



Release Notes for the Cisco ATA 186 and Cisco ATA 188 Release 3.2

Nov. 19, 2004

These release notes describe newly incorporated features, changed features or changed behavior, resolved issues, and open issues for the Cisco ATA 186 and the Cisco ATA 188 for Release 3.2. This release contains new images for two Cisco ATA protocols: SIP and SCCP.

Refer also to the *Release Notes for the Cisco ATA 186 and Cisco ATA 188 Release 3.1(2)* for information about that release:

<http://www.cisco.com/univercd/cc/td/doc/product/voice/ata/atarn/atarn312.htm>

In addition, refer to the Cisco ATA administrator's guide for your protocol at the following location:

<http://www.cisco.com/univercd/cc/td/doc/product/voice/ata/ataadm/index.htm>

The term *Cisco ATA* refers to both the Cisco ATA 186 and the Cisco ATA 188.

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Introduction to the Cisco ATA Analog Telephone Adaptor

The Cisco ATA is an analog telephone adaptor that allows traditional analog telephones to operate on IP-based telephony networks. The Cisco ATA supports two voice ports, each with its own independent telephone number.

Two Cisco ATA products are available to Cisco customers—the Cisco ATA 186 and the Cisco ATA 188. Both products run the same software and have two voice ports. The Cisco ATA 186 has one RJ45 port that provides access to an Ethernet network. The Cisco ATA 188 has an Ethernet switch and two RJ45 ports—one for accessing an Ethernet network and one for connecting a downstream Ethernet device such as a PC.

Downloading and Upgrading the Software

Before you can use the Cisco ATA Release 3.2, you must first download and upgrade the Cisco ATA software. You can download the software, after logging in, at:

<http://www.cisco.com/cgi-bin/tablebuild.pl/ata186>



Note

If you are using the Cisco ATA executable-file-upgrade method, check with the administrator of the TFTP server to make sure that the TFTP upgrade method is disabled. Otherwise, the Cisco ATA might downgrade to an old image via TFTP.

For more information about downloading and upgrading software, see the Cisco ATA administrator's guides for the signaling protocol you are using. The administrator's guides can be found at the following location:

<http://www.cisco.com/univercd/cc/td/doc/product/voice/ata/ataadmn/index.htm>

Enhanced Web Configuration and Informational Pages

Enhanced Cisco ATA web pages are available in Release 3.2 for the SIP and SCCP protocols. These pages include more information and are divided into more configuration-parameter types than in previous releases.

The Cisco ATA web pages are divided into the following categories:



Note

<ip_address> refers to the IP address of the Cisco ATA.

- Information pages:
 - Device Information: http://<ip_address>/DeviceInfo
 - Network Configuration: http://<ip_address>/NetworkCfg
 - Ethernet Statistics: http://<ip_address>/NetStat
 - RTP Statistics: http://<ip_address>/rtps
- Configuration pages:
 - Change UIPassword: http://<ip_address>/ChgPasswd
 - Network Parameters: http://<ip_address>/dev or http://<ip_address>/NetCfg
 - SIP Parameters: http://<ip_address>/SIPCfg
 - SCCP Parameters: http://<ip_address>/SCCPCfg
 - Tone Parameters: http://<ip_address>/ToneCfg
 - Audio Parameters: http://<ip_address>/AudioCfg
 - Service Parameters: http://<ip_address>/ServiceCfg
 - Debug Parameters: http://<ip_address>/DebugCfg



Note

For information on all configuration parameters, see the “Parameters and Defaults” section of the Cisco ATA administrator’s guide for the applicable protocol.

- Service pages:
 - Phone Status: http://<ip_address>/stats
 - Phone Service: http://<ip_address>/service
 - Call History: http://<ip_address>/callhistory
- All Cisco ATA configuration settings:
http://<ip_address>/dev.xml

Parameter Name Changes

Some parameter names have changed for the 3.2 release.



Note

Conversion of the old parameter names to the new parameter names is optional. If you would like to retain and use the old parameter names and add the new parameters introduced in release 3.2 to an existing configuration file, you must use the latest ptag.dat file for the cfgfmt.exe program to recognize both old and new parameter names. For complete information about the cfgfmt.exe tool, refer to the Cisco ATA administration guide for your protocol.

[Table 1](#) shows the parameter name changes for the SIP protocol. [Table 2](#) shows the parameter name changes for the SCCP protocol. For complete definitions and descriptions of all Cisco ATA parameters, see the Cisco ATA Administrator's Guide for your protocol; also refer to the sip_example.txt file (for SIP) or the sk_example.txt file (for SCCP), each of which is included with the Cisco ATA software package for their respective protocol.

Table 1 Parameter Name Changes for SIP for Release 3.2

Old Name of Parameter	New Name of Parameter
GKOrProxy	Proxy
AltGk	SecProxy
AltGkTimeOut	SecProxyTimeOut

Table 2 Parameter Name Changes for SCCP for Release 3.2

Old Name of Parameter	New Name of Parameter
CA0orCM0	CallManager0
CA1orCM1	CallManager1
EPID0orSID0	SID0
EPID1orSID1	SID1

Password Protection

The Cisco ATA web configuration pages are password protected.

You can set or change your password on the following web page:

http://<ip_address>/ChgPasswd

When you attempt to access any Cisco ATA configuration page for the first time, the Cisco ATA requests your password. Once you have successfully entered the configuration-screen area, you can navigate to other configuration screens without re-entering your password.

New Features for Release 3.2

Cisco ATA Release 3.2 includes many new features: This section contains the following topics:

- “[New Features for Release 3.2 that Require Configuration](#)” section on page 5
- “[Dial Plan ‘N’ Rule for E.164 Number Normalization and Denormalization](#)” section on page 16
- “[New Features for Release 3.2 that do not Require Configuration](#)” section on page 18

New Features for Release 3.2 that Require Configuration

[Table 3](#) lists new features for Cisco ATA Release 3.2 that require configuration. The “Topic” column lists the applicable protocol.

Table 3 *New Cisco ATA Features that Require Configuration for Release 3.2(0)*

Topic	Behavior Description	Configuration
Proxy-only INVITE acceptance (SIP)	You can configure the Cisco ATA to accept INVITE messages from only the proxy. If the INVITE message does not come from a configured proxy, the Cisco ATA rejects the INVITE request with a 403 “ <i>Forbidden</i> ” response. The method works even if the privacy feature is enabled.	Parameter ConnectMode (bit 5) Values 0= Allow INVITE messages from any SIP user agent (default). 1= Allow INVITE messages from primary, backup or outbound proxies only.
Toll control with PIN (SIP)	The Cisco ATA allows the user to dial a personal identification number (PIN) to override toll restriction on a per-call basis. The configuration of the PIN should be with web-page or TFTP configuration.	The http command to set the PIN is: http://<ip_address>/setpin=<pin>

Table 3 *New Cisco ATA Features that Require Configuration for Release 3.2(0) (continued)*

Topic	Behavior Description	Configuration
Setting Up and Placing a Call Without Using a SIP Proxy	The way this feature gets configured has changed in the Cisco ATA 3.2 release.	<ul style="list-style-type: none"> <li data-bbox="980 304 1474 441">• Old configuration: Refer to the following description in the Cisco ATA administrator's guide for the SIP protocol for the 3.0 release: http://www.cisco.com/univercd/cc/td/doc/product/voice/ata/ataadm/sip30ad/sip88ch4.htm#wp1060839 <li data-bbox="980 556 1474 787">• Configuration change: The Cisco ATA no longer requires that the <i>GkOrProxy</i> parameter be set to 0 in order to place direct IP-to-IP calls. However, the Cisco ATA now requires that the <i>IPDialPlan</i> parameter be set to a non-zero value (either 1 or 2) for direct IP-to-IP calls to work. For information on configuring the <i>IPDialPlan</i> parameter, refer to the following information in the Cisco ATA administrator's guide for SIP: http://www.cisco.com/univercd/cc/td/doc/product/voice/ata/ataadm/sip30ad/sip88ape.htm#wp1117097

Table 3 New Cisco ATA Features that Require Configuration for Release 3.2(0) (continued)



Topic	Behavior Description	Configuration
Anonymous Call Rejection (for SIP)	<p>The Cisco ATA allow users to configure their phone to reject an anonymous call. The default activation code is *77, and the de-activation code is *87. This feature can also be activated and de-activated through the use of the <i>PaidFeature</i> and <i>CallFeature</i> parameters. However, once you have used the activation/deactivation codes to set this feature, that setting overrides any changes made using the <i>PaidFeature</i> and <i>CallFeature</i> parameters and can only be changed by again using the activation/deactivation codes.</p> <p>Optionally, the Cisco ATA allows the system administrator to configure an announcement server number to which the Cisco ATA can forward a rejected call. For information on how to configure an announcement server number, see the next topic in this table: “Forwarding blocked anonymous calls to an Interactive Voice Response (IVR) server.”</p> <p>Related topic in this table</p> <ul style="list-style-type: none"> • “New CallCmd Action Identifiers” • “Remotely reset internal anonymous call-rejection (ACR) state” 	<p>Anonymous Call Rejection (ACR) is disabled/enabled using bits 13/29 of the <i>PaidFeature</i> and <i>CallFeature</i> parameters:</p> <p>Configuring via the Web Interface</p> <ul style="list-style-type: none"> • <i>PaidFeature</i> parameter: <ul style="list-style-type: none"> – 0 = ACR not available – 1(default)= ACR available • <i>CallFeature</i> parameter: <ul style="list-style-type: none"> – 0 = Reject anonymous calls – 1 (default)= Allow anonymous calls <p>Using the Voice Configuration Menu</p> <p>Use the following voice menu codes for ACR:</p> <ul style="list-style-type: none"> • *77 activates ACR and sets <i>CallFeature</i> bit(s) to 0. • *87 deactivates ACR and sets <i>CallFeature</i> bit(s) to 1. <p>CallCmd Parameter</p> <p>The <i>CallCmd</i> parameter example shown below includes the highlighted portion for enabling ACR:</p> <pre>Af;AH;BS;NA;CS;NA;Df;EB;Ff;EP;Kf;EFh;H H;Jf;AFh;EQ;I*67;gA*82;fA#90v#;OI;H#72 v#;bA#74v#;cA#75v#;dA#73;eA*67;gA*82;f A*70;iA*69;DA*99;xA*77;1A*87;mA;Uh;GQ;</pre>
Forwarding blocked anonymous calls to an Interactive Voice Response (IVR) server (SIP)	<p>This feature allows you to forward blocked Anonymous Calls (see the <i>ACR</i> feature described in the previous table entry) to an Interactive Voice Response (IVR) server.</p> <p> Note The local proxy must provide the IVR server.</p>	<p>To configure an Anonymous Call Rejection Directory Number (ACRDN), change the value of the ACRDN parameter (located on the Service Parameters screen of the Cisco ATA web configuration pages) from the default value of 0 to the dialed digits of the announcement server.</p> <p> Note The ACRDN parameter is an alphanumeric string (eight characters maximum), and its voice configuration menu code is 5004.</p>

Table 3 *New Cisco ATA Features that Require Configuration for Release 3.2(0) (continued)*

Topic	Behavior Description	Configuration
Syslog enhancements (SIP)	<p>At the end of a call, the syslog message contains the remote IP address (192.168.2.128 in the example below) and the caller/callee party number (9100 in the example).</p> <p>Example</p> <pre><182>00:02:02 192.168.2.246 [16]:[0][192.168.2.128][9100]RTP-RxStat:Dur=5,P kt=250,Oct=40000 <182>00:02:02 192.168.2.246 [16]:[0][192.168.2.128][9100]RTP-RxStat:Dur=5,P kt=156,Oct=24801,LatePkt=0,LostPkt=0,AvgJit=0</pre> <p>In these RTP statistics messages, the facility value is 22 (local use 6), the severity value is 6 (informational messages), and the priority value is 182.</p> <p>For more information about these messages and how the various values are determined, see the “Using System Diagnostics” section (see the subsection called “Example—RTP Statistic Messages”) in the administrator’s guide:</p> <p>http://www.cisco.com/univercd/cc/td/doc/product/voice/ata/ataadm/sip30ad/sip88ch5.htm#wp1040127</p>	There are no changes in this release for how to configure the <i>SyslogCtrl</i> and <i>SyslogIP</i> parameters.

Table 3 New Cisco ATA Features that Require Configuration for Release 3.2(0) (continued)

Topic	Behavior Description	Configuration
RTP statistics enhancements (SIP, SCCP)	<p>RTP statistics are reported in the following ways:</p> <ul style="list-style-type: none"> From the syslog output, where the statistics appear at the end of the call. From the web page <a href="http://<ip_address>/rtps">http://<ip_address>/rtps (or click on the RTP Statistics link). <p>The web page refreshes every 10 seconds unless you change the refresh interval (0 means to stop the refreshment). You can input the refresh interval, then press Set, or press Stop to stop the refreshment. You can also click on Clear Line 0 or Clear Line 1 to reset the counter for line 0 or line 1.</p> <ul style="list-style-type: none"> If the SIP protocol is used, RTP statistics can also be included in a BYE request and response. (See the “Configuration” column for more information.) <p>The output for both syslog and the web page shows the current/previous RTP statistics for both lines and both channels. If the Rx Status and Tx Status are ON, the statistics apply to the current call; if the Rx Status and Tx Status are OFF, the statistics apply to the previous call.</p>	<p>For the SIP protocol only, the Cisco ATA can be configured to include RTP statistics in a BYE request and response. ConnectMode bit 3 provides the ability to enable or disable this function. If the bit value is 0, no RTP statistics are included in the BYE request and response; if the bit value is 1, RTP statistics are included in the BYE request and response.</p> <p>If this bit is enabled, the Cisco ATA will insert the headers <i>RxStat</i> and <i>TxStat</i> as follows:</p> <pre>RxStat:Dur=a,Pkt=b,Oct=c,LatePkt=d,LostPkt=e,AvgJit=f</pre> <pre>TxStat: Dur=g,Pkt=h,Oct=i</pre> <p>where:</p> <ul style="list-style-type: none"> <i>Dur</i> is the total number of seconds since the beginning of reception or transmission. <i>Pkt</i> is the total number of RTP packets received or transmitted. <i>Oct</i> is the total number of RTP payload octets received or transmitted (not including RTP header). <i>LatePkt</i> is the total number of late RTP packets received. <i>LostPkt</i> is the total number of lost RTP packets received (not including late RTP packets). <i>AvgJit</i> is the average jitter, which is an estimate of the statistical variance of the RTP packet inter-arrival time, measured in timestamp units and calculated according to RFC 1889. <i>a, b, c, d, e, f, g, h, and i</i> are integers

Table 3 New Cisco ATA Features that Require Configuration for Release 3.2(0) (continued)

Topic	Behavior Description	Configuration
Dial plan enhancement (SIP)	The Cisco ATA system administrator can restrict the numbers to which a call can be transferred. The Cisco ATA has a new dial plan rule called the ‘T’ rule to allow this implementation.	<p>Syntax</p> <p><i>Tnnnn</i></p> <p><i>nnnn</i> represents the leading digits of the blocked-call number. <i>nnnn</i> can be composed with selection and range. The rule is triggered when the leading digits of the dialed string match the string <i>nnnn</i>. Negation cannot be used with the T rule.</p> <p>For complete information about using dial plans, see the corresponding section in the Cisco ATA administrator’s guide:</p> <p>http://www.cisco.com/univercd/cc/td/doc/product/voice/ata/ataadm/sip30ad/sip88ape.htm#wp1127834</p> <p>Example</p> <p>T1900 T1888 or T(1900 1888)</p> <p>This example dial plan blocks the transferred numbers that begin with 1900 or 1888.</p> <p>Log</p> <p>Call XFER block: < num> is shown in the prserv log, and a busy tone plays.</p>
Call history for incoming and outgoing calls (SIP)	The Cisco ATA system administrator can obtain the call history of the five most recent incoming and five most recent outgoing calls. The information includes the date, time, the called-party numbers, and the calling-party numbers.	<p>To obtain call history, access the following web page:</p> <p><a href="http://<ip_address>/callhistory">http://<ip_address>/callhistory</p> <p>When there are outgoing and incoming call histories displayed, the line is the active phone line, the date and time are for the initial time of the call, and the caller/callee is the phone number of the remote party.</p>
Instance ID Contact Header Parameter (SIP)	The Cisco ATA supports the Instance ID requirement as specified in the document <i>draft-jennings-sipping-instance-id-00</i> . The Contact header parameter, <i>instance=</i> , is appended to the end of the current list of Contact header parameters. The device MAC address is used as the Instance ID.	To enable this feature, set bit 4 of the ConnectMode parameter to 1.

Table 3 New Cisco ATA Features that Require Configuration for Release 3.2(0) (continued)


Topic	Behavior Description	Configuration
Service Failure User Notification on Dial Plan blocked calls (SIP)	<p>The Cisco ATA supports User Notification of Call Failure on all Dial Plan blocked calls if an Interactive Voice Response system is provided for this purpose; otherwise, standard tone treatments are applied.</p> <p>For complete dial plan information, see the following section in the administrator's guide:</p> <p>http://www.cisco.com/univercd/cc/td/doc/product/voice/ata/ataadmn/sip30ad/sip88ape.htm#wp1116978</p>	<p>You can configure the Cisco ATA to call a voice announcement server whenever an outgoing call attempt fails due to dial-plan restrictions. The telephone number of the server to which the Cisco ATA sends an INVITE request is specified by a configurable base number and a pre-assigned, non-configurable extension number.</p> <p>You must configure the following information:</p> <ul style="list-style-type: none"> • Base Number <p>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', 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.</p> <hr/> <p> Note Each dial plan rule must be partitioned from other rules with a vertical bar ().</p> <hr/> <p>The telephone number that the Cisco ATA calls will always consist of the base number followed by a two-digit extension. The extension corresponds to the service failure.</p> <p>The extension is 10 for dial plan blocked calls.</p> <p>If the administrator has configured a base number of 1234 and an outgoing call placed by the Cisco ATA user is blocked at the dial plan level, the called number would be 123410.</p> <ul style="list-style-type: none"> • Bit 24 of the OpFlags Parameter—Set the value of this bit to 1.

Table 3 *New Cisco ATA Features that Require Configuration for Release 3.2(0) (continued)*

Topic	Behavior Description	Configuration
Visual Message Waiting Indicator (VMWI) support for Denmark (SIP, SCCP)	The Cisco ATA supports the DTMF-based VMWI interface, and conforms to the <i>TeleDanmark TDK-TS 900 301-5</i> .	If Bits 0-1 of the <i>CallerIdMethod</i> parameter are set to the value of 1 (thus enabling DTMF caller ID), then Bits 30-31 of <i>CallerIdMethod</i> are defined as follows: <ul style="list-style-type: none"> • 0 = Disables VMWI based on TeleDanmark MWI, TDK-TS 900 301-5 • 1 = Enables VMWI based on TeleDanmark MWI, TDK-TS 900 301-5 • 2-3 = Reserved bits To enable caller ID and VMWI (conforming to TeleDanmark specification), set the <i>CallerIdMethod</i> parameter to 0x4000f5e1 or 0x433ef5e1.
Denmark Call Waiting Caller ID Presentation (CWCLIP) (SIP)	The Cisco ATA supports the dialing of R9 upon receipt of a call-waiting call to obtain the caller identification of the call-waiting caller. You must modify the <i>CallCmd</i> parameter of the Cisco ATA configuration to recognize an R9 input sequence. This feature is applicable for Denmark customers only, and adheres to the reference document <i>TeleDanmark TDK-TS 900 301-2</i> . Related topic in this table “New CallCmd Action Identifiers”	1. Set the Denmark <i>CallerIdMethod</i> parameter to one of the following values: <ul style="list-style-type: none"> – 0x0000f5e1 – 0x033ef5e1 – 0x4000f5e1 – 0x433ef5e1 2. Add an f9;SA call command sequence to the F (CONNECTED_ALERTING) Context-Identifier as in the following example: <pre>BS;NA;CS;NA;DE;EB;Ff0;ARf1;HPf2;EPf3;APf9;SA;Kf1;HFf2;EFF3;AFf4;EQ;Jf1;HFf2;EFF3;AFf4;EQ;Af4;EQ;I*31#;gA#31#;gA*90*v#;OI;H*21*v#;bA*61*v#;dA*67*v#;cA#21#;eA#61#;eA#67#;eA*31#;gA#31#;gA*43#;hA#43#;iA*69#;DA*99#;xA#37#;kA;Uh;GQ;S5;jA</pre> For complete information on call commands, see the Cisco ATA administrator’s guide for the SIP protocol: http://www.cisco.com/univercd/cc/td/doc/product/voice/ata/ataadm/sip30ad/sip88api.htm

Table 3 New Cisco ATA Features that Require Configuration for Release 3.2(0) (continued)




Topic	Behavior Description	Configuration
Configurable Ethernet port and speed for the Cisco ATA 188 (SIP, SCCP)	<p>This feature is applicable for the following Cisco ATA models:</p> <ul style="list-style-type: none"> • ATA188-I1-A • ATA188-I2-A <p> Note To make sure your Cisco ATA supports this feature, the hardware version number on your Cisco ATA must be 0x0009. Verify this by checking on the Device Information page (<a href="http://<ip_address>/DeviceInfo">http://<ip_address>/DeviceInfo).</p> <hr/> <p>You can configure the port speed and duplex of the Cisco ATA 188 PC and uplink ports.</p>	<p>Configuration Methods</p> <p>Configuration of this feature is available using any of the following methods:</p> <ul style="list-style-type: none"> • Using the TFTP server—<i>SwitchPorts</i> parameter. • Using the Voice Configuration Menu—Voice code 73876787. • Using the Cisco ATA web configuration page—<i>SwitchPorts</i> field on the Network Parameters screen. <p> Note See the Cisco ATA administrator's guide for your protocol for complete information about configuration methods.</p> <hr/> <p>Configuration Values</p> <p>Set the values of the applicable <i>SwitchPorts</i> parameter bits as follows to set the port speed and duplex of the PC and uplink ports of the Cisco ATA.</p> <p>Applicable Bits</p> <ul style="list-style-type: none"> • Bits 0-3 are for the PC port • Bits 4-7 are for the uplink port <p>Bit Values</p> <ul style="list-style-type: none"> • 0=10 Mbps, half duplex • 1=10 Mbps, full duplex • 2=100 Mbps, half duplex • 3=100 Mbps, full duplex • 4=AUTO detect • Other bit values are not valid <p> Note If the PC port value is set to a higher speed and duplex than that of the uplink port, the Cisco ATA automatically sets the PC port value to the maximum speed and duplex allowed by the uplink port.</p>

Table 3 New Cisco ATA Features that Require Configuration for Release 3.2(0) (continued)

Topic	Behavior Description	Configuration
Dial plan rule that handles number mapping for call services (SIP)	The Cisco ATA dial plan rule 'R' allows number mapping for call services.	<p>You can use the 'R' rule to map, for example, 411 service to a regional number configured by the service provider.</p> <p>Example R---5551234(411) maps 411 to 5551234.</p> <p>For complete information about dial plans, see the "Parameters and Defaults" section of the Cisco ATA administrator's guide: http://www.cisco.com/univercd/cc/td/doc/product/voice/ata/ataadm/sip30ad/sip88ape.htm#wp1019004</p>
Remotely set call forwarding number (SIP)	The Cisco ATA allows the system administrator to set the call forwarding number and the style by using an <i>http</i> command.	<p>Command syntax</p> <pre>http://<ip_address>/setcfwd?line<num>=<phone number>&style=<style></pre> <p>where:</p> <ul style="list-style-type: none"> • <num> is 0/1 for line 0/line 1 on the Cisco ATA • <phone number> is the call forwarding number • <style> is "None", "UnCond", "Busy" or "NoAns". <p>Example</p> <pre>http://<ip_address>/setcfwd?line0=5551234&style=Busy</pre> <p>sets the call forwarding number on busy for line 0 to 5551234.</p>
Remotely set call waiting state (SIP)	The Cisco ATA allows the system administrator to set the internal call waiting state if the permanent call waiting configuration is enabled. This is performed with an <i>http</i> command.	<p>Command syntax</p> <pre>http://<ip_address>/setcwait?line<num>=<value></pre> <p>where</p> <ul style="list-style-type: none"> • <num> is 0/1 for line 0/line 1 • <value> is 0/1 to denote that call waiting is disabled/enabled. <p>Example</p> <pre>http://<ip_address>/setcwait?line1=0</pre> <p>disables call waiting for line 1.</p>

Table 3 New Cisco ATA Features that Require Configuration for Release 3.2(0) (continued)

Topic	Behavior Description	Configuration
Remotely reset internal anonymous call-rejection (ACR) state (SIP)	<p>The Cisco ATA allows the system administrator to reset its internal anonymous call rejection state if the Anonymous Call Rejection feature is enabled.</p>	<ol style="list-style-type: none"> Set the PaidFeature parameter to either 0x00002000 or 0xffffffff. Include the sequence *77;1A*87;mA in the CallCmd parameter, as shown in the following example: <pre data-bbox="1052 499 1521 636">Af;AH;BS;NA;CS;NA;DE;EB;Ff;EP;Kf;EFh;HH;Jf;AFh;EQ;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*77;1A*87;mA;Uh;GQ;</pre> Issue the command: <pre data-bbox="1052 709 1521 730">http://<ip_address>/resetACR?line<num></pre> <p>where <num> is 0/1 for line 0/line 1.</p> <p>Example</p> <pre data-bbox="1052 846 1521 867">http://<ip_address>/resetACR?line0</pre> <p>resets the internal Anonymous Call Rejection value for line 0.</p> <p>After the setting takes effect, the web page displays either <i>OK</i> in the browser window (to indicate that the Cisco ATA internal ACR setting has been reset) or <i>Denied</i> (if the internal ACR settings on the Cisco ATA could not be reset because the ACR feature is not set in the PaidFeature bitmap parameter).</p>
New CallCmd Action Identifiers	<p>CallCmd action identifiers tell the Cisco ATA what action to perform. This action is based on the Input-Sequence that the user enters and the context in which the sequence is entered.</p> <p>The following three new CallCmd action identifiers are related to the new Anonymous Call Rejection feature and the new Denmark Call Waiting Caller ID feature:</p> <ul style="list-style-type: none"> Identifier <i>I</i> = Activate anonymous call rejection Identifier <i>m</i> = Deactivate anonymous call rejection Identifier <i>S</i> = Display DTMF-based Call Waiting Caller ID <p>For a list of the other CallCmd action identifiers, see the following section of the administrator's guide:</p> <p>http://www.cisco.com/univercd/cc/td/doc/product/voice/ata/ataadmn/sip30ad/sip88api.htm#wp1067084</p>	<p>See the configuration information for the two features that relate to these new CallCmd action identifiers:</p> <ul style="list-style-type: none"> “Anonymous Call Rejection” topic “Denmark Call Waiting Caller ID” topic

Dial Plan 'N' Rule for E.164 Number Normalization and Denormalization

This feature applies only to the SIP Protocol. The Cisco ATA provides a new dial plan rule—the 'N' rule—to facilitate ENUM mapping of URLs to E.164 formats and vice versa (normalizing a called number on outgoing calls and denormalizing calling number on incoming calls to the E.164 numbering format).

This section contains the following topics:

- [“Definitions of Terms” section on page 16](#)
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Definitions of Terms

The following is a list of terms that are applicable to this topic or are used in the sections that follow:

- ENUM—A protocol developed for fetching Universal Resource Identifiers (URIs) given an E.164 number.
- NDD (National Direct Dialing) Prefix—The access code used to make a call within that country from one city to another (when calling another city in the same vicinity, this may not be necessary). This is also referred to as the National Prefix.
- NPA—Numbering Plan Area (also known as an area code).
- NXX—The three digits after an NPA in a phone number (also called a prefix, exchange, or central office code).
- NANP—North American Numbering Plan. This refers to the area code/prefix concept, which applies in the United States, most U.S. territories, Canada, and some island countries in the Atlantic and Caribbean.
- CC (Country Code)—A one-, two-, or three-digit number used to specify the destination country for international calls outside the North American Numbering Plan (NANP).
- IDD (International Direct Dialing) Prefix—The prefix needed to dial a call FROM the country listed TO another country. This is followed by the country code for the country you are calling. This is also referred to as the International Prefix.

Behavior Description

This dial plan rule allows users to specify their international prefix, country code, national prefix, and local area code. It also provides the option to turn on/off the normalization and denormalization process, and the option of putting a leading plus sign (+) in the normalized outgoing number.

Three types of calls are subject to this process—Local calls, long distance calls and international calls.

Normalization Rules

The Cisco ATA follows these rules to normalize a called-party number on outgoing calls when the N-rule **NF011+1+1+408** is applied:

- A local call is normalized as:

[+]**<NDD><NPA><NXX-XXXX>**

For example, if a local call for a number of 5556666 is made (and the NPA is 408), the Cisco ATA would normalize the number to +14085556666.

- A long distance call is normalized as:

[+]**<NDD><NPA><NXX-XXXX>**

For example, if a long distance call for a number of 13106667777 is made, the Cisco ATA normalizes the number to +13106667777.

- An international call is normalized as:

[+]**<CC><NPA><NXX-XXXX>**

For example, if an international call for a number of 011886277778888 is made, the Cisco ATA normalizes the number to +886277778888.

Denormalization Rules

The Cisco ATA follows these rules to de-normalize calling-party numbers on incoming calls (the leading plus sign is removed) when the N-rule **NF011+1+1+408** is applied.

- A local call is denormalized as:

<NXX-XXXX>

For example, if the calling party number is +14085556666, the Cisco ATA would display a caller ID of 5556666.

- A long distance call is denormalized as:

<NDD><NPA><NXX-XXXX>

If the calling party number is +13106667777, the Cisco ATA would display a caller ID of 13106667777.

- An international call is denormalized as:

<IDD><CC><NPA><NXX-XXXX>

For example, if the calling party number is +886277778888, the Cisco ATA would display a caller ID of 011886277778888.

Syntax for 'N' Rule

The syntax for the 'N' rule follows this format:

NTn1+n2+n3+n4

where:

- T is the letter used to indicate the available options in applying the 'N' rule:
 - A: No normalization, no denormalization
 - B: No normalization, denormalization only
 - C: Normalization without leading plus sign only, no denormalization
 - D: Normalization without leading plus sign, denormalization
 - E: Normalization with leading plus sign only, no denormalization
 - F: Normalization with leading plus sign, denormalization
- n1 is the international prefix.
- n2 is the local country code number.
- n3 is the national prefix.
- n4 is the local area code.

Example 1

NF011+1+1+408

011 is the international prefix, the first 1 is the country code of the United States, the second 1 is the national prefix in the United States, and 408 is the area code for San Jose, CA.

Example 2

NF011+886+1+2

where 011 is the international prefix, 886 the country code of Taiwan, 1 is the national prefix in Taiwan, and 2 is the area code for Taipei, Taiwan.

New Features for Release 3.2 that do not Require Configuration

The following list contains new features for Cisco ATA Release 3.2 that do not require configuration. This list applies only to the SIP protocol.

- RFC 3261 Enhancements:
 - Loose-routing support
 - Addition of branch parameter in Via header
- RFC 3262 PRACK Support:
 - The Cisco ATA supports PRACK based on RFC 3262.
- RFC 3264 Offer/Answer Support:
 - The Cisco ATA supports the offer and answer model based on RFC 3264.

- RFC 3311 UPDATE Method Support:
The Cisco ATA supports handling a SIP UPDATE request but does not initiate an UPDATE request.
- Registration and line status reporting:
The Cisco ATA provides a web page that includes SIP registration status and call-state status for both Cisco ATA lines.

The web page is:

`http://<ip_address>/stats`

For each line, this page shows the status of channels 1 and 2 of the Cisco ATA. For example, their status might be:

```
Line 0 states IDLE,IDLE  
Line 1 states IDLE,IDLE
```

Possible states for each channel are:

- IDLE
- PREDIAL
- DIALING
- CALLING
- CONNECTED
- ALERTING
- ANSWERING
- HOLDING
- HELD
- CANCELING
- DISCONNECTING
- WAITHOOK
- CONFIG
- TRYING
- RETRYING
- N/A

Resolved Issues for Release 3.2

Table 4 lists DDTS issues that have been resolved in Cisco ATA Release 3.2

Table 4 Resolved Issues for Release 3.2

DDTS Number	Summary	Protocol(s)
CSCee58660	After receiving a <i>487 Request Cancelled</i> message, the Cisco ATA does not generate an ACK with the correct <i>Req-URI</i> destination.	SIP
CSCee89902	A Cisco ATA using both the Sweden locale and G.726 codec encounters unexpected audio after a hook flash is performed.	SIP
CSCee91360	Loose Routing causes subsequent calls to fail.	SIP
CSCee94128	A Cisco ATA using the SCCP protocol with Cisco CallManager version 4.0(1) SR2 does not provide a reminder ring for those parties in a three-way call.	SCCP
CSCef01154	The Cisco ATA does not release a call-cancel request if a new call occurs within five seconds.	SIP
CSCef04850	Some supplementary services failed on a Cisco ATA running SCCP with Cisco CallManager version 4.x.	SCCP
CSCef15297	When the Consultation Transfer feature is disabled in the PaidFeature parameter, and the user goes on hook during a three-way call, the last party dialed gets placed on hook instead of being dropped.	SIP
CSCef16936	DNS Query is not re-executed if an SRV query times out.	SIP
CSCef19907	The Cisco ATA does not report Quality of Service (QoS) to CAR.	SCCP
CSCef20551	When attached to a cable modem, the Cisco ATA crashes after several hours online.	SIP
CSCef22627	The Cisco ATA does not send a reminder ring to the phone.	SCCP
CSCef24305	While performing loose routing, the Cisco ATA may drop the port information in the Request URI.	SIP
CSCef44462	The Cisco ATA Reorder tone cannot be disabled.	SIP
CSCef54264	In a Cisco CallManager Express environment, the Cisco ATA cannot send out digits after the user presses **3 (pickup).	SCCP
CSCef56665	DTMF-relay tone generation is susceptible to packet loss.	SIP
CSCef57781	The Cisco ATA crashes while trying to re-establish audio using a bogus audio codec value of -1.	SIP
CSCef58130	Clicking the Apply button on the web configuration page should invoke a warning that the Cisco ATA will get reset.	SCCP
CSCef60921	Multiple media-level sections in an SDP are not supported.	SIP
CSCef94946	Upon receiving an unknown telephone number, the Cisco ATA does not correctly send calling line identification presentation (CLIP) information to a TeleDanmark-compliant caller ID device.	SIP, SCCP
CSCeg23884	The Cisco ATA has poor voice quality when <i>NumTXFrames</i> is set with a high value.	SIP, SCCP

Related Documentation

Use these release notes in conjunction with the documents located at this index:

<http://www.cisco.com/univercd/cc/td/doc/product/voice/ata/index.htm>

Obtaining Documentation

Cisco documentation and additional literature are available on Cisco.com. Cisco also provides several ways to obtain technical assistance and other technical resources. These sections explain how to obtain technical information from Cisco Systems.

Cisco.com

You can access the most current Cisco documentation at this URL:

<http://www.cisco.com/univercd/home/home.htm>

You can access the Cisco website at this URL:

<http://www.cisco.com>

You can access international Cisco websites at this URL:

http://www.cisco.com/public/countries_languages.shtml

Ordering Documentation

You can find instructions for ordering documentation at this URL:

http://www.cisco.com/univercd/cc/td/doc/es_inpk/pdi.htm

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San Jose, CA 95134-9883

We appreciate your comments.

Obtaining Technical Assistance

For all customers, partners, resellers, and distributors who hold valid Cisco service contracts, Cisco Technical Support provides 24-hour-a-day, award-winning technical assistance. The Cisco Technical Support Website on Cisco.com features extensive online support resources. In addition, Cisco Technical Assistance Center (TAC) engineers provide telephone support. If you do not hold a valid Cisco service contract, contact your reseller.

Cisco Technical Support Website

The Cisco Technical Support Website provides online documents and tools for troubleshooting and resolving technical issues with Cisco products and technologies. The website is available 24 hours a day, 365 days a year at this URL:

<http://www.cisco.com/techsupport>

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<http://tools.cisco.com/RPF/register/register.do>

Submitting a Service Request

Using the online TAC Service Request Tool is the fastest way to open S3 and S4 service requests. (S3 and S4 service requests are those in which your network is minimally impaired or for which you require product information.) After you describe your situation, the TAC Service Request Tool automatically provides recommended solutions. If your issue is not resolved using the recommended resources, your service request will be assigned to a Cisco TAC engineer. The TAC Service Request Tool is located at this URL:

<http://www.cisco.com/techsupport/servicerequest>

For S1 or S2 service requests or if you do not have Internet access, contact the Cisco TAC by telephone. (S1 or S2 service requests are those in which your production network is down or severely degraded.) Cisco TAC engineers are assigned immediately to S1 and S2 service requests to help keep your business operations running smoothly.

To open a service request by telephone, use one of the following numbers:

Asia-Pacific: +61 2 8446 7411 (Australia: 1 800 805 227)

EMEA: +32 2 704 55 55

USA: 1 800 553 2447

For a complete list of Cisco TAC contacts, go to this URL:

<http://www.cisco.com/techsupport/contacts>

Definitions of Service Request Severity

To ensure that all service requests are reported in a standard format, Cisco has established severity definitions.

Severity 1 (S1)—Your network is “down,” or there is a critical impact to your business operations. You and Cisco will commit all necessary resources around the clock to resolve the situation.

Severity 2 (S2)—Operation of an existing network is severely degraded, or significant aspects of your business operation are negatively affected by inadequate performance of Cisco products. You and Cisco will commit full-time resources during normal business hours to resolve the situation.

Severity 3 (S3)—Operational performance of your network is impaired, but most business operations remain functional. You and Cisco will commit resources during normal business hours to restore service to satisfactory levels.

Severity 4 (S4)—You require information or assistance with Cisco product capabilities, installation, or configuration. There is little or no effect on your business operations.

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- The Cisco *Product Catalog* describes the networking products offered by Cisco Systems, as well as ordering and customer support services. Access the Cisco Product Catalog at this URL:
<http://cisco.com/univercd/cc/td/doc/pcat/>
- *Cisco Press* publishes a wide range of general networking, training and certification titles. Both new and experienced users will benefit from these publications. For current Cisco Press titles and other information, go to Cisco Press at this URL:
<http://www.ciscopress.com>
- *Packet* magazine is the Cisco Systems technical user magazine for maximizing Internet and networking investments. Each quarter, Packet delivers coverage of the latest industry trends, technology breakthroughs, and Cisco products and solutions, as well as network deployment and troubleshooting tips, configuration examples, customer case studies, certification and training information, and links to scores of in-depth online resources. You can access Packet magazine at this URL:
<http://www.cisco.com/packet>

- *iQ Magazine* is the quarterly publication from Cisco Systems designed to help growing companies learn how they can use technology to increase revenue, streamline their business, and expand services. The publication identifies the challenges facing these companies and the technologies to help solve them, using real-world case studies and business strategies to help readers make sound technology investment decisions. You can access iQ Magazine at this URL:

<http://www.cisco.com/go/iqmagazine>

- *Internet Protocol Journal* is a quarterly journal published by Cisco Systems for engineering professionals involved in designing, developing, and operating public and private internets and intranets. You can access the Internet Protocol Journal at this URL:

<http://www.cisco.com/ipj>

- World-class networking training is available from Cisco. You can view current offerings at this URL:

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