



## Parameters and Defaults

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

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

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

The following list contains general configuration information:

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

- Bitmap value—unsigned hexadecimal integer (for specifying bits in a 32-bit integer)




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

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**Note** A tool called bitaid.exe is bundled with your Cisco ATA software. You can use this tool to help you configure values of Cisco ATA bitmap parameters. The tool prompts you for the necessary information.

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- Extended IP address—IP address followed by port number (for example, 192.168.2.170.9001)
- IP address (e.g. 192.168.2.170)
- Integer (32-bit integer)
- Numeric digit string




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

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**Note** This section contains recommended values for the United States and other countries as configuration examples for certain parameters. For detailed recommendations of tone-parameter values by country, see [Appendix E, “Recommended Cisco ATA Tone Parameter Values by Country.”](#)

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## Configuration Text File Template

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

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

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

```
#txt
UIPassword:0
UseTftp:1
TftpURL:0
cfgInterval:3600
EncryptKey:0
EncryptKeyEx:0
upgradecode:0,0x301,0x0400,0x0200,0.0.0.0,69,0,none
upgradelang:0,0x301,0x0400,0x0200,0.0.0.0,69,0,none
Dhcp:1
StaticIp:0
StaticRoute:0
StaticNetMask:0
DNS1IP:0.0.0.0
DNS2IP:0.0.0.0
NTPIP:0.0.0.0
AltNTPIP:0.0.0.0
```

```

VLANSetting:0x0000002b
GkOrProxy:0
AltGk:0
AltGkTimeOut:0
UID0:0
UID1:0
PWD0:0
PWD1:0
LoginID0:0
LoginID1:0
UseLoginID:0
SIPPort:5060
SIPRegOn:0
SIPRegInterval:120
MaxRedirect:5
SipOutBoundProxy:0
NATIP:0
NatServer:0
NatTimer:0
MsgRetryLimits:0x00000000
SessionTimer:0x00000000
SessionInterval:1800
MinSessionInterval:1800
DisplayName0:0
DisplayName1:0
MediaPort:16384
RxCCodec:1
TxCodec:1
LBRCCodec:0
AudioMode:0x00150015
NumTxFrames:2
TOS:0x0000A8B8
CallFeatures:0xffffffff
PaidFeatures:0xffffffff
CallCmd:Af;AH;BS;NA;CS;NA;Df;EB;Ff;EP;Kf;EFh;HH;Jf;AFh;HQ;I*67;gA*82;fA#90v#;OI;H#72v#;
bA#74v#;cA#75v#;dA#73;eA*67;gA*82;fA*70;IA*69;DA*99;xA;Uh;GQ;
FeatureTimer:0x00000000
FeatureTimer2:0x0000001e
SigTimer:0x01418564
ConnectMode:0x00060000
OpFlags:0x2
TimeZone:17
CallerIdMethod:0x00019e60
Polarity: 0
FXSInputLevel:-1
FXSOutputLevel:-4
DialTone:2,31538,30831,1380,1740,1,0,0,1000
BusyTone:2,30467,28959,1191,1513,0,4000,4000,0
ReorderTone:2,30467,28959,1191,1513,0,2000,2000,0
RingBackTone:2,30831,30467,1943,2111,0,16000,32000,0
CallWaitTone:1,30831,0,5493,0,0,2400,2400,4800
AlertTone:1,30467,0,5970,0,0,480,480,1920
SITone:2,30467,28959,1191,1513,0,2000,2000,0
RingOnOffTime:2,4,25
DialPlan:*St4-|#St4-|911|1>#t8.r9t2-|0>#t811.rat4-|^1t4>#.-
IPDialPlan: 1
NPrintf:0
TraceFlags:0x00000000
SyslogIP:0.0.0.0.514
SyslogCtrl:0x00000000

```

The sections that follow describe these parameters.

# User Interface (UI) Security Parameter

This parameter type contains one parameter—UIPassword.

## UIPassword

### Description

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

To clear a password, change the value to 0.

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



### Note

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

---

### Value Type

Alphanumeric string

### Range

Maximum nine characters

### Default

0

### Voice Configuration Menu Access Code

7387277

### Related Parameter

[OpFlags, page 5-45](#)

## Parameters for Configuration Method and Encryption

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

- [UseTFTP, page 5-5](#)
- [TftpURL, page 5-5](#)
- [CfgInterval, page 5-6](#)
- [EncryptKey, page 5-6](#)
- [EncryptKeyEx, page 5-7](#)

## UseTFTP

**Settings**

1—Use the TFTP server for Cisco ATA configuration.

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

**Value Type**

Boolean

**Range**

0 or 1

**Default**

1

**Voice Configuration Menu Access Code**

305

**Related Parameters**

- [TftpURL, page 5-5](#)
- [EncryptKey, page 5-6](#)
- [OpFlags, page 5-45](#) (bits 0 and 3)

## TftpURL

**Description**

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

Optionally, you can include the path prefix to the TFTP file to download.

For example, if the TFTP server IP address is 192.168.2.170 or www.cisco.com, and the path to download the TFTP file is in /ata186, you can specify the URL as 192.168.2.170/ata186 or www.cisco.com/ata186.

**Note**

---

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

---

**Value Type**

Alphanumeric string

**Range**

Maximum number of characters: 31

**Default**

0

**Voice Configuration Menu Access Code**

905

**Related Parameters**

- [UseTFTP, page 5-5](#)
- [CfgInterval, page 5-6](#)

## CfgInterval

**Description**

Use this parameter to specify the number of seconds between each configuration update. The Cisco ATA will also upgrade its signaling image if it detects that the TFTP server contains an upgraded image.

For example, when using TFTP for configuration, the Cisco ATA contacts TFTP each time the interval expires to get its configuration file.

You can set CfgInterval to a random value to achieve random contact intervals from the Cisco ATA to the TFTP server.

**Value Type**

Decimal

**Range**

60 to 4294967295

**Default**

3600

**Voice Configuration Menu Access Code**

80002

## EncryptKey

**Description**

This parameter specifies the encryption key that is used to encrypt the Cisco ATA configuration file on the TFTP server.

The cfgfmt tool, which is used to create a Cisco ATA binary configuration file (see the [“Using Encryption With the cfgfmt Tool” section on page 3-12](#)), automatically encrypts the binary file when the EncryptKey parameter has a value other than 0. The cfgfmt tool uses the rc4 encryption algorithm.

If this parameter value is set to 0, the Cisco ATA configuration file on the TFTP server is not encrypted.

**Note**


---

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

---

For examples on how to upgrade from the EncryptKey parameter to the stronger encryption method that uses the EncryptKeyEx parameter, see the [“Examples of Upgrading to Stronger Encryption Key” section on page 3-15](#).

**Value Type**

Hexadecimal string

**Range**

Maximum number of characters: 8

**Default**

0

**Voice Configuration Menu Access Code**

320

**Related Parameters**

- [UseTFTP](#), page 5-5
- [TftpURL](#), page 5-5
- [EncryptKeyEx](#), page 5-7

## EncryptKeyEx

**Description**

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

**Note**

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

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

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

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

**Note**

---

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

---

For examples on how to upgrade from the `EncryptKey` parameter to the stronger encryption method that uses the `EncryptKeyEx` parameter, see the “[Examples of Upgrading to Stronger Encryption Key](#)” section on page 3-15.

**Value Type**

Hexadecimal string of the form:

Rc4PasswdInHex/macInHex\_12

- rc4KeyInHex\_n is a hexadecimal string of one to 64 characters.
- /macInHex\_12 is the optional extension consisting of a forward slash (/) followed by the six-byte MAC address of the Cisco ATA to which the configuration file will be downloaded.

**Range**

Maximum number of characters: 64

**Default**

0

**Voice Configuration Menu Access Code**

Not applicable for this parameter.

**Related Parameters**

- [UseTFTP, page 5-5](#)
- [TftpURL, page 5-5](#)
- [EncryptKey, page 5-6](#)

## Network Configuration Parameters

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

- [DHCP, page 5-8](#)
- [StaticIp, page 5-9](#)
- [StaticRoute, page 5-9](#)
- [StaticNetMask, page 5-10](#)
- [DNS1IP, page 5-10](#)
- [DNS2IP, page 5-11](#)
- [NTPIP, page 5-11](#)
- [AltNTPIP, page 5-12](#)
- [VLANSetting, page 5-12](#)

## DHCP

**Description**

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

- 1—Enable DHCP
- 0—Disable DHCP



**Value Type**

Boolean

**Range**

0 or 1

**Default**

1

**Voice Configuration Menu Access Code**

20

**Related Parameters**

- [StaticIp](#), page 5-9
- [StaticRoute](#), page 5-9
- [StaticNetMask](#), page 5-10
- [OpFlags](#), page 5-45 (bits 3 and 11)

## StaticIp

**Description**

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

**Value Type**

IP address

**Default**

0.0.0.0

**Voice Configuration Menu Access Code**

1

**Related Parameters**

- [DHCP](#), page 5-8
- [StaticRoute](#), page 5-9
- [StaticNetMask](#), page 5-10

## StaticRoute

**Description**

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

**Value Type**

IP address

**Default**

0.0.0.0

**Voice Configuration Menu Access Code**

2

**Related Parameters**

- [DHCP, page 5-8](#)
- [StaticIp, page 5-9](#)
- [StaticNetMask, page 5-10](#)

## StaticNetMask

**Description**

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

**Value Type**

IP address

**Default**

255.255.255.0

**Voice Configuration Menu Access Code**

10

**Related Parameters**

- [DHCP, page 5-8](#)
- [StaticIp, page 5-9](#)
- [StaticRoute, page 5-9](#)

## DNS1IP

**Description**

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

**Value Type**

IP address

**Default**

0.0.0.0

**Voice Configuration Menu Access Code**

916

**Related Parameter**[DHCP, page 5-8](#)

## DNS2IP

**Description**

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

**Value Type**

IP address

**Default**

0.0.0.0

**Voice Configuration Menu Access Code**

917

**Related Parameter**[DHCP, page 5-8](#)

## NTPIP

**Description**

This parameter is the NTP IP address, required if DHCP server does not provide one.

The Cisco ATA requires an NTP Server from which to obtain Coordinated Universal Time (UTC) to time-stamp incoming calls (H.323 and SIP) to drive an external Caller-ID device.

DHCP may also supply a NTP server. If NTPIP is specified, it overwrites the value supplied by DHCP. NTPIP is ignored if its value is 0 or 0.0.0.0.

The user *must not* specify a port parameter. The Cisco ATA uses the default NTP port only.

**Value Type**

IP address

**Default**

0.0.0.0

**Voice Configuration Menu Access Code**

141

## AltNTPIP

**Description**

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

**Value Type**

IP address

**Default**

0.0.0.0

**Voice Configuration Menu Access Code**

142

## VLANSetting

**Description**

This parameter is for specifying VLAN-related settings.

Bitmap definitions are as follows for the VLANSetting parameter:

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

**Value Type**

Bitmap

**Default**

0x0000002b

**Voice Configuration Menu Access Code**

324

**Related Parameter**

[OpFlags, page 5-45](#)

# SIP Configuration Parameters

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

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

## GkOrProxy

### Description

This parameter is the proxy address or registrar address.

For a SIP proxy server, this can be an IP address with or without a port parameter such as 123.123.110.45, 123.123.110.45.5060, or 123.123.110.45:5061, or a URL such as sip.cisco.com, or sip.ata.cisco.com:5061. For an IP address, a '.' or ':' can be used to delimit a port parameter. For a URL, a ':' must be used to indicate a port.

The default port is port number 5060.

**Note**

If the SIP proxy server and registration server reside on separate hardware, enter the SIP registration server address in this field.

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

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

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

**Value Type**

Alphanumeric string

**Range**

Maximum number of characters: 31

**Default**

0—Disables proxy registration and proxy-routed calls.

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

The following list shows some examples of direct SIP IP dialing:

- 1234\*\*192\*168\*1\*10\*5060
- 102\*210\*9\*101\*5061
- 4084281002\*\*100\*123\*89\*10

**Voice Configuration Menu Access Code**

5

## AltGk

**Description**

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

**Value Type**

Alphanumeric string

**Range**

Maximum number of characters: 31

**Default**

0

**Voice Configuration Menu Access Code**

6

## AltGkTimeOut

**Description**

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

**Value Type**

Integer

**Default**

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

**Range**

30 to 4294967295 seconds

**Voice Configuration Menu Access Code**

251

## UID0

**Description**

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

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

**Value Type**

Alphanumeric string

**Range**

Maximum number of characters: 31

**Default**

0

**Voice Configuration Menu Access Code**

3

**Related Parameter**[LoginID0, page 5-17](#)

## PWDO

**Description**

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

**Value Type**

Alphanumeric string

**Range**

Maximum number of characters: 31

**Default**

0

**Voice Configuration Menu Access Code**

4

## UID1

**Description**

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

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

**Value Type**

Alphanumeric string

**Range**

Maximum number of characters: 31

**Default**

0

**Voice Configuration Menu Access Code**

13

**Related Parameter**[LoginID1, page 5-18](#)



## PWD1

**Description**

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

**Value Type**

Alphanumeric string

**Range**

Maximum number of characters: 31

**Default**

0

**Voice Configuration Menu Access Code**

14

## LoginID0

**Description**

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

**Note**

---

UID0 is used for authentication if UseLoginID is 0.

---

**Value Type**

Alphanumeric string

**Range**

Maximum number of characters: 51

**Default**

0

**Voice Configuration Menu Access Code**

46

**Related Parameters**

- [UID0, page 5-15](#)
- [UseLoginID, page 5-18](#)

## LoginID1

### Description

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



### Note

---

UID1 is used for authentication if UseLoginID is 0.

---

### Value Type

Alphanumeric string

### Range

Maximum number of characters: 51

### Default

0

### Voice Configuration Menu Access Code

47

### Related Parameters

- [UID1, page 5-16](#)
- [UseLoginID, page 5-18](#)

## UseLoginID

### Description

0—Use UID0 and UID1 as the authentication ID.

1—Use LoginID0 and LoginID1 as the authentication ID.

### Value Type

Boolean

### Range

0 or 1

### Default

0

### Voice Configuration Menu Access Code

93

## SIPPort

**Description**

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

**Value Type**

Integer

**Range**

1 to 65535

**Default**

5060

**Voice Configuration Menu Access Code**

201

## SIPRegInterval

**Description**

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

**Value Type**

Integer

**Range**

1 to 86400

**Default**

3600

**Voice Configuration Menu Access Code**

203

## SIPRegOn

**Description**

0—Disable SIP registration.

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

**Value Type**

Boolean

**Range**

0 or 1

**Default**

0

**Voice Configuration Menu Access Code**

204

## MAXRedirect

**Description**

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

**Value Type**

Integer

**Range**

0 to 10

**Default**

5

**Voice Configuration Menu Access Code**

205

## SipOutBoundProxy

### Description

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

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

The OutBoundProxy parameter can be an IP address with or without a port parameter, such as 123.123.110.45, 123.123.110.45.5060, or 123.123.110.45:5061, or a URL such as sip.cisco.com, sip.ata.cisco.com:5061. For IP addresses, a period (.) or colon (:) can be used to delimit a port parameter. For a URL, a colon (:) must be used to indicate a port. If no port parameter is specified, the port 5060 is assumed.

### Value Type

Alphanumeric string

### Range

Maximum number of characters: 31

### Default

0

### Voice Configuration Menu Access Code

206

## NATIP

### Description

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

### Value Type

IP address

### Default

0.0.0.0

### Voice Configuration Menu Access Code

200

## NatServer

### Description

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

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

### Syntax

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

The following syntax rules apply:

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

### Value Type

IP address or FQDN format

### Range

Maximum number of characters: 47

### Default

5060 is the default port if no port is specified.

### Voice Configuration Menu Access Code

207

## NatTimer

### Description

The NatTimer parameter provides the following configuration:

- Interval of Keep-Alive packets
- STUN mode to use
- Destination of keep-alive packets.

See [Table 5-1](#) for definitions of each bit.

### Value Type

Bitmap

### Default

0x00000000

## Voice Configuration Menu Access Code

208

Table 5-1 NatTimer Parameter Bit Definitions

Bit Number	Definition
0-11	<p>Use these bits to specify the interval in seconds for sending keep-alive packets from the SIP port and RTP port.</p> <p>Keep-alive packets are sent in the following manner:</p> <ul style="list-style-type: none"> <li>• The packets are sent constantly for the SIP port once the keep-alive interval is reached.</li> <li>• The packets are sent only during a call for the media ports once the keep-alive interval is reached.</li> </ul> <p>Range: 0 to 2047 seconds</p> <p>Default: 0 (no keep-alive packets are sent).</p>
12-15	Reserved. These bits must be set to 0.
16-17	<p>Use these bits with one of the following values to specify NAT traversal mode:</p> <ul style="list-style-type: none"> <li>• 0—Send keep-alive packets only <ul style="list-style-type: none"> <li>The Cisco ATA sends keep-alive packets only and does not perform any discovery or substitution of the NAT WAN IP address or port in SIP messages.</li> </ul> </li> <li>• 1—STUN <ul style="list-style-type: none"> <li>The Cisco ATA obtains IP address and port mappings from the STUN server specified in the NatServer parameter and substitutes these mappings into SIP messages. The Cisco ATA automatically obtains port mappings, as needed, for the following parameters: <ul style="list-style-type: none"> <li>– SIPPort</li> <li>– MediaPort</li> <li>– MediaPort + 2</li> <li>– MediaPort + 4</li> </ul> </li> </ul> </li> <li>• 2—NATIP auto-mapping <ul style="list-style-type: none"> <li>The Cisco ATA obtains only the IP address mapping from the STUN server specified in the NatServer parameter and substitutes the mapping into SIP messages. With this mode, you must manually map the Cisco ATA port to an external port.</li> </ul> </li> <li>• 3—Reserved. Do not use this value.</li> </ul>
18	<p>Use this bit to set the destination of keep-alive UDP packets that are sent from the SIP port:</p> <ul style="list-style-type: none"> <li>• 0—Keep-alive UDP packets are sent to the server specified in the NatServer parameter. This setting is normally used when only keep-alive packets are sent (see bits 16-17).</li> <li>• 1—Keep-alive UDP packets are sent to the server specified in the GkOrProxy parameter. This setting is normally used when STUN mode is selected (see bits 16-17).</li> </ul> <p><b>Related Parameters</b></p> <ul style="list-style-type: none"> <li>• <a href="#">NatServer</a>, page 5-22</li> <li>• <a href="#">GkOrProxy</a>, page 5-13</li> </ul>
19	Reserved.

Table 5-1 NatTimer Parameter Bit Definitions

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

## MsgRetryLimits

### Description

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

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

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

- REGISTER
- INVITE
- BYE



- CANCEL
- REFER
- NOTIFY

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

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

See [Table 5-2](#) for definitions of each bit.

#### Value Type

Bitmap

#### Default

0x00000000

#### Voice Configuration Menu Access Code

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

Bit Number	Definition
0-3	<p>Number of times to retransmit the following Cisco ATA SIP requests or responses to the SIP user agent:</p> <ul style="list-style-type: none"> <li>• NOTIFY request</li> <li>• Response to an OPTIONS request</li> <li>• Final response to an INVITE request</li> </ul> <p>Range: 0-15</p> <p>Default: 0. This means the following numbers of retransmission attempts are used:</p> <ul style="list-style-type: none"> <li>• NOTIFY request: 6</li> <li>• OPTIONS response: 10</li> <li>• INVITE final response: 7</li> </ul>
4-7	<p>Number of times to retransmit REGISTER request.</p> <p>Default: 0 (Number of attempts is 10.)</p>
8-11	<p>Number of times to retransmit INVITE request.</p> <p>Default: 0 (Number of attempts is five.)</p>
12-15	<p>Number of times to retransmit BYE request.</p> <p>Default: 0 (Number of attempts is four.)</p>
16-19	<p>Number of times to retransmit CANCEL request.</p> <p>Default: 0 (Number of attempts is four.)</p>
20-23	<p>Number of times to retransmit REFER request.</p> <p>Default: 0 (Number of attempts is five.)</p>
24-31	<p>Reserved. The value of these bits must be 0.</p>

# SessionTimer

**Description**

Use this parameter to configure various options of the SIP session timer. See [Table 5-3](#) for bit definitions.

**Value Type**

Bitmap

**Default**

0x00000000

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**Examples**

- SessionTimer: 0x00000001

With this setting, the Cisco ATA accepts requests from the far end for a session timer but never initiates requests for a session timer.

By default, the Cisco ATA will not be the session refresher except if one of the following conditions is true:

- If the caller has already specified the Cisco ATA as the refresher in the initial call setup request.
- If the party that the Cisco ATA called has specified that the Cisco ATA be the refresher.

- SessionTimer: 0x00000003

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

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

By default, the Cisco ATA will not be the session refresher except if one of the following conditions is true:

- If the caller has already specified the Cisco ATA as the refresher in the initial call setup request.
- If the party that the Cisco ATA called has specified that the Cisco ATA be the refresher.

- SessionTimer: 0x00000005

With this setting, the Cisco ATA only accept requests from the far end for a session timer but never initiates requests for a session timer.

By default, the Cisco ATA will not be the session refresher except if one of the following conditions is true:

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

- SessionTimer: 0x00000007

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

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


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

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

#### Related Parameters

- [SessionInterval](#), page 5-28
- [MinSessionInterval](#), page 5-28

**Table 5-3 SessionTimer Parameter Bit Definitions**

Bit Number	Definition
0	Set this bit to 1 to enable the SIP session-timer feature.
1	Set this bit to 1 to request that a SIP session timer be in effect on every call if the far endpoint also supports this feature.
2	Use this bit to select the party responsible for initiating session refreshes if the caller has not specified the party. This field is applicable only when the Cisco ATA is the callee. Set the bit value as follows: <ul style="list-style-type: none"> <li>• 0—The caller will perform session refreshes.</li> <li>• 1—The callee will perform session refreshes.</li> </ul>
	 <p><b>Note</b> If the caller specifies which party will perform session refreshes (by means of the SIP Session-Expires field of the initial INVITE request), that selection takes precedence over the configured value of this bit.</p>
3	Reserved.
4	Set this bit to 1 to enable the collection of session debug messages.
5-31	Reserved.

## SessionInterval

### Description

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

### Conditions for Usage

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

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

### Value Type

Integer

### Range

60 to 86400 seconds

### Default

1800 seconds (one half hour)

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### Related Parameters

- [SessionTimer](#), page 5-26
- [MinSessionInterval](#), page 5-28

## MinSessionInterval

### Description

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

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

If the Session-Expires value in the INVITE request sent to the Cisco ATA is smaller than the value of the MinSessionInterval parameter, the Cisco ATA will reject the INVITE request and return a *422 Session Interval Too Small* message to the caller.

If the Session-Expires value in the received INVITE request is greater than or equal to the Cisco ATA MinSessionInterval, the Cisco ATA will accept this interval by sending a *200 OK* message to the caller with the refresh interval listed in the Session-Expires field and the *Required:timer* line included in the Cisco ATA response.

**Value Type**

Integer

**Range**

60 to 86400 seconds

**Default**

1800

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364

**Related Parameters**

- [SessionTimer](#), page 5-26
- [SessionInterval](#), page 5-28

## DisplayName0

**Description**

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

**Value Type**

Alphanumeric string

**Range**

Maximum 31 characters

**Default**

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

**Voice Configuration Menu Access Code**

5002

## DisplayName1

**Description**

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

**Value Type**

Alphanumeric string

**Range**

Maximum 31 characters

**Default**

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

**Voice Configuration Menu Access Code**

5003

## Audio Configuration Parameters

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

- [MediaPort](#), page 5-30
- [RxCodec](#), page 5-31
- [TxCodec](#), page 5-31
- [LBRCodec](#), page 5-32
- [AudioMode](#), page 5-32
- [NumTxFrames](#), page 5-34
- [TOS](#), page 5-34

## MediaPort

**Description**

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

**Value Type**

Integer

**Range**

1 to 65535

**Default**

16384

**Voice Configuration Menu Access Code**

202

**Related Parameters**

- [TOS, page 5-34](#)
- [VLANSetting, page 5-12](#)

## RxCodec

**Description**

Use this parameter to specify receiving-audio codec preference. The following values are valid:

- 0—G.723 (can be selected only if LBRCCodec is set to 0)
- 1—G.711A-law
- 2—G.711 $\mu$ -law
- 3—G.729a (can be selected only if LBRCCodec is set to 3)
- 6—G.726-32kbps

**Value Type**

Integer

**Range**

0-3, 6

**Default**

2

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## TxCodec

**Description**

Use this parameter to specify the transmitting-audio codec preference. The following values are valid:

- 0—G.723 (can be selected only if LBRCCodec is set to 0)
- 1—G.711A-law
- 2—G.711 $\mu$ -law
- 3—G.729A (can be selected only if LBRCCodec is set to 3)
- 6—G.726-32kbps

**Value Type**

Integer

**Range**

0-3, 6

**Default**

2

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37

## LBRCodec

**Description**

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

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

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

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

The following values are valid:

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

**Value Type**

Integer

**Range**

0, 3 or 6

**Default**

0

**Voice Configuration Menu Access Code**

300

## AudioMode

**Description**

This parameter represents the audio operating mode. The lower 16 bits are for the **Phone 1** port, and the upper 16 bits are for the **Phone 2** port. [Table 5-4 on page 5-33](#) provides definitions for each bit.



**Value Type**

Bitmap

**Default**

0x00150015

**Voice Configuration Menu Access Code**

312

**Related Parameter**[ConnectMode, page 5-41](#)**Table 5-4 AudioMode Parameter Bit Definitions**

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

## NumTxFrames

**Description**

Use this parameter to select the default RTP packet size in number of frames per packet. The Cisco ATA default frame sizes are as follows:

- G.711 and G.729—10 ms
- G.723.1—30 ms

For example, to receive 20 ms of G.729 packets, set the parameter to 2.

**Value Type**

Integer

**Range**

1-6

**Default**

2

**Voice Configuration Menu Access Code**

35

## TOS

**Description**

This parameter allows you to configure Type of Service (ToS) bits by specifying the precedence and delay of audio and signaling IP packets, as follows:

- Bits 0-7—These bits are for the ToS value for voice data packets.  
Range: 0-255  
Default: 184
- Bits 8-15—These bits are for the ToS value for signaling-data packets  
Range: 0-255  
Default: 168
- Bits 16-31—Reserved.

**Value Type**

Bitmap

**Default**

0x0000A8B8

**Voice Configuration Menu Access Code**

255

**Note**

This parameter is called UDPTOS in previous Cisco ATA releases. If you are performing a Cisco ATA upgrade, the previous value of the UDPTOS parameter is carried forward to the TOS parameter.

## Operational Parameters

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

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

## CallFeatures

### Description

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

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

**Note**

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

### Value Type

Bitmap

### Default

0xffffffff

### Voice Configuration Menu Access Code

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**Table 5-5 CallFeatures Parameter Bit Definitions**

Bit Number	Definition
0 and 16	Forward unconditionally
1 and 17	Forward on busy
2 and 18	Forward on no answer
3 and 19	Caller Line Identification Restriction (CLIR)—0; Caller Line Identification Presentation (CLIP)—1
4 and 20	Call waiting
5 and 21	three-way calling
6 and 22	Blind transfer
7 and 23	Transfer with consultation. This service allows the user to transfer the remote party to a different number by first calling that number and consulting with the callee.
8 and 24	Caller ID. This service enables the Cisco ATA 186 to generate a Caller ID signal to drive a Caller ID display device attached to the FXS line.
9 and 25	Call return
10 and 26	Message waiting indication
11 and 27	Call Waiting Caller ID. This is available only if the Method bit in CallerIdMethod is set to Bellcore (FSK).
12-14 and 28-30	Reserved.
15 and 31	Fax mode. This service allows the user to set the Cisco ATA to Fax mode on a per-call basis. For Fax mode, use the following settings: <ul style="list-style-type: none"> <li>• G711 codec only</li> <li>• No silence suppression</li> <li>• No FAX tone detection</li> </ul>

## PaidFeatures

### Description

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

### Value Type

Bitmap

### Default

0xffffffff

### Voice Configuration Menu Access Code

315

**Table 5-6 PaidFeatures Parameter Bit Definitions**

Bit Number	Definition
0 and 16	Forward unconditionally
1 and 17	Forward on busy
2 and 18	Forward on no answer
3 and 19	Caller Line Identification Restriction_Caller Line Identification Presentation (CLIP_CLIR)
4 and 20	Call waiting
5 and 21	three-way calling
6 and 22	Blind transfer
7 and 23	Transfer with consultation. This service allows the user to transfer the remote party to a different number by first calling that number and consulting with the callee.
8 and 24	Caller ID. This service enables the Cisco ATA 186 to generate a Caller ID signal to drive a Caller ID display device attached to the FXS line.
9 and 25	Call return
10 and 26	Message waiting indication
11 and 27	Call Waiting Caller ID. This is available only if the Method bit in CallerIdMethod is set to Bellcore (FSK).
12-14 and 28-30	Reserved.
15	Fax mode. This service allows the user to set the Cisco ATA to Fax mode on a per-call basis. For Fax mode, use the following settings: <ul style="list-style-type: none"> <li>• G.711 codec only</li> <li>• No silence suppression</li> <li>• No FAX tone detection</li> </ul>

## CallCmd

### Description

Command table that controls call commands such as turning on/off caller ID.

For detailed information on the CallCmd parameter, see [Chapter 6, “Call Commands.”](#)

### Value Type

Alphanumeric string

### Range

Maximum of 248 characters

**Default**

US command table:

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

**Voice Configuration Menu Access Code**

930

## FeatureTimer

**Description**

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

- Bits 0-3—Maximum time to spend redialing if line is busy
  - Range: 0 - 15
  - Factor: five-minute increments
  - Values: 0 - 75 minutes
  - Default: 0 (equals 30 minutes)
- Bits 4-7—Retry interval if line is busy
  - Range: 0 - 15
  - Factor: 15-second increments
  - Values: 0 - 225 seconds
  - Default: 0 (equals 15 seconds)
- Bits 8-12—On-hook delay before a call is disconnected. This feature works only when the Cisco ATA is the terminating endpoint of the call. The user can hang up the phone in one room and pick up the phone in another room without disconnecting the line.
  - Range: 0 - 31
  - Factor: five-second increments
  - Values: 0 - 155 seconds
  - Default: 0 (no delay)
- Bits 13-15—Amount of time the Cisco ATA waits for a "486 Busy Here" response from a PSTN gateway after receiving a "183 Session Progress" response.
  - Range: 0 - 7
  - Factor: one-second increments
  - Values: 0 to 7 seconds
  - Default: 0 (no waiting)
- Bits 16-18—Configurable call waiting ring timeout. When a call arrives for a Cisco ATA port that is in use and has call-waiting enabled, the Cisco ATA plays a call-waiting tone. If the incoming call is not answered within a specified period of time, the Cisco ATA can reject the call by returning a "486 Busy" response to the remote user agent.

You can configure FeatureTimer parameter bits 16-18 to specify the ringing period for incoming call-waiting calls.

This feature can be disabled by either using the default value 0 or by setting bits 16-18 to a value greater than the standard timeout for an incoming call as specified in SigTimer parameter bits 14-19. When this feature is disabled, a "480 Temporarily Not Available" response is returned to the remote user agent when the standard ring times out.

- Range: 0 - 7
- Factor: 10-second increments
- Values: 0 to 70 seconds
- Default: 0 (never timeout)
- Bits 19-31—Reserved.

**Value Type**

Bitmap

**Default**

0x00000000

**Voice Configuration Menu Access Code**

317

## FeatureTimer2

**Description**

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

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

**Note**

---

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

---

**Value Type**

Bitmap

**Default**

0x0000001e

**Voice Configuration Menu Access Code**

361

## SigTimer

**Description**

This parameter controls various timeout values. [Table 5-7](#) contains bit definitions of this parameter.

**Value Type**

Bitmap

**Default**

0x01418564

**Voice Configuration Menu Access Code**

318

**Related Parameter**

[CallWaitTone](#), page 5-62

**Table 5-7 SigTimer Parameter Bit Definitions**

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



**Table 5-7 SigTimer Parameter Bit Definitions (continued)**

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

## ConnectMode

### Description

This parameter is a 32-bit bitmap used to control the connection mode of the selected call signaling protocol. [Table 5-8](#) provides bit definitions for this parameter.

### Value Type

Bitmap

### Default

0x00060400

### Voice Configuration Menu Access Code

311

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

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

Table 5-8 ConnectMode Parameter Bit Definitions (continued)


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

Table 5-8 ConnectMode Parameter Bit Definitions (continued)



Bit Number	Definition
22—SIP only	<p>0/1—Disable/enable SIP to process a <i>received=</i> tag in the VIA header to extract the external IP addresses used by the Network Address Translation (NAT) router.</p> <p>When the Cisco ATA is operating behind a NAT, the NATIP parameter must be set to the external IP address of the NAT router. This allows the correct IP address to be placed in the Contact and SDP headers.</p> <p>In release 2.14 or later, you may leave the NAT IP address at the default value of "0" or "0.0.0.0" and let the ATA automatically scan the Via header for a "received=" parameter. The parameter, if present, would indicate that the Cisco ATA is operating behind a firewall.</p> <p>The Cisco ATA proceeds as follows:</p> <ol style="list-style-type: none"> <li>1. If the "received=" parameter is in an INVITE response, the current INVITE is canceled and a new INVITE is sent with the new IP address extracted from the "received=&lt;NAT IP address&gt;" parameter in the Contact and SDP headers.</li> </ol> <p>In addition, the Cisco ATA will cancel all previous registrations and re-register with the new IP address in the Contact header. This step is performed only if registration is currently in an idle state.</p> <ol style="list-style-type: none"> <li>2. If the "received=" parameter is in a REGISTER response as a result of a REGISTER command, the Cisco ATA will cancel all previous registrations and re-register with the new IP address extracted from the "received=&lt;NAT IP address&gt;" parameter in the Contact header.</li> </ol> <p>Default: 0</p> <hr/> <p> <b>Note</b> For the Cisco ATA to automatically detect its presence behind a NAT, the SIP proxy server or remote user agent server <i>must</i> include the "received=" parameter in the Via header in the responses to the Cisco ATA if the proxy detects that the source address and port do not match those in the Via header.</p>
23	<p>0/1—Disable/enable user-configurable setting for call-waiting default.</p> <p>If this value is 0 (default), the end user cannot configure the permanent default call-waiting setting for every call. Instead, the service provider's default call-waiting setting is used for every call.</p> <p>If this value is 1, the end-user can configure the permanent default call-waiting setting for every call, thus overriding the value set by the service provider.</p> <p>Default: 0</p>
24	<p>0/1—Disable/enable the mixing of audio and call waiting tone during a call.</p> <p>Default: 0</p>
25	<p>0—Both the call-waiting call and active call will be dropped the instant the call-waiting user hangs up the phone.</p> <p>1—This setting enables the call-waiting hang-up alert feature, which alerts call-waiting users that an active call is still on hold once the user has hung up from another call. The phone will continue to ring until the party on hold has hung up.</p> <hr/> <p> <b>Note</b> For the call-waiting hang-up feature to work, the call-waiting call command sequence must be set appropriately. For more information on how to set this sequence, see the <a href="#">“Enabling the Call-Waiting Hang-Up Alert Feature”</a> section on page 4-6.</p> <hr/> <p>Default: 0</p>

Table 5-8 ConnectMode Parameter Bit Definitions (continued)

Bit Number	Definition
26	Reserved.
27	<p>0—The Cisco ATA does not change the user name to <i>Anonymous</i> in the SIP URL of the FROM and CONTACT headers in any INVITE requests even when the display name is sent as <i>Anonymous</i>. Also, the Cisco ATA does not change the user name to <i>Anonymous</i> in the <i>o=</i> line (also called the <i>origin</i> line) of the Session Description Protocol (SDP) header in SIP INVITE requests.</p> <p>1—The Cisco ATA exhibits the following behavior, provided that the Caller Line Identification Restriction (CLIR) feature is enabled:</p> <ul style="list-style-type: none"> <li>• The Cisco ATA uses a user name of <i>Anonymous</i> in the following SIP headers and requests: <ul style="list-style-type: none"> <li>– FROM header</li> <li>– CONTACT header</li> <li>– The <i>o=</i> line of the SDP header in SIP INVITE requests</li> </ul> </li> </ul> <p>For more information and example requests showing the usage of the <i>Anonymous</i> user name, see the <a href="#">“Anonymous User Name Support for SIP INVITE Requests”</a> section on page 4-11.</p> <ul style="list-style-type: none"> <li>• The Diversion header in the INVITE request contains the <i>privacy=[full off]</i> field, and this field will have the value of <i>full</i>. For more information, see the <a href="#">“Privacy Token Support for SIP Diversion Header”</a> section on page 4-12.</li> </ul> <p>Default: 0</p> <p><b>Enabling the CLIR feature</b></p> <p>CLIR is enabled in one of two ways:</p> <ul style="list-style-type: none"> <li>• By setting bit 3 of both the CallFeatures and PaidFeatures parameters to 0. (For the <b>Phone2</b> port of the Cisco ATA, you would set bit 19 to 0 for each parameter.) For more information on these parameters, see the <a href="#">“CallFeatures”</a> section on page 5-35 and the <a href="#">“PaidFeatures”</a> section on page 5-36.</li> <li>• By enabling CLIR on a per-call basis using the call command dial string. For more information, see <a href="#">Chapter 6, “Call Commands.”</a> If CLIR is enabled in the call command string, this takes precedence over the setting of bits 3 or 19 of the CallFeatures and PaidFeatures parameters.</li> </ul>
28	<p>0/1—Disable/enable conference warning-tone setting.</p> <p>If you set this value to 1, the Cisco ATA, when acting as a conference bridge, sends a three-party-conference warning tone to all parties in the conference call.</p> <p>Default: 0</p> <p><b>Specifications of the Warning Tone:</b></p> <ul style="list-style-type: none"> <li>• Cadence: 0.33 seconds in duration; played every 15 seconds</li> <li>• Duration: The full duration of the conference call</li> <li>• Frequency: 1400 Hz</li> <li>• Level: -33 dBm to -53 dBm</li> </ul>

Table 5-8 ConnectMode Parameter Bit Definitions (continued)

Bit Number	Definition
29	<p>0—The Cisco ATA performs only a DNS A-record query on the domain name.</p> <p>1—The Cisco ATA first performs a DNS SRV query on the domain name. If the response is empty, the Cisco ATA then performs a DNS A-record query on the domain name.</p> <p>Default: 0</p> <p><b>Related Parameters</b></p> <ul style="list-style-type: none"> <li>• <a href="#">GkOrProxy, page 5-13</a></li> <li>• <a href="#">AltGk, page 5-14</a></li> </ul>
30	<p>0—The Cisco ATA does not change the user name to <i>Anonymous</i> in the TO header of any INVITE requests.</p> <p>1—The Cisco ATA replaces all occurrences of a user <i>ID</i> in the TO header in all INVITE requests with <i>Anonymous</i>, regardless of whether the calling party has enabled or disabled the Caller Line Identification Restriction (CLIR) feature.</p> <p>Default: 0</p>
31	Reserved

## OpFlags

### Description

This parameter enables/disables various operational features.

See [Table 5-9 on page 5-46](#) for bit definitions of this parameter.

### Value Type

Bitmap

### Default

0x2

### Voice Configuration Menu Access Code

323


### Related Parameters

- [TftpURL, page 5-5](#)
- [DHCP, page 5-8](#)
- [VLANSetting, page 5-12](#)

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

Bit Number	Definition
0	<p>If Bit 0 = 0, the TFTP configuration filename supplied by the DHCP server overwrites the default filename for each Cisco ATA.</p> <p>If Bit 0 = 1, the default Cisco ATA filename is always used.</p> <p>Default: 0</p>
1	<p>If Bit 1 = 0, the Cisco ATA probes the static network router during the power-up process.</p> <p>If Bit 1 = 1, static network router probing is disabled.</p> <p>Default: 1</p>
2	Reserved.
3	<p>If Bit 3=1, the Cisco ATA does not request DHCP option 150 in the DHCP discovery message; some DHCP server do not respond if option 150 is requested.</p> <p>Default: 0</p>
4	<p>If Bit 4 = 1, the Cisco ATA uses the VLAN ID specified in the VLANSetting parameter for VLAN IP encapsulation (see the “<a href="#">VLANSetting</a>” section on page 5-12).</p> <p>Default: 0</p>
5	<p>If Bit 5=1, the Cisco ATA does not use VLAN IP encapsulation.</p> <p>Default: 0</p>
6	<p>If Bit 6=1, the Cisco ATA does not perform CDP discovery.</p> <p>Default: 0</p>
7	<p>If Bit 7=1, the Cisco ATA does not allow web configuration. Once the web server is disabled, you must configure the Cisco ATA with the TFTP or voice configuration menu methods.</p> <p>Default: 0</p> <p><b>Examples</b></p> <ol style="list-style-type: none"> <li>If the existing OpFlags value is 0x2, select menu option 323 from the voice configuration menu and enter the value 130 (0x82). This disables web configuration. If you later attempt to access the Cisco ATA web configuration page, the following error messages will be displayed. <ul style="list-style-type: none"> <li>Netscape: The document contained no data. Try again later, or contact the server's administrator.</li> <li>Internet Explorer: The page cannot be displayed.</li> </ul> </li> <li>If the existing OpFlags value is 0x82, select menu option 323 from the voice configuration menu and enter the value 2 (0x2). This disables web configuration.</li> </ol>
8	<p>If Bit 8=1, the Cisco ATA does not allow HTTP refresh access with the http://ip/refresh command.</p> <p>Default: 0</p>
9	<p>If Bit 9=1, the Cisco ATA does not allow HTTP reset access with the http://ip/reset command.</p> <p>Default: 0</p>
10	Reserved.

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

Bit Number	Definition
11	<p>If Bit 11=0, the Cisco ATA requests the device hostname from the DHCP server.</p> <p>If Bit 11=1, the Cisco ATA uses the device hostname that is specified in DHCP option 12.</p> <p>Default: 0</p>
12	Reserved.
13	<p>DNS Servers For Name Resolution</p> <p>If Bit 13=0 (default), use statically configured DNS IP addresses, if available, for name resolution. If statically configured DNS servers are not available, use DHCP-provided DNS IP addresses for name resolution.</p> <p>If Bit 13=1, use both statically configured DNS IP addresses and as many as two DHCP-provided DNS IP addresses. Therefore, the Cisco ATA can query as many as four DNS IP addresses in one DNS query.</p> <p>Default: 0</p> <p>For more information about statically configured DNS IP addresses, see the <a href="#">“DNS1IP” section on page 5-10</a> and the <a href="#">“DNS2IP” section on page 5-11</a>.</p>
14	<p>DNS Servers For Name Resolution 2</p> <p>If Bit 14=0 (default), use statically configured DNS IP addresses (DNS1IP and DNS2IP), if available, for name resolution; otherwise, use DHCP-provided DNS IP addresses.</p> <p>If Bit 14=1, use both statically configured (DNS1IP and DNS2IP) and DHCP-provided DNS IP addresses (maximum of two) for name resolution.</p> <p> <b>Note</b> This configuration bit gives precedence to statically provided DNS IP addresses over DHCP-provided DNS IP addresses. This bit also overrides the value of OpFlags parameter bit 13.</p>
15	<p>Disable UDP Checksum Generation</p> <p>If Bit 13=0, generate UDP checksum in outgoing UDP packets.</p> <p>If Bit 13=1, disable generation of of UDP checksum in outgoing UDP packets.</p> <p>Default: 0</p>
16	<p>Set this bit to 1 for voice prompt confirmation for the following services:</p> <ul style="list-style-type: none"> <li>• Call Forward All Enable</li> <li>• Call Forward Busy Enable</li> <li>• Call Forward No Answer Enable</li> </ul> <p>For more information, see the <a href="#">“Voice Prompt Confirmation for Call Waiting and Call Forwarding” section on page 4-20</a>.</p>
17	<p>Set this bit to 1 for voice prompt confirmation for the following services:</p> <ul style="list-style-type: none"> <li>• Call Forward All Disable</li> <li>• Call Forward Busy Disable</li> <li>• Call Forward No Answer Disable</li> </ul> <p>For more information, see the <a href="#">“Voice Prompt Confirmation for Call Waiting and Call Forwarding” section on page 4-20</a>.</p>

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

Bit Number	Definition
18	Set this bit to 1 for voice prompt confirmation for the following service: <ul style="list-style-type: none"> <li>• Call Waiting Enable</li> </ul> For more information, see the <a href="#">“Voice Prompt Confirmation for Call Waiting and Call Forwarding”</a> section on page 4-20.
19	Set this bit to 1 for voice prompt confirmation for the following service: <ul style="list-style-type: none"> <li>• Call Waiting Disable</li> </ul> For more information, see the <a href="#">“Voice Prompt Confirmation for Call Waiting and Call Forwarding”</a> section on page 4-20.
20-27	Reserved.
28-31	To configure the Cisco ATA to prompt the user for the UIPassword when the user attempts to perform a factory reset or upgrade using the voice configuration menu, configure bits 28 to 31 with the value of 6. Any other value for these bits means that the Cisco ATA will not prompt the user for the UIPassword in these cases. The default value for these bits is 0.

## TimeZone

### Description

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

Local time is generated by the following formula:

- Local Time=GMT + TimeZone, if TimeZone <= 12
- Local Time=GMT + TimeZone - 25, if TimeZone > 12

### Value Type

Integer

### Range

0-24

### Default

17

### Voice Configuration Menu Access Code

302

### Additional Description

Use the following list to select Timezone offset (in minutes) from GMT for the following cities and countries that have 30-minute-factor and 45-minute-factor time zone offsets. These values are integers and can range from -720 through -60, and from 60 through 780.

- Tehran—210 = GMT + 3:30



- Kabul— 270 = GMT + 4:30
- Calcutta, Chennai, Mumbai, and New Delhi—330 = GMT + 5:30
- Kathmandu—345 = GMT + 5:45
- Rangoon—390 = GMT + 6:30
- Darwin and Adelaide—570 = GMT + 9:30
- Newfoundland— -210 = GMT - 3:30

**Note**

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

## Telephone Configuration Parameters

This section includes the following parameters, which allow you to configure items such as generating caller ID format and controlling line polarity:

- [CallerIdMethod](#), page 5-49
- [Polarity](#), page 5-51
- [FXSInputLevel](#), page 5-52
- [FXSOutputLevel](#), page 5-52

## CallerIdMethod

### Description

This 32-bit parameter specifies the signal format to use for both FXS ports for generating Caller ID format. Possible values are:

- Bits 0-1 (method)—0=Bellcore (FSK), 1=DTMF, 2=ETSI, and 3 is reserved.

If *method*=0 (default), set the following bits:

- Bit 2—Reserved.
- Bit 3 to 8—Maximum number of digits in phone number (valid values are 1 to 20; default is 12).
- Bit 9 to 14—Maximum number of characters in name (valid values are 1 to 20; default is 15).
- Bit 15—If this bit is enabled (it is enabled by default), send special character **O** (out of area) to CID device if the phone number is unknown.
- Bit 16—If this bit is enabled (it is enabled by default), send special character **P** (private) to CID device if the phone number is restricted.
- Bits 17 to 27—Reserved.

**Note**

The Cisco ATA supports the Bellcore FSK method to turn on/off the visual message waiting indicator (VMWI) on a phone when the Cisco ATA receives MWI messages from a server. The Bellcore FSK VMWI is enabled automatically if the *CallerIdMethod* parameter is configured to use the Bellcore

method and if the PaidFeatures and CallFeatures parameters (bits 11 and 27) are configured to enable message waiting indication. (See the “PaidFeatures” section on page 5-36 and the “CallFeatures” section on page 5-35.)

If *method=1*, set the following bits:

- Bit 2—Reserved.
- Bits 3-6—Start digit for known numbers (valid values are **12** for “A,” **13** for “B,” **14** for “C,” and **15** for “D.”)
- Bits 7-10—End digit for known numbers (valid values are **11** for “#,” **12** for “A,” **13** for “B,” **14** for “C,” and **15** for “D.”)
- Bits 11—Polarity reversal before and after Caller ID signal (value of 0/1 disables/enables polarity reversal)
- Bits 12-16—Maximum number of digits in phone number (valid values are 1 to 20; default is 15).
- Bits 17 to 19—Start digit for unknown or restricted numbers (valid values are **4** for “A,” **5** for “B,” **6** for “C,” and **7** for “D.”)
- Bits 20 to 22—End digit for unknown or restricted numbers (valid values are **3** for “#,” **4** for “A,” **5** for “B,” **6** for “C,” and **7** for “D.”)
- Bits 23 to 24—Code to send to the CID device if the number is unknown (valid values are **0** for “00,” **1** for “0000000000,” and **2** for “3.” **3** is reserved and should not be used.
- Bits 25 to 26—Code to send to the CID device if the number is restricted (valid values are **0** for “10,” and **1** for “1.” **2** and **3** are reserved and should not be used.
- Bits 27 to 31—Reserved.

If *method=2*, set the following bits:

- Bit 2—Set to 0 to have the Cisco ATA transmit data prior to ringing by using the Ring-Pulse Alerting Signal (RP-AS); set to 1 to have the Cisco ATA transmit data after the first ring.
- Bits 3-8—Maximum number of digits in a phone number (valid values are 1 to 20; default is 12).
- Bits 9-14—Maximum number of characters in a name (valid values are 1 to 20; default is 15).
- Bit 15—If this bit is enabled (it is enabled by default), send special character **O** (out of area) to CID device if telephone number is unknown.
- Bit 16—If this bit is enabled (it is enabled by default), send special character **P** (private) to CID device if telephone number is restricted.
- Bits 17-27 are reserved.

**Examples**

The following examples are recommended values for the CallerID Method parameter:

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

**Value Type**

Bitmap

**Default**

0x00019e60

**Voice Configuration Menu Access Code**

316

## Polarity

**Description**

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

- Bit 0: CALLER\_CONNECT\_POLARITY. Polarity to use when the Cisco ATA is the caller and the call is connected.
  - 0 =Use forward polarity (Default)
  - 1 =Use reverse polarity
- Bit 1: CALLER\_DISCONNECT\_POLARITY. Polarity to use when the Cisco ATA is the caller and the call is disconnected.
  - 0 =Use forward polarity (Default)
  - 1 =Use reverse polarity
- Bit 2: CALLEE\_CONNECT\_POLARITY. Polarity to use when the Cisco ATA is the callee and the call is connected.
  - 0 =Use forward polarity (Default)
  - 1 =Use reverse polarity
- Bit 3: CALLEE\_DISCONNECT\_POLARITY. Polarity to use when the Cisco ATA is the callee and the call is disconnected.
  - 0 =Use forward polarity (Default)
  - 1 =Use reverse polarity

**Note**

Bits 4-31 are reserved.

**Value Type**

Bitmap

**Default**

0x00000000

**Voice Configuration Menu Access Code**

304

## FXSInputLevel

**Description**

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

**Value Type**

Integer

**Range**

-9 to 2 dB

**Default**

-1

**Voice Configuration Menu Access Code**

370

**Related Parameter**[FXSOutputLevel, page 5-52](#)

## FXSOutputLevel

**Description**

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

**Value Type**

Integer

**Range**

-9 to 2 dB

**Default**

-4

**Voice Configuration Menu Access Code**

371

**Related Parameter**[FXSInputLevel](#), page 5-52

## Tone Configuration Parameters

The Cisco ATA supports the following tone parameters:

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

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

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

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

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

- [RingOnOffTime](#), page 5-64

**Note**

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

## Tone Parameter Syntax—Basic Format

Each tone is specified by nine integers, as follows:

*parametername*: *NumOfFreqs*, *Tfreq1*, *Tfreq2*, *Tamp1*, *Tamp2*, *Steady*, *OnTime*, *OffTime*, *TotalToneTime*

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

$$32767 * \cos(2 * \pi * F / 8000)$$

where *F* is the desired frequency in Hz. Set this value to 0 if the frequency does not exist.

The range of each value is -32768 to 32767.

For negative values, use the 16-bit 2's complement value. For example, enter **-1** as 65535 or as 0xffff.

- *Tamp1* and *Tamp2* are the transformed amplitudes of the first and second frequencies, respectively. Their values are calculated with the following formulas:

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

$$A \text{ (amplitude factor)} = 0.5 * 10^{((k+10-(n-1)*3)/20)}$$

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

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

- *OnTime* controls the length of time the tone is played in milliseconds (ms).

Specify each value as a number of samples with a sampling rate of 8 kHz. The range of each value is 0 to 0xffff. For example, for a length of 0.3 seconds, set the value to 2400.

- *OffTime* controls the length of time between audible tones in milliseconds (ms).

Specify each value as a number of samples with a sampling rate of 8 kHz. The range of each value is 0 to 0xffff. For example, for a length of 0.3 seconds, set the value to 2400.

- *TotalToneTime* controls the length of time the tone is played. If this value is set to 0, the tone will play until another call event stops the tone. For *DialTone*, *DialTone2*, *BusyTone*, *ReorderTone*, and *RingBackTone*, the configurable value is the number of 10 ms (100 = 1 second) units.

For the remaining tones, the configurable value is the number of samples with a sampling rate of 8 kHz.


**Note**


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

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## Tone Parameter Syntax—Extended Formats

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

- [Extended Format A, page 5-55](#)—This format can be used for the following tone parameters:
  - *DialTone*
  - *BusyTone*
  - *RingbackTone*
  - *CallWaitTone*
  - *AlertTone*
- [Extended Format B, page 5-56](#)—This format can be used for the following tone parameters:
  - *ReorderTone*
  - *SITone*

## Extended Format A

Each tone is specified by 11 integers, as follows:

*parametername*:*NumOfFreqs*,*Tfreq1*,*Tamp1*,*Tfreq2*,*Tamp2*,*NumOfOnOffPairs*,*OnTime1*,  
*OffTime1*,*OnTime2*,*OffTime2*,*TotalToneTime*

- *parametername* is the name of the tone.
- *NumOfFreqs* = 100 + the number of frequencies in the tone. (Therefore, *NumOfFreqs* = 101 for one frequency, and 102 for two frequencies.)
- *Tfreq1* and *Tfreq2* are the transformed frequencies of the first and second frequencies, respectively. Their values are calculated with the following formula:

$$32767 * \cos(2 * \pi * F / 8000)$$

where *F* is the desired frequency in Hz. Set this value to 0 if the frequency does not exist.

The range of each value is -32768 to 32767.

For negative values, use the 16-bit 2's complement value. For example, enter -1 as 65535 or as 0xffff.

- *Tamp1* and *Tamp2* are the transformed amplitudes of the first and second frequencies, respectively. Their values are calculated with the following formula:

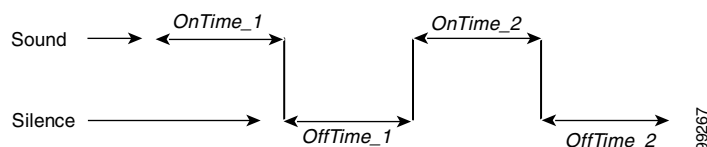
$$32767 * A * \sin(2 * \pi * F / 8000)$$

$$A \text{ (amplitude factor)} = 0.5 * 10^{((k+10-(n-1)*3)/20)}$$

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

- *NumOfOnOffPairs* is the number of on-off pairs in the cadence of the tone. Valid values are 0, 1 and 2. Use 0 if the tone is steady.
- *OnTime1* and *OnTime2* values are the lengths of time the tone is played for the first and second on-off pairs of a cadence, respectively. (See [Figure 5-1](#) for a graphical representation.) Specify each value as a number of samples with a sampling rate of 8 kHz. The range of each value is 0 to 0xffff. For example, for a length of 0.3 seconds, set the value to 2400.
- *OffTime1* and *OffTime2* values are the lengths of time that silence is played for the first and second on-off pairs of a cadence, respectively. (See [Figure 5-1](#) for a graphical representation.) Specify each value as a number of samples with a sampling rate of 8 kHz. The range of each value is 0 to 0xffff. For example, for a length of 0.3 seconds, set the value to 2400.

**Figure 5-1 Cadence With Two On-Off Pairs**



- *TotalToneTime* controls the length of time the tone is played. If this value is set to 0, the tone will play until another call event stops the tone. For *DialTone*, *DialTone2*, *BusyTone*, *ReorderTone*, and *RingBackTone*, the configurable value is the number of 10 ms (100 = 1 second) units.

For the remaining tones, the configurable value is the number of samples with a sampling rate of 8 kHz.

**Note**

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

**Extended Format B**

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

- Up to three frequencies played simultaneously and a cadence of up to three on-off pairs. The first on-off pair can repeat multiple times before the second on-off pair plays.  
For example, a 400 Hz frequency plays four times for 0.75 second followed by 0.1 second of silence after each play and then plays one time for 0.75 second followed by 0.4 second of silence. This pattern can be set to repeat until another call event stops the pattern.
- Up to three frequencies played sequentially with a cadence of up to three on-off pairs  
For example, the frequencies 900 Hz, 1400 Hz, and 1800 Hz play sequentially for 0.33 seconds each with no silence after the first and second frequencies but one second of silence after the third frequency.

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

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

where:

- *parameter* is either *ReorderTone* or *SITone*.
- *Sequential* specifies whether multiple frequencies in a tone play simultaneously (100) or sequentially (101). Set to 100 for a tone with one frequency. If *Sequential* is 101, the number of frequencies (*NumOfFreqs*) has to be the same value as the number of on-off pairs in a cadence (*NumOfOnOffPairs*).
- *NumOfFreqs* is the number of frequencies in the tone (1, 2, or 3). The frequencies can play simultaneously or sequentially, depending on the *Sequential* setting.
- *TFreq1*, *TFreq2*, and *TFreq3* are the transformed frequencies of the first, second, and third frequencies, respectively. Calculate each value with the following formula:

$$32767 * \cos(2 * \pi * F/8000)$$

where *F* is the desired frequency in Hz. Set this value to 0 if the frequency does not exist.

The range of each value is -32768 to 32767.

For negative values, use the 16-bit 2's complement value. For example, enter -1 as 65535 or as 0xffff.

- *Tamp1*, *Tamp2* and *Tamp3* are the transformed amplitudes of the first, second and third frequencies, respectively. Their values are calculated with the following formula:

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

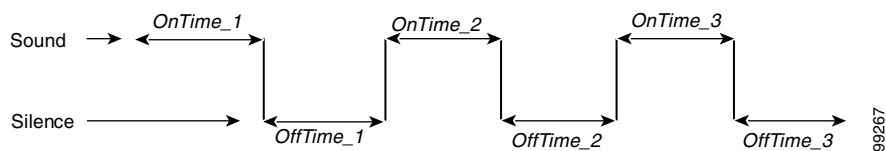
$$A \text{ (amplitude factor)} = 0.5 * 10^{((k+10-(n-1)*3)/20)}$$

where *F* is the desired frequency in Hz, *k* is the desired volume in dBm, and *n* is the number of frequencies (If *Sequential* is set to 101, *n* is equal to 1). The ^ symbol means *to the order of*.



- *NumOfOnOffPairs* is the number of on-off pairs in the cadences of the tone (0, 1, 2, or 3). For a steady tone, use 0.  
If this value is 0, the *OnTime1*, *OnTime2*, *OnTime3*, *OffTime1*, *OffTime2*, and *OffTime3* values must also be 0.
- *OnTime1*, *OnTime2*, and *OnTime3* are the lengths of time that the first, second, and third on-off pairs of a cadence play a sound, respectively. (See [Figure 5-2](#) for a graphical representation.)  
Specify each value as a number of samples with the sampling rate of 8 kHz. The range of each value is 0 to 0xffff.  
For example, for a length of 0.3 seconds, set a value to 2400.
- *OffTime1*, *OffTime2*, and *OffTime3* are the lengths of silence after the sound of the first, second, and third on-off pairs of a cadence, respectively.  
Specify each value as a number of samples with the sampling rate of 8 kHz. The range of each value is 0 to 0xffff.  
For example, for a length of 0.3 seconds, set a value to 2400. (See [Figure 5-2](#) for a graphical representation.)

**Figure 5-2 Cadence with Three On-Off Pairs**



- *NumOfRepeats* is the number of times that the first on-off pair of the cadence (specified by *OnTime1*, *OffTime1*) repeats before the second on-off pair (specified by *OnTime2*, *OffTime2*) plays.  
For example, if *NumOfRepeats* is 2, the first on-off pair will play three times (it will play once and then repeat two times), then the second on-off pair will play.
- *TotalToneTime* is the total length of time that the tone plays. If this value is 0, the tone will play until another call event stops the tone.  
This value is in 10 ms units (100 ms = 1 second).

Two examples of Extended Format B, both using the Reorder tone, follow.

#### ReorderTone Parameter Example1

Assume that you want a reorder tone in which:

- The frequencies 900 Hz, 1400 Hz, and 1800 Hz play sequentially.
- Each frequency plays once for 0.33 seconds.
- There is no silence after the first and the second frequencies.
- There is 1 second of silence after the third frequency (before the first frequency starts again)
- The volume of each frequency is -19 dBm.
- The tone plays until another call event stops the tone.

For this reorder tone, make the following setting. See [Table 5-10](#) for a detailed explanation.

```
ReorderTone:101,3,24917,3405,14876,4671,5126,5178,3,2640,0,2640,0,
2640,8000,0,0
```

**Table 5-10 Reorder Tone Parameter Example 1 Explanation**

Component	Setting	Explanation
Sequential	101	Frequencies play sequentially
NumOfFreqs	3	Three frequencies in the tone
TFreq1	24917	First frequency is 900 Hz
TAmp1	3405	First frequency volume is -19 dBm
TFreq2	14876	Second frequency is 1400 Hz
TAmp2	4671	Second frequency volume is -19 dBm
TFreq3	5126	Third frequency is 1800 Hz
TAmp3	5178	Third frequency volume is -19 dBm
NumOfOnOffPairs	3	Three on-off pairs in the cadence of the tone
OnTime1	2640	Sound in first on-off pair plays for 0.33 seconds
OffTime	0	No silence after the first sound (the second sound plays immediately)
OnTime2	2640	Sound in second on-off pair plays for 0.33 seconds
OffTime2	0	No silence after the second sound (the third sound plays immediately)
OnTime3	2640	Sound in third on-off pair plays for 0.33 seconds
OffTime3	8000	1 second of silence after the sound in the third on-off pair (before the pattern repeats, beginning with the first on-off pair)
NumOfRepeats	0	First on-off pair of the cadence plays once (does not repeat), then the second on-off pair plays
TotalToneTime	0	Tone plays continuously (set of three on-off pairs of the cadence repeat continuously) until another call event stops the tone

**ReorderTone Parameter Example 2**

Assume that you want a reorder tone in which:

- The only frequency is 400 Hz.
- The frequency plays six times, each time for 0.1 second followed by 0.9 second of silence.
- The frequency then plays once for 0.3 second followed by 0.7 second of silence.
- The volume of the frequency is -19 dBm.
- The tone plays until another call event stops the tone.

For this reorder tone, make the following setting. See [Table 5-11](#) for a detailed explanation.

```
ReorderTone:100,1,31164,1620,0,0,0,0,2,800,7200,2400,5600,0,0,5,0
```

**Table 5-11 Reorder Tone Parameter Example 2 Explanation**

Component	Setting	Explanation
Sequential	100	Required setting for a tone with one frequency
NumOfFreqs	1	One frequency in the tone
TFreq1	31164	First frequency is 400 Hz
TAmp1	1620	First frequency volume is -19 dBm
TFreq2	0	No second frequency
TAmp2	0	No second frequency
TFreq3	0	No third frequency
TAmp3	0	No third frequency
NumOfOnOffPairs	2	Two on-off pairs in the cadence of the tone
OnTime1	800	Sound in first on-off pair plays for 0.1 second
OffTime1	7200	Sound in first on-off pair is followed by 0.9 second of silence
OnTime2	2400	Sound in second on-off pair plays for 0.3 seconds
OffTime2	5600	Sound in second on-off pair is followed by 0.7 second of silence
OnTime3	0	No third on-off pair in the cadence
OffTime3	0	No third on-off pair in the cadence
NumOfRepeats	5	First on-off pair of the cadence plays six times (plays once and then repeats five times), then the second on-off pair plays
TotalToneTime	0	Tone plays continuously (set of two on-off pairs of the cadence repeat continuously) until another call event stops the tone

## Recommended Values

The following settings are recommended for the US:

- DialTone = "2,31538,30831,1380,1740,1,0,0,1000" (approximately -17 dBm)
- BusyTone = "2,30467,28959,1191,1513,0,4000,4000,0" (approximately -21 dBm)
- ReorderTone = "2,30467,28959,1191,1513,0,2000,2000,0" (approximately -21 dBm)
- RingBackTone = "2,30831,30467,1943,2111,0,16000,32000,0" (approximately -16 dBm)
- CallWaitTone = "1,30831,0,5493,0,0,2400,2400,4800" (approximately -10 dBm)
- AlertTone = "1,30467,0,5970,0,0,480,480,1920"
- SITone = "2,30467,28959,1191,1513,0,2000,2000,0" (approximately -21 dBm)



### Note

For detailed recommendations of tone-parameter values by country, see [Appendix E, "Recommended Cisco ATA Tone Parameter Values by Country."](#)

## Specific Tone Parameter Information

Brief descriptions, and lists of default values and the voice configuration menu code for each Cisco ATA tone parameter, appear in the following sections:

- [DialTone](#), page 5-60
- [BusyTone](#), page 5-61
- [ReorderTone](#), page 5-61
- [RingbackTone](#), page 5-62
- [CallWaitTone](#), page 5-62
- [AlertTone](#), page 5-63
- [SITone](#), page 5-63

### DialTone

#### Description

The Cisco ATA plays the dial tone when it is ready to accept the first digit of a remote address to make an outgoing call.

#### Default values (using the Basic format)

- NumOfFreqs—2
- Tfreq1—31538
- Tfreq2—30831
- Tamp1—1380
- Tamp2—1740
- Steady—1
- OnTime—0
- OffTime—0
- TotalToneTime—1000

#### Voice Configuration Menu Access Code

920

## BusyTone

### Description

The Cisco ATA plays the busy tone when the callee is busy.

### Default values (using the Basic format)

- NumOfFreqs—2
- Treq1—30467
- Tfreq2—28959
- Tamp1—1191
- Tamp2—1513
- Steady—0
- OnTime—4000
- OffTime—4000
- TotalToneTime—0

### Voice Configuration Menu Access Code

921

## ReorderTone

### Description

The Cisco ATA plays the reorder tone (also known as congestion tone) if the outgoing call failed for reasons other than busy. This is a fast-busy tone.

### Default values (using the Basic format)

- NumOfFreqs—2
- Treq1—30467
- Treq2—28959
- Tamp1—1191
- Tamp2—1513
- Steady—0
- OnTime—2000
- OffTime—2000
- TotalToneTime—0

### Voice Configuration Menu Access Code

922

## RingbackTone

### Description

The Cisco ATA plays the ring-back tone when the callee is being alerted by the called device.

### Default values (using the Basic format)

- NumOfFreqs—2
- Tfreq1—30831
- Tfreq2—30467
- Tamp1—1943
- Tamp2—2111
- Steady—0
- OnTime—16000
- OffTime—32000
- TotalToneTime—0

### Voice Configuration Menu Access Code

923

## CallWaitTone

### Description

The Cisco ATA plays the call-waiting tone when an incoming call arrives while the user is connected to another party.

### Default values (using the Basic format)

- NumOfFreqs—1
- Tfreq1—30831
- Tfreq2—0
- Tamp1—5493
- Tamp2—0
- Steady—0
- OnTime—2400
- OffTime—2400
- TotalToneTime—4800

### Voice Configuration Menu Access Code

924

## AlertTone

### Description

The Cisco ATA plays the alert tone as a confirmation tone that a special event, such as call forwarding, is in effect.

### Default values (using the Basic format)

- NumOfFreqs—1
- Tfreq1—30467
- Tfreq2—0
- Tamp1—5970
- Tamp2—0
- Steady—0
- OnTime—480
- OffTime—480
- TotalToneTime—1920

### Voice Configuration Menu Access Code

925

## SITone

### Description

The Cisco ATA plays the SITone (special information tone) if the Cisco ATA receives a *404 (Not Found)* response, which indicates that the called party cannot be located on the network.

### Default values (using the Basic format)

- NumOfFreqs—2
- Tfreq1—30467
- Tfreq2—28959
- Tamp1—1191
- Tamp2—1513
- Steady—0
- OnTime—2000
- OffTime—2000
- TotalToneTime—0

### Voice Configuration Menu Access Code

926

## RingOnOffTime

### Description

This parameter specifies the ringer cadence pattern, expressed as a triplet of integers “a,b, and c”.

- a—Number of seconds to turn the ring ON.
- b—Number of seconds to turn the ring OFF.
- c—The ring frequency, fixed at 25.

### Value Type

List of three integer values, separated by commas

### Range

1-65535

### Default

2, 4, 25

### Recommended Values:

- United States — 2,4,25
- Sweden — 1,5,25

### Voice Configuration Menu Access Code

929

## Dial Plan Parameters

This section describes the configurable parameters related to dial plans:

- [DialPlan](#), page 5-64
- [DialPlanEx](#), page 5-72
- [IPDialPlan](#), page 5-72

## DialPlan

### Description

The programmable dial plan is designed for the service provider to customize the behavior of the Cisco ATA for collecting and sending dialed digits. The dial plan allows the Cisco ATA user to specify the events that trigger the sending of dialed digits. These events include the following:

- The termination character has been entered.
- The specified dial string pattern has been accumulated.
- The specified number of dialed digits has been accumulated.
- The specified inter-digit timer has expired.



**Value Type**

Alphanumeric string

**Range**

Maximum number of characters is 199.

**Note**

If the dial plan exceeds 199 characters, use the DialPlanEx parameter instead of the DialPlan parameter. For more information, see the “DialPlanEx” section on page 5-72.

**Default**

```
*St4-!#St4-!91111!>#t8.r9t2-!0>#t811.rat4-!^1t4>#.-
```

**Voice Configuration Menu Access Code**

926

**Additional DialPlan Information**

The DialPlan section contains the following additional topics that describe commands and rules for creating your own dial plan, and includes many examples:

- [Dial Plan Commands, page 5-65](#)
- [Dial Plan Rules, page 5-66](#)
- [Dial Plan Examples, page 5-70](#)

## Dial Plan Commands

The following list contains commands that can be used to create you own dial plans:

- . —Wildcard, match any digit entered.
- - —Additional digits can be entered. This command can be used only at the end of a dial plan rule (for example, 1408t5- is legal usage of the - command, but 1408t5-3... is illegal).
- [ ]—Range, which means to match any single digit in the list. Use an underscore ( \_ ) to indicate a range of digits. For example, [135] matches the digits 1, 3, and 5. Also, [1\_5] matches the digits 1, 2, 3, 4 and 5. The pound key (#) and asterisk (\*) are not allowed in the Range command. Also, the Repeat (rn) command does not apply to range, and range cannot include the Subrule matching command.
- (subrule0| subrule1| ...|subruleN)—Subrule matching. Using the ( ) and | operators allows you to specify multiple subrules within a dial plan rule so that a subrule match is reached if the entered digits fit one of the subrules. This can be used to reduce the length of the desired dial plan rule by concatenating the group of the subrules with the common rule.

For example, a dial plan rule of (1900|1800|17..)555.r3 or three dial plan rules of 1900555.r3|1800555.r3|17..555.r3 are equivalent. A match is reached if 11 digits are entered and the first three digits are either 1900, 1800, or 17..., and the fifth, sixth, and seventh digits are all 5.

- >#—Defines the # character as a termination character. When the termination character is entered, the dial string is automatically sent. The termination character can be entered only after at least one user-entered digit matches a dial plan rule. Alternatively, the command >\* can be used to define \* as the termination character.
- tn— Defines the timeout value **n**, in the unit of seconds, for the interdigit timer. Valid values are 0-9 and a-z, where a-z indicates a range of 10 to 35.

- *rn*—Repeat the last pattern *n* times, where *n* is 0-9 or a-z. The values a-z indicate a range of 10 to 26. Use the repeat modifier to specify more rules in less space.




---

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

---

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




---

**Note**

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

---




---

**Note**

No syntax check is performed by the actual implementation. The administrator has the responsibility of making sure that the dial plan is syntactically valid.

---

## Dial Plan Rules

The Cisco ATA supports the following dial plan rules:

- [\(In Rule\) for Dial Plan Blocking, page 5-66](#)
- [‘H’ Rule to Support Hot/Warm Line, page 5-67](#)
- [‘P’ Rule to Support Dial Prefix, page 5-67](#)
- [‘B’ Rule for Base Number, page 5-68](#)
- [‘R’ Rule for Enhanced Prefix, page 5-68](#)
- [‘C’ Rule for Call Blocking, page 5-69](#)
- [‘F’ Rule for Call Forwarding Blocking, page 5-69](#)
- [‘X’ Rule for Call Blocking and Call Forwarding Blocking, page 5-70](#)
- [‘D’ Rule for Displaying Caller ID, page 5-70](#)

### (In Rule) for Dial Plan Blocking

Dial plan blocking can be used to reduce the occurrences of invalid dialed digits being sent and can prevent the dialed string of a specified pattern from being sent. By adding dial plan blocking, dialed digits are discarded after the interdigit timer expires unless one of the specified matching rules is met.

In addition, the default nine-second global interdigit timeout value is also modified with the value specified in the dial plan blocking command:

#### Syntax

*In*

where  $n$  specifies the global interdigit timeout and the valid values are 1-9 and a-z (10-35).

### Example

```
Ic| 911
```

This command specifies an interdigit timeout of 12 seconds, and will discard dialed digits unless 911 is entered.

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

## 'H' Rule to Support Hot/Warm Line

Hotline/Warmline, also known as Private Line Automatic Ringdown (PLAR), is a line used for priority telephone service. If the Hotline feature is configured, the Cisco ATA immediately dials a pre-configured number as soon as the handset goes off hook. If the Warmline feature is configured, the Cisco ATA dials a pre-configured number if no digits were entered before the specified timer value expired when the handset went offhook.

### Syntax

```
Hdnnnn
```

where  $d$  is a delay-in-seconds parameter 0-9,a-z (to support 0 to 35 seconds delay), and  $nnnn$  is the variable-length phone number to call when no digits are entered for  $d$  seconds after offhook.

### Example 1

```
H05551212
```

This is a hotline configuration; the Cisco ATA immediately dials 555-1212 when the handset goes off hook.

### Example 2

```
H55551212
```

This is a warmline configuration; the Cisco ATA waits for five seconds and dials 555-1212 if no digits were entered when the handset went off hook.

## 'P' Rule to Support Dial Prefix

This rule is for automatic pre-pending the dial string as entered by the user with a specified prefix.

### Syntax

```
Ptnnnn
```

where  $t$  is a single leading trigger character; if  $t$  is the *first* entered digit when making a new call, it triggers the prepending of a variable-length prefix (as specified by  $nnnn$ ) in the dial string. The  $t$  character can take one of the following values:

0-9,\*,#, 'n' (= any of 1-9), 'N' (any of 'n' and 0), 'a' (any of 'n',\* and #), or 'A' (any of 'a' and 0);

### Example

```
Pn12345
```

This rule prepends 12345 to the dial string when the first entered digit is any of 1-9. The triggered digit is not removed from the dial string.

### 'B' Rule for Base Number

A base number is the first part of a phone number that the Cisco ATA dials when Rule 'B' is used in a dial plan. The telephone number that the Cisco ATA calls consists of the base number followed by a two-digit extension.

Rule 'B' is used for voice prompt confirmation for the call-waiting and call-forwarding features. For more information, see the ["Voice Prompt Confirmation for Call Waiting and Call Forwarding"](#) section on page 4-20.

To set a base number, use 'B' followed by the desired base number. For example, if the desired base number is 1234, you would add the rule 'B1234' to your dial plan.

#### Syntax

*Bnnnn*

where *nnnn* is the base number.

#### Example

If the administrator has configured a base number of 1234 and call forward on busy (extension 03) is enabled, the called number is 123403.

### 'R' Rule for Enhanced Prefix

This enhanced prefix rule matches entire strings, whereas the 'P' rule matches only a single digit. The 'R' rule is for automatically prepending a specified prefix to the dialed string. The string must be an exact match to trigger the rule. If more than one 'R' rule matches, the first matched 'R' rule is triggered.

The 'R' rule also uses negation to exclude one or more leading digits before prepending the defined prefix string.

The number of dashes (-) after the *R* represents the number of leading digits that will be removed preceding the prefix.

#### Syntax

*Rnnnn (tttt)*

where *tttt* is a trigger string. If the dialed numbers match this string, this match triggers the prepending of a variable-length prefix (as specified by *nnnn*) to the dial string. The triggered string is not removed from the dial string. The negation, subrule matching and range patterns can be applied to the trigger strings.

#### Example 1

*R1212 ([2\_9]-)*

This rule prepends 1212 to dial strings that have a leading digit of 2 to 9.



#### Note

Note: 'R' rules can replace most 'P' rules; for example, *Pn12345* is the same as *R12345([1\_9]-)*.

#### Example 2

*R-0033 (0[1-9].x7)*

This removes the first dialed digit, then prepends 0033 to the dialed string. For example, if the number 0148336134 is dialed, the resulting string becomes 0033148336134.

**Example 3**

```
R----0(0033[1-9].r7)
```

This removes the first four dialed digits, then prepends 0 to the dialed string. For example, if the number 0033148336134 is dialed, the resulting string becomes 0148336134.

**Log Information**

The *Call Prefix* <prefix>+<num> is shown in the prserv log.

**'C' Rule for Call Blocking**

This rule is for blocking call numbers.

**Syntax**

```
Cnnnn
```

where *nnnn* is the leading set of digits of the blocked call number; *nnnn* can be composed with subrule matching and range. The rule is triggered when the leading digits of a dialed string match the string *nnnn*.

The 'C' rule does not work with negation.

**Example:**

```
C1900|C1888 or C(1900|1888)
```

This rule blocks call numbers beginning with 1900 or 1888.

**Log Information**

The *Call Block* <num> is shown in the prserv log, and a busy tone is being played.

**'F' Rule for Call Forwarding Blocking**

This rule is for blocking call forwarding numbers.

**Syntax**

```
Fnnnn
```

where *nnnn* is the leading set of digits of the blocked call forwarding number; *nnnn* can be composed with subrule matching and range. The rule is triggered when the leading digits of a dialed forwarding number match the string *nnnn*. The 'F' rule does not work with negation.

**Example:**

```
F1900|F1888 or F(1900|1888)
```

These rules block call forwarding numbers beginning with 1900 or 1888.

**Log Information**

The *CFWD Block*:<num> is shown in the prserv log, and a busy tone is being played.

## 'X' Rule for Call Blocking and Call Forwarding Blocking

This rule is for blocking call numbers and call forwarding numbers.

### Syntax

*Xnnnn*

where *nnnn* is the leading set of digits of the blocked call number and blocked call forwarding number; *nnnn* can be composed with subrule matching and range. The rule is triggered when the leading set of digits of a dialed call number or forwarding number match the string *nnnn*. The 'F' rule does not work with negation.

### Example

`x1900|x1888` or `x(1900|1888)`

This rule blocks the call numbers and call forwarding numbers beginning with 1900 or 1888.

## 'D' Rule for Displaying Caller ID

This rule is for displaying caller ID at the remote site. The number must be an exact match to trigger the rule.

### Syntax

*Dnnnn*

where *nnnn* is the callee number. The caller ID will show to the callee; *nnnn* automatically becomes a valid calling number. Also, *nnnn* can be composed with negation, subrule matching and range. The 'D' rule is checked before the 'R' and 'P' rules.

### Example

`D911`

This rule shows the caller ID at the remote side when if the call number is 911.

### Log Information

`SCC Cmd[:CLIP` or `CLIP:<num>` are shown in the prserv log.

## Dial Plan Examples

This section contains three dial plan examples that use many different rules and commands.

### Dial Plan Example 1 (Default Dial Plan)

The following dial plan:

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

consists of the following rules:

- **\*St4**—If the first digit entered is \*, all other dial plan rules are voided. Additional digits can be entered after the initial \* digit, and the timeout before automatic dial string send is four seconds.
- **#St4**—Same as above, except with # as the initial digit entered.
- **911**—If the dial string 911 is entered, send it immediately.

- **1>#t8.r9t2**—If the first digit entered is 1, the timeout before automatic send is eight seconds. The terminating character # can be entered at any time to manually send the dial string. After the 11th digit is entered, the timeout before an automatic send changes to two seconds. The user can enter more digits until the dial string is sent by the timeout or by the user entering the # character.
- **0>#t811.rat4**—If the first digit entered is 0, the timeout before automatic send is eight seconds, and the terminating character # can be entered at any time to manually send the dial string. If the first three digits entered are 011, then, after an additional 11 digits are entered, the timeout before an automatic send changes to four seconds. The user can enter more digits until the dial string is sent by the timeout or by the user entering the # character.
- **^1t4>#.**—If the first digit entered is anything other than 1, the timeout before an automatic send is four seconds. The terminating character # can be entered at any time to manually send the dial string. The user can enter more digits until the dial string is sent by the timeout or by the user entering the # character.

### Dial Plan Example 2

The following dial plans:

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

or

```
.t7>#r6t4-|911|1t7>#.x9t1-|0t4>#.t7-
```

consist of the following rules:

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

### Dial Plan Example 3

The following dial plan:

```
R1408([2_9].r5|[2_9].r6)|R9^(911|.r4)|X(1900|1888)|F011
```

consists of the following rules:

- **R1408([2\_9].r5|[2\_9].r6)**—The prefix 1408 will be added to any call numbers with seven or eight digits where the leading digit is in the range of 2 to 9. For example, 5551234 will become 14085551234, but 555123 does not match this rule.
- **R9^(911|.r4)**—The prefix 9 will be added to any numbers except 911 and five-digit numbers. For example, 911 will still be 911, and 51234 will still be 51234.
- **X(1900|1888)**—There will be no calls or call forwarding to numbers beginning with 1900 or 1888.

- **F011**—There will be no call forwarding to numbers beginning with 011.

In Dial Plan Example 3, there are two ‘R’ rules, so the first matched rule is triggered. Therefore, 5551234 becomes 14085551234. However, 555123 will then become 9555123 because it matches the second rule.

## DialPlanEx

If your dial plan exceeds 199 characters, then you must use the DialPlanEx parameter to configure your dial plans. The DialPlanEx parameter supports dial plans up to 499 characters in length. This range in the number of characters is the only difference between the DialPlanEx and DialPlan parameters. Therefore, all the information about the DialPlan parameter applies to the DialPlanEx parameter. For more information, see the “DialPlan” section on page 5-64.



### Note

---

If you are not using this parameter for dial plan configuration, be sure this parameter is set to 0.

---

## IPDialPlan

### Description

This parameter allows for detection of IP-like destination address in DialPlan. Three values are valid:

- 0—String is dialed as is and not treated as an IP address.
- 1—When the Cisco ATA detects two asterisks (\*\*), IPDialPlan takes over. The user enters the pound (#) key to terminate the digit collection, and the interdigit timeout default is not used.
- 2—When IPDialPlan is set to 2, three asterisks (\*\*\*) are required for IPDialPlan to take effect.

All other values are currently undefined.

### Value Type

Integer

### Range

0, 1 or 2

### Default

1

### Voice Configuration Menu Access Code

310



# Diagnostic Parameters

This section describes the following parameters, which are used for diagnostic purposes:

- [NPrintf, page 5-73](#)
- [TraceFlags, page 5-73](#)
- [SyslogIP, page 5-74](#)
- [SyslogCtrl, page 5-75](#)

## NPrintf

### Description

Use this parameter to specify the IP address and port of a host to which all Cisco ATA debug messages are sent. The program *prserv.exe*, which comes bundled with the Cisco ATA software, is needed to capture the debug information.

### Syntax

<HOST\_IP>, <HOST\_PORT>

### Example

If the program *prserv.exe* is running on a host with IP address 192.168.2.170 and listening port 9001, set NPrintf to 192.168.2.170.9001. This causes the Cisco ATA to send all debug traces to that IP address.

### Value Type

Extended IP address

### Default

0

### Voice Configuration Menu Access Code

81

## TraceFlags

### Description

Use this parameter to turn on specific trace features for diagnostic use. Bit values are as follows:

- Bits 0 to 1:
  - 0 (default) is for simple debug messages.
  - 1 is for detailed debug messages.
  - 2 and 3 are reserved.
- Bits 2-7—Reserved
- Bit 8—RTP statistics log (values **0/1** to disable/enable with default of **0**) has the following format:  
Recv[channel number]: <call duration in seconds> <number of rcv packets>  
<number of rcv octets> <number of late packets>

<number of lost packets> <average network jitter in 1/8 ms>  
 <counts in speeding up local clock (adjustment for 10 ms each time)>  
 <counts in slowing down local clock (adjustment for 10 ms each time)>




---

**Note** Bit 8 is not used at this time.

---

- Bits 9 to 31—Reserved.

#### Value Type

Bitmap

#### Default

0x00000000

#### Voice Configuration Menu Access Code

313

## SyslogIP

#### Description

Use this parameter for diagnostic purposes; specify the IP address and port number to which the Cisco ATA should send its *syslog* output information.

The program *prserv.exe*, which is included in all Cisco ATA software upgrade packages, can be used to capture syslog information if you do not have a syslog server.

#### Syntax

<HOST\_IPaddress>.<HOST\_PORT>

#### Example

If you want to send syslog information to the host at IP address 192.168.2.170 and port number 514, do the following:

- Configure the value of this parameter as 192.168.2.170.514
- On your PC, run the command:

```
prserv 514
```

#### Value Type

Extended IP address

**Default**

0.0.0.0.514

**Voice Configuration Menu Access Code**

7975640

**Related Parameter**[SyslogCtrl, page 5-75](#)

## SyslogCtrl

**Description**

Use this parameter to turn on specific syslog traces. All traces are sent to the syslog server specified in the SyslogIP parameter.

See [Table 5-12](#) for bit values and the corresponding types of messages to turn on for tracing.

**Value Type**

Bitmap

**Default**

0x00000000

**Voice Configuration Menu Access Code**

7975641

**Related Parameter**[SyslogIP, page 5-74](#)**Table 5-12 SyslogCtrl Parameter Definitions**

Bit Number	Type of Messages to Trace
0	ARP messages.
1	DHCP messages
2	TFTP messages
3	Cisco ATA configuration-update messages.
4	System reboot messages
5-7	Reserved.
8	SIP messages
9	Cisco ATA event messages.
10	FAX messages.
11-15	Reserved.
16	RTP statistics messages.
17-31	Reserved.

# CFGID—Version Parameter for Cisco ATA Configuration File

**Description**

CFGID is a 32-bit unsigned-value parameter whose purpose is to allow the local administrator to track the version of the Cisco ATA configuration file. This parameter-value assignment is entirely the responsibility of the local administrator, and has no significance to the operation of the Cisco ATA.

**Value Type**

Bitmap

**Default**

0x00000000