



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 4-12](#)—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 the UseSIP parameter is set to 1 (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.

Table 4-1 Required SIP Parameters and Defaults

Parameter	Value Type	Description	Voice Menu Access Code	Minimum Value	Maximum Value	Default
SIPRegInterval , page 5-15	Integer	Seconds between registration renewal	203	1	86400	3600
MAXRedirect , page 5-15	Integer	Maximum number of times to try redirection	202	0	10	5
SIPRegOn , page 5-16	Integer	Enable SIP registration	204	0	1	0
NATIP , page 5-16	IP address	WAN address of the attached router/NAT; currently only used to support SIP behind a NAT.	200	0	255	0.0.0.0
SIPPort , page 5-17	Integer	Port to listen for incoming SIP requests	201	1	65535	5060
MediaPort , page 5-17	Integer	Base port to receive RTP media; only used to support SIP behind a NAT	202	1	65535	16384
OutBoundProxy , page 5-17	Alphanumeric string	Proxy server for all outbound SIP requests. All SIP requests are sent to OutBoundProxy, when configured, instead of to the configured GkOrProxy.	206	—	—	0
GkOrProxy , page 5-10	Alphanumeric string	SIP proxy server address or registrar address.	5	—	—	0
UseSIP , page 5-14	Boolean	Set to 1 for SIP mode.	38	—	—	0
ToConfig , page 5-4	Boolean	Set to 0 after you have completed configuration of the Cisco ATA. If this value remains at 1, the Cisco ATA will unnecessarily continue to contact the TFTP server.	80001	—	—	1

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 5-11
- [UID0](#), page 5-9
- [UID1](#), page 5-9

- [LoginID0, page 5-12](#)
- [LoginID1, page 5-12](#)
- [PWD0, page 5-9](#)
- [PWD1, page 5-10](#)

Setting the Codec

The `LBRCCodec` (low-bit-rate codec) parameter determines whether the G.723 or G.729A codec, in addition to G.711A-law and G.711 μ -law, can be used for receiving and transmitting. For configuration information, see the “[LBRCCodec](#)” section on page 5-20.

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 5-5.

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](#)
- [Comfort Noise During Silence Period When Using G.711, page 4-5](#)
- [Configurable Hook Flash Timing, page 4-5](#)
- [Configurable Mixing of Call Waiting Tone and Audio, page 4-5](#)
- [Configurable On-hook delay, page 4-5](#)
- [Debugging Diagnostics, page 4-5](#)
- [Dial Plan, page 4-6](#)
- [Disabling Access To The Web Interface, page 4-6](#)
- [Distinctive Ringing, page 4-6](#)
- [DNS SRV Support, page 4-6](#)
- [Hardware Information Display, page 4-7](#)
- [NAT Gateway, page 4-7](#)
- [NAT/PAT Translation, page 4-7](#)

- [Network Timing](#), page 4-8
- [OutBoundProxy Support](#), page 4-8
- [Progress Tones](#), page 4-8
- [Receiver-tagged VIA header](#), page 4-9
- [Repeat Dialing on Busy Signal](#), page 4-9
- [SIP Proxy Server Redundancy](#), page 4-10
- [Stuttering Dial Tone on Unconditional Call Forward](#), page 4-10
- [User Configurable Call Waiting Permanent Default Setting](#), page 4-10
- [User Configurable Timeout On No Answer for Call Forwarding](#), page 4-10
- [Setting Up and Placing a Call Without Using a SIP Proxy](#), page 4-11

Advanced Audio Configuration

The UDPTOS (specifies the default IP precedence of UDP packets) and AudioMode (audio operating mode) parameters allow you to tune audio configuration.

Related Parameters

[UDPTOS](#), page 5-32

[AudioMode](#), page 5-20

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 4-2 Billable Features and Related Parameters

Feature	Related Parameters
Call Conferencing	PaidFeatures , page 5-24, CallFeatures , page 5-23
Call Forwarding	PaidFeatures , page 5-24, CallFeatures , page 5-23, ConnectMode , page 5-28, SigTimer , page 5-32
Call Transfer	PaidFeatures , page 5-24, CallFeatures , page 5-23
Call Waiting	PaidFeatures , page 5-24, CallFeatures , page 5-23, SigTimer , page 5-32
Caller ID	PaidFeatures , page 5-24, CallFeatures , page 5-23, CallerIdMethod , page 5-25
Call Return	ConnectMode , page 5-28, PaidFeatures , page 5-24, CallFeatures , page 5-23
Polarity	Polarity , page 5-27
Voice Mail Indicator	PaidFeatures , page 5-24, CallFeatures , page 5-23



Note

CallWaitCallerID is an obsolete parameter. Do not use it.

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 5-20—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 5-32—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 5-28—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 5-26—Bits 8 to 12

Debugging Diagnostics

You can use the following parameters to troubleshoot operation issues:

- [NPrintf](#), page 5-36—Specify the IP address and port where debug information is sent.
- [TraceFlags](#), page 5-36—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 Parameter

[DialPlan](#), page 5-38

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 5-34—Bit 7

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 5-10
- [AltGk](#), page 5-13

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 5-7—Enter the LAN IP address of the NAT through which the Cisco ATA will communicate.
- [NATIP](#), page 5-16—Enter the WAN IP address of the NAT through which all external SIP user agents will communicate.
- [SIPPort](#), page 5-17—Enter a new port for SIP messages (optional).
- [MediaPort](#), page 5-17—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 5-19—Bits 0 to 11 are for specifying the retransmission period.
- [NatServer](#), page 5-18—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 5-30](#)—Use for time-stamping incoming calls (offset from Greenwich Mean Time) with local time.
- [NTPIP, page 5-30](#)—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 5-31](#)—Use to configure an alternate NTP server IP address.
- [ConnectMode, page 5-28](#)—Used to control the connection mode of the SIP protocol.

OutBoundProxy Support

If the OutBoundProxy 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 OutBoundProxy 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 OutBoundProxy is enabled, the Cisco ATA determines whether to retry to connect with the backup OutBoundProxy 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

- [OutBoundProxy, page 5-17](#)
- [AltGkTimeOut, page 5-13](#)

Progress Tones

Values for the following parameters (all defined in the “[Call-Progress Tone Parameters](#)” section on [page 5-42](#)) must be determined based on the country in which the Cisco ATA is located:

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

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 5-28—Bit 22

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 5-26—Bits 0 to 3 control the maximum time the Cisco ATA redials a number.
- [FeatureTimer](#), page 5-26—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 5-47—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."](#)

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 5-10](#)
- [AltGk, page 5-13](#)
- [AltGkTimeOut, page 5-13](#)

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.

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 5-28](#)—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:

```
<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 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 5-32](#)—Bits 20 to 25

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 4-11](#)
- [Placing an IP Call, page 4-12](#)

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 5-10](#)—Set to the value of 0 (zero).
 - [UID0, page 5-9](#)—Set to the unique telephone number of the **Phone 1** port of the Cisco ATA.
 - [UID1, page 5-9](#)—Set to the unique telephone number of the **Phone 2** port of the Cisco ATA.
 - [UseSIP, page 5-14](#)—Set to 1 to enable SIP mode.
 - [SIPRegOn, page 5-16](#)—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

```
<phone number>**<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#
```

Complete Reference Table of all Cisco ATA SIP Services

Table 4-3 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 4-3 Configurable Features and Related Parameters

Configurable Feature	Related Parameter
802.1Q packet tagging	VLANSetting , page 5-35
Audio compression and decompression	LBRCodec , page 5-20
Backup proxy configuration	AltGk , page 5-13
Backup proxy timeout	AltGkTimeOut , page 5-13
Call forward enable/disable	ConnectMode , page 5-28
Call forwarding—Maximum times allowed	MAXRedirect , page 5-15
Call commands	CallCmd , page 5-47, Chapter 6, “Call Commands”
Call features	CallFeatures , page 5-23
Caller ID format	CallerIdMethod , page 5-25
Call waiting	SigTimer , page 5-32
Call-waiting call ring timeout	FeatureTimer , page 5-26
Call-waiting state specified	ConnectMode , page 5-28
Cisco Discovery Protocol	OpFlags , page 5-34
CNG tone detection	AudioMode , page 5-20
Configuration update interval	CfgInterval , page 5-5
Debug messages—configuring host	NPrintf , page 5-36

Table 4-3 Configurable Features and Related Parameters (continued)

Configurable Feature	Related Parameter
Dial plan commands	DialPlan, page 5-38
Domain name server	DNS1IP, page 5-31
DNS hostname lookup	ConnectMode, page 5-28
DTMF method	AudioMode, page 5-20
Encryption	EncryptKey, page 5-6
Fax CED tone	AudioMode, page 5-20
Fax mode on a per-call basis	CallFeatures, page 5-23, PaidFeatures, page 5-24
Fax pass-through	AudioMode, page 5-20, ConnectMode, page 5-28
G.711 codec	AudioMode, page 5-20
Hook flash	AudioMode, page 5-20, SigTimer, page 5-32
IDs for phone lines	UID0, page 5-9, UID1, page 5-9
IP-like address in dial plan	IPDialPlan, page 5-38
Login ID	LoginID0, page 5-12, LoginID1, page 5-12
Low bit-rate codec	LBRCCodec, page 5-20
Mixing of tones	ConnectMode, page 5-28
Network Address Translation (NAT) server—Maintain during session	NatServer, page 5-18
NSE payload number	ConnectMode, page 5-28
NTP IP address	NATIP, page 5-16
On-hook delay	FeatureTimer, page 5-26
Outbound proxy	OutBoundProxy, page 5-17
Paid features	PaidFeatures, page 5-24
Passwords for phone lines	PWD0, page 5-9, PWD1, page 5-10
Polarity	Polarity, page 5-27
Polarity reversal before and after caller ID signal	CallerIdMethod, page 5-25
<i>Received</i> = tag enable/disable	ConnectMode, page 5-28
Receiving-audio codec preference	RxCodec, page 5-21
Redial time if line is busy	FeatureTimer, page 5-26
Refresh Cisco ATA using Web server	OpFlags, page 5-34
REGISTER messages	ConnectMode, page 5-28
Registration removal	ConnectMode, page 5-28
Reset Cisco ATA using Web server	OpFlags, page 5-34
Retransmission interval for NAT server	NatTimer, page 5-19

Table 4-3 Configurable Features and Related Parameters (continued)

Configurable Feature	Related Parameter
Retry interval if line is busy	FeatureTimer, page 5-26
Ringback tone—send to caller	ConnectMode, page 5-28
Ring-cadence pattern	RingOnOffTime, page 5-37
RTP media port	MediaPort, page 5-17
RTP packet size	NumTxFrames, page 5-22
RTP statistics	TraceFlags, page 5-36
Secondary domain name server	DNS2IP, page 5-31
Silence compression	AudioMode, page 5-20
SIP call return	ConnectMode, page 5-28
SIP mode	UseSIP, page 5-14
SIP proxy registrar address	GkOrProxy, page 5-10
SIP proxy registration renewal	SIPRegInterval, page 5-15
SIP registration enable/disable	SIPRegOn, page 5-16
SIP-request listening port	SIPPort, page 5-17
Static network router probe	OpFlags, page 5-34
TFTP file—not using internally generated name	OpFlags, page 5-34
Timeout values	SigTimer, page 5-32
Time zone offset	TimeZone, page 5-30
Tones: BusyTone, CallWaitTone AlertTone, DialTone, ReorderTone, and RingBackTone parameters	Call-Progress Tone Parameters, page 5-42
Tracing	TraceFlags, page 5-36
Transmitting-audio codec preference	TxCodec, page 5-22
UDP packet default IP precedence	UDPTOS, page 5-32
VLAN encapsulation	OpFlags, page 5-34
VLAN mode	OpFlags, page 5-34
WAN address of NAT	NATIP, page 5-16
Web configuration—disallowing	OpFlags, page 5-34

