGING state).  
  • AFE hangs up—Go to WAITHOOK_ALERTING state, continue to play CWT.  
  • WFE cancels the call—Stop CWT and revert to CONNECTED state. |
Table 6-4  Call Command Behavior (continued)

<table>
<thead>
<tr>
<th>Cisco ATA State and its Definition</th>
<th>Summary of Commands (Input Sequence and Actions)</th>
</tr>
</thead>
</table>
| CALL WAITING: Cisco ATA is connected to two far end users on the same line; one 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 is connected to one of the far ends while the other far end calls into the Cisco ATA. | United States:  
• R—Place the AFE on-hold and retrieve the WFE.  
• ONH—The default behavior is to disconnect both the current call and the held call, then go to the PREDIAL state.  
The enhanced behavior (with the call-waiting hang-up alert feature enabled) is to disconnect the current call, retrieve the held call, go to the CONNECTED state, then alert the user of the call on hold by ringing the phone.  
Sweden:  
• R1—Disconnect current call, answer the waiting call, and go to CONNECTED state.  
• R2—Place the AFE on-hold and retrieve the WFE.  
• R3—Retrieve the WFE, and go to CONFERENCE state.  
• R4—Transfer the WFE to the AFE, drop out of the call, and go to PREDIAL state. |
| 3WAYCALLING: Cisco ATA 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 is connected to one of the far ends, then places this far end on hold and calls the second far end. | United States:  
• R—Retrieve the WFE and go to CONFERENCE state.  
• ONH—Transfer the WFE to the AFE, drop out of the call, and go to PREDIAL state.  
Sweden:  
• Same as for CALLWAITING state |
| CONFERENCE: Cisco ATA is connected to two active far ends simultaneously; Cisco ATA performs audio mixing such that every party can hear the other two parties but not themselves. | United States:  
• R—Disconnect the last callee and stay connected with the first party, and revert to CONNECTED state.  
Sweden:  
• R4—Transfer one FE to the other, drop out of the call, and go to PREDIAL state. |
| PREDIAL_HOLDING: Cisco ATA user places a connected call on-hold and prepares to dial a second number; Cisco ATA plays dial-tone. | United States and Sweden:  
• *,#—Stop dial-tone, go to CONFIG_HOLDING state, and prepare to collect a configuration command.  
• 0-9—Stop dial-tone, go to DIALING_HOLDING state, and prepare to complete dialing a second phone number.  
All Regions:  
• Stop dial-tone, retrieve the WFE, and revert to CONNECTED state. |
Table 6-4  Call Command Behavior (continued)

<table>
<thead>
<tr>
<th>Cisco ATA State and its Definition</th>
<th>Summary of Commands (Input Sequence and Actions)</th>
</tr>
</thead>
</table>
| CONFIG_HOLDING: A connected FE is placed on hold, while the Cisco ATA is entering a configuration command. | United States:  
- *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 go to PREDIAL state.  

Sweden:  
- #31# or *31#—CLIR in the next call  
- *90*v#—Blind transfer to the number specified in 'v'; disconnect the call and go to PREDIAL (non-standard) state.  

All Regions:  
- R or any unrecognized sequence—Abort configuration, restart dial tone, and go to PREDIAL_HOLDING state.  
- A complete configuration sequence—Carry out the command, and go to PREDIAL_HOLDING state. |
| DIALING_HOLDING: Cisco ATA user is 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 go to CALLING_HOLDING state  
- R—Abort dialing and revert to PREDIAL_HOLDING state. |
| CALLING_HOLDING: Cisco ATA is waiting for a second far end to respond while placing a connected call on hold | R—Cancel the call and revert to PREDIAL_HOLDING state.  
ONH—Cancel the call and transfer the waiting party to the callee, and revert back to PREDIAL state. |
| 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 go to CONNECTED state. |
| AITHOOK_ALERTING: The AFE hangs up while a waiting call alerts | R—Stop CWT, answer the waiting call, and go to CONNECTED state.  
WFE: Cancel the call; stop CWT, go to WAITHOOK state.  
ONH—Go to IDLE state (in which Cisco ATA automatically starts ringing the phone, and goes to RINGING state). |
Configuring and Debugging Fax Services

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

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

With fax mode, the Cisco ATA presents itself as a device capable of using only G.711 codecs; therefore, no codec renegotiation or switchover is required. This places minimum functionality and configuration requirements on remote gateways. Fax mode is recommended for environments in which G.711 fax upspeed is not available for the supporting Cisco gateways.

This section contains the following topics:

- Using Fax Pass-through Mode, page 7-1
- Using FAX Mode, page 7-6
- Debugging the Cisco ATA 186/188 Fax Services, page 7-7

Note

The term Cisco ATA is used throughout this manual to refer to both the Cisco ATA 186 and the Cisco ATA 188, unless differences between the Cisco ATA 186 and Cisco ATA 188 are explicitly stated.

Using Fax Pass-through Mode

Fax pass-through mode allows for maximum codec flexibility because users may set up a voice call using any voice codec, then renegotiate to a G.711 codec for the fax session. To use fax pass-through mode, first configure the Cisco ATA and supporting Cisco gateways to support the Cisco-proprietary G.711 fax upspeed method. Then, disable fax relay on the far-end gateway—for the entire gateway or for the dial peer engaged in the fax call with the Cisco ATA.

The fax upspeed method allows you to use low bit-rate codecs such as G.723 and G.729 for voice calls, and G.711 codecs for fax calls. With a fax call, the Cisco ATA detects a 2100-Hz CED tone or V.21 preamble flag, then informs the remote gateway of its intent to switchover to G.711 via a peer-to-peer message. This type of message, carried as a Named Signaling Event (NSE) within the RTP stream, is used for all fax event signaling. The Cisco ATA can initiate and respond to NSEs and can function as either an originating or terminating gateway.
The Cisco ATA can also accept standard-based protocol-level codec switch requests, but cannot send such requests. Therefore, to interoperate with a Cisco gateway, use the Cisco-proprietary codec switch.

This section contains the following topics:
- Configuring the Cisco ATA for Fax Pass-through mode, page 7-2
- Configuring Cisco IOS Gateways to Enable Fax Pass-through, page 7-3

**Configuring the Cisco ATA for Fax Pass-through mode**

*Fax Pass-through mode* requires configuring two configuration parameters:
- AudioMode, page 7-2
- ConnectMode, page 7-3

### AudioMode

**Description**
The AudioMode parameter is a 32-bit value. The lower 16 bits apply to the Phone 1 port of the Cisco ATA and the upper 16 bits apply to the Phone 2 port of the Cisco ATA.

**Example**
The following is an example of configuring the Phone 1 port of the Cisco ATA for fax pass-through mode:

```
0xXXXX0015
```

**Translation**
This setting translates to the following bitmap:

```
xxxx xxxx xxxx xxxx 0000 0000 0001 0101
```

- Bit 0 = 1—Enables G.711 silence suppression (VAD)
- Bit 2 = 1—Enables Fax CED tone detection and switchover upon detection
- Bit 4 = 1, Bit 5 = 0—DTMF transmission method = out-of-band through negotiation
- Bit 6 = Bit 7 = 0—Hookflash transmission method = disable sending out hookflash

**Note**
The values xxxx in the example apply to the Phone 2 port of the Cisco ATA.

To configure the same value for the Phone 2 port of the Cisco ATA, the value would be 0x0015XXXX. The configuration of one port is independent from the configuration of the other port.
ConnectMode

Description
The ConnectMode parameter is a 32-bit value. The parameter settings apply to both lines of the Cisco ATA. Configure ConnectMode after configuring AudioMode for fax pass-through mode. Cisco recommends you use the following ConnectMode setting to interoperate with a Cisco IOS gateway.

Recommended Setting
0x90000400

Translation
This setting translates to the bitmap:
1001 0000 0000 0000 0000 0100 0000 0000

Bit 2 and bits 7 through 15 are the only relevant bits for fax pass-through mode. These bits from the example are isolated below:
xxxx xxxx xxxx xxxx 0000 0100 0xxx x0xx

- Bit 2 = 0—Uses RTP payload number 126/127 for fax upspeed to G.711μ-law/G.711A-law. Set this value to 1 if you want to use RTP payload number 0/8 for fax upspeed.
- Bit 7 = 0—Disables fax pass-through redundancy. Set this bit to 1 to enable redundancy. With redundancy enabled, the Cisco ATA sends each packet twice. Because of bandwidth and transmission time costs, use this option only if network quality is poor and all other gateways used in the network support this feature.
- Bits {12, 11, 10, 9, 8} = {0, 0, 1, 0, 0}—Sets the offset to NSE payload-type number 96 to 4. Setting the offset to 4 results in the Cisco ATA sending an NSE payload-type value of 100 by default. Valid offset values range from 2 to 23 (NSE payload type value of 98 to 119). Set this value to match the value for your Cisco gateways.
Most Cisco MGCP-based gateways, such as Cisco 6608, use NSE payload type 101 by default. Most Cisco H.323/SIP-based gateways use NSE payload type 100 by default.
- Bit 13 = 0—Uses G.711μ-law for fax pass-through upspeed. Set this bit to 1 to use G.711A for fax pass-through upspeed.
- Bit 14 = Bit 15 = 0—Enables fax pass-through mode using the Cisco proprietary method (recommended). Set both of these bits to 1 to disable fax pass-through mode.

Configuring Cisco IOS Gateways to Enable Fax Pass-through

To configure your IOS gateways to network with Cisco ATA, do the following:

Procedure

<table>
<thead>
<tr>
<th>Step 1</th>
<th>Enable Fax Pass-through Mode, page 7-4</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Disable Fax Relay Feature, page 7-5</td>
</tr>
</tbody>
</table>
Using Fax Pass-through Mode

For detailed information on setting up your IOS gateways and on feature availability, refer to the document Cisco Fax Services over IP.

Enable Fax Pass-through Mode

The supporting Cisco gateway can enable fax pass-through mode using system-level or dial-peer-level commands.

System Level commands

Enable the fax pass-through feature using the following system-level commands:

Procedure

Step 1 Run the following command:

```text
voice service voip
```

Step 2 Run the following command:

```text
modem passthrough NSE [payload-type number] codec {g711μ/law | g711alaw} [redundancy] [maximum-sessions value]
```

The definitions of the command parameters are as follows:

- The **payload-type** parameter default is 100. Valid values are from 98 to 119.
  The NSE payload number must be the same on both the Cisco ATA and the Cisco gateway.
- The **codec** parameter must be G.711μ-law for faxes sent over a T1 trunk or G.711A-law for faxes sent over an E1 trunk.
- The **redundancy** parameter enables RFC 2198 packet redundancy. It is disabled by default.
- The **maximum sessions** parameter defines the number of simultaneous fax pass-through calls with redundancy. The default is 16. Valid values are 1 to 26.

Step 3 For the Cisco ATA ConnectMode parameter, turn off bits 14 and 15. This enables the sending of fax pass-through signals and the detection of incoming fax pass-through signals using the Cisco proprietary method.

Note The NSE payload-type number, fax pass-through codec (G.711μ-law or G.711A-law) and redundancy parameters must have the same settings for the Cisco ATA that they have for supporting Cisco gateways.
Dial-Peer Level Commands

You can enable *fax pass-through mode* for communication between a Cisco IOS gateway and the specified Cisco ATA using the following dial-peer level commands:

**Procedure**

<table>
<thead>
<tr>
<th>Step 1</th>
<th>Perform the command:</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>dial-peer voice tag voip</strong></td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Step 2</th>
<th>Perform the command:</th>
</tr>
</thead>
<tbody>
<tr>
<td>**modem passthrough {NSE [payload-type number] codec {g711μlaw</td>
<td>g711alaw} [redundancy]</td>
</tr>
<tr>
<td>a. The default of this command is:</td>
<td></td>
</tr>
<tr>
<td><strong>modem passthrough system</strong></td>
<td></td>
</tr>
<tr>
<td>When using the default configuration, the dial-peer fax pass-through configuration is defined by the <strong>voice service voip</strong> command. When the <strong>system</strong> option is used, no other parameters are available.</td>
<td></td>
</tr>
<tr>
<td>When the NSE is configured in the fax pass-through command at the dial-peer level, the fax pass-through definition in the <strong>dial-peer</strong> command takes priority over the definition in the <strong>voice service voip</strong> command.</td>
<td></td>
</tr>
<tr>
<td>b. The <strong>payload-type number</strong>, <strong>codec</strong>, and <strong>redundancy</strong> parameters can also be used.</td>
<td></td>
</tr>
<tr>
<td>For example, the command:</td>
<td></td>
</tr>
<tr>
<td><strong>modem passthrough NSE codec g711μlaw</strong></td>
<td></td>
</tr>
<tr>
<td>means that the Cisco ATA will use the NSE payload-type number 100, G.711μ-law codec, and no redundancy in <em>fax pass-through mode</em>.</td>
<td></td>
</tr>
</tbody>
</table>

| Step 3 | When setting up dial-peer for fax pass-through, it is necessary to set up a pair of dial-peers for inbound and outbound calls between the Cisco ATA and Cisco IOS gateways. You do this by specifying the **destination-pattern** and **incoming-called number**. The **destination-pattern** should point to the Cisco ATA, while the incoming-called number should apply to all numbers that the Cisco ATA is allowed to dial. |

**Disable Fax Relay Feature**

Fax relay may be enabled by default for some IOS gateways. If you do not disable the fax relay feature, it may override the precedence of fax/modem pass-through and cause the fax transmission to fail. It is necessary to disable fax relay at the dial-peer or system level with the following command:

**fax rate disable**
Using FAX Mode

Use fax mode when the gateways in the network do not support fax pass-through mode or dial-peer configuration.

You can set one or both lines of the Cisco ATA to G.711-only fax mode. This mode allows the fax machine connected to the Cisco ATA to communicate directly with the far endpoint with no fax signaling event occurring between the two gateways.

This section contains the following topics:
- Configuring the Cisco ATA for Fax Mode, page 7-6
- Configuring the Cisco ATA for Fax Mode on a Per-Call Basis, page 7-7
- Configuring the Cisco IOS Gateway for Fax Mode, page 7-7

Configuring the Cisco ATA for Fax Mode

G.711-only fax mode operation requires configuration of one parameter—AudioMode.

Description
The AudioMode parameter is a 32-bit value. The lower 16 bits apply to the Phone 1 port of the Cisco ATA, and the upper 16 bits to the Phone 2 port. The following is an example of the Phone 1 port of the Cisco ATA configured for G.711-only fax mode:

Example
0xXXXX0012

Translation
This setting translates to the bitmap:
xxxx xxxx xxxx xxxx 0000 0000 0001 0010

- Bit 0 = 0—Disables G.711 silence suppression (VAD).
- Bit 1 = 1—Uses G.711 only, does not use the low bit-rate codec.
- Bit 2 = 0—Disables Fax CED tone detection.
- Bit 4 = 1, Bit 5 = 0—DTMF transmission method: out-of-band through negotiation
- Bit 6 = Bit 7 = 0—Hookflash transmission method: disables sending out hookflash

Note
The values xxxx in the example do not apply to the Phone 1 port of the Cisco ATA.

To configure the same value for the Phone 2 port of the Cisco ATA, the value would be 0x0012XXXX.
The configuration of one port is independent from the configuration of the other port.

Note
The AudioMode configuration overrides the values of the following three parameters: RxCodec, TxCodec, and LBRCodec. For example, if these three parameters are each set to 0 (for G.723), the Cisco ATA would still use G.711 if AudioMode is set to 0x00120012. With this configuration, the Cisco ATA sends both G.711μ-law and G.711A-law as preferred codecs to a peer voice gateway.
Configuring the Cisco ATA for Fax Mode on a Per-Call Basis

**Note**
The per-call-basis *fax mode* feature is only available for the H.323 and SIP protocols.

If you want to activate *fax mode* on a per-call basis, configure the following parameters:

**Procedure**

1. **CallFeatures** and **PaidFeatures** Bit 15 (for line1—mask 0x8000) and Bit 31 (for line2—mask 0x80000000) = 1: This sets the default to enable *fax mode* on a per-call basis.
2. **AudioMode** Bit 2 = 0: This disables fax CED tone detection.
3. **CallCmd** includes *99;xA* (99 is the default; the value can be changed to any prefix code.)

To activate a call from your fax machine, enter *99* (default), then enter the telephone number to which you want to send the fax. The next call will automatically revert to normal mode.

Configuring the Cisco IOS Gateway for Fax Mode

On the Cisco gateway, disable both fax relay and fax pass-through at the dial-peer level or system level with the following commands:

**Procedure**

1. Run the command:
   ```
   fax rate disable
   ```
2. Run the command:
   ```
   no modem passthrough
   ```

Debugging the Cisco ATA 186/188 Fax Services

This section includes the following debugging topics for fax services:

- **Common Problems When Using IOS Gateways**, page 7-7
- **Using prserv for Diagnosing Fax Problems**, page 7-9
- **Using rtpcatch for Diagnosing Fax Problems**, page 7-12

Common Problems When Using IOS Gateways

Table 7-1 lists typical problems and actions that might solve these problems for situations in which the Cisco ATA is using fax over a Cisco IOS gateway.
### Table 7-1  Solving Common Fax Problems

<table>
<thead>
<tr>
<th>Problem</th>
<th>Action</th>
</tr>
</thead>
</table>
| The far-end gateway is not loaded with correct software image.         | Cisco recommends IOS version 12.2 (11)T or higher for the Cisco 2600 and Cisco 3600, and IOS version 12.1 (3)T or higher for Cisco AS5300.  
|                                                                        | The Cisco 6608 supports both the NSE and NTE methods of fax pass-through mode, beginning with software version D004030145S16608. To use fax pass-through mode with the Cisco 6608, the user must select 6608 NSE mode, and the NSE payload type must be reconfigured to match the Cisco ATA. |
| The Cisco IOS gateway is not configured using the external T1 clock.   | Perform these steps:  
|                                                                        | 1. Enter the following CLI commands:  
|                                                                        | Controller T1 0  
|                                                                        | clock source line  
|                                                                        | 2. On the Cisco CallManager Gateway Configuration page, choose the T1 line connection port. Set the clock as “external primary.” |
| The Cisco ATA is not loaded with the proper software.                  | Cisco recommends using software version 2.14 or higher.                                                                                                                                                 |
| User is operating Cisco ATA software on an outdated model.             | Cisco recommends using Cisco ATA models 186-I1, 186-I2, 188-I1, or 188-I2 (hardware platforms).                                                                                                         |
| The Cisco ATA is not configured for fax mode or fax pass-through mode. | For fax mode, the AudioMode configuration parameter should be set to 0xXXXX0012 ($X =$ value not applicable) for the Phone 1 port of the Cisco ATA, and 0x0012XXXX for the Phone 2 port.  
|                                                                        | For fax pass-through mode, AudioMode should be set to 0xXXXX0015 for the Phone 1 port of the Cisco ATA, and 0x0015XXXX for the Phone 2 port.                                                                |
| The remote gateway is not configured for modem/fax pass-through mode.  | When the Cisco ATA is configured for fax pass-through mode, all remote gateways must be configured with modem/fax pass-through mode either on a dial-peer level or system level.                        |
| Fax relay is not disabled on the remote gateway.                       | Fax relay is enabled by default on some Cisco gateways. When fax relay is enabled, it can override fax pass-through mode and cause fax failure. Examples of the CLI commands to disable fax relay for IOS gateways are as follows:  
|                                                                        | • fax rate disable for H.323/SIP gateways  
|                                                                        | • mgcp fax t38 inhibit for MGCP gateways |
Using prserv for Diagnosing Fax Problems

This section contains the following topics:

- prserv Overview, page 7-9
- Analyzing prserv Output for Fax Sessions, page 7-10

prserv Overview

prserv is a tool that runs on a Microsoft Windows-based PC and serves as a log server that captures debug information that the Cisco ATA sends to your PC IP address/port. The debug information is saved into a readable text file.

To enable your Cisco ATA to send debug information, you need to set the NPrintf configuration parameter to your PC IP address and an available port, as shown in the following procedure:

Procedure

Step 1

<IP address>.<port>

<IP address> is the IP address of your PC.

<port> is any unused port (any number from 1024 to 65535) on your PC.

Note

You can the Nprintf parameter on the Cisco ATA configuration web page or with the TFTP-based configuration method.

Step 2

To operate the debug capture program prserv.exe, place the prserv program in a folder on your PC. At the DOS prompt, enter:

C:>prserv <port>

<port> is the port number you have selected. If <port> is omitted, the default port number is 9001.
Chapter 7  Configuring and Debugging Fax Services

Debugging the Cisco ATA 186/188 Fax Services

As prserv receives debug information from the Cisco ATA, it displays the information on the DOS screen and saves it to the output file <port>.log.

Once you are finished capturing debug information, you can stop prserv by entering Ctrl-C at the DOS prompt. If you restart the process without changing the name of the log file, any new debug information is appended to the end of the original file.

Analyzing prserv Output for Fax Sessions

The debug log obtained from prserv is for detecting simple configuration problems.

A comprehensive understanding of the fax events requires the use of the rtpcatch tool (see the “Using rtpcatch for Diagnosing Fax Problems” section on page 7-12).

Table 7-2 lists log events relevant to analyzing a fax session.

Table 7-2  Debug Log Examples

<table>
<thead>
<tr>
<th>Log event</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>[ch] Enable encoder &lt;pt&gt;</td>
<td>Voice encoder type pt is enabled for the channel ch, where pt can be 0 for G.711µ-law, 4 for G.723.1, 8 for G.711A-law, and 18 for G.729. For example, [0]Enable encoder 4 indicates that the Cisco ATA transmitted G.723.1-encoded voice packets.</td>
</tr>
<tr>
<td>[ch] DPKT 1st: &lt;&lt;timestamp1&gt;&gt;</td>
<td>The first voice packet that the Cisco ATA received was of RTP payload type pt for the channel ch with timestamp of timestamp1, and the local decoding timestamp was set to timestamp2. For example, [0]DPKT 1st: 1491513359 1491512639, pt 4 indicates that the first RTP packet that the Cisco ATA received was G.723.1-encoded for channel 0.</td>
</tr>
<tr>
<td></td>
<td></td>
</tr>
<tr>
<td>[ch] codec: &lt;pt1&gt; =&gt; &lt;pt2&gt;</td>
<td>Voice codec switchover occurred. The voice encoder type switched from pt1 to pt2 for the channel ch. For example, [0]codec: 4 =&gt; 0 indicates that the local voice encoder on the Cisco ATA switched from G.723.1 to G.711µ-law.</td>
</tr>
<tr>
<td>[ch] Rx MPT PT=&lt;NSEpt&gt; NSE pkt &lt;event&gt;</td>
<td>Channel ch received an NSE packet of event with payload type of NSEpt. For event, c0XXXXXX indicates a CED tone event, and c1XXXXXX indicates a phase reversal event. For example, [0]Rx MPT PT=100 NSE pkt c0000000 indicates that the Cisco ATA received a CED tone event NSE packet with payload type of 100.</td>
</tr>
<tr>
<td>[ch] Tx MPT PT=&lt;pt&gt; NSE pkt &lt;event&gt;</td>
<td>Channel ch transmitted an NSE packet of event with payload type of NSEpt. For event, c0XXXXXX indicates a CED tone event, and c1XXXXXX indicates a phase reversal event. For example, [0]Tx MPT PT=100 NSE pkt c0000000 indicates that the ATA transmitted a CED tone event NSE packet with payload type of 100.</td>
</tr>
</tbody>
</table>
Chapter 7      Configuring and Debugging Fax Services

Debugging the Cisco ATA 186/188 Fax Services

Debugging FAX Pass-through Mode
When the Cisco ATA is configured to use fax pass-through mode, the fax call session can be established with an arbitrary voice codec. Once the voice call has been established, fax machines can signal their presence by means of a CED tone or V.21 preamble flag, after which the gateways send NSE packets to initiate switchover.

Note
For fax pass-through mode, check the Cisco ATA debug log to verify that it is acting as an originating gateway as well as a terminating gateway.

Terminating-Gateway Example
When the Cisco ATA is used as a terminating gateway for a fax session, make sure the following conditions are true:

- The Cisco ATA transmits CED-tone-event NSE packets.
- The encoder switchover to G.711 occurs during the NSE-packet transaction.

An example debug log for a terminating gateway scenario is show below:

```
[0]Tx MPT PT=100 NSE pkt c0000000
[0]codec: 4 => 0
[0]Rx MPT PT=100 NSE pkt c0000000
```

Note
The NSE response to the CED tone event is not mandatory; some gateways may not send back an NSE response.

Originating-Gateway Example
When the Cisco ATA is used as an originating gateway for a fax session, make sure that the following conditions are true:

- The Cisco ATA receives and responds to CED-tone-event NSE packets.
- The NSE payload type is the same for the received and transmitted NSE packets.
- The encoder switchover to G.711 occurs during NSE-packet transaction.

An example debug log for an originating gateway scenario is shown below:

```
[0]Rx MPT PT=100 NSE pkt c0000000
[0]Tx MPT PT=100 NSE pkt c0000000
[0]codec: 4 => 0
[0]Rx MPT PT=100 NSE pkt c0000000
[0]Rx MPT PT=100 NSE pkt c0000000
```

Note
If your gateway is using a legacy IOS software image, it may not send NSE packets but instead may rely on a straightforward codec switchover mechanism. In this case, a codec switchover event occurs rather than an NSE packet transaction.
Possible Reasons for Failure

If your Cisco ATA does not receive CED-tone-event NSE packets and codec switchover does not occur, the failure may be due to the following reasons:

- The terminating gateway is not configured with fax/modem pass-through.
- The *fax pass-through mode* used by the terminating gateway may not be compatible with the Cisco NSE method.

If the log shows proper NSE packet transaction and G.711 upspeed for your fax session but the session still fails, check that the following conditions are true:

- The Cisco ATA software image version is 2.14 or above.
- The Cisco ATA model number is ATA186-I1, ATA186-I2, ATA188-I1, or ATA188-I2.
- The fax relay option for the remote gateways has been disabled.

Debugging FAX Mode

When the Cisco ATA is configured with *fax mode*, only G.711 codecs are used. You must confirm that only 0 (for G.711µ-law) or 8 (for G.711A-law) appear in the `Enable encoder` and `DPKT 1st` debug lines. The following example of a debug log shows that G.711µ-law is used:

```
[0]Enable encoder 0
[0]DPKT 1st: 1491513359 1491512639, pt 0
```

If the numeric codes for the G.711 codecs do not appear in the log, you need to check your *AudioMode* parameter setting on the Cisco ATA.

If the correct G.711 codecs appear in the log but your fax sessions still fail, check that the following conditions are true:

- The Cisco ATA software image version is 2.14 or above.
- The Cisco ATA model number is ATA186-I1, ATA186-I2, ATA188-I1, or ATA188-I2.
- The fax relay option for the remote gateways has been disabled.

Using rtpcatch for Diagnosing Fax Problems

This section contains the following topics:

- rtpcatch Overview, page 7-12
- Example of rtpcatch, page 7-14
- Analyzing rtpcatch Output for Fax Sessions, page 7-16
- Using rtpcatch to Analyze Common Causes of Failure, page 7-18
- rtpcatch Limitations, page 7-20

rtpcatch Overview

*rtpcatch* is a tool that provides comprehensive information for a VoIP connection. The tool runs on a Microsoft Windows-based PC and is capable of parsing an output capture file from Network Associates (NAI) Sniffer Pro and identifies significant fax pass-through and fax relay events.
Major functions

**rtpcatch** includes the following major functions:

- Reads session data from Sniffer Pro capture files.
- Analyzes media streams.
- Stores media streams to files.
- Reports RTP statistics such as the number of RTP packets, the number of RTP frames, the number of lost packets, the number of filler packets during silence suppression periods, and the number of erased packets.

How to Use

To use **rtpcatch**, follow these steps:

**Procedure**

1. Create a working directory for **rtpcatch** and place the executable file rtpcatch.exe in this directory.
2. Copy your Network Associates Sniffer Pro capture files into this directory.
3. At the DOS prompt of this directory, enter the following command:

   ```
   :>rtpcatch <cap_file> [prefix] [options]
   ```

   - `<cap_file>` is the NAI Sniffer capture file.
   - `<prefix>` is the prefix prepended to the output filenames.

Output Files

The output files of **rtpcatch** include a summary file and audio stream files.

The summary file is `<prefix>.sum` if `<prefix>` is specified, otherwise it is `file.sum`. Stream files are labeled with an integer tag beginning with 00. Stream files are also tagged with the extension `pcm` for G.711A/G.711µ-law, `723` for G723.1, `729` for G729, `t38` for T.38, and `cfr` for Cisco Fax Relay.

Options

**rtpcatch** options include:

- `-fax`—to output the fax events for a connection.
  
  The output includes "FAX summary 1" as the interleaved event list for all directions, and "FAX summary 2" as the event list for each direction. The reported events include voice codec change, NSE signalling, and fax relay events.

- `-port <port0> <port1>`—to discard any packets sent from/to this port.

  If the NAI Sniffer capture file includes Cisco ATA prserv packets, these packets can interfere with **rtpcatch** analysis. Some prserv packets might be interpreted as NTE or NSE events. To prevent such interference, you can either disable debugging output on the Cisco ATA (do this by setting the Nprintf configuration parameter to 0), configure your NAI Sniffer to filter out the prserv packets, or run **rtpcatch** with the -port options.
Note  \texttt{rtpcatch} works best for analyzing a single VoIP session. Command-line options can be entered in any order.

Example of \texttt{rtpcatch}

The section contains an example of using \texttt{rtpcatch} and includes an explanation of its output:

\textbf{Output}

\begin{verbatim}
C:\>rtpcatch faxpassthru -fax

    [ 25] open file: 00.723, (G723) 2.213:10000 => 2.116:10002
    [ 26] open file: 01.723, (G723) 2.116:10002 => 2.213:10000
    [ 29] <00> 1 silence pkts from TS 1760 (seq# 3)
    [ 42] <00> 2 silence pkts from TS 4400 (seq# 9)
    [ 47] <00> 2 silence pkts from TS 5600 (seq# 11)
    [ 55] <00> 2 silence pkts from TS 7760 (seq# 15)
    [ 101] open file: 02.pcm, (G711u) 2.116:10002 => 2.213:10000
    [ 106] <02> 2 lost pkts from seq# 39
    [ 107] open file: 03.pcm, (G711u) 2.213:10000 => 2.116:10002
    [ 110] <03> 1 silence pkts from TS 19440 (seq# 41)

------------ Summary --------------
Input file: faxpassthru.cap
<00.723>: (G723) 2.213:10000 => 2.116:10002
    total 38 pkts(70 frames), lost 0 pkts, fill 7 silence pkts
<01.723>: (G723) 2.116:10002 => 2.213:10000
    total 38 pkts(76 frames), lost 0 pkts, fill 0 silence pkts
<02.pcm>: (G711u) 2.116:10002 => 2.213:10000
    total 2181 pkts(2181 frames), lost 2 pkts, fill 0 silence pkts
<03.pcm>: (G711u) 2.213:10000 => 2.116:10002
    total 2179 pkts(2179 frames), lost 0 pkts, fill 1 silence pkts

---------- FAX Summary 1 ----------

[ 25]<2.213=>2.116> Codec G723
[ 26]<2.116=>2.213> Codec G723
[ 101]<2.116=>2.213> Codec G711u/D
[ 102]<2.116=>2.213> NSE PT 100, EVT 192: Up-Speed, CED tone Detected
[ 103]<2.116=>2.213> NSE PT 100, EVT 193: ECAN OFF, Phase Reversal Detected
[ 105]<2.213=>2.116> NSE PT 100, EVT 192: Up-Speed, CED tone Detected
[ 107]<2.213=>2.116> Codec G711u/D

---------- FAX Summary 2 ----------

PATH: 2.213:10000 => 2.116:10002
[ 25]Codec G723
[ 105]NSE PT 100, EVT 192: Up-Speed, CED tone Detected
[ 107]Codec G711u/D

PATH: 2.116:10002 => 2.213:10000
\end{verbatim}
Codec G723
Codec G711u/D
NSE PT 100, EVT 192: Up-Speed, CED tone Detected
NSE PT 100, EVT 193: ECAN OFF, Phase Reversal Detected

Explanation

The output is printed on screen and saved in the file file.sum.

The following lines are described:

- [ 25] open file: 00.723, (G723) 2.213:10000 => 2.116:10002
  This indicates that rtpcatch reached NAI Sniffer packet number 25 and opened a new file named 00.723 to store an audio stream consisting of G.723-compressed data. The audio path originates from the IP address ending with 2.213 and port 10000 (written as <2.213:1000>) and terminates at the IP address ending with 2.116 and port 10002.

- [ 29] <00> 1 silence pkts from TS 1760 (seq# 3)
  This indicates that rtpcatch detected one silence RTP packet in the audio path <00> and the silence packet began at timestamp 1760. This occurred at packet number 29 with the RTP sequence number 3.

- [ 106] <02> 2 lost pkts from seq# 39
  This indicates that rtpcatch detected two lost RTP packets in the audio path <02>. The missing packets began with sequence number 39. This occurred at packet number 106.

--- Summary ---

Input file: faxpassthru.cap
<00.723>: (G723) 2.213:10000 => 2.116:10002
total 38 pkts(70 frames), lost 0 pkts, fill 7 silence pkts
This indicates that the input filename is faxpassthru.cap. The output file 00.723 contains the G.723-compressed stream from <2.123:10000> to <2.116:10002>; 38 packets (70 frames) were processed by rtpcatch. No lost packets were detected and seven silence packets were found.

--- FAX Summary 1 ---
[ 25]<2.213=>2.116> Codec G723
[ 26]<2.116=>2.213> Codec G723
[ 101]<2.116=>2.213> Codec G711u/D
[ 102]<2.116=>2.213> NSE PT 100, EVT 192: Up-Speed, CED tone Detected
[ 103]<2.116=>2.213> NSE PT 100, EVT 193: ECAN OFF, Phase Reversal Detected
[ 105]<2.213=>2.116> NSE PT 100, EVT 192: Up-Speed, CED tone Detected
[ 107]<2.213=>2.116> Codec G711u/D
This indicates that the audio streams originating at <2.213> and <2.216> are G.723-compressed. The audio stream from <2.116> was then up-spded to G.711µ-law at packet number 101. The NSE signaling packets were sent at packet number 102, 103 and 105. Finally, the audio stream from <2.113> was up-spded to G.711µ-law.

--- FAX Summary 2 ---
PATH: 2.213:10000 => 2.116:10002
[ 25]<2.213=>$2.116> Codec G723
[ 105]<2.116=>2.213> Codec G711u/D
This indicates that the audio streams originating at <2.213> and <2.216> are G.723-compressed. The audio stream from <2.116> was then up-spded to G.711µ-law at packet number 101. The NSE signaling packets were sent at packet number 102, 103 and 105. Finally, the audio stream from <2.113> was up-spded to G.711µ-law.
Chapter 7  Configuring and Debugging Fax Services

Debugging the Cisco ATA 186/188 Fax Services

PATH: 2.116:10002 => 2.213:10000
[ 26]Codec G723
[ 101]Codec G711u/D
[ 102]NSE PT 100, EVT 192: Up-Speed, CED tone Detected
[ 103]NSE PT 100, EVT 193: ECAN OFF, Phase Reversal Detected

This summarizes the fax events for each path.

The audio stream events reported by rtpcatch include:
- beginning of new audio codec
- silence packets
- lost packets
- erased packets (as in G.729)

The NSE events reported by rtpcatch include:
- event 32, Fax Mode, CED tone Detected (RFC2833)
- event 34, Modem Mode, ANSam tone Detected (RFC2833)
- event 192, Up-Speed, CED tone Detected
- event 193, ECAN OFF, Phase Reversal Detected
- event 194, ECAN ON, Silence Detected
- event 200, T38 Fax Mode, V.21 Detected
- event 201, T38 Fax Mode ACK
- event 202, T38 Fax Mode NACK
- event 203, Modem Relay Mode, CM Tone Detected
- event Cisco Fax Relay (with RTP payload type 96)
- event Cisco Fax Relay ACK (with RTP payload type 97)

Analyzing rtpcatch Output for Fax Sessions

The following examples show the proper fax events when gateways are configured to operate in the following modes:
- Cisco ATA fax mode
- Cisco ATA fax pass-through mode
- T.38 fax relay mode
- Cisco fax relay mode

Example 7-1  Fax Mode

---------- FAX Summary 1 ----------
[ 25]<2.131=>3.200> Codec G711u
[ 26]<3.200=>2.131> Codec G711u

Analysis
Both sides use G.711 for the entire fax session.
Example 7-2 Fax Pass-through Mode

---------- FAX Summary 1 ----------
[ 25]<2.213=>2.116> Codec G723
[ 26]<2.116=>2.213> Codec G723
[ 101]<2.116=>2.213> Codec G711u/D
[ 102]<2.116=>2.213> NSE PT 100, EVT 192: Up-Speed, CED tone Detected
[ 103]<2.116=>2.213> NSE PT 100, EVT 193: ECAN OFF, Phase Reversal Detected
[ 105]<2.213=>2.116> NSE PT 100, EVT 192: Up-Speed, CED tone Detected
[ 107]<2.213=>2.116> Codec G711u/D

Analysis
- Both sides initially use G.723.
- <2.116> switches to G.711µ-law using a dynamic payload type.
- NSE signaling packets are sent from <2.116>.
- An optional NE signaling packet is sent from <2.213>.
- <2.113> switches to G.711µ-law using a dynamic payload type.

Note
EVT 193 may not appear for some fax transmission.

Example 7-3 Fax Pass-through Mode

---------- FAX Summary 1 ----------
[ 37]<3.200=>2.53> Codec G723
[ 41]<2.53=>3.200> Codec G723
[ 136]<3.200=>2.53> Codec G711u/D
[ 137]<3.200=>2.53> NSE PT 100, EVT 192: Up-Speed, CED tone Detected
[ 140]<2.53=>3.200> Codec G711u/D

Analysis
- Both sides initially use G.723.
- <3.200> switches to G.711µ-law using a dynamic payload type.
- NSE signaling packets are sent from <3.200>.
- <2.53> switches to G.711µ-law using a dynamic payload type.

Example 7-4 T38 Fax Relay Mode

---------- FAX Summary 1 ----------
[ 486]<3.99=>2.53> Codec G711u
[ 1277]<3.99=>2.53> Codec T38
[ 1278]<2.53=>3.99> Codec T38

Analysis
- Both sides initially use G.711µ-law.
- Both sides switch to T.38
Example 7-5  Cisco Fax Relay

---------- FAX Summary 1 ----------
[  8]<2.53=>3.99> Codec G711u
[ 248]<3.99=>2.53> Codec G711u
[ 798]<2.53=>3.99> NSE PT 96, Cisco Fax Relay
[ 799]<3.99=>2.53> NSE PT 97, EVT 192: Up-Speed, CED tone Detected
[ 800]<2.53=>3.99> NSE PT 97, Cisco Fax Relay ACK
[ 801]<2.53=>3.99> Codec C_FxRly
[ 803]<3.99=>2.53> NSE PT 96, EVT 192: Up-Speed, CED tone Detected
[ 804]<2.53=>3.99> NSE PT 97, Cisco Fax Relay ACK
[ 805]<3.99=>2.53> Codec C_FxRly

Analysis
- Both sides initially use G.711µ-law.
- NSE signaling packets are sent between <2.53> and <3.99>.
- Both sides switch to Cisco fax relay.

Using rtpcatch to Analyze Common Causes of Failure

The following examples show the rtpcatch output of failed fax sessions. <3.200> is ATA; <2.53> is a Cisco gateway.

Example 7-6  Cisco ATA Configuration Failure

---------- FAX Summary 1 ----------
[  37]<2.53=>3.200> Codec G723
[  39]<3.200=>2.53> Codec G723

Analysis
- <2.53> is the originating gateway and <3.200> is the terminating Cisco ATA.
- The Cisco ATA and the <2.53> gateway use G.723 codec.

Possible Causes for Failure
- The Cisco ATA is not configured with fax mode or fax pass-through mode.
- If the Cisco ATA is the gateway for a fax sender, the remote gateway is not configured with fax pass-through mode.

Example 7-7  Fax Mode Failure

---------- FAX Summary 1 ----------
[  37]<2.53=>3.200> Codec G711
[  39]<3.200=>2.53> Codec G711
[ 1820]<2.53=>3.200> NSE PT 96, Cisco Fax Relay
[ 1966]<2.53=>3.200> NSE PT 96, Cisco Fax Relay

Analysis
- <2.53> is the originating gateway and <3.200> is the terminating Cisco ATA.
- The Cisco ATA and the <2.53> gateway begin with G.711 codec.
• The <2.53> gateway sends Cisco fax relay event packets.

**Possible Cause for Failure**
• Cisco fax relay option is not disabled on the gateway.

**Example 7-8 Fax Pass-through Mode Failure**

```
---------- FAX Summary 1 ----------
[  2]<2.53=>3.200> Codec G723
[  4]<3.200=>2.53> Codec G723
[ 106]<3.200=>2.53> Codec G711u/D
[ 107]<3.200=>2.53> NSE PT 100, EVT 192: Up-Speed, CED tone Detected
[ 1436]<3.200=>2.53> NSE PT 100, EVT 192: Up-Speed, CED tone Detected
```

**Analysis**
• <2.53> is the originating gateway, and <3.200> is the terminating Cisco ATA.
• The Cisco ATA upspeeds to G.711µ-law and sends G.711 upspeed NSE signaling packets.
• The <2.53> gateway does not respond to the NSE signaling packets.

**Possible Causes for Failure**
• Fax/modem pass-through option is not enabled on the gateway.
• Fax/modem pass-through NSE payload type are configured differently on the Cisco ATA and the gateway.

**Example 7-9 Fax Pass-through Mode Failure**

```
---------- FAX Summary 1 ----------
[  37]<2.53=>3.200> Codec G723
[  39]<3.200=>2.53> Codec G723
[  143]<3.200=>2.53> Codec G711u/D
[  144]<3.200=>2.53> NSE PT 100, EVT 192: Up-Speed, CED tone Detected
[ 1602]<3.200=>2.53> NSE PT 100, EVT 192: Up-Speed, CED tone Detected
[ 1604]<2.53=>3.200> Codec G711u/D
[ 1820]<2.53=>3.200> NSE PT 96, Cisco Fax Relay
[ 1966]<2.53=>3.200> NSE PT 96, Cisco Fax Relay
```

**Analysis**
• <2.53> is the originating gateway, and <3.200> is the terminating Cisco ATA.
• The Cisco ATA upspeeds to G.711µ-law and sends G.711 upspeed NSE signaling packets.
• The <2.53> gateway upspeeds to G.711µ-law and then sends Cisco fax relay event packets.

**Possible Cause for Failure**
• Cisco fax relay option is not disabled on the gateway.

**Example 7-10 Fax Pass-through Mode Failure**

```
---------- FAX Summary 1 ----------
[  33]<3.200=>2.53> Codec G729
[  39]<2.53=>3.200> Codec G729
[  562]<2.53=>3.200> NTE PT 101, EVT 34: Modem Mode, ANSAm tone Detected (RFC2833)
```
[ 563]<2.53=>3.200> NTE PT 101, EVT 34: Modem Mode, ANSam tone Detected (RFC2833)
[ 565]<2.53=>3.200> NTE PT 101, EVT 34: Modem Mode, ANSam tone Detected (RFC2833)
[ 566]<2.53=>3.200> Codec G711u/D
[ 568]<2.53=>3.200> NTE PT 101, EVT 34: Modem Mode, ANSam tone Detected (RFC2833)
[ 580]<3.200=>2.53> Codec G711u/D

Analysis

- <3.200> is the originating Cisco ATA, and <2.53> is the terminating gateway.
- Both sides initially use G.729.
- <2.53> gateway sends NTE signaling packets, then upspeeds to G.711µ-law.
- <3.200> The Cisco ATA switches to G.711µ-law also, but never sends NTE signaling packets.
- Fax transmission fails because <2.53> gateway does not receive any NTE packets, and it drops the fax call.

Possible Cause for Failure

- The Cisco ATA does not support the NTE signaling method and requires that the gateways use the NSE signaling method.

rtpcatch Limitations

- rtpcatch performs optimally when analyzing capture files containing only one VoIP session.
- rtpcatch detects only G.711A, G.711µ-law, G.723, G.729, T.38, Cisco fax relay, modem pass-through with or without redundancy packets, RTCP packets and NSE packets.
- rtpcatch can handle a maximum of 20 prserv ports using the -port option.
- rtpcatch may not detect T.38 packets correctly.
Upgrading the Cisco ATA Signaling Image

This section describes two methods for upgrading the Cisco ATA software for the SIP protocol:

- **Upgrading the Signaling Image from a TFTP Server, page 8-1**—This is the Cisco-recommended method for the SIP protocol. This method is the most efficient method and requires only a one-time configuration change.

- **Upgrading the Signaling Image Manually, page 8-2**—This method can be used if you must manually upgrade the image of one Cisco ATA. However, this method is not the recommended upgrade method because it is not as simple as the TFTP method.

This section also describes procedures for verifying a successful image upgrade:

- **Confirming a Successful Signaling Image Upgrade, page 8-5**—Procedures for using your Web browser or the voice configuration menu are included.

⚠️ **Caution**

Do not unplug the Cisco ATA while the function button is blinking. Doing so can cause permanent damage to the device. The function button blinks during an upgrade.

🔍 **Note**

The term *Cisco ATA* is used throughout this manual to refer to both the Cisco ATA 186 and the Cisco ATA 188, unless differences between the Cisco ATA 186 and Cisco ATA 188 are explicitly stated.

Upgrading the Signaling Image from a TFTP Server

You can configure the Cisco ATA to automatically download the latest signaling image from the TFTP server. You do this configuring the parameter `upgradecode` in your Cisco ATA configuration file. (You also would use this procedure if you wanted to perform a cross-protocol signaling image upgrade.) For more information about setting up the configuration file, see the “Creating Unique and Common Cisco ATA Configuration Files” section on page 3-9.

**Syntax of upgradecode Parameter**

```
upgradecode:3,0x301,0x0400,0x0200,tftp_server_ip,69,image_id,image_file_name
```

**Definitions**

- The hexadecimal values that precede the `tftp_server_ip` variable must always be the values shown in the syntax.
Upgrading the Signaling Image Manually

This section describes how to manually upgrade the Cisco ATA with the most recent signaling image. The executable file that you need is called sata186us.exe, and is bundled in the Cisco ATA release-software zip file.

This section contains the following topics:

- **Preliminary Steps, page 8-3**
- **Running the Executable File, page 8-3**
Preliminary Steps

Before you run the executable file, be sure to complete the following procedure:

Procedure

Step 1 If you are a registered CCO user, go to the following URL:
http://www.cisco.com/cgi-bin/tablebuild.pl/ata186

Step 2 Locate the zip file that contains the software for the applicable release and signaling image you are using. The contents of each file are described next to the file name. Extract the signaling image file (this file has an extension of .zup—for example, ata186-v2-15-020723a.zup) and store it on the PC that has connectivity with the Cisco ATA.

Step 3 Set the Cisco ATA parameter UseTftp to 0.

Note Remember to set this parameter back to 1 before you use the TFTP upgrade method at a later time.

Step 4 Follow the instructions in the “Running the Executable File” section on page 8-3.

Running the Executable File

This section includes the procedure for running the executable file and using the voice configuration menu to complete the upgrade process. First check to make sure the upgrade requirements are met and determine the syntax to use when running the program.

This section contains the following topics:

- “Upgrade Requirements” section on page 8-3
- “Syntax” section on page 8-3
- “Upgrade Procedure” section on page 8-4

Upgrade Requirements

The following list contains the requirements for using the sata186us.exe file and the voice configuration menu to upgrade the Cisco ATA to the latest signaling image:

- A network connection between the PC from which you will invoke the executable file and the Cisco ATA
- A PC running Microsoft Windows 9X/ME/NT/2000

Syntax

sata186us [-any] [-h[host_ip]] [-p[port]] [-quiet] [-d1 -d2 -d3] <image file>
Definitions

- `--any`—Allow upgrade regardless of software and build versions (recommended).
- `--h [host_ip]`—Set the upgrade server to a specific IP address in cases where there may be more than one IP address for the host. The default behavior is that the program will use the first IP address it obtains when it runs the `gethostbyname` command.
- `--p [port]`—Set the server port to a specific port number (the default port number is 8000; use a different port number only if you are setting up an upgrade server other than the default).
- `--quiet`—Quiet mode; send all output to log file named as `[port].log` (useful when running the upgrade server as a daemon).
- `--d[1,2,3]`—Choose a verbosity level for debugging, with -d3 being the most verbose.
- `image file`—This is the name of the signaling image file to which the Cisco ATA will upgrade.

Example

To upgrade the Cisco ATA to the signaling image `ata186-v2-15-020723a.zup`, you can use the following syntax:

```
sata186us --any --d1 ata186-v2-15-020723a.zup
```

Upgrade Procedure

To perform the upgrade, follow these steps:

**Procedure**

**Step 1** Run the executable file (see the “Syntax” section on page 8-3) from the Microsoft Windows DOS or command prompt. You will receive instructions on how to upgrade.

**Step 2** On the Cisco ATA, press the function button to invoke the voice configuration menu.

**Step 3** Using the telephone keypad, enter the following:

```
100# ip_address_of_PC * port #
```

This is the IP address of the PC and the port number at the DOS prompt where you invoked the sata186us.exe file.

For example, if the IP address is 192.168.1.10, and the port number is 8000 (the default), then enter:

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

When the upgrade is complete, the "Upgrade Successful" prompt will sound.

**Note**

When upgrading many Cisco ATAs manually, you can save the software-upgrade dial-pad sequence in your telephone's speed-dial, and use this sequence repeatedly.
Confirming a Successful Signaling Image Upgrade

You can verify that you have successfully upgraded the Cisco ATA signaling image by using one of the following methods:

- Using a Web Browser, page 8-5
- Using the Voice Configuration Menu, page 8-5

Using a Web Browser

To use your web browser to verify a successful image upgrade, perform the following steps:

Procedure

<table>
<thead>
<tr>
<th>Step 1</th>
<th>Open your web browser.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Enter the IP address of your Cisco ATA Web configuration page:</td>
</tr>
<tr>
<td></td>
<td>http://&lt;IP address&gt;/dev</td>
</tr>
<tr>
<td>Step 3</td>
<td>Refresh the page to clear the cache.</td>
</tr>
<tr>
<td></td>
<td>The image version number and its build date should appear at the bottom-left corner of the Cisco ATA Web configuration page.</td>
</tr>
</tbody>
</table>

Using the Voice Configuration Menu

To use the voice configuration menu to verify a successful image upgrade, perform the following steps:

Procedure

<table>
<thead>
<tr>
<th>Step 1</th>
<th>Pick up the telephone handset attached to the Phone1 port of the Cisco ATA.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 2</td>
<td>Press the function button on the Cisco ATA.</td>
</tr>
<tr>
<td>Step 3</td>
<td>Press 123# on the telephone keypad to play out the image version number.</td>
</tr>
<tr>
<td>Step 4</td>
<td>Press 123123# on the telephone keypad to play out the image build date.</td>
</tr>
</tbody>
</table>
Troubleshooting

This section describes troubleshooting procedures for the Cisco ATA:

- General Troubleshooting Tips, page 9-1
- Symptoms and Actions, page 9-2
- Installation and Upgrade Issues, page 9-3
- Debugging, page 9-4
- Using System Diagnostics, page 9-6
- Local Tone Playout Reporting, page 9-10
- Obtaining Network Status Prior to Getting IP Connectivity, page 9-11
- Obtaining Network Status After Getting IP Connectivity, page 9-12
- DHCP Status HTML Page, page 9-13
- Frequently Asked Questions, page 9-14
- Contacting TAC, page 9-15

Note

The term Cisco ATA is used throughout this manual to refer to both the Cisco ATA 186 and the Cisco ATA 188, unless differences between the Cisco ATA 186 and Cisco ATA 188 are explicitly stated.

General Troubleshooting Tips

The suggestions in this section are general troubleshooting tips.

- Make sure that the DHCP server is operating correctly. Note that the function button blinks slowly when the Cisco ATA attempts to acquire the DHCP configuration.
- If the green activity LED is not flashing after you connect the Ethernet cable, make sure that both the power cord and the Ethernet connection are secure.
- If there is no dial tone, make sure that the telephone line cord from the telephone is plugged into the appropriate port on the Cisco ATA. Make sure that your Cisco ATA is properly registered on your Call Control system. Test another phone; if this phone does not work either, there may be a problem with the current configuration or with the Cisco ATA.
Symptoms and Actions

Symptom  Parameters with values set by using the web server interface or voice configuration menu revert to their original settings.

Possible Cause  You are using TFTP for configuration (the UseTFTP parameter is set to 1). The Cisco ATA has a cached version of its configuration file stored in its flash memory; this is what displayed or played through the web server interface or voice configuration menu. If UseTFTP is set to 1, then the cached value of the Cisco ATA configuration file is synchronized with its configuration file located at the TFTP server. This synchronization update of the cached value occurs at approximate intervals determined by the CFGInterval parameter value as well as when the Cisco ATA powers up or resets.

Recommended Action  If you are using TFTP for configuration, do not use the web server interface or voice configuration menu to modify the value of the Cisco ATA configuration file. Use the web server interface or voice configuration menu only to initially configure the Cisco ATA to contact the TFTP server for the Cisco ATA configuration file.

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 voice configuration menu.

Recommended Action  Upgrade the software.

Symptom  The Cisco ATA does not seem to be configured using the TFTP server.

Possible Cause  The TFTP server address is not properly set.

Recommended Action  Ensure that the TftpURL is correctly set to the URL or IP address of the TFTP server that is hosting the configuration file for the Cisco ATA. If you are using DHCP to supply the TFTP server IP address, make sure that the TftpURL is set to 0. Also, unless the TftpURL is an IP address, be sure that the DNS1IP and DNS2IP values are properly set to resolve the TftpURL supplied by DHCP.
Symptom  The Cisco ATA contacts the TFTP server more often than specified in the CfgInterval parameter.

   **Possible Cause**  The ToConfig parameter is not set to 0.

   **Recommended Action**  After the Cisco ATA has a valid configuration file, the ToConfig parameter must be set to 0. If it is not set to 0, the Cisco ATA will attempt to contact the TFTP server too frequently.

Symptom  Cannot place call.

   **Possible Cause**  Equipment failure on the network.

   **Recommended Action**  Replace defective network equipment.

   **Possible Cause**  Recipient has not registered the IP phone.

   **Recommended Action**  Register the IP phone.

   **Possible Cause**  Ethernet cable is not connected.

   **Recommended Action**  Make sure that all cables are connected.

Symptom  Fast busy tone.

   **Possible Cause**  Authentication credential is incorrect.

   **Recommended Action**  Verify authentication credential, and revise if necessary.

   **Possible Cause**  Recipient has not registered the IP phone.

   **Recommended Action**  Register the IP phone.

   **Possible Cause**  No common codec between the Cisco ATA and remote end.

   **Recommended Action**  Change codec to one that is common with the Cisco ATA and the remote end.

   **Possible Cause**  Recipient is in a call with call waiting disabled.

   **Recommended Action**  Attempt to place the call at a later time.

---

### Installation and Upgrade Issues

**Note**  The following issues apply to the manual image-upgrade process only. Image upgrades must be performed separately.
Symptom  The red LED is flashing slowly on the function button.

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

   **Possible Cause**  The Ethernet cable is unplugged.

   **Recommended Action**  Plug in the Ethernet cable.

Symptom  Voice prompt returns *Upgrade not available* message. This can only occur if you are using the executable-file upgrade method.

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

   **Recommended Action**  You do not need to upgrade.

Symptom  Voice prompt returns *Upgrade failed* message. This can only occur if you are using the executable-file upgrade method.

   **Possible Cause**  You have entered an incorrect IP address.

   **Recommended Action**  Enter the correct IP address.

   **Possible Cause**  Software image is corrupted.

   **Recommended Action**  Upgrade software image.

Symptom  No dial tone.

   **Possible Cause**  No user ID was entered.

   **Recommended Action**  Enter the correct user ID.

Symptom  Incorrect dial tone.

   **Possible Cause**  Check the web interface for your DialTone setting. The default is *U.S.*

   **Recommended Action**  Set the correct country DialTone value.

### Debugging

The MS-DOS Windows-based debugging program tool, preserv.exe, is included in every software upgrade package. The tool is also available from Cisco TAC. The preserv program is used in conjunction with the NPrintf configuration parameter. This file serves as an upgrade server that captures debug information sent by the Cisco ATA software to your PC’s IP address and port number. This debug file (prserv.exe) compiles the information from the Cisco ATA into a readable log file. To capture this "NPRINTF" information, you must know the IP address of the PC using the prserv program, illustrated as follows:

```
IP address.port
```
where IP address is the IP address of your PC, and port is 9001. If another process on your PC already uses port 9001, you may use some other value (legal values are from 1024 to 65535). If no port value is entered, the default value is 9001.

To enter the IP address and port number, use voice menu option 81#. You must enter the IP address and port number in alphanumeric format, which requires entering the * key after every character entered. To enter the "." character, you must enter the sequence 1 1#.

For example, for a computer with the IP address 172.28.78.90 and port number 9001 (172.28.78.90.9001), you would enter the following on your telephone handset:

1* 7* 2* 1 1* 2* 8* 1 1* 7* 8* 1 1* 9* 0* 1 1* 9* 0* 1* 1* *

To operate the debug capture program prserv.exe, place the prserv program in a folder on your PC; then at the DOS prompt of the folder where you have placed it, enter:

C:> prserv [-t] port.log

where port is the port number you have selected, and -t, which is optional, means that a time stamp will be included with each message in the form yy:mm:dd:hh (two-digit years, two-digit months, two-digit days, two-digit hours). If you do not enter port.log, debug information still appears on your screen, but it is not saved to a log file.

After you finish capturing debug information, you can stop the log program by entering Ctrl-C at the DOS prompt. The log file created is named port.log. If you restart the process without changing the name of the log file, any new debug information is appended to the end of the original file.

Contact Cisco TAC for more information. See the “Obtaining Technical Assistance” section on page 19 for instructions.

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, using the UIPassword parameter, before the Cisco ATA is moved from the service provider site.
Using System Diagnostics

The Cisco ATA uses functionality of the syslog protocol for system diagnostics. For detailed information on syslog, see RFC-3164.

**Note**
Because the Cisco ATA does not have an internal clock, syslog messages provide the time offset from the most recent Cisco ATA reset. The system administrator should make sure that the syslog relay or syslog server adds the local timestamps upon receiving syslog messages.

**Message Syntax**

\(<\text{Priority}>\text{Time\_Offset ATa\_IP [tag] : [ch]Message}

**Syntax Definitions**

- **Priority** means the facility and severity values for a specific syslog message.
  
  Priority = (facility value) * 8 + (severity value). Facility and severity definitions and values are supplied in RFC-3164; these values can be calculated if you know the priority value.

- **Time\_offset** means the time elapsed since the most recent Cisco ATA reset.
  
  If the time offset is less than 24 hours, this value is shown as:

  \(<\text{hh:mm:ss}\>

  If the time offset is more than 24 hours, this value is shown as:

  \(<\text{dd hh:mm:ss}\>

  where the first \(d\) is the number of days elapsed since the most recent reset, and the second \(d\) is the letter \(d\).

- **ATA\_IP** means the IP address of ATA.

- **tag** means the tag number of the syslog message. Each tag number corresponds to a particular type of message, such as an ARP message. You can turn on tracing for each type of message you want captured by configuring the Cisco ATA parameter syslogCtrl. For more information about the syslogCtrl parameter and for a complete listing of tag numbers and their corresponding message types, see the “SyslogCtrl” section on page 5-75.

  Syslog information is sent to the syslog server that you configure by means of the Cisco ATA syslogIP parameter. For more information, see the “SyslogIP” section on page 5-74.

- **ch** means the active line of the Cisco ATA.

  System-level messages do not contain a \(ch\) field.

- **Message** means the syslog message. (See RFC-3164 for message formats and how to interpret the meaning of each syslog message.)

The following examples show some of the different types of messages that syslog reports.
Example—ARP Message

<62>00:00:51 192.168.3.169 [00]:ARP Update: MAC:080017014e00, IP:192.168.2.81

This message includes the following information:

- Priority=62, which means that the facility value is 7 (network new subsystem) and the severity value is 6 (informational messages). You can derive this information from RFC-3164.
- The time offset is 00:00:51, which means that the most recent Cisco ATA reset was 51 seconds earlier.
- The IP address of the Cisco ATA is 192.168.3.169
- The tag value is 00, which corresponds to ARP messages. This is derived from Table 5-12 on page 5-75.
- The message itself begins with *ARP Update* and can be interpreted by means of RFC-3164.

Example—DHCP Messages

<62>00:04:00 192.168.3.140 [01]:DHCP Reg: Srv:192.168.2.1 lease:120
<62>00:02:31 192.168.2.253 [01]:DHCP's sm: 255.255.254.0
<62>00:02:31 192.168.2.253 [01]:DHCP's rt: 192.168.3.254

These messages include the following information:

- Priority=62, which means that the facility value is 7 (network new subsystem) and the severity value is 6 (informational messages). You can derive this information from RFC-3164.
- The time offset of the first message is 00:04:00, which means that the most recent Cisco ATA reset was four minutes earlier.
- The tag value is 01, which corresponds to DHCP messages. This is derived from Table 5-12 on page 5-75.
- The messages include the DHCP server IP, lease time, subnet mask and router.

Example—TFTP messages

<94>00:04:35 192.168.3.237 [02]:Rx TFTP file:ata00012d010828(684) ok
<94>00:00:02 192.168.3.237 [02]:Rx TFTP file:ata00012d010828(-10) fail

These messages include the following information:

- Priority=94, which means that the facility value is 11 (FTP daemon) and the severity value is 6 (informational messages). You can derive this information from RFC-3164.
- The time offset of the first message is 00:04:35, which means that the most recent Cisco ATA reset was four minutes and 35 seconds earlier.
- The tag value is 02, which corresponds to TFTP messages. This is derived from Table 5-12 on page 5-75.
- The messages include TFTP filename, file size and transmission result.
Example—Cisco ATA Configuration Update Message

<30>00:00:01 192.168.3.237 [03]:ATA Config Update OK

This message includes the following information:

- Priority=30, which means that the facility value is 3 (system daemon) and the severity value is 6 (informational messages). You can derive this information from RFC-3164.
- The time offset of the message is 00:00:01, which means that the most recent Cisco ATA reset was one second earlier.
- The tag value is 03, which corresponds to Cisco ATA configuration-update messages. This is derived from Table 5-12 on page 5-75.
- This message shows indicates the status of the Cisco ATA configuration-file update.

Example—System Reboot Message

<31>00:00:00 192.168.3.220 [04]:Reboot from ata00012d010829(HWVersion1)
0192.168.3.220 (warmStart:0)

This message includes the following information:

- Priority=31, which means that the facility value is 3 (system daemon) and the severity value is 7 (debug-level messages). You can derive this information from RFC-3164.
- The time offset of the message is 0.
- The tag value is 04, which corresponds to system-reboot messages. This is derived from Table 5-12 on page 5-75.
- This message includes the MAC address, hardware version and IP address of the Cisco ATA, and the reason for the reboot.

Example—SIP Messages

<134>00:01:51 192.168.3.169 [08]:[0]Response=100; Trying
<134>00:01:51 192.168.3.169 [08]:[0]Response=407; Proxy Authentication Required
<134>00:01:51 192.168.3.169 [08]:[0]Response=100; Trying
<134>00:01:51 192.168.3.169 [08]:[0]Response=180; Ringing
<134>00:01:53 192.168.3.169 [08]:[0]Response=200; OK
<134>02:20:33 192.168.3.199 [08]:[0]Reg Resp 100; Trying
<134>02:20:33 192.168.3.199 [08]:[0]Reg Resp 401; Unauthorized
<134>02:20:33 192.168.3.199 [08]:[1]Reg Resp 100; Trying
<134>02:20:33 192.168.3.199 [08]:[1]Reg Resp 401; Unauthorized

These messages include the following information:

- Priority=134, which means that the facility value is 16 (local use 0) and the severity value is 6 (informational messages). You can derive this information from RFC-3164.
- The time offset of the first four messages is one minute and 51 seconds.
- The tag value is 08, which corresponds to Cisco ATA SIP messages. This is derived from Table 5-12 on page 5-75.
- The ch (active line of the Cisco ATA) for the first several messages is line 0, which is the Phone 1 port of the Cisco ATA. The active line for the last two messages is line 1, which is the Phone 2 port of the Cisco ATA.
- These SIP messages are displayed in the short format. To obtain more detailed SIP messages, see the “TraceFlags” section on page 5-73, and use the prserv tool to capture logging information.
Example—Cisco ATA Event Messages

<142>00:00:40 192.168.3.169 [09]:[0]OFFHOOK
<142>00:00:44 192.168.3.169 [09]:[0]ONHOOK
<142>00:01:00 192.168.3.169 [09]:[0]OFFHOOK
<142>00:01:50 192.168.3.169 [09]:[0]DTMF 2 , insum:830200
<142>00:01:50 192.168.3.169 [09]:[0]DTMF 1 , insum:868411
<142>00:01:50 192.168.3.169 [09]:[0]DTMF 2 , insum:861215
<142>00:01:50 192.168.3.169 [09]:[0]DTMF 0 , insum:858638
<142>00:01:51 192.168.3.169 [09]:[0]DTMF # , insum:845590
<142>00:01:51 192.168.3.169 [09]:[0]CLIP 22120

These messages include the following information:

- **Priority=142**, which means that the facility value is 17 (local use 1) and the severity value is 6 (informational messages). You can derive this information from RFC-3164.
- The time offset of the first message is 40 seconds.
- The tag value is 09, which corresponds to Cisco ATA event messages. This is derived from Table 5-12 on page 5-75.
- The **ch** (active line of the Cisco ATA) is line 0, which is the **Phone 1** port of the Cisco ATA.
- The messages include DTMF debugging (showing the key and the insum number), on/off hook, Caller ID (CLIP/CLIR) and the callee number.

Example—Fax Event Messages

<150>00:00:11 192.168.3.169 [10]:[1]MPT mode 0
<150>01:07:27 192.168.3.169 [10]:[1]Rx FAX
<150>01:07:27 192.168.3.169 [10]:[1]Tx MPT PT=100 NSE pkt c0000000
<150>01:07:27 192.168.3.169 [10]:[1]MPT mode 2
<150>01:07:27 192.168.3.169 [10]:[1]MPT mode 3
<150>01:07:27 192.168.3.169 [10]:[1]Rx MPT PT=100 NSE pkt c0000000

These messages include the following information:

- **Priority=150**, which means that the facility value is 18 (local use 2) and the severity value is 6 (informational messages). You can derive this information from RFC-3164.
- The time offset of the first message is 11 seconds.
- The tag value is 10, which corresponds to Cisco ATA event messages. This is derived from Table 5-12 on page 5-75.
- The **ch** (active line of the Cisco ATA) is line 1, which is the **Phone 2** port of the Cisco ATA.
- The messages include fax detection, transmit/receive NSE packet status and Fax codec switch information.

Example—RTP Statistic Messages

<182>00:01:58 192.168.3.169 [16]:[0]RTP Tx dur:5, pkt:275, byte:44000
<182>00:01:58 192.168.3.169 [16]:[0]RTP Rx dur:7, pkt:226, byte:35921, latePkt:0 lostPkt:0 avgJitter:0

These messages include the following information:

- **Priority=182**, which means that the facility value is 22 (local use 6) and the severity value is 6 (informational messages). You can derive this information from RFC-3164.
- The time offset of the first message is one minute and 58 seconds.
• The tag value is 16, which corresponds to RTP statistics messages. This is derived from Table 5-12 on page 5-75.

• The ch (active line of the Cisco ATA) is line 0, which is the Phone 1 port of the Cisco ATA.

• The transmission statistics include the duration, packet count and byte count. The receiving statistics include the duration, packet count, byte count, last packet count, lost packet count and average jitter.

Local Tone Playout Reporting

Local tones are tones that the Cisco ATA plays to its FXS port. Each of these tones corresponds to an identifier, and these tone type identifiers are placed into the prserv debug log. These identifiers supply information that administrators can use to help analyze call flows for debugging purposes.

Local tones are different from other tones because local tones are not carried within the inband audio. Instead, the Cisco ATA is prompted by a network event to play the tone, and the Cisco ATA generates the tone for the exclusive purpose of playing it to the attached telephone handset. For example, during a call between the Cisco ATA and a far-end phone, the far-end user might press a digit on the dial pad, thus sending an AVT Named Signaling Event to the Cisco ATA. This event prompts the Cisco ATA to generate a DTMF tone and to play the tone locally to the Cisco ATA phone.

Table 9-1 lists the tone type identifier and its description for local tone reporting.

### Table 9-1 Tone Type Identifiers

<table>
<thead>
<tr>
<th>Tone Type ID</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>0</td>
<td>Dial tone</td>
</tr>
<tr>
<td>1</td>
<td>Reorder tone</td>
</tr>
<tr>
<td>2</td>
<td>Ringback tone</td>
</tr>
<tr>
<td>3</td>
<td>Call-waiting tone</td>
</tr>
<tr>
<td>4</td>
<td>Warning tone</td>
</tr>
<tr>
<td>5</td>
<td>Special information tone (SIT)</td>
</tr>
<tr>
<td>6</td>
<td>Secondary dial tone</td>
</tr>
<tr>
<td>7</td>
<td>DTMF digit 0</td>
</tr>
<tr>
<td>8</td>
<td>DTMF digit 1</td>
</tr>
<tr>
<td>9</td>
<td>DTMF digit 2</td>
</tr>
<tr>
<td>10</td>
<td>DTMF digit 3</td>
</tr>
<tr>
<td>11</td>
<td>DTMF digit 4</td>
</tr>
<tr>
<td>12</td>
<td>DTMF digit 5</td>
</tr>
<tr>
<td>13</td>
<td>DTMF digit 6</td>
</tr>
<tr>
<td>14</td>
<td>DTMF digit 7</td>
</tr>
<tr>
<td>15</td>
<td>DTMF digit 8</td>
</tr>
<tr>
<td>16</td>
<td>DTMF digit 9</td>
</tr>
<tr>
<td>17</td>
<td>DTMF digit A</td>
</tr>
<tr>
<td>18</td>
<td>DTMF digit B</td>
</tr>
</tbody>
</table>
Chapter 9  Troubleshooting

Obtaining Network Status Prior to Getting IP Connectivity

Using voice configuration menu code 3123#, you can obtain basic network status to use for diagnostic purposes. After you enter this code, the Cisco ATA announces a message in the following format:

e123.D.0xX

where:

- D is the VLAN ID (this is a non-zero value if the Cisco ATA has entered a VLAN)
- 0xX is a bitmap value in hexadecimal format. The definition of each bit is shown in Table 9-2.

<table>
<thead>
<tr>
<th>Bit Number</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>0</td>
<td>Cisco ATA sent CDP request</td>
</tr>
<tr>
<td>1</td>
<td>VLAN ID acquired via CDP</td>
</tr>
<tr>
<td>2</td>
<td>Cisco ATA sent DHCP request</td>
</tr>
<tr>
<td>3</td>
<td>DHCP server offered IP address</td>
</tr>
<tr>
<td>4</td>
<td>Cisco ATA obtained IP address from DHCP server</td>
</tr>
<tr>
<td>5</td>
<td>Cisco ATA web server is ready</td>
</tr>
</tbody>
</table>

Example

If the hexadecimal value provided by the voice configuration menu is 0x1d, the network status of the Cisco ATA is shown in Table 9-3.

<table>
<thead>
<tr>
<th>Bit Number</th>
<th>Description</th>
<th>Boolean Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>0</td>
<td>Cisco ATA sent CDP request</td>
<td>True</td>
</tr>
<tr>
<td>1</td>
<td>VLAN ID acquired via CDP</td>
<td>False</td>
</tr>
<tr>
<td>2</td>
<td>Cisco ATA sent DHCP request</td>
<td>True</td>
</tr>
<tr>
<td>3</td>
<td>DHCP server offered IP address</td>
<td>True</td>
</tr>
</tbody>
</table>

Table 9-1  Tone Type Identifiers  (continued)

<table>
<thead>
<tr>
<th>Tone Type ID</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>19</td>
<td>DTMF digit C</td>
</tr>
<tr>
<td>20</td>
<td>DTMF digit D</td>
</tr>
<tr>
<td>21</td>
<td>DTMF digit *</td>
</tr>
<tr>
<td>22</td>
<td>#</td>
</tr>
<tr>
<td>23</td>
<td>CPE alerting tone (CAS)</td>
</tr>
<tr>
<td>24</td>
<td>Prompt tone/Conference warning tone</td>
</tr>
<tr>
<td>25</td>
<td>Beep tone</td>
</tr>
</tbody>
</table>
Chapter 9  Troubleshooting

Obtaining Network Status After Getting IP Connectivity

Use the Cisco ATA Stats Web page (http://<Cisco ATA IP address>/stats) to display the following information:

- **VLAN ID:** D0
- **tftpFile:** S
- **NTP:** D1,D2,D3
- **tftp:** 0xX

where:

- D0 is the VLAN ID. It should be non-zero if the Cisco ATA has entered a VLAN.
- S is the tftp filename, which can be either ata<macaddress> or the filename supplied by the DHCP server.
- D1 is the local time on the Cisco ATA.
- D2 is the last NTP contact time.
- D3 is the last successful NTP contact time.

D1, D2, D3 values are shown in number of seconds since 00:00:00 UTC, 1970-01-01. If no NTP response has been received from the NTP server, the values of D1, D2, and D3 are 0.

- 0xX is a bitmap value in hexadecimal format. The definition of each bit is shown in Table 9-4.

### Table 9-3  Voice Configuration Menu Example Network Status (continued)

<table>
<thead>
<tr>
<th>Bit Number</th>
<th>Description</th>
<th>Boolean Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>4</td>
<td>Cisco ATA obtained IP address from DHCP server</td>
<td>True</td>
</tr>
<tr>
<td>5</td>
<td>Cisco ATA web server is ready</td>
<td>False</td>
</tr>
</tbody>
</table>

### Table 9-4  Web Configuration Menu Network Status Bitmap

<table>
<thead>
<tr>
<th>Bit Number</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>0</td>
<td>Cisco ATA sent request for configuration file, ata&lt;macaddress&gt;, to TFTP server</td>
</tr>
<tr>
<td>1</td>
<td>Cisco ATA sent request for configuration file, atadefault.cfg, to TFTP server</td>
</tr>
<tr>
<td>4</td>
<td>Cisco ATA sent request for image file to TFTP server</td>
</tr>
<tr>
<td>5</td>
<td>Cisco ATA failed to upgrade to the downloaded image file</td>
</tr>
<tr>
<td>8</td>
<td>Configuration file is not found</td>
</tr>
<tr>
<td>9</td>
<td>Bad configuration file</td>
</tr>
<tr>
<td>10</td>
<td>Checksum error for configuration file</td>
</tr>
<tr>
<td>11</td>
<td>Decode error for configuration file (encryption related)</td>
</tr>
<tr>
<td>12</td>
<td>Configuration file is processed successfully</td>
</tr>
</tbody>
</table>

**Example**

If the hexadecimal value provided by the web configuration menu is 0x1011, the network status of the Cisco ATA is shown in Table 9-5.
DHCP Status HTML Page

You can use the following command to check the status DHCP-related information:

http://ipaddress/stats/

where ipaddress is the IP address of the Cisco ATA.

The information you receive includes the following:

- Elapsed time since most recent renewal of Cisco ATA IP address.
- Elapsed time since most recent successful Cisco ATA registration.
- IP address of the proxy to which the Cisco ATA is registered.

Real-Time Transport Protocol (RTP) Statistics Reporting

To monitor the quality of service for the media stream, you can access RTP packet statistics of the two voice ports and their channels by opening the following page on the Cisco ATA Web server:

<Cisco ATA IP address>/rtps

The following RTP packet statistics are reported:

- rxDuration—the number of seconds since the beginning of reception
- rxPktCnt—the total number of RTP packets received
- rxOctet—the total number of RTP payload octets received (not including RTP header)
- latePktCnt—the total number of late RTP packets received
- totalLostPktCnt—the total number of lost RTP packets received (not including late RTP packets)
- avgJitter—an estimate of statistical variance of the RTP packet inter-arrival time, measured in timestamp unit. (Calculation is based on the formula in RFC1889.)
- txDuration—the number of seconds since the beginning of transmission

<table>
<thead>
<tr>
<th>Bit Number</th>
<th>Description</th>
<th>Boolean Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>0</td>
<td>Cisco ATA sent request for configuration file, ata&lt;macaddress&gt;, to TFTP server</td>
<td>True</td>
</tr>
<tr>
<td>1</td>
<td>Cisco ATA sent request for configuration file, atadefault.cfg, to TFTP server</td>
<td>False</td>
</tr>
<tr>
<td>4</td>
<td>Cisco ATA sent request for image file to TFTP server</td>
<td>True</td>
</tr>
<tr>
<td>5</td>
<td>Cisco ATA failed to upgrade to the downloaded image file</td>
<td>False</td>
</tr>
<tr>
<td>8</td>
<td>Configuration file is not found</td>
<td>False</td>
</tr>
<tr>
<td>9</td>
<td>Bad configuration file</td>
<td>False</td>
</tr>
<tr>
<td>10</td>
<td>Checksum error for configuration file</td>
<td>False</td>
</tr>
<tr>
<td>11</td>
<td>Decode error for configuration file (encryption related)</td>
<td>False</td>
</tr>
<tr>
<td>12</td>
<td>Configuration file is processed successfully</td>
<td>True</td>
</tr>
</tbody>
</table>
Frequently Asked Questions

Q. How can I recover the box if I forgot the password?
A. There are two important passwords. One is the UIPassword, which protects access to the Cisco ATA Web Server interface; the other is the EncryptKey, which protects access to the TFTP configuration file. If you forget the value for the UIPassword but still have access to TFTP-stored configuration file, you can modify the UIPassword via TFTP. However, if you are not configuring the Cisco ATA via TFTP, or if you forget both passwords, the only way you can recover the box is to have physical access to the box and do a factory reset on the box via the box voice configuration menu interface (Access Code: FACTRESET#).

Q. What is the maximum distance from which I can drive an analog device with a Cisco ATA?
A. Table 9-6 provides maximum distances for this question.

<table>
<thead>
<tr>
<th>Ring Load (per RJ-11 FXS Port)</th>
<th>Maximum Distance</th>
</tr>
</thead>
<tbody>
<tr>
<td>5 REN</td>
<td>200 feet (61 m)</td>
</tr>
<tr>
<td>4 REN</td>
<td>1000 feet (305 m)</td>
</tr>
<tr>
<td>3 REN</td>
<td>1700 feet (518 m)</td>
</tr>
<tr>
<td>2 REN</td>
<td>2500 feet (762 m)</td>
</tr>
<tr>
<td>1 REN</td>
<td>3200 feet (975 m)</td>
</tr>
</tbody>
</table>

The Cisco ATA, however, is not designed for long distance. The simple test is to determine if the phone or phones that are connected to the Cisco ATA work properly in their environment.

Pay attention to the following questions:
1. Can the Cisco ATA detect on/off hook from the analog phone?
2. Can the Cisco ATA detect the DTMF signal?
3. Can you dial the remote side?
4. Can the Cisco ATA ring the phone?
5. Is voice quality satisfactory?

If you answer no to any of the above questions, you may have a loop impedance greater than 400 ohm. In this case, perform the following procedure.

**Procedure**

<table>
<thead>
<tr>
<th>Step</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>Increase the wire gauge to reduce the impedance until the Cisco ATA can detect on/off hook and DTMF signal.</td>
</tr>
<tr>
<td>Step 2</td>
<td>If the Cisco ATA cannot ring the phone, find a phone that can ring at a lower ringing voltage. Also, try to use only one phone instead of multiple phones in parallel.</td>
</tr>
</tbody>
</table>

**Q.** Does the Cisco ATA support an overhead paging system, and, if so, does the Cisco ATA support power denial?

**A.** The Cisco ATA supports an overhead paging system *only* if that system does not require power denial (battery removal) when a call is disconnected. However, the Cisco ATA can be configured to reverse the voltage polarity when a call is connected or disconnected. For more information, see the “Polarity” section on page 5-51.

**Contacting TAC**

Qualified customers who need to contact the Cisco Technical Assistance Center (TAC) must 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. The MAC address can also be obtained using voice menu option 24.
- Software build information—To identify the software build information, use the voice menu option 123123.
- Cisco ATA serial number.

See the “Obtaining Technical Assistance” section on page 19 for instructions on contacting TAC.

**Note**

Customers who obtained their equipment through service providers, independent dealers and other third parties must contact their equipment provider for technical assistance.
Using SIP Supplementary Services

SIP supplementary services are services that you can use to enhance your telephone service. These services include call forward, call return, call forwarding and conference calling. Use the following parameters to enable and subscribe to supplementary services:

- **CallFeatures, page 5-35**—Use this parameter to enable desired features.
- **PaidFeatures, page 5-36**—Use this parameter to subscribe or unsubscribe to enabled features.

This section contains the following topics:

- Changing Call Commands, page A-1
- Cancelling a Supplementary Service, page A-1
- Common Supplementary Services, page A-1

### Changing Call Commands

To change the command for a supplementary service (for example, to change *69 to *100), change the context identifiers in the Call Command field on the Web configuration page. For more information, see Chapter 6, “Call Commands.”

**Note**

You cannot change supplementary services by means of the voice configuration menu.

### Cancelling a Supplementary Service

You can deactivate some supplementary services by pressing *70 before making a call. You can also configure your system to have services disabled by default and enabled on a call-by-call basis. Use the 32-bit Call Features plan to handle your services in this manner. For more information, see the “CallFeatures” section on page 5-35.

### Common Supplementary Services

The supplementary services described in this section, and their configuration and implementation, depend on the system of the country in which the service is activated. For information about your country’s implementation of services, contact your local Cisco equipment provider.
This section contains the following topics:
- Caller ID, page A-2
- Call-Waiting Caller ID, page A-2
- Voice Mail Indication, page A-2
- Unattended Transfer, page A-3
- Attended Transfer, page A-4
- Making a Conference Call in the United States, page A-4
- Making a Conference Call in Sweden, page A-4
- Call Waiting in the United States, page A-5
- Call Waiting in Sweden, page A-5
- About Call Forwarding, page A-5
- Call Forwarding in the United States, page A-5
- Call Forwarding in Sweden, page A-6
- Call Return in the United States, page A-6
- Call Return in Sweden, page A-6
- Calling Line Identification Presentation, page A-6
- About Calling Line Identification Restriction, page A-6
- Calling Line Identification Restriction in the United States, page A-7
- Calling Line Identification Restriction in Sweden, page A-7

**Caller ID**

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

**Call-Waiting Caller ID**

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

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

**Voice Mail Indication**

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

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

- Semi-unattended Transfer, page A-3
- Fully Unattended Transfer, page A-3

Semi-unattended Transfer

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

Procedure

| Step 1 | Press the flash button on the telephone handset to put the other party on hold and get a dial tone. |
| Step 2 | Dial the telephone number to which you would like to transfer the other party. |
| Step 3 | Wait for at least one ring and then hang up your phone to transfer the other party. |

Fully Unattended Transfer

Perform the following steps to complete a fully unattended transfer:

Procedure

| Step 1 | Press the flash button on the telephone handset to put the other party on hold and get a dial tone. |
| Step 2 | Press #90 (the transfer service activation code) on your telephone keypad, then enter the phone number to which you want to transfer the other party, then press #. |
| Step 3 | Hang up your phone. |
Attended Transfer

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

Procedure

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

Making a Conference Call in the United States

Procedure

- **Step 1**: Dial the first number.
- **Step 2**: When the person you called answers, press the flash or receiver button on the telephone handset. This will put the first person you called on hold and you will receive 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.
- **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 can call an additional person, that new person can then call someone else, and so on. This is known as daisy-chaining.

Making a Conference Call in Sweden

Procedure

- **Step 1**: Dial the first number.
- **Step 2**: When the person you called answers, press the flash or receiver button on the telephone handset. This will put the first person you called on hold and a dial tone will sound.
- **Step 3**: Dial the second person and speak normally when that person answers.
- **Step 4**: Perform a hook flash, then press 2 on your telephone keypad to return to the first person. You can continue to switch back and forth between the two callers.
Step 5  To conference with both callers at the same time, perform a hook flash, then press 3 on the telephone keypad. 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 can call an additional person, that new person can call someone else, and so on. This is known as “daisy-chaining.”

Call Waiting in the United States

If someone calls you while you are speaking on the telephone, you can answer by performing a hook flash. You cannot conference in all three callers, but the first person you called could call someone else and daisy-chain them into the conference.

When the Cisco ATA is configured to use Call Waiting by default, press *70 on your telephone keypad 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 then pressing 2 on your telephone keypad, or you can conference them with the person to whom you are already speaking by performing a hook flash then pressing 3. You can also perform a hook flash then press 3 later during the call to create a conference call.

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

To enable call waiting for Sweden, press *43#. When the Cisco ATA 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 SIP, the Cisco ATA can control call forwarding and call return.

There are three 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.

Call Forwarding in the United States

Forward Unconditional

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

Forward When Busy

Press #74 on your telephone keypad; enter the number to forward the calls to; then press # again.
Forward On No Answer
Press #75 on your telephone keypad; enter the number you want to forward the calls to; then press # again.

Cancelling Call Forwarding
To cancel call forwarding, press #73 on your telephone keypad

Call Forwarding in Sweden
Forward Unconditional
Press *21* on your telephone keypad; enter the number you want to forward calls to; then press #. To cancel, press #21#.

Forward When Busy
Press *67* on your telephone keypad; enter the number to forward the calls to; then press #. To cancel, press #67#.

Forward On No Answer
Press *61* on your telephone keypad; enter the number you want to forward the calls to; then press #. To cancel, press #61#.

Forward On No Answer with a Specified Call Forward Delay
Press *61* on your telephone keypad; enter the number you want to forward the calls to; then press # and the number of seconds for the call forward delay; then press # again. To cancel, press #61# on your telephone keypad.

Call Return in the United States
Press *69* on your telephone keypad to activate call return in the United States.

Call Return in Sweden
Press *69# on your telephone keypad to activate call return in Sweden.

Calling Line Identification Presentation
Calling Line Identification Presentation (CLIP) shows your identity to callers with Caller ID.
Press *82* on your telephone keypad to activate CLIP.

About Calling Line Identification Restriction
Calling Line Identification Restriction (CLIR) hides your identity from callers with Caller ID.
Calling Line Identification Restriction in the United States

Press *67 on your telephone keypad to activate CLIR. This feature is disabled when you hang up.

Calling Line Identification Restriction in Sweden

Press *31# on your telephone keypad to activate CLIR. This feature is disabled when you hang up.
Voice Menu Codes

This section contains a quick-reference list of the voice configuration menu options for the Cisco ATA. This section contains the following tables about voice menu codes:

- **Table B-1**—Information Options
- **Table B-2**—Configuration Parameters
- **Table B-3**—Software Upgrade Codes

**Note**

Follow each voice menu code with #.

**Note**

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

Table B-1 lists codes to return basic Cisco ATA information.

**Table B-1  Cisco ATA Voice Menu Codes—Information Options**

<table>
<thead>
<tr>
<th>Option</th>
<th>Voice Menu Code</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Build information</td>
<td>123123</td>
<td>Build date of the Cisco ATA software</td>
</tr>
<tr>
<td>Review IP address</td>
<td>21</td>
<td>Returns IP address of the Cisco ATA</td>
</tr>
<tr>
<td>Review MAC address</td>
<td>24</td>
<td>Returns media access control (MAC) address of the Cisco ATA</td>
</tr>
<tr>
<td>Review network route IP address</td>
<td>22</td>
<td>Returns IP address of the network route</td>
</tr>
<tr>
<td>Review subnet mask</td>
<td>23</td>
<td>Returns subnet mask of the network route</td>
</tr>
<tr>
<td>Version number</td>
<td>123</td>
<td>Returns version number of the Cisco ATA software</td>
</tr>
</tbody>
</table>
Table B-2 lists configuration codes.

<table>
<thead>
<tr>
<th>Option</th>
<th>Voice Menu Code</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Alternate NTP IP address</td>
<td>78</td>
<td>IP address of the alternate NTP server</td>
</tr>
<tr>
<td>Audio mode</td>
<td>312</td>
<td>Allows finer control of the audio component to suit certain user applications</td>
</tr>
<tr>
<td>Call features</td>
<td>314</td>
<td>Subscribed features statically enabled by the user</td>
</tr>
<tr>
<td>Caller ID method</td>
<td>316</td>
<td>Specifies the signal format when generating the Caller ID format to use</td>
</tr>
<tr>
<td>TFTP Configuration Interval</td>
<td>80002</td>
<td>Interval (in seconds) between configuration updates when TFTP configuration is used,</td>
</tr>
<tr>
<td>Connection mode</td>
<td>311</td>
<td>Controls the connection mode of the call signaling protocol</td>
</tr>
<tr>
<td>Dynamic Host Configuration Protocol (DHCP)</td>
<td>20</td>
<td>Controls whether the Cisco ATA can automatically obtain configuration parameters from a server over the network</td>
</tr>
<tr>
<td>DNS 1 IP</td>
<td>916</td>
<td>IP address of the primary DNS server</td>
</tr>
<tr>
<td>DNS 2 IP</td>
<td>917</td>
<td>IP address of the secondary DNS server</td>
</tr>
<tr>
<td>Encrypt key</td>
<td>320</td>
<td>Encrypts the configuration file on the TFTP server</td>
</tr>
<tr>
<td>Num Tx frames</td>
<td>35</td>
<td>Number of frames transmitted per packet</td>
</tr>
<tr>
<td>Gatekeeper/proxy server IP address</td>
<td>5</td>
<td>SIP registration proxy server IP address</td>
</tr>
<tr>
<td>IP address</td>
<td>1</td>
<td>IP address of the Cisco ATA</td>
</tr>
<tr>
<td>LBR codec</td>
<td>300</td>
<td>Low bit rate codec selection</td>
</tr>
<tr>
<td>Login ID 0</td>
<td>46</td>
<td>Alternate user ID used for authentication</td>
</tr>
<tr>
<td>Login ID 1</td>
<td>47</td>
<td>Alternate user ID used for authentication</td>
</tr>
<tr>
<td>Media port</td>
<td>202</td>
<td>Specifies which base port the Cisco ATA uses to receive RTP media streams</td>
</tr>
<tr>
<td>Network route address</td>
<td>2</td>
<td>Network router address</td>
</tr>
<tr>
<td>NPrintf address</td>
<td>81</td>
<td>IP address of a host to which all Cisco ATA debug messages are sent</td>
</tr>
<tr>
<td>NTP server address</td>
<td>141</td>
<td>IP address of the NTP server</td>
</tr>
<tr>
<td>Paid features</td>
<td>315</td>
<td>Features subscribed to by the user</td>
</tr>
<tr>
<td>Polarity</td>
<td>304</td>
<td>Controls connect and disconnect polarity</td>
</tr>
<tr>
<td>PWD 0</td>
<td>4</td>
<td>Password associated with the primary phone line (UID0 or LoginID0)</td>
</tr>
<tr>
<td>PWD 1</td>
<td>14</td>
<td>Password associated with the secondary phone line (UID1 or LoginID1)</td>
</tr>
<tr>
<td>Option</td>
<td>Voice Menu Code</td>
<td>Description</td>
</tr>
<tr>
<td>-----------------------------</td>
<td>-----------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Rx codec</td>
<td>36</td>
<td>Selects 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.</td>
</tr>
<tr>
<td>Set password</td>
<td>7387277</td>
<td>Configuration interface password</td>
</tr>
<tr>
<td>Signal timers</td>
<td>318</td>
<td>Timeout values controlling the starting or stopping of a signaling event</td>
</tr>
<tr>
<td>SIP max number of redirects</td>
<td>205</td>
<td>Maximum number of redirections the Cisco ATA will attempt to reach a callee when making a call</td>
</tr>
<tr>
<td>SIP NAT IP address</td>
<td>200</td>
<td>WAN IP of the NAT</td>
</tr>
<tr>
<td>SIP outbound proxy</td>
<td>206</td>
<td>IP address of the outbound proxy server to which all outgoing SIP requests are sent</td>
</tr>
<tr>
<td>SIP port</td>
<td>201</td>
<td>Specifies which port the Cisco ATA listens to for incoming SIP messages</td>
</tr>
<tr>
<td>SIP protocol</td>
<td>38</td>
<td>Selects the signaling protocol</td>
</tr>
<tr>
<td>SIP registration On</td>
<td>204</td>
<td>Enables SIP registration</td>
</tr>
<tr>
<td>SIP registration period</td>
<td>203</td>
<td>Interval (in seconds) between each registration renewal to the SIP registration server</td>
</tr>
<tr>
<td>Subnet mask</td>
<td>10</td>
<td>Specifies the subnet mask for the Cisco ATA</td>
</tr>
<tr>
<td>TFTP URL</td>
<td>905</td>
<td>IP address of the TFTP server when TFTP configuration is used</td>
</tr>
<tr>
<td>Timezone</td>
<td>302</td>
<td>Specifies offset to GMT—used to time-stamp incoming calls for caller ID</td>
</tr>
<tr>
<td>ToConfig</td>
<td>80001</td>
<td>Identifies unconfigured or already-configured Cisco ATAs</td>
</tr>
<tr>
<td>Trace Fflags</td>
<td>313</td>
<td>Enables logging of debug information</td>
</tr>
<tr>
<td>Tx Codec</td>
<td>37</td>
<td>Selects transmitting audio codec preference</td>
</tr>
<tr>
<td>UDP TOS bits</td>
<td>255</td>
<td>Determines the precedence and delay of UDP IP packets</td>
</tr>
<tr>
<td>UID 0</td>
<td>3</td>
<td>User ID (telephone number) for the PHONE 1 port</td>
</tr>
<tr>
<td>UID 1</td>
<td>13</td>
<td>User ID (telephone number) for the PHONE 2 port</td>
</tr>
<tr>
<td>Use login ID</td>
<td>93</td>
<td>Determines which pair (UIDx, PWDx or LoginIDx, PWDx) to use for authentication</td>
</tr>
<tr>
<td>Use TFTP</td>
<td>305</td>
<td>Enables TFTP as configuration method</td>
</tr>
</tbody>
</table>
Table B-3 lists codes used in the software upgrade process. For information about these codes, see Appendix , “Upgrading the Cisco ATA Signaling Image.”

**Table B-3   Cisco ATA Voice Menu Codes—Software Upgrade**

<table>
<thead>
<tr>
<th>Option</th>
<th>Voice Menu Code</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Upgrade software</td>
<td>100</td>
<td>Used in the software process to enter the IP address of the PC</td>
</tr>
<tr>
<td>Upgrade language to English</td>
<td>101</td>
<td>When upgrading software, changes or upgrades the voice prompt language to English</td>
</tr>
</tbody>
</table>
Cisco ATA Specifications

This section describes Cisco ATA specifications:

- Physical Specifications, page C-1
- Electrical Specifications, page C-2
- Environmental Specifications, page C-2
- Physical Interfaces, page C-2
- Ringing Characteristics, page C-3
- Software Specifications, page C-3
- SIP Compliance Reference Information, page C-5

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

Physical Specifications

<table>
<thead>
<tr>
<th>Description</th>
<th>Specification</th>
</tr>
</thead>
<tbody>
<tr>
<td>Dimensions</td>
<td>1.5 x 6.5 x 5.75 in. (3.8 x 16.5 x 14.6 cm) (H x W x D)</td>
</tr>
<tr>
<td>Weight</td>
<td>15 oz (425 g)</td>
</tr>
</tbody>
</table>
Electrical Specifications

Table C-2  Electrical Specifications

<table>
<thead>
<tr>
<th>Description</th>
<th>Specification</th>
</tr>
</thead>
<tbody>
<tr>
<td>Power</td>
<td>0.25 to 7.5W (idle to peak)</td>
</tr>
<tr>
<td>DC input voltage</td>
<td>+5.0 VDC at 1.5A maximum</td>
</tr>
<tr>
<td>Power adaptor</td>
<td>Universal AC/DC</td>
</tr>
<tr>
<td></td>
<td>~3.3 x 2.0 x 1.3 in. (~8.5 x 5.0 x 3.2 cm)</td>
</tr>
<tr>
<td></td>
<td>~4.8 oz (135 g) for the AC-input external power adaptor</td>
</tr>
<tr>
<td></td>
<td>~4 ft (1.2 m) DC cord</td>
</tr>
<tr>
<td></td>
<td>6 ft (1.8 m) cord</td>
</tr>
<tr>
<td></td>
<td>UL/CUL, CE approved</td>
</tr>
<tr>
<td></td>
<td>Class II transformer</td>
</tr>
</tbody>
</table>

Environmental Specifications

Table C-3  Environmental Specifications

<table>
<thead>
<tr>
<th>Description</th>
<th>Specification</th>
</tr>
</thead>
<tbody>
<tr>
<td>Operating temperature</td>
<td>41 to 104° F (5 to 40° C)</td>
</tr>
<tr>
<td>Storage temperature</td>
<td>–4 to 140° F (–20 to 65° C)</td>
</tr>
<tr>
<td>Relative humidity</td>
<td>10 to 90% noncondensing, operating, and nonoperating/storage</td>
</tr>
</tbody>
</table>

Physical Interfaces

Table C-4  Physical Interfaces

<table>
<thead>
<tr>
<th>Description</th>
<th>Specification</th>
</tr>
</thead>
<tbody>
<tr>
<td>Ethernet</td>
<td>Two RJ-45 connectors, IEEE 802.3 10BaseT standard</td>
</tr>
<tr>
<td>Analog telephone</td>
<td>Two RJ-11 FXS voice ports</td>
</tr>
<tr>
<td>Power</td>
<td>5 VDC power connector</td>
</tr>
<tr>
<td>Indicators</td>
<td>Function button with integrated status indicator</td>
</tr>
<tr>
<td></td>
<td>Activity LED indicating network activity</td>
</tr>
</tbody>
</table>
Ringing Characteristics

Table C-5  Ringing Characteristics

<table>
<thead>
<tr>
<th>Description</th>
<th>Specification</th>
</tr>
</thead>
<tbody>
<tr>
<td>Tip/ring interfaces for each RJ-11 FXS port (SLIC)</td>
<td></td>
</tr>
<tr>
<td>Ring voltage</td>
<td>40V&lt;sub&gt;RMS&lt;/sub&gt; (typical, balanced ringing only)</td>
</tr>
<tr>
<td>Ring frequency</td>
<td>25 Hz</td>
</tr>
<tr>
<td>Ring waveform</td>
<td>Trapezoidal with 1.2 to 1.6 crest factor</td>
</tr>
<tr>
<td>Ring load</td>
<td>1400 ohm + 40μF</td>
</tr>
<tr>
<td>Ringer equivalence number (REN)</td>
<td>Up to 5 REN per RJ-11 FXS port</td>
</tr>
<tr>
<td>Loop impedance</td>
<td>Up to 200 ohms (plus 430-ohm maximum telephone DC resistance)</td>
</tr>
</tbody>
</table>

On-hook/off-hook characteristics

| On-hook voltage (tip/ring)                                                 | –50V                                              |
| Off-hook current                                                           | 27 mA (nominal)                                  |
| RJ-11 FXS port terminating impedance option                                |                                                   |

Software Specifications

Table C-6  Software Specifications (All Protocols)

<table>
<thead>
<tr>
<th>Description</th>
<th>Specification</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call progress tones</td>
<td>Configurable for two sets of frequencies and single set of on/off cadence</td>
</tr>
<tr>
<td>Dual-tone multifrequency (DTMF)</td>
<td>DTMF tone detection and generation</td>
</tr>
<tr>
<td>Fax</td>
<td>G.711 fax pass-through and G.711 fax mode. Enhanced fax pass-through is supported on the Cisco ATA. Success of fax transmissions up to 14.4 kbps depends on network conditions, and fax modem/fax machine tolerance to those conditions. The network must have reasonably low network jitter, network delay, and packet-loss rate.</td>
</tr>
</tbody>
</table>
### Table C-6  Software Specifications (All Protocols) (continued)

<table>
<thead>
<tr>
<th>Description</th>
<th>Specification</th>
</tr>
</thead>
</table>
| Line-echo cancellation             | • Echo canceller for each port  
• 8 ms echo length  
• Nonlinear echo suppression (ERL > 28 dB for frequency = 300 to 2400 Hz)  
• Convergence time = 250 ms  
• ERLE = 10 to 20 dB  
• Double-talk detection |
| Out-of-band DTMF                    | • H.245 out-of-band DTMF for H.323  
• RFC 2833 A VT tones for SIP, MGCP, SCCP |
| Configuration                      | • DHCP (RFC 2131)  
• Web configuration via built-in Web server  
• Touch-tone telephone keypad configuration with voice prompt  
• Basic boot configuration (RFC 1350 TFTP Profiling)  
• Dial plan configuration  
• Cisco Discovery Protocol |
| Quality of Service                 | • Class-of-service (CoS) bit-tagging (802.1P)  
• Type-of-service (ToS) bit-tagging |
| Security                           | • H.235 for H.323  
• RC4 encryption for TFTP configuration files |
| Voice coder-decoders (codecs)      | • G.723.1  
• G.729, G.729A, G.729AB  
• G.723.1  
• G.711A-law  
• G.711µ-law |

**Note**

In simultaneous dual-port operation, the second port is limited to G.711 when using G.729.

- G.723.1
- G.729, G.729A, G.729AB
- G.723.1
- G.711A-law
- G.711µ-law
Table C-6  Software Specifications (All Protocols) (continued)

<table>
<thead>
<tr>
<th>Description</th>
<th>Specification</th>
</tr>
</thead>
<tbody>
<tr>
<td>Voice features</td>
<td>• Voice activity detection (VAD)</td>
</tr>
<tr>
<td></td>
<td>• Comfort noise generation (CNG)</td>
</tr>
<tr>
<td></td>
<td>• Dynamic jitter buffer (adaptive)</td>
</tr>
<tr>
<td>Voice-over-IP (VoIP) protocols</td>
<td>• H.323 v2</td>
</tr>
<tr>
<td></td>
<td>• SIP (RFC 2543 bis)</td>
</tr>
<tr>
<td></td>
<td>• MGCP 1.0 (RFC 2705)</td>
</tr>
<tr>
<td></td>
<td>• MGCP 1.0/network-based call signalling (NCS) 1.0 profile</td>
</tr>
<tr>
<td></td>
<td>• MGCP 0.1</td>
</tr>
<tr>
<td></td>
<td>• SCCP</td>
</tr>
</tbody>
</table>

SIP Compliance Reference Information

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

SIP Call Flows

This section describes some basic call flows for the Cisco ATA:

- Supported SIP Request Methods, page D-1
- Call Flow Scenarios for Successful Calls, page D-2

Note

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

Supported SIP Request Methods

The Cisco ATA supports the following SIP request methods:

- INVITE—Indicates a user or service is being invited to participate in a call session.
- ACK—Confirms that the client has received a final response to an INVITE request.
- BYE—Terminates a call and can be sent by either the caller or the callee.
- CANCEL— Cancels any pending searches but does not terminate a call that has already been accepted.
- REGISTER—Registers the address listed in the To header field with a SIP proxy.
- NOTIFY—Notifies the user of the status of a transfer using REFER. Also used for remote reset.
- OPTIONS

The following types of responses are used by SIP and generated by the Cisco SIP gateway:

- SIP 1xx—Informational responses
- SIP 2xx—Successful responses
- SIP 3xx—Redirection responses
- SIP 4xx—Client Failure responses
- SIP 5xx—Server Failure responses
- SIP 6xx—Global Failure responses
Call Flow Scenarios for Successful Calls

This section describes call flows for the following scenarios:

- Cisco ATA-to-SIP Server—Registration without Authentication, page D-2
- Cisco ATA-to-SIP Server—Registration with Authentication, page D-3
- Cisco ATA-to-Cisco ATA—Basic SIP to SIP Call without Authentication, page D-6
- Cisco ATA-to-Cisco ATA—Basic SIP to SIP Call with Authentication, page D-12

Each of the call flows includes a call diagram, action descriptions table, and a sample log file.

Cisco ATA-to-SIP Server—Registration without Authentication

Figure D-1 illustrates the Cisco ATA registering with the SIP server. Authentication is not required for registration.

The call flow is as follows:
1. Cisco ATA requests registration.
2. Registration is completed.

Figure D-1 Cisco ATA-to-SIP Server—Registration without Authentication

<table>
<thead>
<tr>
<th>Step</th>
<th>Action</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>REGISTER—Cisco ATA to SIP server</td>
<td>Cisco ATA sends a REGISTER message to the SIP server to register the address in the To header field.</td>
</tr>
<tr>
<td>Step 2</td>
<td>100 Trying—SIP Server to Cisco ATA</td>
<td>SIP server returns a 100 Trying message, indicating that the REGISTER request has been received.</td>
</tr>
<tr>
<td>Step 3</td>
<td>200 OK—SIP server to Cisco ATA</td>
<td>SIP server returns a final 200 OK response, confirming that registration is complete.</td>
</tr>
</tbody>
</table>
Table D-2 Log Listings

1. REGISTER sip:192.168.2.97 SIP/2.0
   Via: SIP/2.0/UDP 192.168.2.194
   From: <sip:9301@192.168.2.97;user=phone>
   To: <sip:9301@192.168.2.97;user=phone>
   Call-ID: 883972530192.168.2.194
   CSeq: 1 REGISTER
   Contact: <sip:9301@192.168.2.194;user=phone;transport=udp>;expires=3600
   User-Agent: Cisco ATA v2.10 ata186 (0705a)
   Content-Length: 0

2. SIP/2.0 100 Trying
   Via: SIP/2.0/UDP 192.168.2.194
   Call-ID: 883972530192.168.2.194
   From: <sip:9301@192.168.2.97;user=phone>
   To: <sip:9301@192.168.2.97;user=phone>
   CSeq: 1 REGISTER
   Content-Length: 0

3. SIP/2.0 200 OK
   Via: SIP/2.0/UDP 192.168.2.194
   Call-ID: 883972530192.168.2.194
   From: <sip:9301@192.168.2.97;user=phone>
   To: <sip:9301@192.168.2.97;user=phone>
   CSeq: 1 REGISTER
   Contact: <sip:9301@192.168.2.194;user=phone;transport=udp>;expires="tue, 23 Oct 2001 14:24:57 GMT"
   Expires: 3600
   Content-Length: 0

Cisco ATA-to-SIP Server—Registration with Authentication

Figure D-2 illustrates the Cisco ATA registering with the SIP server. Authentication is required for registration.

The call flow is as follows:
1. Cisco ATA requests registration.
2. SIP server requests authentication credential.
3. Authentication is received and registration is completed.
Appendix D      SIP Call Flows

Figure D-2  Cisco ATA-to-SIP Server—Registration with Authentication

Table D-3  Action Descriptions

<table>
<thead>
<tr>
<th>Step</th>
<th>Action</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>REGISTER—Cisco ATA to SIP server</td>
<td>Cisco ATA sends a REGISTER message to the SIP server to register the address in the To header field.</td>
</tr>
<tr>
<td>Step 2</td>
<td>100 Trying—SIP server to Cisco ATA</td>
<td>SIP server returns a 100 trying message, indicating that the REGISTER request has been received.</td>
</tr>
<tr>
<td>Step 3</td>
<td>407 Proxy authentication required—SIP server to Cisco ATA</td>
<td>SIP server returns a request for authentication.</td>
</tr>
<tr>
<td>Step 4</td>
<td>REGISTER—Cisco ATA to SIP server</td>
<td>Cisco ATA attempts to register using its authentication credential.</td>
</tr>
<tr>
<td>Step 5</td>
<td>100 Trying—SIP server to Cisco ATA</td>
<td>SIP server returns a 100 trying message, indicating that the new REGISTER request has been received.</td>
</tr>
<tr>
<td>Step 6</td>
<td>200 OK—SIP server to Cisco ATA</td>
<td>SIP server returns a final 200 OK response, confirming that the authentication credential has been verified and registration is complete.</td>
</tr>
</tbody>
</table>
### Log Listings

<table>
<thead>
<tr>
<th></th>
<th>Log Entry</th>
</tr>
</thead>
</table>
| 1. | REGISTER sip:192.168.2.81 SIP/2.0  
Via: SIP/2.0/UDP 192.168.2.194  
From: <sip:93010192.168.2.81;user=phone>  
To: <sip:93010192.168.2.81;user=phone>  
Call-ID: 3113168420192.168.2.194  
CSeq: 1 REGISTER  
Contact: <sip:93010192.168.2.194;user=phone;transport=udp>;expires=3600  
User-Agent: Cisco ATA v2.10 ata186 (0705a)  
Content-Length: 0 |
| 2. | SIP/2.0 100 Trying  
Via: SIP/2.0/UDP 192.168.2.194  
Call-ID: 3113168420192.168.2.194  
From: <sip:93010192.168.2.81;user=phone>  
To: <sip:93010192.168.2.81;user=phone>  
CSeq: 1 REGISTER  
Content-Length: 0 |
| 3. | SIP/2.0 407 Proxy Authentication Required  
Via: SIP/2.0/UDP 192.168.2.194  
Call-ID: 3113168420192.168.2.194  
From: <sip:93010192.168.2.81;user=phone>  
To: <sip:93010192.168.2.81;user=phone>  
CSeq: 1 REGISTER  
Proxy-Authenticate: DIGEST realm="CISCO", nonce="3bd5e334"  
Content-Length: 0 |
| 4. | REGISTER sip:192.168.2.81 SIP/2.0  
Via: SIP/2.0/UDP 192.168.2.194  
From: <sip:93010192.168.2.81;user=phone>  
To: <sip:93010192.168.2.81;user=phone>  
Call-ID: 3113168420192.168.2.194  
CSeq: 2 REGISTER  
Contact: <sip:93010192.168.2.194;user=phone;transport=udp>;expires=3600  
User-Agent: Cisco ATA v2.10 ata186 (0705a)  
Proxy-Authorization: Digest  
username="9301",realm="CISCO",nonce="3bd5e334",uri="sip:192.168.2.81",response="87ac0afeb08222af706f9e8b5c566ce2"  
Content-Length: 0 |
| 5. | SIP/2.0 100 Trying  
Via: SIP/2.0/UDP 192.168.2.194  
Call-ID: 3113168420192.168.2.194  
From: <sip:93010192.168.2.81;user=phone>  
To: <sip:93010192.168.2.81;user=phone>  
CSeq: 2 REGISTER  
Content-Length: 0 |
| 6. | SIP/2.0 200 OK  
Via: SIP/2.0/UDP 192.168.2.194  
Call-ID: 3113168420192.168.2.194  
From: <sip:93010192.168.2.81;user=phone>  
To: <sip:93010192.168.2.81;user=phone>  
CSeq: 2 REGISTER  
Contact: <sip:93010192.168.2.194;user=phone;transport=udp>;expires="tue, 23 Oct 2001 22:37:56 GMT"  
Expires: 3600  
Content-Length: 0 |
Cisco ATA-to-Cisco ATA—Basic SIP to SIP Call without Authentication

Figure D-3 illustrates a call from one Cisco ATA to another. Authentication by the SIP server is not required.

The call flow is as follows:
1. Call is established between Cisco ATA A and Cisco ATA B.
2. Call is terminated.

Figure D-3  Cisco ATA-to-Cisco ATA—Basic SIP to SIP Call without Authentication

<table>
<thead>
<tr>
<th>Step</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>1.</td>
<td>INVITE</td>
</tr>
<tr>
<td>2.</td>
<td>100 Trying</td>
</tr>
<tr>
<td>3.</td>
<td>INVITE</td>
</tr>
<tr>
<td>4.</td>
<td>100 Trying</td>
</tr>
<tr>
<td>5.</td>
<td>180 Ringing</td>
</tr>
<tr>
<td>6.</td>
<td>180 Ringing</td>
</tr>
<tr>
<td>7.</td>
<td>200 OK</td>
</tr>
<tr>
<td>8.</td>
<td>200 OK</td>
</tr>
<tr>
<td>9.</td>
<td>ACK</td>
</tr>
<tr>
<td>10.</td>
<td>ACK</td>
</tr>
<tr>
<td>11.</td>
<td>BYE</td>
</tr>
<tr>
<td>12.</td>
<td>100 Trying</td>
</tr>
<tr>
<td>13.</td>
<td>BYE</td>
</tr>
<tr>
<td>14.</td>
<td>200 OK</td>
</tr>
<tr>
<td>15.</td>
<td>200 OK</td>
</tr>
</tbody>
</table>

2-way voice path
### Table D-5  Action Descriptions

<table>
<thead>
<tr>
<th>Step</th>
<th>Action</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>INVITE—Cisco ATA A to SIP server</td>
<td>Cisco ATA A sends a call session INVITE request to the SIP server to pass on to Cisco ATA B.</td>
</tr>
<tr>
<td>Step 2</td>
<td>100 Trying—SIP server to Cisco ATA A</td>
<td>SIP server returns a 100 trying message, indicating that the INVITE request has been received.</td>
</tr>
<tr>
<td>Step 3</td>
<td>INVITE—SIP server to Cisco ATA B</td>
<td>SIP server sends the call session INVITE request to Cisco ATA B.</td>
</tr>
<tr>
<td>Step 4</td>
<td>100 Trying—Cisco ATA B to SIP server</td>
<td>Cisco ATA B returns a 100 trying message indicating that the INVITE request has been received.</td>
</tr>
<tr>
<td>Step 5</td>
<td>180 Ringing—Cisco ATA B to SIP server</td>
<td>Cisco ATA B sends a 180 ringing response to the SIP server to pass on to Cisco ATA A.</td>
</tr>
<tr>
<td>Step 6</td>
<td>180 Ringing—SIP server to Cisco ATA A</td>
<td>SIP server sends the 180 ringing response to Cisco ATA A.</td>
</tr>
<tr>
<td>Step 7</td>
<td>200 OK—Cisco ATA B to SIP server</td>
<td>Cisco ATA B sends a 200 OK message to the SIP server indicating that a connection has been established.</td>
</tr>
<tr>
<td>Step 8</td>
<td>200 OK—SIP server to Cisco ATA A</td>
<td>SIP server passes the 200 OK message to Cisco ATA A.</td>
</tr>
<tr>
<td>Step 9</td>
<td>ACK—Cisco ATA A to SIP server</td>
<td>Cisco ATA A sends acknowledgement of the 200 OK response to the SIP server to pass on to Cisco ATA B.</td>
</tr>
<tr>
<td>Step 10</td>
<td>ACK—SIP server to Cisco ATA B</td>
<td>SIP server passes ACK response to Cisco ATA B.</td>
</tr>
<tr>
<td>Step 11</td>
<td>BYE—Cisco ATA A to SIP server</td>
<td>Cisco ATA A terminates the call session and sends a BYE request to the SIP server indicating that Cisco ATA A wants to terminate the call.</td>
</tr>
<tr>
<td>Step 12</td>
<td>100 Trying—SIP server to Cisco ATA A</td>
<td>SIP server returns a 100 trying message indicating that the BYE request has been received.</td>
</tr>
<tr>
<td>Step 13</td>
<td>BYE—SIP server to Cisco ATA B</td>
<td>SIP server passes the BYE request to Cisco ATA B.</td>
</tr>
<tr>
<td>Step 14</td>
<td>200 OK—Cisco ATA B to SIP server</td>
<td>Cisco ATA B sends a 200 OK message to the SIP server indicating that Cisco ATA B has received the BYE request.</td>
</tr>
<tr>
<td>Step 15</td>
<td>200 OK—SIP server to Cisco ATA A</td>
<td>SIP server passes the BYE request to Cisco ATA A.</td>
</tr>
</tbody>
</table>

A two-way voice path is established between Cisco ATA A and Cisco ATA B.
Table D-6  Log Listings

1. INVITE sip:9000@192.168.2.97;user=phone SIP/2.0
   Via: SIP/2.0/UDP 192.168.3.175
   From: <sip:8000@192.168.2.97;user=phone>;tag=2819471139
   To: <sip:9000@192.168.2.97;user=phone>
   Call-ID: 488372010192.168.3.175
   CSeq: 1 INVITE
   Contact: <sip:8000@192.168.3.175;user=phone;transport=udp>
   User-Agent: Cisco ATA v2.12 ata186 (0928a)
   Expires: 300
   Content-Length: 253
   Content-Type: application/sdp
   v=0
   o=8000 206154 206154 IN IP4 192.168.3.175
   s=ATA186 Call
   c=IN IP4 192.168.3.175
   t=0 0
   m=audio 10000 RTP/AVP 0 18 8 101
   a=rtpmap:0 PCMU/8000/1
   a=rtpmap:18 G729/8000/1
   a=rtpmap:8 PCMA/8000/1
   a=rtpmap:101 telephone-event/8000
   a=fmtp:101 0-15

2. SIP/2.0 100 Trying
   Via: SIP/2.0/UDP 192.168.3.175
   Call-ID: 488372010192.168.3.175
   From: <sip:8000@192.168.2.97;user=phone>;tag=2819471139
   To: <sip:9000@192.168.2.97;user=phone>
   CSeq: 1 INVITE
   Content-Length: 0

3. INVITE sip:9000@192.168.2.194;user=phone SIP/2.0
   Record-Route: <sip:9000@192.168.2.97;user=phone;maddr=192.168.2.97>
   Via: SIP/2.0/UDP 192.168.2.97:5060;branch=140fed6e-f61cbd1a-52f223b1-9beb149a-1
   Via: SIP/2.0/UDP 192.168.3.175
   From: <sip:9000@192.168.2.97;user=phone>;tag=2819471139
   To: <sip:9000@192.168.2.97;user=phone>
   Call-ID: 488372010192.168.3.175
   CSeq: 1 INVITE
   Contact: <sip:8000@192.168.3.175;user=phone;transport=udp>
   User-Agent: Cisco ATA v2.12 ata186 (0928a)
   Expires: 300
   Content-Length: 253
   Content-Type: application/sdp
   v=0
   o=8000 206154 206154 IN IP4 192.168.3.175
   s=ATA186 Call
   c=IN IP4 192.168.3.175
   t=0 0
   m=audio 10000 RTP/AVP 0 18 8 101
   a=rtpmap:0 PCMU/8000/1
   a=rtpmap:18 G729/8000/1
   a=rtpmap:8 PCMA/8000/1
   a=rtpmap:101 telephone-event/8000
   a=fmtp:101 0-15
Table D-6  Log Listings (continued)

4. SIP/2.0 100 Trying  
   Via: SIP/2.0/UDP 192.168.2.97:5060;branch=140fed6e-f61cbdl1a-52f223b1-9beb149a-1  
   Via: SIP/2.0/UDP 192.168.3.175  
   Record-Route: <sip:9000@192.168.2.97;user=phone;maddr=192.168.2.97>  
   From: <sip:8000@192.168.2.97;user=phone>;tag=2819471139  
   To: <sip:9000@192.168.2.97;user=phone>;tag=909616993  
   Call-ID: 488337201@192.168.3.175  
   CSeq: 1 INVITE  
   Server: Cisco ATA v2.12 ata186 (0928a)  
   Content-Length: 0

5. SIP/2.0 180 Ringing  
   Via: SIP/2.0/UDP 192.168.2.97:5060;branch=140fed6e-f61cbdl1a-52f223b1-9beb149a-1  
   Via: SIP/2.0/UDP 192.168.3.175  
   Record-Route: <sip:9000@192.168.2.97;user=phone;maddr=192.168.2.97>  
   From: <sip:8000@192.168.2.97;user=phone>;tag=2819471139  
   To: <sip:9000@192.168.2.97;user=phone>;tag=909616993  
   Call-ID: 488337201@192.168.3.175  
   CSeq: 1 INVITE  
   Server: Cisco ATA v2.12 ata186 (0928a)  
   Content-Length: 0

6. SIP/2.0 180 Ringing  
   Via: SIP/2.0/UDP 192.168.2.97:5060;branch=140fed6e-f61cbdl1a-52f223b1-9beb149a-1  
   Via: SIP/2.0/UDP 192.168.3.175  
   Record-Route: <sip:9000@192.168.2.97;user=phone;maddr=192.168.2.97>  
   From: <sip:8000@192.168.2.97;user=phone>;tag=2819471139  
   To: <sip:9000@192.168.2.97;user=phone>;tag=909616993  
   Call-ID: 488337201@192.168.3.175  
   CSeq: 1 INVITE  
   Server: Cisco ATA v2.12 ata186 (0928a)  
   Content-Length: 0

7. SIP/2.0 200 OK  
   Via: SIP/2.0/UDP 192.168.2.97:5060;branch=140fed6e-f61cbdl1a-52f223b1-9beb149a-1  
   Via: SIP/2.0/UDP 192.168.3.175  
   Record-Route: <sip:9000@192.168.2.97;user=phone;maddr=192.168.2.97>  
   From: <sip:8000@192.168.2.97;user=phone>;tag=2819471139  
   To: <sip:9000@192.168.2.97;user=phone>;tag=909616993  
   Call-ID: 488337201@192.168.3.175  
   CSeq: 1 INVITE  
   Contact: <sip:9000@192.168.2.194;user=phone;transport=udp>  
   Server: Cisco ATA v2.12 ata186 (0928a)  
   Content-Length: 199  
   Content-Type: application/sdp  
   v=0  
   o=9000 206275 206275 IN IP4 192.168.2.194  
   s=ATA186 Call  
   c=IN IP4 192.168.2.194  
   t=0 0  
   m=audio 10000 RTP/AVP 0 101  
   a=rtpmap:0 PCMU/8000/1  
   a=rtpmap:101 telephone-event/8000  
   a=fmtp:101 0-15
### Call Flow Scenarios for Successful Calls

<table>
<thead>
<tr>
<th>Step</th>
<th>Request/Response</th>
</tr>
</thead>
<tbody>
<tr>
<td>8.</td>
<td>SIP/2.0 200 OK</td>
</tr>
<tr>
<td></td>
<td>Via: SIP/2.0/UDP 192.168.3.175</td>
</tr>
<tr>
<td></td>
<td>Record-Route: <a href="">sip:9000@192.168.2.97;branch=v0;transport=udp</a></td>
</tr>
<tr>
<td></td>
<td>From: <a href="">sip:8000@192.168.2.97;transport=udp</a></td>
</tr>
<tr>
<td></td>
<td>To: <a href="">sip:9000@192.168.2.97;transport=udp</a></td>
</tr>
<tr>
<td></td>
<td>Call-ID: 488337201@192.168.3.175</td>
</tr>
<tr>
<td></td>
<td>CSeq: 1 INVITE</td>
</tr>
<tr>
<td></td>
<td>Contact: <a href="">sip:9000@192.168.2.97;transport=udp</a></td>
</tr>
<tr>
<td></td>
<td>Server: Cisco ATA v2.12 ata186 (0928a)</td>
</tr>
<tr>
<td></td>
<td>Content-Type: application/sdp</td>
</tr>
<tr>
<td></td>
<td>v=0</td>
</tr>
<tr>
<td></td>
<td>o=9000 206275 206275 IN IP4 192.168.2.194</td>
</tr>
<tr>
<td></td>
<td>s=ATA186 Call</td>
</tr>
<tr>
<td></td>
<td>c=IN IP4 192.168.2.194</td>
</tr>
<tr>
<td></td>
<td>t=0 0</td>
</tr>
<tr>
<td></td>
<td>m=audio 10000 RTP/AVP 0 101</td>
</tr>
<tr>
<td></td>
<td>a=rtpmap:0 PCMU/8000/1</td>
</tr>
<tr>
<td></td>
<td>a=rtpmap:101 telephone-event/8000</td>
</tr>
<tr>
<td></td>
<td>a=fmtp:101 0-15</td>
</tr>
<tr>
<td>9.</td>
<td>ACK sip:9000@192.168.2.97;user=phone SIP/2.0</td>
</tr>
<tr>
<td></td>
<td>Route: <a href="">sip:9000@192.168.2.97;branch=v0;transport=udp</a></td>
</tr>
<tr>
<td></td>
<td>Via: SIP/2.0/UDP 192.168.3.175</td>
</tr>
<tr>
<td></td>
<td>From: <a href="">sip:8000@192.168.2.97;transport=udp</a></td>
</tr>
<tr>
<td></td>
<td>To: <a href="">sip:9000@192.168.2.97;transport=udp</a></td>
</tr>
<tr>
<td></td>
<td>Call-ID: 488337201@192.168.3.175</td>
</tr>
<tr>
<td></td>
<td>CSeq: 1 ACK</td>
</tr>
<tr>
<td></td>
<td>User-Agent: Cisco ATA v2.12 ata186 (0928a)</td>
</tr>
<tr>
<td></td>
<td>Content-Length: 0</td>
</tr>
<tr>
<td>10.</td>
<td>ACK sip:9000@192.168.2.97;user=phone SIP/2.0</td>
</tr>
<tr>
<td></td>
<td>Route: <a href="">sip:9000@192.168.2.97;branch=v0;transport=udp</a></td>
</tr>
<tr>
<td></td>
<td>Via: SIP/2.0/UDP 192.168.3.175</td>
</tr>
<tr>
<td></td>
<td>From: <a href="">sip:8000@192.168.2.97;transport=udp</a></td>
</tr>
<tr>
<td></td>
<td>To: <a href="">sip:9000@192.168.2.97;transport=udp</a></td>
</tr>
<tr>
<td></td>
<td>Call-ID: 488337201@192.168.3.175</td>
</tr>
<tr>
<td></td>
<td>CSeq: 1 ACK</td>
</tr>
<tr>
<td></td>
<td>User-Agent: Cisco ATA v2.12 ata186 (0928a)</td>
</tr>
<tr>
<td></td>
<td>Content-Length: 0</td>
</tr>
<tr>
<td>11.</td>
<td>BYE sip:9000@192.168.2.97;user=phone SIP/2.0</td>
</tr>
<tr>
<td></td>
<td>Route: <a href="">sip:9000@192.168.2.97;branch=v0;transport=udp</a></td>
</tr>
<tr>
<td></td>
<td>Via: SIP/2.0/UDP 192.168.3.175</td>
</tr>
<tr>
<td></td>
<td>From: <a href="">sip:8000@192.168.2.97;transport=udp</a></td>
</tr>
<tr>
<td></td>
<td>To: <a href="">sip:9000@192.168.2.97;transport=udp</a></td>
</tr>
<tr>
<td></td>
<td>Call-ID: 488337201@192.168.3.175</td>
</tr>
<tr>
<td></td>
<td>CSeq: 2 BYE</td>
</tr>
<tr>
<td></td>
<td>User-Agent: Cisco ATA v2.12 ata186 (0928a)</td>
</tr>
<tr>
<td></td>
<td>Content-Length: 0</td>
</tr>
<tr>
<td>12.</td>
<td>SIP/2.0 100 Trying</td>
</tr>
<tr>
<td></td>
<td>Via: SIP/2.0/UDP 19.168.3.175</td>
</tr>
<tr>
<td></td>
<td>Call-ID: 488337201@192.168.3.175</td>
</tr>
<tr>
<td></td>
<td>From: <a href="">sip:8000@192.168.2.97;transport=udp</a></td>
</tr>
<tr>
<td></td>
<td>To: <a href="">sip:9000@192.168.2.97;transport=udp</a></td>
</tr>
<tr>
<td></td>
<td>CSeq: 2 BYE</td>
</tr>
<tr>
<td></td>
<td>Content-Length: 0</td>
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</table>
### Table D-6  Log Listings (continued)

<table>
<thead>
<tr>
<th>Call Number</th>
<th>Message</th>
<th>Details</th>
</tr>
</thead>
<tbody>
<tr>
<td>13.</td>
<td>BYE sip:9000@192.168.2.194;user=phone SIP/2.0</td>
<td>Via: SIP/2.0/UDP 192.168.2.97:5060;branch=b499b4be-d7995db7-980cd8af-e5ba35f5-1</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Via: SIP/2.0/UDP 192.168.3.175</td>
</tr>
<tr>
<td></td>
<td></td>
<td>From: <a href="">sip:8000@192.168.2.97;user=phone</a>;tag=2819471139</td>
</tr>
<tr>
<td></td>
<td></td>
<td>To: <a href="">sip:9000@192.168.2.97;user=phone</a>;tag=909616993</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Call-ID: 488337201@192.168.3.175</td>
</tr>
<tr>
<td></td>
<td></td>
<td>CSeq: 2 BYE</td>
</tr>
<tr>
<td></td>
<td></td>
<td>User-Agent: Cisco ATA v2.12 ata186 (0928a)</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Content-Length: 0</td>
</tr>
<tr>
<td>14.</td>
<td>SIP/2.0 200 OK</td>
<td>Via: SIP/2.0/UDP 192.168.2.97:5060;branch=b499b4be-d7995db7-980cd8af-e5ba35f5-1</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Via: SIP/2.0/UDP 192.168.3.175</td>
</tr>
<tr>
<td></td>
<td></td>
<td>From: <a href="">sip:8000@192.168.2.97;user=phone</a>;tag=2819471139</td>
</tr>
<tr>
<td></td>
<td></td>
<td>To: <a href="">sip:9000@192.168.2.97;user=phone</a>;tag=909616993</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Call-ID: 488337201@192.168.3.175</td>
</tr>
<tr>
<td></td>
<td></td>
<td>CSeq: 2 BYE</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Server: Cisco ATA v2.12 ata186 (0928a)</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Content-Length: 0</td>
</tr>
<tr>
<td>15.</td>
<td>SIP/2.0 200 OK</td>
<td>Via: SIP/2.0/UDP 192.168.3.175</td>
</tr>
<tr>
<td></td>
<td></td>
<td>From: <a href="">sip:8000@192.168.2.97;user=phone</a>;tag=2819471139</td>
</tr>
<tr>
<td></td>
<td></td>
<td>To: <a href="">sip:9000@192.168.2.97;user=phone</a>;tag=909616993</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Call-ID: 488337201@192.168.3.175</td>
</tr>
<tr>
<td></td>
<td></td>
<td>CSeq: 2 BYE</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Server: Cisco ATA v2.12 ata186 (0928a)</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Content-Length: 0</td>
</tr>
</tbody>
</table>
Cisco ATA-to-Cisco ATA—Basic SIP to SIP Call with Authentication

Figure D-4 illustrates a call from one Cisco ATA to another. Authentication by the SIP server is required.

The call flow is as follows:
1. Authentication is requested for call initiated by Cisco ATA A.
2. Call is established between Cisco ATA A and Cisco ATA B.
3. Call is terminated.
### Call Flow Scenarios for Successful Calls

**Table D-7 Action Descriptions**

<table>
<thead>
<tr>
<th>Step</th>
<th>Action</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Step 1</td>
<td>INVITE—Cisco ATA A to SIP server</td>
<td>Cisco ATA A sends a call session INVITE request to the SIP server to pass on to Cisco ATA B.</td>
</tr>
<tr>
<td>Step 2</td>
<td>100 Trying—SIP server to Cisco ATA A</td>
<td>SIP server returns a 100 Trying message, indicating that the INVITE request has been received.</td>
</tr>
<tr>
<td>Step 3</td>
<td>407 Proxy Authentication Required—SIP server to Cisco ATA A</td>
<td>SIP server returns a request for authentication to Cisco ATA A.</td>
</tr>
<tr>
<td>Step 4</td>
<td>ACK—Cisco ATA A to SIP server</td>
<td>Cisco ATA A acknowledges the request for authentication.</td>
</tr>
<tr>
<td>Step 5</td>
<td>INVITE—Cisco ATA A to SIP server</td>
<td>Cisco ATA A sends a call session INVITE request along with authentication credential to the SIP server to pass on to Cisco ATA B.</td>
</tr>
<tr>
<td>Step 6</td>
<td>100 Trying—SIP server to Cisco ATA A</td>
<td>SIP server returns a 100 Trying message, indicating that the INVITE request has been received.</td>
</tr>
<tr>
<td>Step 7</td>
<td>INVITE—SIP server to Cisco ATA B</td>
<td>SIP server sends the call session INVITE request to Cisco ATA B.</td>
</tr>
<tr>
<td>Step 8</td>
<td>100 Trying—Cisco ATA B to SIP server</td>
<td>Cisco ATA B returns a 100 trying message indicating that the INVITE request has been received.</td>
</tr>
<tr>
<td>Step 9</td>
<td>180 Ringing—Cisco ATA B to SIP server</td>
<td>Cisco ATA B sends a 180 ringing response to the SIP server to pass on to Cisco ATA A.</td>
</tr>
<tr>
<td>Step 10</td>
<td>180 Ringing—SIP server to Cisco ATA A</td>
<td>SIP server sends the 180 ringing response to Cisco ATA A.</td>
</tr>
<tr>
<td>Step 11</td>
<td>200 OK—Cisco ATA B to SIP server</td>
<td>Cisco ATA B sends a 200 OK message to the SIP server indicating that a connection has been established.</td>
</tr>
<tr>
<td>Step 12</td>
<td>200 OK—SIP server to Cisco ATA A</td>
<td>SIP server passes the 200 OK message to Cisco ATA A.</td>
</tr>
<tr>
<td>Step 13</td>
<td>ACK—Cisco ATA A to SIP server</td>
<td>Cisco ATA A sends acknowledgment of the 200 OK response to the SIP server to pass on to Cisco ATA B.</td>
</tr>
<tr>
<td>Step 14</td>
<td>ACK—SIP server to Cisco ATA B</td>
<td>SIP server passes ACK response to Cisco ATA B.</td>
</tr>
<tr>
<td>Step 15</td>
<td>BYE—Cisco ATA A to SIP server</td>
<td>Cisco ATA A terminates the call session and sends a BYE request to the SIP server indicating that Cisco ATA A wants to terminate the call.</td>
</tr>
<tr>
<td>Step 16</td>
<td>100 Trying—SIP server to Cisco ATA A</td>
<td>SIP server returns a 100 trying message indicating that the BYE request has been received.</td>
</tr>
<tr>
<td>Step 17</td>
<td>BYE—SIP server to Cisco ATA B</td>
<td>SIP server passes the BYE request to Cisco ATA B.</td>
</tr>
<tr>
<td>Step 18</td>
<td>200 OK—Cisco ATA B to SIP server</td>
<td>Cisco ATA 186 B sends a 200 OK message to the SIP server indicating that Cisco ATA 186 B has received the BYE request.</td>
</tr>
<tr>
<td>Step 19</td>
<td>200 OK—SIP server to Cisco ATA A</td>
<td>SIP server passes the BYE request to Cisco ATA A.</td>
</tr>
</tbody>
</table>

A two-way voice path is established between Cisco ATA A and Cisco ATA B.
### Table D-8 Log Listings

1. **INVITE sip:9000@192.168.2.81;user=phone SIP/2.0**  
   Via: SIP/2.0/UDP 192.168.3.175  
   From: <sip:8000@192.168.2.81;user=phone>;tag=3515135869  
   To: <sip:9000@192.168.2.81;user=phone>  
   Call-ID: 5571886500192.168.3.175  
   CSeq: 1 INVITE  
   Contact: <sip:8000@192.168.3.175;user=phone;transport=udp>  
   User-Agent: Cisco ATA v2.12 ata186 (0928a)  
   Expires: 300  
   Content-Length: 253  
   Content-Type: application/sdp  
   v=0  
   o=8000 177731 177731 IN IP4 192.168.3.175  
   s=ATA186 Call  
   c=IN IP4 192.168.3.175  
   t=0 0  
   m=audio 10000 RTP/AVP 0 18 8 101  
   a=rtpmap:0 PCMU/8000 1  
   a=rtpmap:18 G729/8000 1  
   a=rtpmap:8 PCMA/8000 1  
   a=rtpmap:101 telephone-event/8000  
   a=fmtp:101 0-15

2. **SIP/2.0 100 Trying**  
   Via: SIP/2.0/UDP 192.168.3.175  
   Call-ID: 5571886500192.168.3.175  
   From: <sip:8000@192.168.2.81;user=phone>;tag=3515135869  
   To: <sip:9000@192.168.2.81;user=phone>  
   CSeq: 1 INVITE  
   Content-Length: 0

3. **SIP/2.0 407 Proxy Authentication Required**  
   Via: SIP/2.0/UDP 192.168.3.175  
   Call-ID: 5571886500192.168.3.175  
   From: <sip:8000@192.168.2.81;user=phone>;tag=3515135869  
   To: <sip:9000@192.168.2.81;user=phone>;tag=600eaef7-7c3a549a  
   CSeq: 1 INVITE  
   Proxy-Authenticate: DIGEST realm="CISCO", nonce="3bd76584"  
   Content-Length: 0

4. **ACK sip:9000@192.168.2.81:5060 SIP/2.0**  
   Via: SIP/2.0/UDP 192.168.3.175  
   From: <sip:8000@192.168.2.81;user=phone>;tag=3515135869  
   To: <sip:9000@192.168.2.81;user=phone>;tag=600eaef7-7c3a549a  
   Call-ID: 5571886500192.168.3.175  
   CSeq: 1 ACK  
   User-Agent: Cisco ATA v2.12 ata186 (0928a)  
   Content-Length: 0
Table D-8  Log Listings (continued)

5. INVITE sip:9000@192.168.2.81;user=phone SIP/2.0
Via: SIP/2.0/UDP 192.168.3.175
From: <sip:8000@192.168.2.81;user=phone>;tag=3515135869
To: <sip:9000@192.168.2.81;user=phone>
Call-ID: 5571886508192.168.3.175
CSeq: 2 INVITE
Contact: <sip:8000@192.168.3.175;user=phone;transport=udp>
User-Agent: Cisco ATA v2.12 ata186 (0928a)
Proxy-Authorization: Digest username="8000",realm="CISCO",nonce="3bd76584", uri="sip:9000@192.168.2.81",response="6e91de67ad976997ffac76f0398ef224"
Expires: 300
Content-Length: 253
Content-Type: application/sdp
v=0
o=8000 177738 177738 IN IP4 192.168.3.175
s=ATA186 Call
c=IN IP4 192.168.3.175
t=0 0
m=audio 10000 RTP/AVP 0 18 8 101
a=rtpmap:0 PCMU/8000/1
a=rtpmap:18 G729/8000/1
a=rtpmap:8 PCMA/8000/1
a=rtpmap:101 telephone-event/8000
a=fmt:101 0-15

6. SIP/2.0 100 Trying
Via: SIP/2.0/UDP 192.168.3.175
Call-ID: 5571886508192.168.3.175
From: <sip:8000@192.168.2.81;user=phone>;tag=3515135869
To: <sip:9000@192.168.2.81;user=phone>
CSeq: 2 INVITE
Content-Length: 0

7. INVITE sip:9000@192.168.2.194;user=phone SIP/2.0
Record-Route: <sip:9000@192.168.2.194:5060;user=phone>maddr=192.168.2.194
Via: SIP/2.0/UDP 192.168.2.194:5060:branch=c0b3510a-819f9d1a-ea43c345-9604747f-1
Via: SIP/2.0/UDP 192.168.3.175
From: <sip:8000@192.168.2.194;user=phone>;tag=3515135869
To: <sip:9000@192.168.2.194;user=phone>
Call-ID: 5571886508192.168.3.175
CSeq: 2 INVITE
Contact: <sip:8000@192.168.3.175;user=phone;transport=udp>
User-Agent: Cisco ATA v2.12 ata186 (0928a)
Proxy-Authorization: Digest username="8000",realm="CISCO",nonce="3bd76584", uri="sip:9000@192.168.2.194",response="6e91de67ad976997ffac76f0398ef224"
Expires: 300
Content-Length: 253
Content-Type: application/sdp
v=0
o=8000 177738 177738 IN IP4 192.168.3.175
s=ATA186 Call
c=IN IP4 192.168.3.175
t=0 0
m=audio 10000 RTP/AVP 0 18 8 101
a=rtpmap:0 PCMU/8000/1
a=rtpmap:18 G729/8000/1
a=rtpmap:8 PCMA/8000/1
a=rtpmap:101 telephone-event/8000
a=fmt:101 0-15
### Table D-8 Log Listings (continued)

8. **SIP/2.0 100 Trying**
   Via: SIP/2.0/UDP 192.168.2.81:5060;branch=c0b3510a-819f9d1a-ea43c345-9604747f-1
   Via: SIP/2.0/UDP 192.168.3.175
   Record-Route: <sip:9000@192.168.2.81:5060;user=phone;rn=192.168.2.81>
   From: <sip:8000@192.168.2.81;user=phone;rn=351513869>
   To: <sip:9000@192.168.2.81;user=phone;rn=100585329>
   Call-ID: 557188650192.168.3.175
   CSeq: 2 INVITE
   Server: Cisco ATA v2.12 ata186 (0928a)
   Content-Length: 0

9. **SIP/2.0 180 Ringing**
   Via: SIP/2.0/UDP 192.168.2.81:5060;branch=c0b3510a-819f9d1a-ea43c345-9604747f-1
   Via: SIP/2.0/UDP 192.168.3.175
   Record-Route: <sip:9000@192.168.2.81:5060;user=phone;rn=192.168.2.81>
   From: <sip:8000@192.168.2.81;user=phone;rn=351513869>
   To: <sip:9000@192.168.2.81;user=phone;rn=100585329>
   Call-ID: 557188650192.168.3.175
   CSeq: 2 INVITE
   Server: Cisco ATA v2.12 ata186 (0928a)
   Content-Length: 0

10. **SIP/2.0 180 Ringing**
    Via: SIP/2.0/UDP 192.168.2.81:5060;branch=c0b3510a-819f9d1a-ea43c345-9604747f-1
    Via: SIP/2.0/UDP 192.168.3.175
    Record-Route: <sip:9000@192.168.2.81:5060;user=phone;rn=192.168.2.81>
    From: <sip:8000@192.168.2.81;user=phone;rn=351513869>
    To: <sip:9000@192.168.2.81;user=phone;rn=100585329>
    Call-ID: 557188650192.168.3.175
    CSeq: 2 INVITE
    Server: Cisco ATA v2.12 ata186 (0928a)
    Content-Length: 0

11. **SIP/2.0 200 OK**
    Via: SIP/2.0/UDP 192.168.2.81:5060;branch=c0b3510a-819f9d1a-ea43c345-9604747f-1
    Via: SIP/2.0/UDP 192.168.3.175
    Record-Route: <sip:9000@192.168.2.81:5060;user=phone;rn=192.168.2.81>
    From: <sip:8000@192.168.2.81;user=phone;rn=351513869>
    To: <sip:9000@192.168.2.81;user=phone;rn=100585329>
    Call-ID: 557188650192.168.3.175
    CSeq: 2 INVITE
    Contact: <sip:9000@192.168.3.194;user=phone;transport=udp>
    Server: Cisco ATA v2.12 ata186 (0928a)
    Content-Length: 199
    Content-Type: application/sdp
    v=0
    o=9000 179263 179263 IN IP4 192.168.2.194
    s=ATA186 Call
    c=IN IP4 192.168.2.194
    t=0 0
    m=audio 10000 RTP/AVP 0 101
    a=rtpmap:0 PCMU/8000/1
    a=rtpmap:101 telephone-event/8000
    a=fmtp:101 0-15
Table D-8  Log Listings (continued)

12.  SIP/2.0 200 OK
    Via: SIP/2.0/UDP 192.168.3.175
    Record-Route: <sip:9000@192.168.2.81;user=phone>maddr=192.168.2.81
    From: <sip:8000@192.168.2.81;user=phone>;tag=3515135869
    To: <sip:9000@192.168.2.81;user=phone;tag=100585329
    Call-ID: 5571886508192.168.3.175
    CSeq: 2 INVITE
    Content-Type: application/sdp
    v=0
    o=9000 179263 179263 IN IP4 192.168.2.194
    s=ATA186 Call
    c=IN IP4 192.168.2.194
    t=0 0
    m=audio 10000 RTP/AVP 0 101
    a=rtpmap:0 PCMU/8000/1
    a=rtpmap:101 telephone-event/8000
    a=fmtp:101 0-15

13.  ACK sip:9000@192.168.2.81;user=phone SIP/2.0
    Route: <sip:9000@192.168.2.194;5060;user=phone;transport=udp>
    Via: SIP/2.0/UDP 192.168.3.175
    From: <sip:8000@192.168.2.81;user=phone;>;tag=3515135869
    To: <sip:9000@192.168.2.81;user=phone;tag=100585329
    Call-ID: 5571886508192.168.3.175
    CSeq: 2 ACK
    User-Agent: Cisco ATA v2.12 ata186 (0928a)
    Content-Length: 0

14.  ACK sip:9000@192.168.2.194;user=phone SIP/2.0
    Route: <sip:8000@192.168.2.194;5060;user=phone;transport=udp>
    Via: SIP/2.0/UDP 192.168.3.175
    From: <sip:9000@192.168.2.81;user=phone;>;tag=3515135869
    To: <sip:9000@192.168.2.81;user=phone;tag=100585329
    Call-ID: 5571886508192.168.3.175
    CSeq: 2 ACK
    User-Agent: Cisco ATA v2.12 ata186 (0928a)
    Content-Length: 0

15.  BYE sip:9000@192.168.2.81;user=phone SIP/2.0
    Route: <sip:9000@192.168.2.194;5060;user=phone;transport=udp>
    Via: SIP/2.0/UDP 192.168.3.175
    From: <sip:9000@192.168.2.81;user=phone;>;tag=3515135869
    To: <sip:9000@192.168.2.81;user=phone;tag=100585329
    Call-ID: 5571886508192.168.3.175
    CSeq: 3 BYE
    User-Agent: Cisco ATA v2.12 ata186 (0928a)
    Content-Length: 0

16.  SIP/2.0 100 Trying
    Via: SIP/2.0/UDP 19.168.3.175
    Call-ID: 5571886508192.168.3.175
    From: <sip:8000@192.168.2.81;user=phone;>;tag=3515135869
    To: <sip:9000@192.168.2.81;user=phone;>;tag=100585329
    CSeq: 3 BYE
    Content-Length: 0
### Table D-8 Log Listings (continued)

<table>
<thead>
<tr>
<th>Number</th>
<th>Log Entry</th>
</tr>
</thead>
</table>
| 17.    | BYE sip:9000@192.168.2.194;user=phone SIP/2.0  
Via: SIP/2.0/UDP 192.168.2.81:5060;branch=424f3898-9ef87cec-82179ac3-50eebd3-1  
Via: SIP/2.0/UDP 192.168.3.175  
From: <sip:8000@192.168.2.81;user=phone>;tag=3515135869  
To: <sip:9000@192.168.2.81;user=phone>;tag=100585329  
Call-ID: 5571886500192.168.3.175  
CSeq: 3 BYE  
User-Agent: Cisco ATA v2.12 ata186 (0928a)  
Content-Length: 0 |
| 18.    | SIP/2.0 200 OK  
Via: SIP/2.0/UDP 192.168.2.81:5060;branch=424f3898-9ef87cec-82179ac3-50eebd3-1  
Via: SIP/2.0/UDP 192.168.3.175  
From: <sip:8000@192.168.2.81;user=phone>;tag=3515135869  
To: <sip:9000@192.168.2.81;user=phone>;tag=100585329  
Call-ID: 5571886500192.168.3.175  
CSeq: 3 BYE  
Server: Cisco ATA v2.12 ata186 (0928a)  
Content-Length: 0 |
| 19.    | SIP/2.0 200 OK  
Via: SIP/2.0/UDP 192.168.3.175  
From: <sip:8000@192.168.2.81;user=phone>;tag=3515135869  
To: <sip:9000@192.168.2.81;user=phone>;tag=100585329  
Call-ID: 5571886500192.168.3.175  
CSeq: 3 BYE  
Server: Cisco ATA v2.12 ata186 (0928a)  
Content-Length: 0 |
## Recommended Cisco ATA Tone Parameter Values by Country

This section provides tables of recommended tone parameters for the following countries, listed alphabetically:

**Note** The extended tone format used by some countries is available only with Cisco ATA software version 3.0 or later. For more information about tone parameter syntax and formats, see the “Tone Configuration Parameters” section on page 5-53.

<table>
<thead>
<tr>
<th>Country</th>
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<tbody>
<tr>
<td>Argentina</td>
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<tr>
<td>Austria</td>
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<tr>
<td>Belgium</td>
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<tr>
<td>Brazil</td>
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<tr>
<td>Canada</td>
</tr>
<tr>
<td>China</td>
</tr>
<tr>
<td>Columbia</td>
</tr>
<tr>
<td>Czech Republic</td>
</tr>
<tr>
<td>Denmark</td>
</tr>
<tr>
<td>Egypt</td>
</tr>
<tr>
<td>Finland</td>
</tr>
<tr>
<td>France</td>
</tr>
<tr>
<td>Germany</td>
</tr>
<tr>
<td>Greece</td>
</tr>
<tr>
<td>Hong Kong</td>
</tr>
<tr>
<td>Hungary</td>
</tr>
<tr>
<td>Iceland</td>
</tr>
<tr>
<td>India</td>
</tr>
<tr>
<td>Indonesia</td>
</tr>
<tr>
<td>Ireland</td>
</tr>
</tbody>
</table>
- Israel
- Italy
- Japan
- Korea
- Luxembourg
- Malaysia
- Mexico
- Netherlands
- New Zealand
- Norway
- Pakistan
- Panama
- Peru
- Phillippines
- Poland
- Portugal
- Russia
- Saudi Arabia
- Singapore
- Slovakia
- Slovenia
- South Africa
- Spain
- Sweden
- Switzerland
- Taiwan
- Thailand
- Turkey
- United Kingdom
- United States
- Venezuela
### Table E-1  Argentina

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Recommended Values</th>
</tr>
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<tbody>
<tr>
<td>DialTone</td>
<td>1,30958,0,3125,0,1,0,0,0</td>
</tr>
<tr>
<td>BusyTone</td>
<td>1,30958,0,1757,0,0,2400,1600,0</td>
</tr>
<tr>
<td>ReorderTone</td>
<td>1,30958,0,1757,0,0,2400,3200,0</td>
</tr>
<tr>
<td>RingbackTone</td>
<td>1,30958,0,1971,0,0,8000,32000,0</td>
</tr>
<tr>
<td>SITTone</td>
<td>1,30958,0,1757,0,0,2400,1600,0</td>
</tr>
</tbody>
</table>

### Table E-2  Australia

<table>
<thead>
<tr>
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<th>Recommended Values</th>
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<td>DialTone</td>
<td>2,31163,30958,1477,1566,1,0,0,0</td>
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<tr>
<td>BusyTone</td>
<td>1,30958,0,2212,0,0,3000,3000,0</td>
</tr>
<tr>
<td>ReorderTone</td>
<td>1,31163,0,2086,0,0,3000,3000,0</td>
</tr>
<tr>
<td>RingbackTone</td>
<td>102,31163,1477,30742,1654,2,3200,1600,32000,0</td>
</tr>
<tr>
<td>SITTone</td>
<td>1,30958,0,2212,0,0,20000,4000,0</td>
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</tbody>
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### Table E-3  Austria

<table>
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<th>Parameter</th>
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<td>DialTone</td>
<td>1,31000,0,3089,0,1,0,0,0</td>
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<tr>
<td>BusyTone</td>
<td>1,31000,0,1737,0,0,3200,3200,0</td>
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<tr>
<td>ReorderTone</td>
<td>1,31000,0,1737,0,0,1600,1600,0</td>
</tr>
<tr>
<td>RingbackTone</td>
<td>1,31000,0,1949,0,0,8000,40000,0</td>
</tr>
<tr>
<td>SITTone</td>
<td>101,3,24062,3640,14876,4778,5126,5297,3,2664,0,2664,0,2664,8000,0,0</td>
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### Table E-4  Belgium

<table>
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<td>DialTone</td>
<td>1,30958,0,4952,0,1,0,0,0</td>
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<td>BusyTone</td>
<td>1,30958,0,1757,0,0,4000,4000,0</td>
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<tr>
<td>ReorderTone</td>
<td>1,30958,0,1757,0,0,1336,1336,0</td>
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<tr>
<td>RingbackTone</td>
<td>1,30958,0,1971,0,0,8000,24000,0</td>
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<tr>
<td>SITTone</td>
<td>1,30958,0,1757,0,0,1336,1336,0</td>
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### Table E-5  Brazil

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<td>DialTone</td>
<td>1,30958,0,3125,0,1,0,0,0</td>
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<tr>
<td>BusyTone</td>
<td>1,30958,0,1757,0,0,2000,2000,0</td>
</tr>
<tr>
<td>ReorderTone</td>
<td>1,30958,0,1757,0,0,2000,2000,0</td>
</tr>
<tr>
<td>RingbackTone</td>
<td>1,30958,0,1971,0,0,8000,32000,0</td>
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<tr>
<td>SITTone</td>
<td>100,1,30958,1757,0,0,0,0,2,6000,2000,2000,0,0,0,0</td>
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### Table E-6  Canada

<table>
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<td>DialTone</td>
<td>2,31537,30830,1490,1859,1,0,0,0</td>
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<tr>
<td>BusyTone</td>
<td>2,30466,28958,1246,1583,0,4000,4000,0</td>
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<tr>
<td>ReorderTone</td>
<td>2,30466,28958,1246,1583,0,2000,2000,0</td>
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<tr>
<td>RingbackTone</td>
<td>2,30830,30466,793,862,0,8000,24000,0</td>
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<td>SITTone</td>
<td>2,30466,28958,1246,1583,0,2000,2000,0</td>
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### Table E-7  China

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<td>ReorderTone</td>
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<td>RingbackTone</td>
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### Table E-8  Columbia

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<tr>
<td>RingbackTone</td>
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### Table E-9  Czech Republic

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<tr>
<td>BusyTone</td>
<td>1,30958,0,1757,0,0,2664,2664,0</td>
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<td>reorderTone</td>
<td>1,30958,0,1757,0,0,1336,1336,0</td>
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<td>RingbackTone</td>
<td>1,30958,0,1971,0,0,8000,32000,0</td>
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<td>SITTone</td>
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### Table E-10  Denmark

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<tr>
<td>BusyTone</td>
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<td>reorderTone</td>
<td>1,30958,0,1757,0,0,2000,2000,0</td>
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<tr>
<td>RingbackTone</td>
<td>1,30958,0,1757,0,0,8000,32000,0</td>
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<tr>
<td>SITTone</td>
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### Table E-11  Egypt

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### Table E-12  Finland

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<td>BusyTone</td>
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## Table E-13 France

<table>
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<td>RingbackTone</td>
<td>1,30830,0,2038,0,0,12000,28000,0</td>
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## Table E-14 Germany

<table>
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<td>RingbackTone</td>
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<td>SITTone</td>
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## Table E-15 Greece

<table>
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## Table E-16 Hong Kong

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### Table E-22 Israel

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### Table E-23 Italy

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### Table E-24 Japan

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### Table E-25 Korea

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### Table E-26 Luxembourg

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### Table E-27 Malaysia

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### Table E-28 Mexico

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<td>RingbackTone</td>
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### Table E-29 Netherlands

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### Table E-30 New Zealand

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### Table E-31 Norway

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### Table E-32 Pakistan

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### Table E-33 Panama

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### Table E-34 Peru

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### Table E-35 Philippines

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### Table E-36 Poland

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### Appendix E  Recommended Cisco ATA Tone Parameter Values by Country

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#### Table E-38  Russia

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#### Table E-39  Saudi Arabia

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<td>BusyTone</td>
<td>1,30958,0,1757,0,0,2640,2640,0</td>
</tr>
<tr>
<td>ReorderTone</td>
<td>1,30958,0,1757,0,0,1320,1320,0</td>
</tr>
<tr>
<td>RingbackTone</td>
<td>1,30958,0,1971,0,0,8000,32000,0</td>
</tr>
<tr>
<td>SITTone</td>
<td>1,30958,0,1757,0,0,1320,1320,0</td>
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</table>

### Table E-42  Slovenia

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Recommended Values</th>
</tr>
</thead>
<tbody>
<tr>
<td>DialTone</td>
<td>101,30958,3125,0,0,2,1600,2400,5600,6400,0</td>
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<tr>
<td>BusyTone</td>
<td>1,30958,0,1757,0,0,4000,4000,0</td>
</tr>
<tr>
<td>ReorderTone</td>
<td>1,30958,0,1757,0,0,1600,1600,0</td>
</tr>
<tr>
<td>RingbackTone</td>
<td>1,30958,0,1971,0,0,8000,32000,0</td>
</tr>
<tr>
<td>SITTone</td>
<td>1,30958,0,1757,0,0,1600,1600,0</td>
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### Table E-43  South Africa

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Recommended Values</th>
</tr>
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<tbody>
<tr>
<td>DialTone</td>
<td>2,31415,30890,1919,2252,1,0,0,0</td>
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<tr>
<td>BusyTone</td>
<td>1,31163,0,1657,0,0,4000,4000,0</td>
</tr>
<tr>
<td>ReorderTone</td>
<td>1,31163,0,1657,0,0,2000,2000,0</td>
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<tr>
<td>RingbackTone</td>
<td>102,31415,1079,30890,1266,2,3200,1600,3200,16000,0</td>
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<tr>
<td>SITTone</td>
<td>1,31163,0,1657,0,0,20000,4000,0</td>
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### Table E-44  Spain

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Recommended Values</th>
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<tbody>
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<td>DialTone</td>
<td>1,30958,0,4895,0,1,0,0,0</td>
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<tr>
<td>BusyTone</td>
<td>1,30958,0,1757,0,0,1600,1600,0</td>
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<td>ReorderTone</td>
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<tr>
<td>RingbackTone</td>
<td>1,30958,0,1757,0,0,12000,24000,0</td>
</tr>
<tr>
<td>SITTone</td>
<td>100,1,30958,1757,0,0,0,0,2,1600,1600,1600,4800,0,0,0,0</td>
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### Table E-45  Sweden

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Recommended Values</th>
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<tbody>
<tr>
<td>DialTone</td>
<td>1,30958,0,3889,0,1,0,0,0</td>
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<td>BusyTone</td>
<td>1,30958,0,1757,0,0,2000,2000,0</td>
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<tr>
<td>ReorderTone</td>
<td>1,30958,0,1757,0,0,2000,6000,0</td>
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<tr>
<td>RingbackTone</td>
<td>1,30958,0,1927,0,0,8000,40000,0</td>
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<tr>
<td>SITTone</td>
<td>101,3,24062,3640,14876,4778,5126,5297,3,2664,0,2664,0,2664,8000,0,0</td>
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<tr>
<td>CallWaitTone</td>
<td>1,30958,0,1757,0,0,1600,4000,11200</td>
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<tr>
<td>AlertTone</td>
<td>1,30467,0,4385,0,0,480,480,1920</td>
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### Table E-46  Switzerland

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<tr>
<th>Parameter</th>
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<td>DialTone</td>
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<td>BusyTone</td>
<td>1,30958,0,1757,0,0,4000,4000,0</td>
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<td>RingbackTone</td>
<td>1,30958,0,1927,0,0,8000,32000,0</td>
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<td>SITTone</td>
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### Table E-47  Taiwan

<table>
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<th>Parameter</th>
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<td>2,31537,30830,1833,2287,1,0,0,0</td>
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<tr>
<td>BusyTone</td>
<td>2,30466,28958,1398,1777,0,4000,4000,0</td>
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<tr>
<td>ReorderTone</td>
<td>2,30466,28958,1398,1777,0,2000,2000,0</td>
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<tr>
<td>RingbackTone</td>
<td>2,30830,1443,30466,1568,2,3200,1600,3200,16000,0</td>
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<td>SITTone</td>
<td>2,30466,28958,1398,1777,0,2000,2000,0</td>
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### Table E-48  Thailand

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<td>1,31163,0,1657,0,0,4000,4000,0</td>
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<td>ReorderTone</td>
<td>1,30742,0,1856,0,0,2640,2640,0</td>
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<td>RingbackTone</td>
<td>1,31163,0,1657,0,0,8000,32000,0</td>
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<td>SITTone</td>
<td>100,1,31163,1657,0,0,0,2,800,7200,2400,5600,0,0,5,0</td>
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### Table E-49 Turkey

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<td>ReorderTone</td>
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<td>1,30742,0,2083,0,0,16000,32000,0</td>
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<td>SITTone</td>
<td>1,30742,0,1856,0,0,1600,1600</td>
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### Table E-50 United Kingdom

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<tr>
<th>Parameter</th>
<th>Recommended Values</th>
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<tr>
<td>DialTone</td>
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<td>BusyTone</td>
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<tr>
<td>ReorderTone</td>
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<td>RingbackTone</td>
<td>102,31163,1173,30742,1314,2,3200,1600,3200,16000,0</td>
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<tr>
<td>SITTone</td>
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### Table E-51 United States

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<tr>
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<td>2,30466,28958,1246,1583,0,2000,2000</td>
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### Table E-52 Venezuela

<table>
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<tr>
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<th>Recommended Values</th>
</tr>
</thead>
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<tr>
<td>BusyTone</td>
<td>1,30958,0,1757,0,0,2000,2000,0</td>
</tr>
<tr>
<td>ReorderTone</td>
<td>1,30958,0,1757,0,0,2000,2000,0</td>
</tr>
<tr>
<td>RingbackTone</td>
<td>1,30958,0,1757,0,0,8000,32000,0</td>
</tr>
<tr>
<td>SITTone</td>
<td>1,30958,0,1757,0,0,2000,2000,0</td>
</tr>
</tbody>
</table>
# Glossary

## Numerics

<table>
<thead>
<tr>
<th>Term</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>10BaseT</td>
<td>10-Mbps baseband Ethernet specification using two pairs of twisted-pair cabling (Categories 3, 4, or 5); one pair for transmitting data and the other for receiving data. 10BASET, which is part of the IEEE 802.3 specification, has a distance limit of approximately 328 feet (100 meters) per segment.</td>
</tr>
<tr>
<td>A-law</td>
<td>ITU-T companding standard used in the conversion between analog and digital signals in PCM systems. A-law is used primarily in European telephone networks and is similar to the North American µ-law standard. See also companding and µ-law.</td>
</tr>
<tr>
<td>AVT tones</td>
<td>Out-of-bound signaling as defined in RFC 2833.</td>
</tr>
<tr>
<td>category-3 cable</td>
<td>One of five grades of UTP cabling described in the EIA/TIA-586 standard. Category 3 cabling is used in 10BaseT networks and can transmit data at speeds up to 10 Mbps.</td>
</tr>
<tr>
<td>CED tone detection</td>
<td>Called station identification. A three-second, 2100 Hz tone generated by a fax machine answering a call, which is used in the hand-shaking used to set the call; the response from a called fax machine to a CNG tone.</td>
</tr>
<tr>
<td>CLIP</td>
<td>Calling Line Identification Presentation. Shows your identity to callers with Caller ID.</td>
</tr>
<tr>
<td>CLIR</td>
<td>Calling Line Identification Restriction. Hides your identity from callers with Caller ID.</td>
</tr>
<tr>
<td>CNG</td>
<td>Comfort Noise Generation.</td>
</tr>
<tr>
<td>codec</td>
<td>Coder decoder. In Voice over IP, Voice over Frame Relay, and Voice over ATM, a DSP software algorithm used to compress/decompress speech or audio signals.</td>
</tr>
<tr>
<td>companding</td>
<td>Contraction derived from the opposite processes of compression and expansion. Part of the PCM process whereby analog signal values are rounded logically to discrete scale-step values on a nonlinear scale. The decimal step number then is coded in its binary equivalent prior to transmission. The process is reversed at the receiving terminal using the same nonlinear scale. Compare with compression and expansion. See also a-law and µ-law.</td>
</tr>
<tr>
<td><strong>compression</strong></td>
<td>The running of a data set through an algorithm that reduces the space required to store or the bandwidth required to transmit the data set. Compare with companding and expansion.</td>
</tr>
<tr>
<td><strong>CoS</strong></td>
<td>Class of service. An indication of how an upper-layer protocol requires a lower-layer protocol to treat its messages. In SNA subarea routing, CoS definitions are used by subarea nodes to determine the optimal route to establish a given session. A CoS definition comprises a virtual route number and a transmission priority field.</td>
</tr>
<tr>
<td><strong>D</strong></td>
<td>Dynamic Host Configuration Protocol. Provides a mechanism for allocating IP addresses dynamically so that addresses can be reused when hosts no longer need them.</td>
</tr>
<tr>
<td><strong>DHCP</strong></td>
<td>An addressable call endpoint. In Voice over IP (VoIP), there are two types of dial peers: POTS and VoIP.</td>
</tr>
<tr>
<td><strong>DNS</strong></td>
<td>Domain Name System. System used on the Internet for translating names of network nodes into addresses.</td>
</tr>
<tr>
<td><strong>DSP</strong></td>
<td>digital signal processor. A DSP segments the voice signal into frames and stores them in voice packets.</td>
</tr>
<tr>
<td><strong>DTMF</strong></td>
<td>dual tone multifrequency. Tones generated when a button is pressed on a telephone, primarily used in the U.S. and Canada.</td>
</tr>
<tr>
<td><strong>E</strong></td>
<td>The international public telecommunications numbering plan. A standard set by the ITU-T which addresses telephone numbers.</td>
</tr>
<tr>
<td><strong>E.164</strong></td>
<td>A SIP terminal or gateway. An endpoint can call and be called. It generates and/or terminates the information stream.</td>
</tr>
<tr>
<td><strong>endpoint</strong></td>
<td>The process of running a compressed data set through an algorithm that restores the data set to its original size. Compare with companding and compression.</td>
</tr>
<tr>
<td><strong>F</strong></td>
<td>Router or access server, or several routers or access servers, designated as a buffer between any connected public networks and a private network. A firewall router uses access lists and other methods to ensure the security of the private network.</td>
</tr>
<tr>
<td><strong>firewall</strong></td>
<td>Fax over IP</td>
</tr>
<tr>
<td><strong>FoIP</strong></td>
<td>Fully Qualified Domain (FQDN) format “mydomain.com” or “company.mydomain.com.”</td>
</tr>
<tr>
<td><strong>FQDN</strong></td>
<td>Frequency shift key.</td>
</tr>
</tbody>
</table>
FXO
Foreign Exchange Office. An FXO interface connects to the public switched telephone network (PSTN) central office and is the interface offered on a standard telephone. Cisco FXO interface is an RJ-11 connector that allows an analog connection at the PSTN central office or to a station interface on a PBX.

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

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

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

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

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

H
H.245
An ITU standard that governs H.245 endpoint control.

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

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

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

LDAP  Lightweight Directory Access Protocol

LEC  local exchange carrier.

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

MGCP  Media Gateway Control Protocol.

MWI  message waiting indication.

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

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

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

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

NTP  Network Time Protocol. Protocol built on top of TCP that assures accurate local time-keeping with reference to radio and atomic clocks located on the Internet. This protocol is capable of synchronizing distributed clocks within milliseconds over long time periods.
Plain old telephone service. Basic telephone service supplying standard single-line telephones, telephone lines, and access to the PSTN.

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

Public switched telephone network.

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

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

A registrar server is a server that accepts Register requests. A registrar is typically co-located with a proxy or redirect server and may offer location services.

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

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

Signaling connection control part.

Session Definition Protocol. An IETF protocol for the definition of Multimedia Services. SDP messages can be part of SGCP and MGCP messages.
SIP
Session Initiation Protocol. Protocol developed by the IETF MMUSIC Working Group as an alternative to H.323. SIP features are compliant with IETF RFC 2543, published in March 1999. SIP equips platforms to signal the setup of voice and multimedia calls over IP networks.

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

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

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

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

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

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

TOS
Type of service. See CoS.

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

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

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

VAD
Voice activity detection. When enabled on a voice port or a dial peer, silence is not transmitted over the network, only audible speech. When VAD is enabled, the sound quality is slightly degraded but the connection monopolizes much less bandwidth.
<table>
<thead>
<tr>
<th>term</th>
<th>definition</th>
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<tbody>
<tr>
<td>voice packet</td>
<td>Gateway platforms that enable Internet telephony service providers to offer</td>
</tr>
<tr>
<td>gateway</td>
<td>residential and business-class services for Internet telephony.</td>
</tr>
<tr>
<td>VolP</td>
<td>Voice over IP. The capability to carry normal telephony-style voice over an</td>
</tr>
<tr>
<td></td>
<td>IP-based Internet with POTS-like functionality, reliability, and voice</td>
</tr>
<tr>
<td></td>
<td>quality. VolP enables a router to carry voice traffic (for example,</td>
</tr>
<tr>
<td></td>
<td>telephone calls and faxes) over an IP network. In VolP, the DSP segments the</td>
</tr>
<tr>
<td></td>
<td>voice signal into frames, which then are coupled in groups of two and stored</td>
</tr>
<tr>
<td></td>
<td>in voice packets. VolP is a blanket term, which generally refers to Cisco’s</td>
</tr>
<tr>
<td></td>
<td>standard-based (for example H.323) approach to IP voice traffic.</td>
</tr>
<tr>
<td>XML</td>
<td>eXtensible Markup Language. Designed to enable the use of SGML on the</td>
</tr>
<tr>
<td></td>
<td>World-Wide Web. XML allow you to define your own customized markup language.</td>
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</table>
# Index

## Numerics

<table>
<thead>
<tr>
<th>Page</th>
</tr>
</thead>
<tbody>
<tr>
<td>38</td>
</tr>
<tr>
<td>4</td>
</tr>
<tr>
<td>12</td>
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</tr>
<tr>
<td>37</td>
</tr>
<tr>
<td>2</td>
</tr>
<tr>
<td>61</td>
</tr>
<tr>
<td>1</td>
</tr>
<tr>
<td>29</td>
</tr>
<tr>
<td>2</td>
</tr>
<tr>
<td>36</td>
</tr>
<tr>
<td>37</td>
</tr>
<tr>
<td>49</td>
</tr>
<tr>
<td>35</td>
</tr>
<tr>
<td>35</td>
</tr>
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<td>35</td>
</tr>
<tr>
<td>35</td>
</tr>
<tr>
<td>35</td>
</tr>
<tr>
<td>35</td>
</tr>
<tr>
<td>36</td>
</tr>
<tr>
<td>37</td>
</tr>
<tr>
<td>42</td>
</tr>
<tr>
<td>4</td>
</tr>
<tr>
<td>5</td>
</tr>
<tr>
<td>6</td>
</tr>
<tr>
<td>6</td>
</tr>
</tbody>
</table>

## A

- **Action Identifiers**: 4
- **advanced audio configuration**: 4
- **alphanumeric values**: 22
- **alternate gatekeeper requirement configuration setting**: 41
- **alternate NTP IP address configuration**: 12
- **AltGk parameter**: 16, 14
- **AltNTPIP parameter**: 12
- **anonymity for called third party**: 11
- **Anonymous user name**: 11, 12
- **ata.txt file**: 10
- **atadefault.cfg configuration file**: 17
- **atapname.exe Tool**: 11
- **ata unique configuration file**: 3, 17
- **audio and call waiting tone mixing**: 7
- **audio and call-waiting tone mixing**: 43
- **audio configuration**: 4
- **audio configuration parameters**: 30
- **AudioMode parameter**: 32, 2, 6
- **audio operating mode**: 32
- **authentication**: 2, 17, 18
- **authentication ID**: 18

## B

- **backup proxy configuration parameter**: 14
- **Bellcore (FSK) method**: 36
- **Bellcore method for called ID**: 49
- **billable features**: 4
- **billable features configuration**: 4
- **binary configuration file creation**: 10
- **blind-transfer configuration setting**: 36, 37
- **bootload**: 2
- **boot load behavior**: 2
- **BYE request**: 61
- **CallerIdMethod parameter**: 49
- **CallFeatures**: 35
- **call commands**: 1
- **call commands changing**: 1
- **call forwarding**: 5
- **call forwarding in Sweden**: 6
- **call forwarding in United States**: 5
- **call forwarding types**: 5
- **call forwarding configuration setting**: 42
- **call forwarding setting removal**: 5
- **Calling Line Identification Presentation**: 35
- **calling line identification presentation (CLIP)**: 6
<table>
<thead>
<tr>
<th>Index</th>
</tr>
</thead>
</table>

- Calling Line Identification Restriction 35
  - calling line identification restriction
    - definition 6
    - in Sweden 7
    - in United States 7
- Calling Line Identification Restriction (CLIR) 11
  - call-progress toes 53
  - Call-Progress tone parameters 53, 59
  - Call-progress tones 13
  - call-progress tones 53
- call return
  - in Sweden 6
  - in United States 6
- call return configuration setting 36
- call-return configuration setting 37
- call waiting 20
  - in Sweden 5
  - in United States 5
- call-waiting caller ID 2
- call-waiting-caller ID configuration setting 37
- call-waiting caller ID setting 36
- call-waiting configuration setting 36, 37
- call-waiting hang-up alert 5, 44
- call-waiting period 40
- call-waiting ring timeout 38
- call waiting tone 7
- call-waiting tone 62
- call waiting tone and audio mixing 7
- CallWaitTone 62
- CANCEL request 13
- CDP discovery setting 46
- cfgfmt.exe tool 10, 15
- CFGID 76
- CfgInterval parameter 6
- Cisco ATA route static configuration 9
- Cisco ATA-specific configuration file 3
- Cisco CallManager environment configuration setting 41
- Cisco Discovery Protocol (CDP) 2
- CLIP_CLIR 35
  - CLIP_CLIR setting 36, 37
  - CLIR 11
    - codec negotiation in sending fax 42
    - codecs 3
    - codec setting 3
    - Comfort 6
  - common parameters
    - configuration 9
  - conference call
    - in Sweden 4
    - in United States 4
  - conference warning-tone setting 45
  - configurable reboot of Cisco ATA 7
  - configurable reboot time 39
  - configurable time Ethernet is down before reboot 39
  - configuration
    - advanced audio 4
    - alphabetical listing of features with related parameters 23
    - atadefault.cfg file 12
    - audio 4
    - authentication 2
    - billable features 4
    - call-progress tones 53
    - cfgfmt.exe tool 10, 15
    - codec 3
    - dial plan 7
    - features and related parameters 23
    - hook flash timing 7
  - methods
    - using TFTP and DHCP servers 8, 9
    - Web-based 23, 25
    - Wed-based 26
  - mixing of call waiting tone and audio 7
  - NAT/PAT translation 10
  - NAT gateway 9
  - network timing 10
  - on-hook delay 7
  - parameters
<table>
<thead>
<tr>
<th>Parameter</th>
<th>Page Numbers</th>
</tr>
</thead>
<tbody>
<tr>
<td>AltGk</td>
<td>16, 14</td>
</tr>
<tr>
<td>AltNTPIP</td>
<td>12</td>
</tr>
<tr>
<td>AudioMode</td>
<td>32, 2, 6</td>
</tr>
<tr>
<td>CallCmd</td>
<td>37</td>
</tr>
<tr>
<td>CallerIdMethod</td>
<td>49</td>
</tr>
<tr>
<td>CallFeatures</td>
<td>35</td>
</tr>
<tr>
<td>CfgInterval</td>
<td>6</td>
</tr>
<tr>
<td>ConnectMode</td>
<td>41, 3</td>
</tr>
<tr>
<td>DHCP</td>
<td>8</td>
</tr>
<tr>
<td>DialPlan</td>
<td>64</td>
</tr>
<tr>
<td>DNS1IP</td>
<td>10</td>
</tr>
<tr>
<td>DNS2IP</td>
<td>11</td>
</tr>
<tr>
<td>EncryptKey</td>
<td>6, 14</td>
</tr>
<tr>
<td>FeatureTimer</td>
<td>38</td>
</tr>
<tr>
<td>GkOrProxy</td>
<td>2, 16, 13</td>
</tr>
<tr>
<td>LoginID0</td>
<td>17</td>
</tr>
<tr>
<td>LoginID1</td>
<td>18</td>
</tr>
<tr>
<td>MAXRedirect</td>
<td>2</td>
</tr>
<tr>
<td>MediaPort</td>
<td>9, 30</td>
</tr>
<tr>
<td>MediaPort parameter</td>
<td>2</td>
</tr>
<tr>
<td>NATIP</td>
<td>2, 9, 21, 43</td>
</tr>
<tr>
<td>NatServer</td>
<td>22</td>
</tr>
<tr>
<td>NatTimer</td>
<td>22</td>
</tr>
<tr>
<td>network timing</td>
<td>10</td>
</tr>
<tr>
<td>NPrintF</td>
<td>73</td>
</tr>
<tr>
<td>NTPIP</td>
<td>11</td>
</tr>
<tr>
<td>NumTxFrames</td>
<td>34</td>
</tr>
<tr>
<td>OpFlags</td>
<td>45</td>
</tr>
<tr>
<td>Polarity</td>
<td>51</td>
</tr>
<tr>
<td>PWD0</td>
<td>16</td>
</tr>
<tr>
<td>PWD1</td>
<td>17</td>
</tr>
<tr>
<td>RxCodec</td>
<td>31</td>
</tr>
<tr>
<td>SigTimer</td>
<td>40</td>
</tr>
<tr>
<td>SipOutBoundProxy</td>
<td>2, 16, 21</td>
</tr>
<tr>
<td>SIPPort</td>
<td>9, 19</td>
</tr>
<tr>
<td>SIPPort parameter</td>
<td>2</td>
</tr>
<tr>
<td>SIPRegInterval</td>
<td>2, 19</td>
</tr>
<tr>
<td>SIPRegOn</td>
<td>2, 20</td>
</tr>
<tr>
<td>StaticIp</td>
<td>9</td>
</tr>
<tr>
<td>StaticNetMask</td>
<td>10</td>
</tr>
<tr>
<td>StaticRoute</td>
<td>9</td>
</tr>
<tr>
<td>TftpURL</td>
<td>5</td>
</tr>
<tr>
<td>TimeZone</td>
<td>48</td>
</tr>
<tr>
<td>TraceFlags</td>
<td>73</td>
</tr>
<tr>
<td>TxCodec</td>
<td>31</td>
</tr>
<tr>
<td>UID0</td>
<td>15</td>
</tr>
<tr>
<td>UID1</td>
<td>16</td>
</tr>
<tr>
<td>UIPassword</td>
<td>4, 14</td>
</tr>
<tr>
<td>UseLoginID</td>
<td>18</td>
</tr>
<tr>
<td>UseTFTP</td>
<td>5</td>
</tr>
<tr>
<td>VLAN Setting</td>
<td>12</td>
</tr>
<tr>
<td>refresh interval</td>
<td>3</td>
</tr>
<tr>
<td>repeat dialing on busy signal</td>
<td>14</td>
</tr>
<tr>
<td>required parameters</td>
<td>1</td>
</tr>
<tr>
<td>services and related parameters</td>
<td>23</td>
</tr>
<tr>
<td>silence suppression</td>
<td>6</td>
</tr>
<tr>
<td>SipOutBoundProxy support</td>
<td>16</td>
</tr>
<tr>
<td>SIP proxy server redundancy</td>
<td>16</td>
</tr>
<tr>
<td>timeout on no answer for call forwarding</td>
<td>20</td>
</tr>
<tr>
<td>timing</td>
<td>10</td>
</tr>
<tr>
<td>tones</td>
<td>53</td>
</tr>
<tr>
<td>VIA header</td>
<td>13</td>
</tr>
<tr>
<td>configuration changes taking effect</td>
<td>6</td>
</tr>
<tr>
<td>configuration files generated</td>
<td>13</td>
</tr>
<tr>
<td>configuration in a non-TFTP server environment</td>
<td>7</td>
</tr>
<tr>
<td>configuration in a TFTP server environment</td>
<td>5</td>
</tr>
<tr>
<td>configuration-method parameters</td>
<td>4</td>
</tr>
<tr>
<td>configuration update interval</td>
<td>6</td>
</tr>
<tr>
<td>configuring seconds between registration renewal</td>
<td>2</td>
</tr>
<tr>
<td>congestion tone</td>
<td>61</td>
</tr>
<tr>
<td>ConnectMode parameter</td>
<td>41, 3</td>
</tr>
<tr>
<td>Context-Identifiers</td>
<td>3, 4</td>
</tr>
<tr>
<td>converting MAC address to hexadecimal format</td>
<td>11</td>
</tr>
<tr>
<td>convert unique configuration file to binary format</td>
<td>6</td>
</tr>
<tr>
<td>creating a text configuration file</td>
<td>9</td>
</tr>
</tbody>
</table>
Index

D

debugging 4
debugging, preserv.exe program 4
debugging diagnostics
  TraceFlags 7
Default 2
default boot load behavior 2
default configuration file 17
default values 9, 2
device hostname 47
DHCP
  disabling 8
  enabling 8
DHCP discovery message 46
DHCP option 12 47
DHCP option 150 18, 46
DHCP option 66 18
DHCP options 19
DHCP parameter 8
DHCP server
  configuration without using DHCP 20
  if not under control of Cisco ATA administrator 19
DHCP server-supplied TFTP configuration filename used
  configuration setting 46
DHCP server usage 18
Diagnostic 73
diagnostics 7
  SyslogCtrl 75
  SyslogIP 74
  types of messages to trace 75
diagnostics for debugging 7
DialPlan 7
dial plan blocking 66
Dial Plan Commands 65
dial plan configuration 7
dial plan examples 70
dial plan exceeding 199 characters 72
DialPlanEx parameter 72
DialPlan parameter 64
dial plans 64
dial plan syntax 65
DialTone 60
direct IP-to-IP calls 15
disabling access to Web interface 8
disabling CDP discovery 4
disabling use of DHCP server 21
disabling VLAN encapsulation 4
disabling VLAN IP encapsulation 3
DisplayName0 29
DisplayName1 29
display name for caller ID 8
display name parameters 29
display of hardware information 9
Diversion header 12, 44
DNS1IP parameter 10
DNS2IP parameter 11
DNS A-record query performed 14
DNS SRV 45
DNS SRV lookup 9
DNS SRV performed 14
DNS SRV support 9
downloading Cisco ATA software from CCO 7, 8
DTMF Method
  always out-of-band setting 33
  by negotiation 33
  in band setting 33
DtMfMethod configuration settings 33
DTMF method for caller ID 49
dynamic payload type 41

E

electrical specifications 2
enabling SIP registration 2
enabling user-specified voice VLAN ID 4
EncrpytKeyEx parameter 7
encryption 12
<table>
<thead>
<tr>
<th>Encryption examples</th>
<th>15</th>
</tr>
</thead>
<tbody>
<tr>
<td>Encryption key</td>
<td>6</td>
</tr>
<tr>
<td>Encryption parameter</td>
<td>4</td>
</tr>
<tr>
<td>EncryptKey parameter</td>
<td>6, 14</td>
</tr>
<tr>
<td>Entering alphanumeric values</td>
<td>22</td>
</tr>
<tr>
<td>Environmental specifications</td>
<td>2</td>
</tr>
<tr>
<td>Establishing IP connectivity</td>
<td>20</td>
</tr>
<tr>
<td>Ethernet ports</td>
<td>6</td>
</tr>
<tr>
<td>ETSI method for caller ID</td>
<td>49</td>
</tr>
<tr>
<td>Example configuration text file</td>
<td>2</td>
</tr>
</tbody>
</table>

**F**

<table>
<thead>
<tr>
<th>Factory default reset</th>
<th>23</th>
</tr>
</thead>
<tbody>
<tr>
<td>Failback to primary proxy</td>
<td>15</td>
</tr>
<tr>
<td>Fax CED tone detection configuration setting</td>
<td>33</td>
</tr>
<tr>
<td>Fax mode</td>
<td>1, 6</td>
</tr>
<tr>
<td>Configuration</td>
<td>7</td>
</tr>
<tr>
<td>G.711 codec only setting</td>
<td>36, 37</td>
</tr>
<tr>
<td>No fax tone detection setting</td>
<td>36, 37</td>
</tr>
<tr>
<td>No silence suppression setting</td>
<td>36, 37</td>
</tr>
<tr>
<td>Fax mode configuration</td>
<td>6, 7</td>
</tr>
<tr>
<td>Fax mode configuration settings</td>
<td>36</td>
</tr>
<tr>
<td>Fax-mode configuration settings</td>
<td>37</td>
</tr>
<tr>
<td>Fax pass-through</td>
<td>42</td>
</tr>
<tr>
<td>G.711µ-law</td>
<td>42</td>
</tr>
<tr>
<td>G.711 A-law</td>
<td>42</td>
</tr>
<tr>
<td>Fax pass-through mode</td>
<td>1</td>
</tr>
<tr>
<td>Fax pass-through mode, enabling</td>
<td>4, 5</td>
</tr>
<tr>
<td>Fax Pass-through mode configuration</td>
<td>2</td>
</tr>
<tr>
<td>Fax pass-through redundancy configuration setting</td>
<td>42</td>
</tr>
<tr>
<td>Fax relay, disabling</td>
<td>5</td>
</tr>
<tr>
<td>Fax services</td>
<td>1</td>
</tr>
<tr>
<td>Fax services, debugging</td>
<td>7, 9</td>
</tr>
<tr>
<td>Features and related parameters</td>
<td>23</td>
</tr>
<tr>
<td>FeatureTimer2 parameter</td>
<td>39</td>
</tr>
<tr>
<td>FeatureTimer parameter</td>
<td>38</td>
</tr>
<tr>
<td>Forgotten password</td>
<td>4</td>
</tr>
</tbody>
</table>

**G**

| G.711 silence suppression configuration | 33 |
| G.723.1 | 32 |
| G.726-32kbps | 32 |
| G.729 | 32 |
| Generated configuration files | 13 |
| Generating binary configuration file | 10 |
| GkOrProxy parameter | 2, 16, 13 |
| Greenwich Mean Time offset | 48 |

**H**

| H.245 tunneling configuration setting | 41 |
| Hardware information display | 9 |
| Hook flash time | 41 |
| Hook-flash timing | 7 |
| Hook flash timing configuration | 7 |
| Hotline/Warmline | 67 |
| HTTP refresh | 46 |
| HTTP reset | 47 |

**I**

<p>| Include command | 10 |
| Input level of FXS ports | 52 |</p>
<table>
<thead>
<tr>
<th>Index</th>
</tr>
</thead>
</table>

**installation**
- procedure 3

**INVITE request** 44

**IP address static configuration** 9

**IP dial plan configuration** 72

**ITU G.711** 6

---

**K**
- keepalive mechanism for a SIP session 17
- keep-alive packets 23

---

**L**
- Lightweight Directory Access Protocol (LDAP) 3
- line polarity 51
- local tone playout reporting 10
- LoginID0 parameter 17
- LoginID1 parameter 18
- log listings 3, 5, 8, 14
- low-bit-rate codec 33
- low-bit-rate codecs 32

---

**M**
- MAC address 11
- making configuration changes after boot up 11
- maximum number of digits in phone number 50
- maximum number of times to try redirection 2
- maximum redirects to another number 20
- MAXRedirect parameter 2
- MediaPort 9
- MediaPort parameter 2, 30
- message-waiting-indication setting 36, 37
- message waiting indicator 50
- methods supported 9
- minimum required interval for SIP session timer 28
- mixing of call waiting tone and audio 7
- MsgRetryLimits parameter 24

---

**N**
- NAT/PAT translation 10
- NAT/PAT translation configuration 10
- NAT gateway configuration 9
- NATIP 9, 21
- NATIP auto-mapping 23
- NATIP parameter 2, 21, 43
- NATs 18
- NatServer 19
- NatServer parameter 22
- NatTimer 19
- NatTimer parameter 22
- Network Address Translation (NAT) 9
- network configuration parameters 8
- network requirements 2
- network status 11, 12
- network timing 10
- network timing configuration 10
- no-answer timeout 40
- non-dotted hexadecimal MAC address location 10
- normal start connection mode setting 41
- NPrintf 7
- NPrintF parameter 73
- NTP IP address configuration 11
- NTPIP parameter 11
- NumTxFrames parameter 34

---

**O**
- obtaining network status 11, 12
- obtaining non-dotted hexadecimal MAC address of Cisco ATA 10
- on-hook delay 7
- on-hook delay before call is disconnected 38
- on-hook delay configuration 7
- operating parameters 35
### Index

#### Operational
- OpFlags parameter 45
- outbound proxy server 21
- output level of FXS ports 52

#### P
- paid features 36
- parameters
  - audio configuration 30
  - FeatureTimer2 39
  - MsgRetryLimits 24
  - SIP session timer 26
  - SITone 63
- parameters and defaults 14, 1
- parameters for configuration method 4
- parameter types 1
- password 4
- password for Phone 1 port 16
- password for Phone 2 port 17
- phone service status using HTTP 17
- physical interfaces 2
- physical specifications 1
- placing call without using SIP proxy 15
- Polarity 51
- polarity 51
- Polarity parameter 51
- polarity reversal before and after Caller ID signal 50
- polarity reversal before and after caller ID signal port
  - incoming SIP requests 19
  - outgoing SIP requests 19
- RTP media 30
- port for debug messages
  - configuration 73
- port for incoming SIP requests 2
- port restricted cone NAT 18
- primary domain name server 10
- priority value 6

#### R
- reboot 7
- reboot time configuration 39
- received= tag in VIA header 43
- receiving-audio codec settings 31
- redialing timeout 38
- redirection attempts 2
- redirect server 4
- redirects to another number 20
- redundancy 12
- redundant proxy support for BYE/CANCEL Request 13
- refreshing the Cisco ATA 27
- refresh interval 3, 6
- refresh-interval configuration 6
- refresh procedure 11
- REGISTER message configuration 42
- registrar address 2
- registrar server 4
- registration-renewal configuration setting 42
- registration renewal increment 19
- registration server 14
- registration with authentication 3
- registration without authentication 2
- re-INVITE method 17
- reorder delay 40
- Reorder Tone 61
Index

reorder tone  56, 61
ReorderTone parameter  56
reorder tone parameter example  57, 58
repeat dialing on busy signal  14
reporting
  local tone playout  10
  RTP statistics  13
required parameters  1
resetting the Cisco ATA  27
resetting the Cisco ATA to factory defaults  23
restricted cone NAT  18
retry timeout  38
reviewing default router for Cisco ATA to use  22
reviewing subnet mask of Cisco ATA  22
review IP address of Cisco ATA  21
Ringback Tone  62
ringback tone configuration setting  42
ringer cadence pattern configuration  64
ringing characteristics  3
ring loads  14
ring timeout  40
RJ-45 LED
  Cisco ATA 186  6
  Cisco ATA 188  6
rtpcatch  12, 13, 14, 16, 18, 20
RTP frames  13
RTP media port  2, 30
RTP packets  13
RTP payload type  41
RTP statistics  13
RTP statistics reporting  13
RxCodec parameter  31

S

safety recommendations  2
sample configuration text file  9
secondary domain name server  11
sending SIP requests  24
sending SIP responses  24
send special character O  49
send special character P  50
services
  basic  8
  supplemental  10
services and related parameters  23
Session-Expires value  29
session timing  17
setting a password  4
setting the codec  3
signaling data packets  34
signaling image upgrade  27
SigTimer  40
SigTimer parameter  40
silence suppression  6
Simple Traversal of UDP through NAT (STUN) support  18
SIP  2, 3
  advanced features  3
call flow scenarios  2
clients  4
  request methods  1
  servers  4
sip_example.txt file  10
sip_example.txt sample configuration file  9
sip_example.txt text file  2
SIP call return configuration setting  42
SIP INVITE messages  12
SipOutBoundProxy parameter  2, 16, 21
SipOutBoundProxy support configuration  16
SIPPort  9
SIPPort parameter  2, 19
SIP proxy not used for call  15
SIP proxy server address  2
SIP proxy server configuration parameter  13
SIP Proxy Server redundancy  16
SIPRegInterval  19
SIPRegInterval parameter  2, 19
SIP registration 2
SIP registration enabled 20
SIPRegOn parameter 2, 20
SIP requests 24
SIP responses 24
SIP session interval 28
SIP session refreshes 28
SIP Session Timer 17
SIP session timer 26
minimum required interval 28
SIP session timer parameters 26
SIP to SIP call with authentication 12
SIP to SIP call without authentication 6
SITone parameter 56
SITtone parameter 63
software download from CCO 7, 8
software specifications (all protocols) 3
special information tone 56, 63
specifying a preconfigured VLAN ID 3
specifying number of attempts at sending SIP requests 24
specifying VLAN CoS bit value (802.1P priority) for UDP packets 4
specifying VLAN CoS bit value for TCP packets 4
standard payload type 41
statically configuring various IP addresses 2
StaticIP 20
StaticIP parameter 9
StaticNetMask 20
StaticNetMask parameter 10
static network-router-probing configuration setting 46
StaticRoute 20, 9
StaticRoute parameter 9
status of phone services 17
strongest encryption key 7
STUN 23
stuttering dial tone 19
subnet mask static configuration 10
supplementary services
  cancelling 1
common 1
symmetric NAT 18
syslog 6
  IP address 6
  priority value 6
  tag 6
  time_offset 6
SyslogCtrl 75
SyslogIP 74
system diagnostics 6

T
TFTP server IP address 18
TFTP server IP address or URL specification 5
TFTP server name 18
TFTP server usage 5
TftpURL parameter 5
three-way calling configuration setting 36
three-way-calling configuration setting 37
timeout configuration for failback 15
timeout on no answer for call forwarding 20
timeouts
  redialing 38
  retry 38
time-stamping incoming calls 48
time zone configuration 48
TimeZone parameter 48
timing configuration 10
tone
  reorder 56
tone parameters 53, 54
tone parameter syntax
    basic format 53
    extended formats 54
tones 13, 53
  syntax 53
trace-features configuration 73
TraceFlags parameter 73
Index

transfer with consultation 36
transfer-with-consultation configuration setting 37
transmitting-audio codec settings 31
troubleshooting
general tips 1
installation 3
upgrade issues 3
two-way cut-through configuration setting 41
TxCodec 31
TxCodec parameter 31
Type of Service (ToS) bits 34

U

UID0 parameter 15
UID1 parameter 16
UIPassword parameter 4, 14
UIPassword prompt configuration setting 48
unconditional call forwarding 19
unique configuration file 10, 17
upgrading firmware from TFTP server 1
upgrading software
  using executable file 2
  upgrading software from TFTP server 1
upgrading the signaling image 27
UseLoginID 18
UseLoginID parameter 18
User agent client (UAC) 3
User agent server (UAS) 3
user configurable timeout 20
User ID 15, 16
User Interface (UI) Parameters 4
UseTFTP parameter 5
Using 12

V

version parameter 76
version parameter for configuration file 76
VIA header 13
visual message waiting indicator 50
VLAN CoS bit value (802.1P priority) for TCP packets 12
VLAN CoS bit value (802.1P priority) for UDP packets 12
VLAN ID 12
VLAN ID example 4
VLAN IP encapsulation setting 46
VLAN-related parameters 3
VLAN Setting parameter 12
VLAN tagging 2
voice configuration menu 21
voice data packets 34
Voice Menu Codes
  configuration parameters 2
  information options 1
  software upgrade 4
voice prompt confirmation bit settings 47, 48
Voice prompt confirmation for call waiting and call forwarding 20

W

WAN address of attached router/NAT 2, 21
warnings
  circuit breaker (15A) 5
  installation 2
  lightning activity 2
  main disconnecting device 2
  No. 26 AWG 5
  product disposal 2
  web configuration disabling 46
Web interface access disabled 8

X

XML configuration page 22
XML pages for Cisco ATA information 22