Release Notes for Cisco ATA 186 and Cisco ATA 188 Release 2.15

March 21, 2003

OL-1269-07

These release notes describe newly incorporated features including resolved and open issues for the Cisco ATA 186 (Analog Telephone Adaptor 186) and Cisco ATA 188 (Analog Telephone Adaptor 188) Release 2.15 for SIP and H.323 images.

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

Contents

These release notes provide the following information:

CISCO SYSTEMS

Corporate Headquarters:
Cisco Systems, Inc., 170 West Tasman Drive, San Jose, CA 95134-1706 USA

Introduction to Cisco ATA Analog Telephone Adaptor

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

There are presently two Cisco ATA products 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 difference between these products is that the Cisco ATA 186 has one RJ45 port that provides access to an Ethernet network, while the Cisco ATA 188 has an Ethernet switch and two RJ45 ports. The Cisco ATA 188 has one RJ45 port for access to an Ethernet network and a second RJ45 port for connecting a downstream Ethernet device such as a PC.

Downloading and Upgrading the Software

To take advantage of the features of Cisco ATA Release 2.15, you must upgrade the Cisco ATA software. You can download the software, after logging in, at:

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

For more information, refer to the *Cisco ATA Installation and Configuration Guide (H.323)* and the *Cisco ATA 186 and Cisco ATA 188 Analog Telephone Adaptor Administrator's Guide (SIP)*:

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

New Features and Changes in Release 2.15

This section contains information on new and changed features for Cisco ATA Release 2.15.

New Features in Release 2.15

General Features

This section contains information on new features for Cisco ATA Release 2.15.

- The passwords (PWD0 and PWD1) and encryption key (EncryptKey) are now displayed on the web interface as asterisks instead of plain readable text.
- A network status page is added. The page is http://*ip address*/stats where *ip address* is the IP address of the Cisco ATA 186.

- The default media base port changed from 10000 to 16384 in the example profiles. This is a documentation change only. No software changed.

- To maximize interoperability with other Cisco gateway devices, you can use either 126/127 or 0/8 as RTP dynamic payload type for G.711 /G.711a upspeed during fax passthrough.

    Configuration bit 2 of ConnectMode(mask 0x4)

    > 0: Use 126/127 for G.711 /G.711a upspeed (default)

    > 1: Use 0/8 for G.711 /G.711a upspeed

New Features for H.323

This section contains information on new features for Cisco ATA Release 2.15 for H.323 only.

- The Cisco ATA now has the capability to handle call forward requests from an H.225 Facility-UUIE message.

- When the Cisco ATA 186 GkorProxy parameter is set to 0, setting the IpDialPlan parameter to 0 causes the Cisco ATA to handle a dialed string containing '*' delimiters as an outgoing calledPartyNumber instead of a target IP address. In addition, instead of translating each '*' delimiter to a '.', the Cisco ATA passes these delimiters to the calledPartyNumber parameter intact.

New Features for SIP

This section contains information on new features for Cisco ATA Release 2.15 for SIP only.

- OPTIONS Method Support

    The Cisco ATA does not generate OPTIONS requests; however, it responds to OPTIONS requests.

    When an OPTIONS request is received, the Cisco ATA always returns a 200 response code. The response includes an **Allow** header listing all SIP methods supported and a SDP listing all audio and AVT payload types available.

- DNS SRV Support

    The Cisco ATA now 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 priority are used as the primary and backup proxy servers (AltGk is ignored in this case).

- User Configurable Call Waiting Permanent Default Setting

  This feature allows the user to specify the default call waiting setting for every call on a permanent basis through the service activation and deactivation codes.

  The following parameter controls the activation and deactivation of this feature.

  ```
  Parameter: ConnectMode
  Bit:  23
  Bitmask: 0x00800000
  Values: 0 = Disable ability for the user to configure permanent call waiting default
  setting for all calls  (DEFAULT)
          1 = Enable ability for the user to configure permanent call waiting default
  setting for all calls
  ```

- 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.

- User Configurable Timeout On No Answer for Call Forwarding

  This feature allows the user 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:

  ```
  <Service Activation Code>  * <Phone Number>  * <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 SigTimer parameter bits 20-25, which has a default value of 20 seconds, is in effect.

  For example, using the U.S. Call Command parameter string, the U.S. service activation code is #75 and deactivation code is #73. To forward calls on no answer after 15 seconds to 555-1212, enter the following:

  ```
  #75*5551212*15#
  ```

  To deactivate this feature, enter #73.

  ```
  #73
  ```

New Features for H.323 and SIP

This section contains information on new features for Cisco ATA Release 2.15 for H.323 and SIP.

- Added ITU G.711 comfort noise during silence period.

  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 the user from hearing silent gaps when neither party is talking.

- Configurable hook-flash timing

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

  ```
  Parameter:  SigTimer
  Bits:  26 - 27
  Values:  0 = 60 ms 2 = 200 ms
           1 = 100 ms            3 = 300 ms
  Default:  0
  ```

```
Description:  These bits specify the minimum on-hook time required for a hook-flash
event

Bits:  28 - 31
Values:
0 = 1000 ms   8 = 800 ms
1 = 100 ms              9 = 900 ms
2 = 200 ms             10 = 1000 ms
3 = 300 ms             11 = 1100 ms
4 = 400 ms             12 = 1200 ms
5 = 500 ms             13 = 1300 ms
6 = 600 ms             14 = 1400 ms
7 = 700 ms             15 = 1500 ms
Default:  0
Description:  These bits specify the maximum on-hook time allowed for a hook-flash
event

The following shows the timing for hook-flash and on-hook events.


ignore     |          hook-flash        |    on-hook
--------------+-------------------------+----------------
          Minimum                       Maximum
```
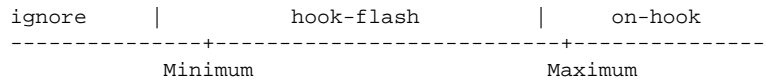
- Configurable mixing of call-waiting tone and audio

  This feature allows the call-waiting tone to be mixed with the audio in an active call. A party receiving an incoming call while on another call hears the call-waiting tone without a pause in the audio.

  The following parameter controls the activation and deactivation of this feature:

  ```
  Parameter: ConnectMode
  Bit: 24
  Bitmask: 0x01000000
  Values: 0 = Disable mixing of call waiting tone with audio (DEFAULT)
          1 = Enable mixing of call waiting tone with audio
  ```

- Configurable on-hook delay

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

  The following parameter controls the timing of this feature:

  ```
  Parameter: FeatureTimer
  Bits: 8 - 12
  Bitmask: 0x00001f00
  Bit Range: 0 - 31         (1 unit = 5 seconds)
  Values: 0 - 155 seconds
  Default: 0 second         (no delay)
  ```

- 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 hearing the busy tone. 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 hears a ringback. If the caller picks up the call first, the caller hears the ringback. This feature is automatically cancelled when the called party rings.

The following parameters control the timing of this feature.

```
Parameter: FeatureTimer
Bits: 0 - 3
Bitmask: 0x0000000f
Bit Range: 0 - 15          (1 unit = 5 minutes)
Values: 0 - 75 minutes
Default: 0                 (1 = 5 min, 2 = 10 min, 3 = 15 min ...)
Description: These bits control the maximum time spent redialing
```

Note A value of zero (0) sets the default redialing timeout to 30 minutes.

```
Parameter: FeatureTimer
Bits: 4 - 7
Bitmask: 0x000000f0
Bit Range: 0 - 15          (1 unit = 15 seconds)
Values: 0 - 225 seconds
Default: 0                 (1 = 15 sec, 2 = 30 sec, 3 = 45 sec ...)
Description: These bits control the interval between each redial.
```

Note A value of zero (0) sets the default redial interval to 15 seconds.

```
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
```

Changes in Release 2.15

This section lists the changes in Cisco ATA Release 2.15.

Changes for SIP

This section lists the changes in Cisco ATA Release 2.15 when using SIP only:

- CSCdx77974

  When a re-INVITE request without an SDP is received, the Cisco ATA lists all supported codecs in the SDP of its 200 OK response. Previously, only the codec selected for the original connection was returned in the body of the 200 OK response.

- The Record-Route header is now ignored in non-2xx responses. Previously, the Cisco ATA would process and use the Record-Route header information that appeared in all responses.

- The Record-Route header is now included in 2xx response to INVITE requests only. Previously, the header was included in every response.

- When a 480 "Temporary Unavailable" response is received for a REGISTER request, the Cisco ATA attempts to register again. Previously, the Cisco ATA would stop at reregistering when a 480 response is returned by the proxy server.

- The REFERRED-BY header is now treated as a REQUESTED-BY header to support the far-end user agent's transfer-by-reference.

Changes for H.323 and SIP

This section lists the change in Cisco ATA Release 2.15 when using SIP or H.323 protocol:

- CDP/VLAN is enabled for SIP/H.323.

The old and new behavior is given for each of the following features.

- Old behavior: G.711 /G.711aupspeed for fax passthrough can only be triggered by the 2100 Hz CED tone.

  New behavior: If CED tone is absent in a fax call, the receiver V.21 preamble flag is detected by Cisco ATA to trigger G.711 /G.711a upspeed for fax passthrough.

- Old behavior: When operated in fax passthrough mode, Cisco ATA can handle only the RTP dynamic payload type of 126/127 for G.711 /G.711a upspeed.

  New behavior: When operated in fax passthrough mode, Cisco ATA can handle RTP dynamic payload type of 126/127 and 0/8 for G.711 /G.711a upspeed.

- Old behavior: When operated in fax passthrough mode, Cisco ATA accepts only NSE event packets, in which both volume and duration parameters are set to 0.

  New behavior: When operated in fax passthrough mode, Cisco ATA accepts NSE event packets with any volume and duration.

- Old behavior: When operated in fax passthrough mode, Cisco ATA uses RTP dynamic payload type of 126/127 for G.711 /G.711a upspeed.

  New behavior: You can use bit 2 in ConnectMode (mask 0x4) to specify the RTP payload type for G711 /G711a upspeed in fax passthrough mode.

  - 0 means use 126/127 for G711 /G711a upspeed (default)
  - 1 means use 0/8 for G711 /G711a upspeed

Resolved Issues in Cisco ATA Release 2.15

This section lists the issues in previous releases of the Cisco ATA that are resolved in Release 2.15.

- Resolved H.323 Issues, page 8
- Resolved SIP Issues, page 9
- Resolved H.323 and SIP Issues, page 12

Resolved H.323 Issues

This section lists the issues in previous releases of the Cisco ATA that are resolved in Release 2.15 for H.323 only.

- CSCdy27753

  ReleaseCompleteReason and Cause IE are both included in the ReleaseComplete message. This causes issues for some customers' CDR applications. This has been resolved—only Cause IE is now included in the ReleaseComplete message.

- CSCdy36046

  One-way audio occurs when the Cisco ATA186 calls the gateway using IOS version 12.2(11)T. This has been resolved.

- CSCdx52024

  Audio channel is not opened between Cisco ATA and Siemens IP phone

  The Cisco ATA fast start element construction method was changed to fix this issue.

- CSCdx66687

  Missing RAS address in keepalive RRQ

  The Cisco ATA is missing the rasAddress in its keepalive (lightweight) RRQ. This issue has been resolved.

- CSCdx76611

  Fast start request from other endpoint result in no audio or failure

  The fast start fix is included in Cisco ATA Release 2.15.

- CSCdx80624

  Cisco ATA fails to make/receive further calls after it rejects an incoming call

  Under rare timing conditions, if the call is disconnected or rejected while in the middle of the H.245 negotiation, the H.245 channel may be left in a state that can prevent the line from making or receiving further calls until it resets.

Resolved SIP Issues

This section lists the issues in previous releases of the Cisco ATA that are resolved in Release 2.15 for SIP only.

- CSCdx89985

  Registration stops if a 480 or 503 response is returned. The Cisco ATA 186 now continues to retry registration if those response are received.

- CSCdy00898

  The RequestUri in a BYE request is truncated when it is longer than 40 characters. The Cisco ATA 186 can now properly handle RequestUri strings up to 70 characters.

- CSCdy02064

  All port numbers in the route header URIs are set to the same port number as the one in the contact header URI when the Cisco ATA is a user agent server. The Cisco ATA no longer modifies the port numbers in the route header URIs.

- CSCdy49824

  A "488 Not Acceptable Here" response is returned for an INVITE request if there are multiple "m=" SDP headers. The Cisco ATA can now properly parse more than one "m=" SDP header in a request.

- CSCdy56100

  Cisco ATA186 does not release SIP call if it goes onhook after hook flash. This has been resolved.

- CSCdy53544

  Cisco ATA should not modify the To/From header in a Cancel request.

- CSCdy61282

  On-hook delay prevents switching to a call-waiting call.

- CSCdy66651

  A Bye Request that contains an Also header should also contain a Requested-By header.

Release 2.15 Build 030313a

The following list contains SIP issues that have been resolved for Build 030313a of the 2.15 release:

- CSCea48953

  The Cisco ATA can now issue a prompt for the UIPassword value, if enabled, when a factory reset or upgrade is initiated with the voice configuration menu.

  To configure the Cisco ATA to prompt for the UIPassword, configure the OpFlags parameter, bits 28 to 31, with the value of 6. Any other value for these bits means that the Cisco ATA will not prompt you for the UIPassword when you attempt to perform a factory reset or upgrade using the voice configuration menu.

- CSCea26155

  The PROTOS test-suite for SIP can cause the Cisco ATA to operate improperly, such as rebooting or hanging. Cisco recommends that customers with earlier software versions upgrade to this latest release.

  This is now fixed.

- CSCea27473

  In earlier releases, the Cisco ATA accepted a SIP request or response up to 1,536 bytes long; however, if a message was larger than this, the Cisco ATA truncated it and was not able to parse it correctly.

  The Cisco ATA can now accept a SIP message up to 3,000 bytes long.

- CSCdz43469

  The G.711 frame size was fixed at 20 ms and could not be configured through the NumTxFrames configuration parameter.

  In this release, the G.711 frame size is now configurable via the NumTxFrames parameter with a range of 10-60 ms.

- CSCdz87773

  In previous releases, the Cisco ATA sends an INVITE message with a request URI that includes the "user=phone" parameter, but the Cisco ATA does not include this parameter in the corresponding ACK message for a "487 Request Terminated" response.

  The Cisco ATA now includes the "user=phone" parameter as needed in an ACK request.

- CSCea13176

  When the Contact header in the "200 OK" response from the SIP proxy does not contain a User ID, the Route header in the ACK message from the Cisco ATA contains the Contact IP address with an empty user-id field, such as "@cisco.com".

  This has been fixed.

- CSCdz74514

  When a PSTN gateway returns a "486 Busy Here" response after a "183 Session Progress" response, the callback-on-busy feature fails on the Cisco ATA. The Cisco ATA interprets a 183 response as the far end phone ringing and, therefore, terminates the automatic retry even though the far end may actually be busy.

  In this release, you can configure FeatureTimer parameter bits 13-15 to specify the amount of time the Cisco ATA waits for a "486 Busy Here" response after receiving a "183 Session Progress" response. This allows a time period in which the Cisco ATA can receive a "486" response before ringing the phone.

  You can configure the FeatureTimer parameter as follows:

  FeatureTimer Parameter

  Type: Bitmap

  Bits: 13-15

  Values:

  - 0 = No waiting (Default)
  - 1 = 1 second
  - 2 = 2 seconds
  - 3 = 3 seconds
  - 4 = 4 seconds
  - 5 = 5 seconds

- – 6 = 6 seconds
- – 7 = 7 seconds

- CSCdz23724

  Upon receiving an INVITE message, the Cisco ATA performs a DNS lookup of the "From" header hostname for the call return feature. This may delay the Cisco ATA from ringing the local phone and playing the ringback tone to the caller.

  In this release, ConnectMode bit 25 can be used to indicate whether the Cisco ATA or the Proxy will perform the call return. If the bit is set to 0, the Cisco ATA will perform the call return and does a DNS lookup of the "From" header hostname. If the bit is set to 1, the Proxy will perform the call return and the Cisco ATA will not perform a DNS lookup of the "From" header hostname.

- CSCdy53544

  The Cisco ATA incorrectly adds a tag to the "To" header in a CANCEL request to an INVITE message.

  This has been fixed.

- CSCdy61282

  When on-hook delay is active, a Cisco ATA originating a call can temporarily hang up the line and then pick up the line from another phone (or even the same phone) without disconnecting the call. A problem arises if a user is already in a call and an incoming call occurs. The user will receive the call-waiting tone while the second caller will receive the ringback tone. If the user hangs up, the first caller will be temporarily suspended (not disconnected or placed on hold) while the second caller continues to receive ringback. When the user picks up the phone again, both the user and the first caller can communicate with each other; however, the user cannot switch to the call-waiting call.

  This has been fixed.

- CSCdy54007

  The Cisco ATA now supports SDP in MIME multiparts.

- CSCdy58652

  When display name is not used, there is an extra blank space within the header.

  This has been fixed.

- CSCea49262

  The Server and User-Agent headers show incorrect model information for a Cisco ATA 188.

  This has been fixed.

- CSCea49281

  If the DHCP server loses its state due to a power failure, the Cisco ATA ignores broadcast packets and, as a result, the lease time extension will fail.

  With this release, the Cisco ATA now listens for broadcast packets after it requests a lease extension from the DHCP server.

- CSCdz46738

  The HTTP Refresh and Reset commands do not work if access to the web interface is disabled via the OpFlags parameter (bit 7).

  The Cisco ATA now allows the Refresh and Reset commands to work independent of the web interface access setting.

Resolved SIP Issues ■

- CSCea49402

  If you press the Cisco ATA function button at any time, the Cisco ATA will reset. This is a designed behavior similar to the reset button on a PC. However, if the function button is pressed during a call, the Cisco ATA will reset and existing calls will be dropped.

  To prevent accidental resets, the function button is now disabled during a call (H323/SIP only). The voice configuration menu can be accessed only if both phone lines are not in use.

- CSCea49801

  The Cisco ATA uses the wrong payload type number for sending out-of-band (OOB) DTMF packets if the received SDP media line (m=) includes more than one dynamic payload type and the last payload type in the list is not for an OOB DTMF.

  The Cisco ATA now correctly parses the SDP media line (m=) and checks each dynamic payload type with the corresponding definition in the attribute line (a=).

- CSCea49799

  Out-of-band DTMF packets are sent with fixed duration and incremental timestamp instead of fixed timestamp and incremental duration. This may cause some servers to detect multiple transmission of the same digit even when the digit had been pressed only once.

  The ATA now correctly sends OOB DTMF packets with fixed timestamp and incremental duration.

- CSCea49843

  The registration timing has been improved in this release.

  For registration requests, the T2 backoff time has been changed from two seconds to four seconds.

  Upon receiving a provisional response for a registration request, the Cisco ATA will resend a registration request at a fixed four-second interval instead of using an exponential backoff algorithm capping off at two seconds. Therefore, re-registration now occurs twice as early. For example, if re-registration occurred one second before expiration in previous releases, it now would occur two second before expiration.

Resolved H.323 and SIP Issues

This section lists the issues in previous releases of the Cisco ATA that are resolved in Release 2.15 for both H.323 and SIP.

- CSCdy23449

  Cisco ATA 186 is not responding to DHCP Offer packet. This has been resolved.

- CSCdy42827

  Cisco ATA does not detect all DTMF digits during dialing stage. This has been resolved.

- CSCdy42849

  Calls failing on Cisco ATA 186 and AS5350 gateways with Cisco CallManager using the G723 codec. This has been resolved.

- CSCdy47333

  When silence suppression is enabled, IP ringback and conferencing may become silent when using the G711 codec. This has been resolved.

- CSCdy47351

  Repeat dialing does not resume after pausing to place a call. This has been resolved.

Open Issues in Cisco ATA Release 2.15

This section contains the following topics:

Open Issues for H.323

No H.323-specific open issues exist in Cisco ATA Release 2.15.

Open Issues for SIP

No SIP-specific open issues exist in Cisco ATA Release 2.15.

Open Issues for H.323 and SIP

This section lists open issues for Cisco ATA that are for both H.323 and SIP.

- CSCdy23439

  Description: When the Cisco ATA sends Cisco Discovery Protocol (CDP) frames to the switch, the Device ID TLV has a value of *SEPxxxxxxxxxxxx* instead of *ATAxxxxxxxxxxxx*. The Device ID TLV value is the same as the device name used to register with Cisco CallManager. Change the value accordingly.

  Workaround: None.

Note Repeat dialing upon receiving a busy signal fails when calling a Primary Rate Interface (PRI) gateway or an IPRingBack-enabled Cisco ATA. This problem exists in H.323 and SIP for PRI. The problem usually occurs when the far end sends ringback tone to the caller. The only cases where the "call-back-on-busy" feature works are on-net calls without enabling IP ringback from the called device.

Related Documentation

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

- *ATA 186 and ATA 188 Analog Telephone Adaptor*

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

Obtaining Documentation

The following 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 the following URL:

http://www.cisco.com

Translated documentation is available at the following URL:

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

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:

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

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.

This document is to be used in conjunction with the documents listed in the "Related Documentation" section on page 13.

CCIP, the Cisco Arrow logo, the Cisco *Powered* Network mark, the Cisco Systems Verified logo, Cisco Unity, Follow Me Browsing, FormShare, Internet Quotient, iQ Breakthrough, iQ Expertise, iQ FastTrack, the iQ logo, iQ Net Readiness Scorecard, Networking Academy, ScriptShare, SMARTnet, TransPath, and Voice LAN are trademarks of Cisco Systems, Inc.; Changing the Way We Work, Live, Play, and Learn, Discover All That's Possible, The Fastest Way to Increase Your Internet Quotient, and iQuick Study are service marks of Cisco Systems, Inc.; and Aironet, ASIST, BPX, Catalyst, CCDA, CCDP, CCIE, CCNA, CCNP, Cisco, the Cisco Certified Internetwork Expert logo, Cisco IOS, the Cisco IOS logo, Cisco Press, Cisco Systems, Cisco Systems Capital, the Cisco Systems logo, Empowering the Internet Generation, Enterprise/Solver, EtherChannel, EtherSwitch, Fast Step, GigaStack, IOS, IP/TV, LightStream, MGX, MICA, the Networkers logo, Network Registrar, *Packet*, PIX, Post-Routing, Pre-Routing, RateMUX, Registrar, SlideCast, StrataView Plus, Stratm, SwitchProbe, TeleRouter, and VCO are registered trademarks of Cisco Systems, Inc. and/or its affiliates in the U.S. and certain other countries.

All other trademarks mentioned in this document or Web site are the property of their respective owners. The use of the word partner does not imply a partnership relationship between Cisco and any other company. (0206R)

Copyright © 2001–2003
Cisco Systems, Inc.
All rights reserved.