Cisco ATA 191 and ATA 192 Analog Telephone Adapter Provisioning Guide for Multiplatform Firmware

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CHAPTER 1

Deployment and Provisioning

Deployment and Provisioning Overview

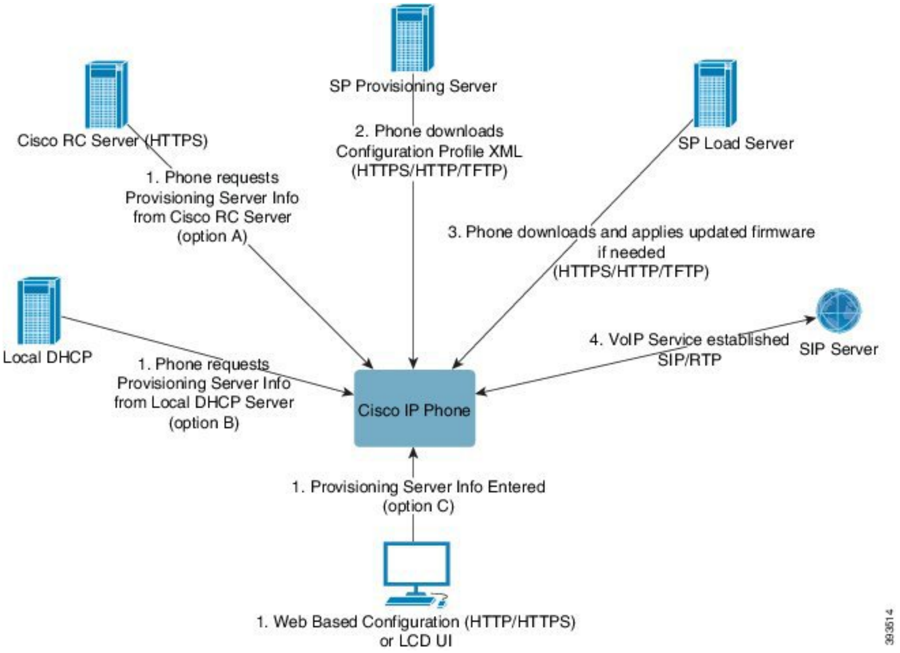
Cisco ATA 191 and 192 Analog Telephone Adapters (ATA) are intended for high-volume deployments by VoIP service providers to residential and small business customers. In business or enterprise environments, these ATAs can serve as terminal nodes. These devices are widely distributed across the Internet, connected through routers and firewalls at the customer premises.

The ATA can be used as a remote extension of the service provider back-end equipment. Remote management and configuration ensures the proper operation of the IP Telephony device at the customer premises.

This customized, ongoing configuration is supported by the following features:

- Reliable remote control of the endpoint
- Encryption of the communication controlling the endpoint
- Streamlined endpoint account binding

Phones can be provisioned to download configuration profiles or updated firmware from a remote server. Downloads can happen when the phones are connected to a network, when they are powered up, and at set intervals. Provisioning is typically part of high-volume, Voice-over-IP (VoIP) deployments common to service providers. Configuration profiles or updated firmware are transferred to the device using TFTP, HTTP, or HTTPS.


SP Provisioning Server
Cisco RC Server (HTTPS)
2. Phone downloads Configuration Profile XML (HTTPS/HTTP/TFTP)
SP Load Server
1. Phone requests Provisioning Server Info from Cisco RC Server (option A)
3. Phone downloads and applies updated firmware if needed (HTTPS/HTTP/TFTP)
Local DHCP
1. Phone requests Provisioning Server Info from Local DHCP Server (option B)
4. VoIP Service established SIP/RTP
SIP Server
Cisco IP Phone
1. Provisioning Server Info Entered (option C)
1. Web Based Configuration (HTTP/HTTPS) or LCD UI
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At a high level, the phone provisioning process is:

1. If the phone is not yet configured, the provisioning server information is applied to the phone via one of the following options:
    a. Downloaded from the Cisco EDOS RC server via HTTPS.
    b. Queried from the local DHCP server.
    c. Entered via the Cisco phone web based configuration utility.
2. The phone downloads and applies the configuration XML via HTTPS, HTTP, or TFTP using provisioning server information.
3. The phone downloads and applies the updated firmware, if needed, via HTTPS, HTTP, or TFTP.
4. The VOIP service establishes using the specified configuration and firmware.

In this document, the terms *phone* and *device* mean the ATA.

Deployment

These ATAs provide convenient mechanisms for provisioning, based on two deployment models:

- Bulk distribution—The service provider acquires these ATAs in bulk quantity and either preprovisions them in-house or purchases RC units from Cisco. The devices are then issued to the customers as part of a VoIP service contract.
- Retail distribution—The customer purchases the ATA from a retail outlet and requests VoIP service from the service provider. The service provider must then support the secure remote configuration of the device.

Bulk Distribution

In this model, the service provider issues phones to its customers as part of a VoIP service contract. The devices are either RC units or preprovisioned in-house.

Cisco preprovisions RC units to resynchronize with a Cisco server that downloads the device profile and firmware updates.

A service provider can preprovision phones with the desired parameters, including the parameters that control resynchronization, through various methods:

- In-house by using DHCP and TFTP
- Remotely by using TFTP, HTTP, or HTTPS
- A combination of in-house and remote provisioning

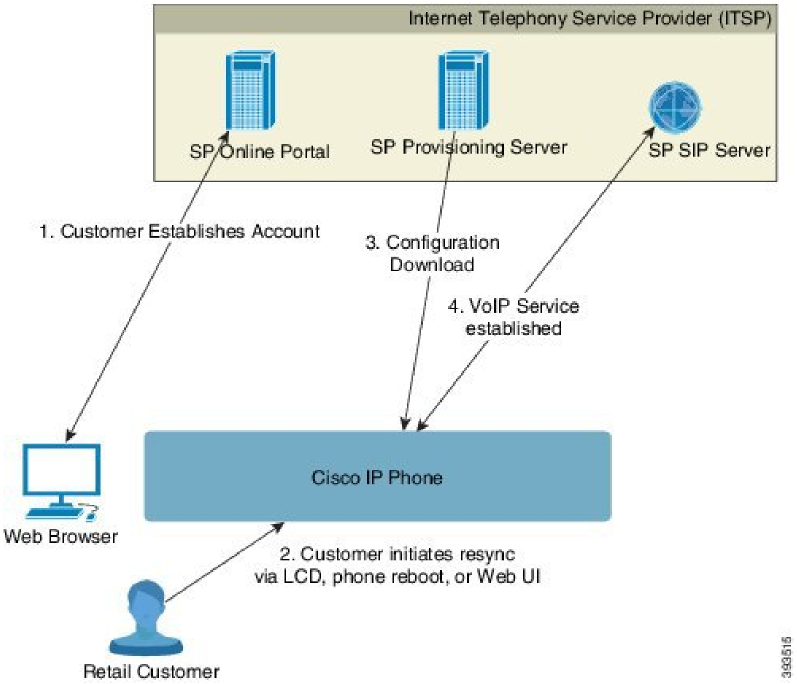
Related Topics

Remote Customization (RC) Distribution, on page 36

Retail Distribution

In a retail distribution model, a customer purchases a phone and subscribes to a particular service. The Internet Telephony Service Provider (ITSP) sets up and maintains a provisioning server, and preprovisions the phone to resynchronize with the service provider server.

Figure 1: Retail Distribution


Internet Telephony Service Provider (ITSP)
SP Online Portal
SP Provisioning Server
SP SIP Server
1. Customer Establishes Account
3. Configuration Download
4. VoIP Service established
Cisco IP Phone
Web Browser
2. Customer initiates resync via LCD, phone reboot, or Web UI
Retail Customer
393515

The customer signs on to the service and establishes a VoIP account, possibly through an online portal, and binds the device to the assigned service account. The unprovisioned phone is instructed to resync with a specific provisioning server through a resync URL command.

In the following example, a device at the DHCP-assigned IP address 192.168.1.102 is instructed to provision itself to the SuperVoIP service:

```
http://192.168.1.102/admin/resync?https://prov.supervoip.com/cisco-init/ata.cfg
```

For both initial and permanent access, the provisioning server relies on the phone client certificate for authentication. The provisioning server supplies correct configuration parameter values based on the associated service account.

When the device is powered up or a specified time elapses, the phone resynchronizes and downloads the latest parameters. These parameters can address goals such as setting up a hunt group, setting speed dial numbers, and limiting the features that a user can modify.

Related Topics

In-House Device Preprovisioning, on page 37

Resynchronization Process

The firmware for each phone includes an administration web server that accepts new configuration parameter values. The phone may be instructed to resynchronize configuration after reboot, or at scheduled intervals with a specified provisioning server through a resync URL command in the device profile.

By default, the web server is enabled. To disable or enable the Web server, use the resync URL command.

If needed, an immediate resynchronization may be requested via a “resync" action URL.

Example

```
http://192.168.1.102/admin/resync?https://prov.supervoip.com/cisco-init/ata.cfg
```

In this example, a device at the DHCP-assigned IP address 192.168.1.102 is instructed to provision itself to the SuperVoIP service at prov.supervoip.com. The Customer ID number for the new account is 1234abcd. The remote provisioning server associates the phone that is performing the resync request with the account, based on the URL and Customer ID.

Through this initial resync operation, the phone is configured in a single step. The phone is automatically directed to resync thereafter to a permanent URL on the server.

For both initial and permanent access, the provisioning server relies on the client certificate for authentication. The server supplies configuration parameter values based on the associated service account.

Provisioning

A phone can be configured to resynchronize its internal configuration state to match a remote profile periodically and on power-up. The phone contacts a normal provisioning server (NPS) or an access control server (ACS).

By default, a profile resync is only attempted when the phone is idle. This practice prevents an upgrade that would trigger a software reboot and interrupt a call. If intermediate upgrades are required to reach a current upgrade state from an older release, the upgrade logic can automate multistage upgrades.

Normal Provisioning Server

The Normal Provisioning Server (NPS) can be a TFTP, HTTP, or HTTPS server. A remote firmware upgrade is achieved by using TFTP or HTTP, or HTTPS, because the firmware does not contain sensitive information.

Although HTTPS is recommended, communication with the NPS does not require the use of a secure protocol because the updated profile can be encrypted by a shared secret key. For more information about utilizing HTTPS, see Communication Encryption, on page 7. Secure first-time provisioning is provided through a mechanism that uses SSL functionality. An unprovisioned phone can receive a 256-bit symmetric key encrypted profile that is targeted for that device.

TR-069

The digital subscriber line (DSL) Forum TR-069, CPE WAN Management Protocol (CWMP), is used for communications between a customer premise equipment (CPE) device and an autoconfiguration server (ACS). The TR-069 Agent manages a collection of CPE devices, with the primary capability for auto-configuration and dynamic service provisioning, software image management, status and performance monitoring, and diagnostics.

It supports multiple scenarios, including:

- Device administration—Authenticates administrators, authorizes commands, and provides an audit trail
- Remote Access—Works with VPN and other remote network access devices to enforce access policies
- Network admission control—Communicates with posture and audit servers to enforce admission control policies

The TR-069 Agent CPE devices must be set up and enabled for TR-069. An ACS used to communicate with the CPE must be TR-069 compliant in order to enable the TR-069 Agent.

Provisioning States

The provisioning process involves these provisioning states.

State	Description
MFG-RESET Manufacturing Reset	The device returns to a fully unprovisioned state; all configurable parameters regain their default values. Manufacturing reset can be performed in these ways: • Through the IVR sequence ******73738#1#**. • Press the reset button on the ATA. • Use the web interface to reset the ATA to factory default settings. On phones that do not support IVR, press the reset button or LCD factory reset entry to reset it to the default values. Allowing the end user to perform a manufacturing reset guarantees that the device can always be returned to an accessible state.

State	Description
SP-CUST Service Provider Customization	The Profile_Rule parameter points to a device-specific configuration profile by using a provisioning server that is specific to the service provider. The methods for initiating resynchronization are: • Autoconfiguration with a local DHCP server. A TFTP server name or IPv4 address is specified by DHCP. The TFTP server includes the Profile_Rule parameter in the configuration file. • Enter a resync URL. The URL starts a web browser and requests a resync to a specific TFTP server by entering the URL syntax: **`http://x.x.x.x/admin/resync?tftp://prvserv/device.cfg`**, where: • **`x.x.x.x`** is the IP address of the IP Telephony device, • **`prvserv`** is the target TFTP server, • **`device.cfg`** is the name of the configuration file on the server. • Edit the Profile_Rule parameter by opening the provisioning pane on the web interface and entering the TFTP URL in the Profile_Rule parameter. For example, **`tftp://prserv/spa112.cfg`**. • Modify the configuration file Profile_Rule and to contact a specific TFTP server and request a configuration file identified by the MAC-address. For example, this entry contacts a provisioning server, requesting a profile unique to the device with a MAC address identified by the $MA parameter: **`Profile_Rule tftp.callme.com/profile/$MA/spa112.cfg;`**

SEC-PRV-1 Secure Provisioning—Initial Configuration	An initial, device-unique CFG file is targeted to a IP Telephony device by compiling the CFG file with the SPC --target option. This provides an encryption that does not require the exchange of keys.
	The initial, device-unique CFG file reconfigures the device profile to enable stronger encryption by programming a 256-bit encryption key and pointing to a randomly generated TFTP directory. For example, the CFG file might contain:
	`Profile_Rule [--key $A] tftp.callme.com/profile/$B/spa112.cfg;`
	`GPP_A 8e4ca259…; # 256 bit key`
	`GPP_B Gp3sqLn…; # random CFG file path directory`

SEC-PRV-2 Secure Provisioning—Full Configuration	Profile resync operations subsequent to the initial SECPRV-1 provisioning retrieve the 256-bit encrypted CFG files that maintain the IP Telephony device in a state synchronized to the provisioning server. The profile parameters are reconfigured and maintained through this strongly encrypted profile. The encryption key and random directory location in the SEC-PRV-2 configuration can be changed periodically for extra security.

Configuration Access Control

The phone firmware provides mechanisms for restricting end-user access to some parameters. The firmware provides specific privileges for sign-in to an **Admin** account or a **User** account. Each can be independently password protected.

- Admin account–Allows the service provider full access to all administration web server parameters.
- User account–Allows the user to configure a subset of the administration web server parameters.

The service provider can restrict the user account in the provisioning profile in the following ways:

- Indicate which configuration parameters are available to the user account when creating the configuration.
- Disable user access to the administration web server.
- Restrict the Internet domains accessed by the device for resync, upgrades, or SIP registration for Line 1.

Related Topics

Element Tag Properties, on page 12
Access Control, on page 14

Communication Encryption

The configuration parameters that are communicated to the device can contain authorization codes or other information that protect the system from unauthorized access. It is in the service provider's interest to prevent unauthorized customer activity. It is in the customer's interest to prevent the unauthorized use of the account. The service provider can encrypt the configuration profile communication between the provisioning server and the device, in addition to restricting access to the administration web server.

Additional Information

Related Documentation

Use the following sections to obtain related information.

Cisco ATA 190 Series Documentation

Refer to publications that are specific to your language and call control system. Navigate from the following documentation URL:

https://www.cisco.com/c/en/us/products/unified-communications/ata-190-series-analog-telephone-adapters/index.html

Cisco ATA 190 Series Firmware Support Policy

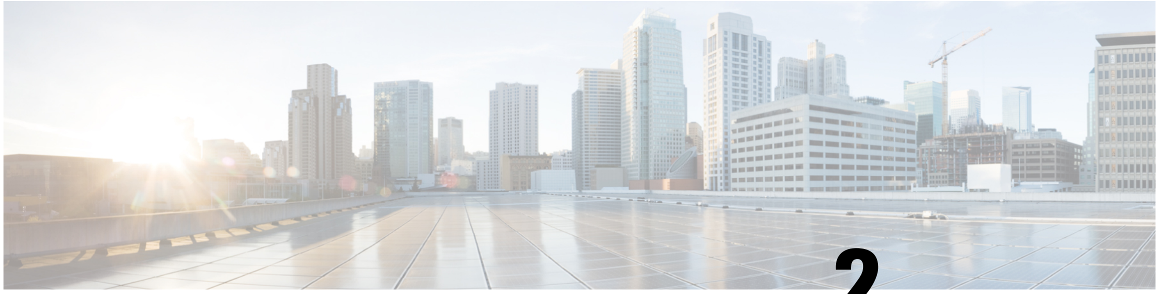
For information on the support policy for ATAs, see http://www.cisco.com/c/en/us/support/docs/collaboration-endpoints/unified-ip-phone-7900-series/116684-technote-ipphone-00.html.

Phone Behavior During Times of Network Congestion

- Administrative tasks, such as an internal port scan or security scan.
- Attacks that occur on your network, such as a Denial of Service attack.

Communications, Services, and Additional Information

- To receive timely, relevant information from Cisco, sign up at Cisco Profile Manager.
- To get the business impact you're looking for with the technologies that matter, visit Cisco Services.
- To submit a service request, visit Cisco Support.
- To discover and browse secure, validated enterprise-class apps, products, solutions and services, visit Cisco Marketplace.
- To obtain general networking, training, and certification titles, visit Cisco Press.
- To find warranty information for a specific product or product family, access Cisco Warranty Finder.



CHAPTER 2

New and Changed Information

New and Changed for Firmware Release 11.2(4)

Revision	**New and Changed**
Updated the default value of the parameters <*FAX_Enable_T38_1*> and <*FAX_Enable_T38_2*>	Voice Parameters, on page 71
Updated the description of the parameters <*NAT_Keep_Alive_Msg_1*> and <*NAT_Keep_Alive_Msg_2*>	Voice Parameters, on page 71
Updated the values of <*NAT_Keep_Alive_Msg_1*> and <*NAT_Keep_Alive_Msg_2*>	XML Open Format Sample, on page 165
Updated the parameters in <*Web_Management*>	XML Open Format Sample, on page 165 Web_Management, on page 154
Updated the description of the parameters <*Proxy_Fallback_Intvl_1*> and <*Proxy_Fallback_Intvl_2*>	Voice Parameters, on page 71
Updated the topic to mention the maximum number of A records for SRV record	Redundant Provisioning Servers, on page 43

New and Changed for Firmware Release 11.2(3)

Revision	**New and Changed**
Added the topic for the `HTTP Proxy Support` feature	HTTP_Proxy Parameters, on page 146

Revision	New and Changed
Updated the topic for the `HTTP Proxy Support` feature	XML Open Format Sample, on page 165
Updated the topic for the `Out of Box Provisioning Enhancement` feature	Remote Customization (RC) Distribution, on page 36
Updated the topic to add the value range of the parameters `Call_Back_Delay`, `VMWI_Refresh_Intvl`, `DTMF_Playback_Length`, and `DTMF_Playback_Level`	Voice Parameters, on page 71

New and Changed for Firmware Release 11.2(2)

Revision	New and Changed
Updated the description of the parameters `SIP_Transport_1` and `SIP_Transport_2`	Voice Parameters, on page 71
Updated the topic to add the new parameters `Secure_Call_Option_1` and `Secure_Call_Option_2`	Voice Parameters, on page 71
Updated the topic to add the new parameters `Secure_Call_Option_1` and `Secure_Call_Option_2`	XML Open Format Sample, on page 165

New and Changed for Firmware Release 11.2(1)

Revision	New and Changed
Updated the topic to show all the log modules	Log_Configuration Parameters, on page 159
Updated the topic to add the new log module options	XML Open Format Sample, on page 165
Updated the topic to add new parameters	Voice Parameters, on page 71
Updated the sample to add new parameters	XML Open Format Sample, on page 165

CHAPTER 3

Provisioning Formats

Configuration Profiles

The phone accepts configuration in an XML format.

For detailed information about your phone, refer to the administration guide for your particular device. Each guide describes the parameters that can be configured through the administration web server.

Configuration Profile Formats

The configuration profile defines the parameter values for the phone.

The configuration profile XML format uses standard XML authoring tools to compile the parameters and values.

Note Only the UTF-8 charset is supported. If you modify the profile in an editor, do not change the encoding format; otherwise, the phone cannot recognize the file.

Each phone has a different feature set and therefore, a different set of parameters.

XML Format (XML) Profile

The open format profile is a text file with XML-like syntax in a hierarchy of elements, with element attributes and values. This format lets you use standard tools to create the configuration file. A configuration file in this format can be sent from the provisioning server to the phone during a resync operation. The file can be sent without compilation as a binary object.

The phone can accept configuration formats that standard tools generate. This feature eases the development of back-end provisioning server software that generates configuration profiles from existing databases.

To protect confidential information in the configuration profile, the provisioning server delivers this type of file to the phone over a channel secured by TLS. Optionally, the file can be compressed by using the gzip deflate algorithm (RFC1951).

The file can be encrypted with 256-bit AES symmetric key encryption.

Example: Open Profile Format

```
<flat-profile>
<Resync_On_Reset> Yes </Resync_On_Reset>
<Resync_Periodic> 7200 </Resync_Periodic>
<Profile_Rule> tftp://prov.telco.com:6900/cisco/config/CP_xxxx_MPP.cfg</Profile_Rule>
</flat-profile>
```

The <flat-profile> element tag encloses all parameter elements that the phone recognizes.

Configuration File Components

A configuration file can include these components:

- Element tags
- Attributes
- Parameters
- Formatting features
- XML comments

Element Tag Properties

- The XML provisioning format and the Web UI allow the configuration of the same settings. The XML tag name and the field names in the Web UI are similar but vary due to XML element name restrictions. For example, underscores (_) instead of " ".
- The phone recognizes elements with proper parameter names that are encapsulated in the special <flat-profile> element.
- Element names are enclosed in angle brackets.
- Most element names are similar to the field names in the administration web pages for the device, with the following modifications:
  - Element names may not include spaces or special characters. To derive the element name from the administration web field name, substitute an underscore for every space or the special characters `[`, `]`, `(`, `)`, or `/`.

    Example: The <Resync_On_Reset> element represents the **Resync On Reset** field.

  - Each element name must be unique. In the administration web pages, the same fields can appear on multiple web pages, such as the Line, User, and Extension pages. Append `[n]` to the element name to indicate the number that is shown in the page tab.

    Example: The <Dial_Plan_1_> element represents the **Dial Plan** for Line 1.

- Each opening element tag must have a matching closing element tag. For example:

```
<flat-profile>
<Resync_On_Reset> Yes
  </Resync_On_Reset>
<Resync_Periodic> 7200
  </Resync_Periodic>
<Profile_Rule>tftp://prov.telco.com: 6900/cisco/config/CP_xxxx_MPP.cfg
  </Profile_Rule>
</flat-profile>
```

- Element tags are case-sensitive.
- Empty element tags are allowed and will be interpreted as configuring the value to be empty. Enter the opening element tag without a corresponding element tag, and insert a space and a forward slash before the closing angle bracket (>). In this example, Profile Rule B is empty:

```
<Profile_Rule_B />
```

- An empty element tag can be used to prevent the overwriting of any user-supplied values during a resync operation. In the following example, the user speed dial settings are unchanged:

```
<flat-profile>
<Speed_Dial_2_Name ua="rw"/>
<Speed_Dial_2_Number ua="rw"/>
<Speed_Dial_3_Name ua="rw"/>
<Speed_Dial_3_Number ua="rw"/>
<Speed_Dial_4_Name ua="rw"/>
<Speed_Dial_4_Number ua="rw"/>
<Speed_Dial_5_Name ua="rw"/>
<Speed_Dial_5_Number ua="rw"/>
<Speed_Dial_6_Name ua="rw"/>
<Speed_Dial_6_Number ua="rw"/>
<Speed_Dial_7_Name ua="rw"/>
<Speed_Dial_7_Number ua="rw"/>
<Speed_Dial_8_Name ua="rw"/>
<Speed_Dial_8_Number ua="rw"/>
<Speed_Dial_9_Name ua="rw"/>
<Speed_Dial_9_Number ua="rw"/>
</flat-profile>
```

- Use an empty value to set the corresponding parameter to an empty string. Enter an opening and closing element without any value between them. In the following example, the GPP_A parameter is set to an empty string.

```
<flat-profile>
<GPP_A>
  </GPP_A>
</flat-profile>
```

- Unrecognized element names are ignored.

Access Control

If the <Phone-UI-User-Mode> parameter is enabled, the phone GUI honors the user access attribute of the relevant parameters when the GUI presents a menu item.

For menu entries that are associated with a single configuration parameter:

- Provisioning the parameter with “ua=na” (“ua” stands for “user access”) attribute makes the entry disappear.
- Provisioning the parameter with “ua=ro” attribute makes the entry read-only and non-editable.

For menu entries that are associated with multiple configuration parameters:

- Provisioning all concerned parameters with “ua=na” attribute makes the entries disappear.

Related Topics

Configuration Access Control, on page 6

Parameter Properties

These properties apply to the parameters:

- Any parameters that are not specified by a profile are left unchanged in the phone.
- Unrecognized parameters are ignored.
- If the Open format profile contains multiple occurrences of the same parameter tag, the last such occurrence overrides any earlier ones. To avoid inadvertent override of configuration values for a parameter, we recommend that each profile specify at most one instance of a parameter.
- The last profile processed takes precedence. If multiple profiles specify the same configuration parameter, the value of the latter profile takes precedence.

String Formats

These properties apply to the formatting of the strings:

- Comments are allowed through standard XML syntax.

```
<!-- My comment is typed here -->
```

- Leading and trailing white space is allowed for readability but is removed from the parameter value.
- New lines within a value are converted to spaces.
- An XML header of the form `<? ?>` is allowed, but the phone ignores it.
- To enter special characters, use basic XML character escapes, as shown in the following table.

Special Character	XML Escape Sequence
& (ampersand)	&amp;
< (less than)	&lt;
> (greater than)	&gt;

Special Character	XML Escape Sequence
’ (apostrophe)	&apos;
” (double quote)	&quot;

In the following example, character escapes are entered to represent the greater than and less than symbols that are required in a dial plan rule. This example defines an information hotline dial plan that sets the <Dial_Plan_1_> parameter (**Admin Login** > **advanced** > **Voice** > **Ext (n)**) equal to (S0 <:18005551212>).

```
<flat-profile>
 <Dial_Plan_1_>
  (S0 &lt;:18005551212&gt;)
 </Dial_Plan_1_>
</flat-profile>
```

- Numeric character escapes, using decimal and hexadecimal values (s.a. `&#40;` and `&#x2e;`), are translated.
- The phone firmware only supports ASCII characters.

Open Profile (XML) Compression and Encryption

The Open configuration profile can be compressed to reduce the network load on the provisioning server. The profile can also be encrypted to protect confidential information. Compression is not required, but it must precede encryption.

Related Topics

Configuration Profile Formats, on page 11

Open Profile Compression

The supported compression method is the gzip deflate algorithm (RFC1951). The gzip utility and the compression library that implements the same algorithm (zlib) are available from Internet sites.

To identify compression, the phone expects the compressed file to contain a gzip compatible header. Invocation of the gzip utility on the original Open profile generates the header. The phone inspects the downloaded file header to determine the file format.

For example, if `profile.xml` is a valid profile, the file `profile.xml.gz` is also accepted. Either of the following commands can generate this profile type:

- `>gzip profile.xml`

  Replaces original file with compressed file.

- `>cat profile.xml | gzip > profile.xml.gz`

  Leaves original file in place, produces new compressed file.

A tutorial on compression is provided in the Compress an Open Profile with Gzip, on page 58 section.

AES-256-CBC Encryption

Symmetric key encryption can be used to encrypt an Open configuration profile, whether the file is compressed or not. The supported encryption algorithm is the American Encryption Standard (AES), using 256-bit keys, applied in cipher block chaining mode.

Note Compression must precede encryption for the phone to recognize a compressed and encrypted Open format profile. Encrypt a Profile with OpenSSL, on page 59 provides a tutorial on encryption.

The OpenSSL encryption tool, available for download from various Internet sites, can perform the encryption. Support for 256-bit AES encryption may require recompilation of the tool to enable the AES code. The firmware has been tested against version openssl-1.1.1d.

For an encrypted file, the profile expects the file to have the same format as generated by the following command:

```
# example encryption key = SecretPhrase1234

openssl enc -e -aes-256-cbc -k SecretPhrase1234 -in profile.xml -out profile.cfg

# analogous invocation for a compressed xml file

openssl enc -e -aes-256-cbc -k SecretPhrase1234 -in profile.xml.gz -out profile.cfg
```

A lowercase -k precedes the secret key, which can be any plain text phrase, and which is used to generate a random 64-bit salt. With the secret specified by the -k argument, the encryption tool derives a random 128-bit initial vector and the actual 256-bit encryption key.

When this form of encryption is used on a configuration profile, the phone must be informed of the secret key value to decrypt the file. This value is specified as a qualifier in the profile URL. The syntax is as follows, using an explicit URL:

```
[--key “SecretPhrase1234”] http://prov.telco.com/path/profile.cfg
```

This value is programmed by using one of the Profile_Rule parameters.

The key must be preprovisioned into the unit at an earlier time. Bootstrap of the secret key can be accomplished securely by using HTTPS.

Pre-encrypting configuration profiles offline, with symmetric key encryption, allows the use of HTTP for resyncing profiles. The provisioning server uses HTTPS to handle initial provisioning of the phone after deployment. This feature reduces the load on the HTTPS server in large-scale deployments.

The final filename does not require a specific format, but a filename that ends with the `.cfg` extension normally indicates a configuration profile.

Related Topics

Encrypt a Profile with OpenSSL, on page 59

Macro Expansion

Several provisioning parameters undergo macro expansion internally prior to being evaluated. This preevaluation step provides greater flexibility in controlling the phone resync and upgrade activities.

These parameter groups undergo macro expansion before evaluation:

- Resync_Trigger_*
- Profile_Rule*
- Log_xxx_Msg
- Upgrade_Rule

Under certain conditions, some general-purpose parameters (GPP_*) also undergo macro expansion, as explicitly indicated in Optional Resync Arguments, on page 19.

During macro expansion, the contents of the named variables replace expressions of the form $NAME and $(NAME). These variables include general-purpose parameters, several product identifiers, certain event timers, and provisioning state values. For a complete list, see Macro Expansion Variables, on page 68.

In the following example, the expression $(MAU) is used to insert the MAC address 000E08012345.

The administrator enters: **`$(MAU)config.cfg`**

The resulting macro expansion for a device with MAC address 000E08012345 is: `000E08012345config.cfg`

If a macro name is not recognized, it remains unexpanded. For example, the name STRANGE is not recognized as a valid macro name, while MAU is recognized as a valid macro name.

The administrator enters: **`$STRANGE$MAU.cfg`**

The resulting macro expansion for a device with MAC address 000E08012345 is: `$STRANGE000E08012345.cfg`

Macro expansion is not applied recursively. For example, $$MAU" expands into $MAU" (the $$ is expanded), and does not result in the MAC address.

The contents of the special purpose parameters, GPP_SA through GPP_SD, are mapped to the macro expressions $SA through $SD. These parameters are only macro expanded as the argument of the **--key** , **--uid**, and **--pwd** options in a resync URL.

Conditional Expressions

Conditional expressions can trigger resync events and select from alternate URLs for resync and upgrade operations.

Conditional expressions consist of a list of comparisons, separated by the **and** operator. All comparisons must be satisfied for the condition to be true.

Each comparison can relate to one of the following three types of literals:

- Integer values
- Software or hardware version numbers
- Doubled-quoted strings

Version Numbers

Cisco ATA 191 and 192—`ATA19x.v1-v2-v3MPP-BN` (where *BN* is the Build Number)

The comparing string must use the same format. Otherwise, a format parsing error results.

In the software version, v1-v2-v3-v4 can specify different digits (0-99). When comparing the software version, v1-v2-v3-v4 is compared in sequence, and the leftmost digits take precedence over the latter ones.

If *v[x]* includes only numeric digits, the digits are compared; if *v[x]* includes numeric digits and alpha characters, the digits are compared first, then the characters are compared in alphabetical order.

Example of Valid Version Number

ATA19x.11-1-0MPP-BN

By contrast: 11.1.0 is an invalid format.

Comparison

ATA19x.11-1-0MPP-BN < ATA19x.11-1-1MPP-BN

Quoted strings can be compared for equality or inequality. Integers and version numbers can also be compared arithmetically. The comparison operators can be expressed as symbols or as acronyms. Acronyms are convenient for expressing the condition in an Open format profile.

Operator	Alternate Syntax	Description	Applicable to Integer and Version Operands	Applicable to Operands
=	eq	equal to	Yes	Yes
!=	ne	not equal to	Yes	Yes
<	lt	less than	Yes	No
<=	le	less than or equal to	Yes	No
>	gt	greater than	Yes	No
>=	ge	greater than or equal to	Yes	No
AND		and	Yes	Yes

It is important to enclose macro variables in double quotes where a string literal is expected. Don't do so where a number or version number is expected.

When used in the context of the Profile_Rule* and Upgrade_Rule parameters, conditional expressions must be enclosed within the syntax “(expr)?” as in this upgrade rule example. Remember to replace *BN* with the build number of your firmware load to upgrade to.

- For Firmware Release 11.1(0)SR3 and previous

  Since the version comparison rule changes in the 11.1(0)SR4 release, use the following conditional expression to upgrade the current firmware to Firmware Release 11.1(0)SR4 or later:

  ```
  ("$SWVER" ne "11-1-0MPP")? http://ps.tell.com/sw/ATA19x.11-1-0MPP-BN.loads
  ```

- For Firmware Release 11.1(0)SR4 and later

  ```
  ($SWVER ne 11-1-0MPP)? http://ps.tell.com/sw/ATA19x.11-1-0MPP-BN.loads
  ```

Do not use the preceding syntax with parentheses to configure the Resync_Trigger_* parameters.

URL Syntax

Use the Standard URL syntax to specify how to retrieve configuration files and firmware loads in Profile_Rule* and Upgrade_Rule parameters, respectively. The syntax is as follows:

`[ scheme:// ] [ server [:port]] filepath`

Where **`scheme`** is one of these values:

- tftp
- http
- https

If **`scheme`** is omitted, tftp is assumed. The server can be a DNS-recognized hostname or a numeric IP address. The port is the destination UDP or TCP port number. The filepath must begin with the root directory (/); it must be an absolute path.

If **`server`** is missing, the tftp server specified through DHCP (option 66) is used.

Note For upgrade rules, the server must be specified.

If **`port`** is missing, the standard port for the specified scheme is used. Tftp uses UDP port 69, http uses TCP port 80, https uses TCP port 443.

A filepath must be present. It need not necessarily refer to a static file, but can indicate dynamic content obtained through CGI.

Macro expansion applies within URLs. The following are examples of valid URLs:

```
/$MA.cfg
/cisco/cfg.xml
192.168.1.130/profiles/init.cfg
tftp://prov.call.com/cpe/cisco$MA.cfg
http://neptune.speak.net:8080/prov/$D/$E.cfg
https://secure.me.com/profile?Linksys
```

When using DHCP option 66, the empty syntax is not supported by upgrade rules. It is only applicable for Profile Rule*.

Optional Resync Arguments

Optional arguments, **`key`**, **`uid`**, and **`pwd`**, can precede the URLs entered in Profile_Rule* parameters, collectively enclosed by square brackets.

key

The **key** option is used to specify an encryption key. Decryption of profiles that have been encrypted with an explicit key is required. The key itself is specified as a (possibly quoted) string following the term **--key**.

Usage Examples

```
[--key VerySecretValue]
[--key “my secret phrase”]
[--key a37d2fb9055c1d04883a0745eb0917a4]
```

The bracketed optional arguments are macro expanded. Special purpose parameters, GPP_SA through GPP_SD, are macro expanded into macro variables, $SA through $SD, only when they are used as key option arguments. See these examples:

```
[--key $SC]
[--key “$SD”]
```

In Open format profiles, the argument to **--key** must be the same as the argument to the **-k** option that is given to **openssl**.

uid and pwd

The **uid** and **pwd** options can be used to specify the userID and password that will be sent in response to HTTP Basic and Digest authentication challenges when the specified URL is requested. The bracketed optional arguments are macro expanded. Special purpose parameters, GPP_SA through GPP_SD, are macro expanded into macro variables, $SA through $SD, only when they are used as key option arguments. See these examples:

```
GPP_SA = MyUserID
GPP_SB = MySecretPassword
```

[--uid $SA --pwd $SB] https://provisioning_server_url/path_to_your_config/your_config.xml

would then expand to:

[--uid MyUserID --pwdMySecretPassword]
https://provisioning_server_url/path_to_your_config/your_config.xml

Application of a Profile to the Phone

After you create an XML configuration script, it must be passed to the phone for application. To apply the configuration, you can either download the configuration file to the phone from a TFTP, HTTP, or HTTPS server using a web browser or by using cURL command line utility.

Download the Configuration File to the Phone from a TFTP Server

Complete these steps to download the configuration file to a TFTP server application on your PC.

Procedure

Step 1 Connect your PC to the phone LAN.

Step 2 Run a TFTP server application on the PC and ensure that the configuration file is available in the TFTP root directory.

Step 3 In a web browser, enter the phone LAN IP address, the IP address of the computer, the filename, and the login credentials. Use this format:

```
http://<WAN_IP_Address>/admin/resync?tftp://<PC_IP_Address>/<file_name>&xuser=admin&xpassword=<password>
```

Example:

```
http://192.168.15.1/admin/resync?tftp://192.168.15.100/my_config.xml&xuser=admin&xpassword=admin
```

Download the Configuration File to the Phone with cURL

Complete these steps to download the configuration to the phone by using cURL. This command-line tool is used to transfer data with a URL syntax. To download cURL, visit:

https://curl.haxx.se/download.html

Note We recommend that you do not use cURL to post the configuration to the phone because the username and password might get captured while using cURL.

Procedure

Step 1 Connect your PC to the LAN port of the phone.

Step 2 Download the configuration file to the phone by entering the following cURL command:

```
curl -d @my_config.xml
"http://192.168.15.1/admin/config.xml&xuser=admin&xpassword=admin"
```

Provisioning Parameter Types

This section describes the provisioning parameters broadly organized according to function:

These provisioning parameter types exist:

- General Purpose
- Enables
- Triggers
- Configurable Schedules
- Profile Rules
- Upgrade Rule

Related Topics

Voice Parameters, on page 71

Router Configuration Parameters, on page 117

General Purpose Parameters

The general-purpose parameters GPP_* (**Admin Login** > **advanced** > **Voice** > **Provisioning**) are used as free string registers when configuring the phone to interact with a particular provisioning server solution. The GPP_* parameters are empty by default. They can be configured to contain diverse values, including the following:

- Encryption keys
- URLs
- Multistage provisioning status information.
- Post request templates
- Parameter name alias maps
- Partial string values, eventually combined into complete parameter values.

The GPP_* parameters are available for macro expansion within other provisioning parameters. For this purpose, single-letter uppercase macro names (A through P) suffice to identify the contents of GPP_A through GPP_P. Also, the two-letter uppercase macro names SA through SD identify GPP_SA through GPP_SD as a special case when used as arguments of the following URL options:

key, **uid**, and **pwd**

These parameters can be used as variables in provisioning and upgrade rules. They are referenced by prefixing the variable name with a '$' character, such as $GPP_A.

Use General Purpose Parameters

For example, if GPP_A contains the string ABC, and GPP_B contains 123, the expression $A$B macro expands into ABC123.

Before you begin

Access the phone administration web page. See Access the Phone Web Interface, on page 34.

Procedure

Step 1 Select **Voice** > **Provisioning**.

Step 2 Scroll to the **General Purpose Parameters** section.

Step 3 Enter valid values in the fields, GPP A through GPP P.

Step 4 Click **Submit All Changes**.

Enable Parameters

The Provision_Enable and Upgrade_Enable parameters control all profile resync and firmware upgrade operations. These parameters control resyncs and upgrades independently of each other. These parameters also control resync and upgrade URL commands that are issued through the administration web server. Both of these parameters are set to **Yes** by default.

The Resync_From_SIP parameter controls requests for resync operations. A SIP NOTIFY event is sent from the service provider proxy server to the phone. If enabled, the proxy can request a resync. To do so, the proxy sends a SIP NOTIFY message that contains the Event: resync header to the device.

The device challenges the request with a 401 response (authorization refused for used credentials). The device expects an authenticated subsequent request before it honors the resync request from the proxy. The Event: reboot_now and Event: restart_now headers perform cold and warm restarts, respectively, which are also challenged.

The two remaining enables are Resync_On_Reset and Resync_After_Upgrade_Attempt. These parameters determine whether the device performs a resync operation after power-up software reboots and after each upgrade attempt.

When Resync_On_Reset is enabled, the device introduces a random delay that follows the boot-up sequence before the reset is performed. The delay is a random time up to the value that the Resync_Random_Delay (in seconds) specifies. In a pool of phones that power up simultaneously, this delay spreads out the start times of the resync requests from each unit. This feature can be useful in a large residential deployment, in the case of a regional power failure.

Triggers

The phone allows you to resync at specific intervals or at a specific time.

Resync at Specific Intervals

The phone is designed to resync with the provisioning server periodically. The resync interval is configured in Resync_Periodic (seconds). If this value is left empty, the device does not resync periodically.

The resync typically takes place when the voice lines are idle. If a voice line is active when a resync is due, the phone delays the resync procedure until the line becomes idle again. A resync can cause configuration parameter values to change.

A resync operation can fail because the phone is unable to retrieve a profile from the server, the downloaded file is corrupt, or an internal error occurred. The device tries to resync again after a time that is specified in Resync_Error_Retry_Delay (seconds). If Resync_Error_Retry_Delay is set to 0, the device does not try to resync again after a failed resync attempt.

If an upgrade fails, a retry is performed after Upgrade_Error_Retry_Delay seconds.

Two configurable parameters are available to conditionally trigger a resync: Resync_Trigger_1 and Resync_Trigger_2. Each parameter can be programmed with a conditional expression that undergoes macro expansion. When the resync interval expires (time for the next resync) the triggers, if set, will prevent resync unless one or more triggers evaluates to true.

The following example condition triggers a resync. In the example, the last phone upgrade attempt has elapsed more than 5 minutes (300 seconds), and at least 10 minutes (600 seconds) have elapsed since the last resync attempt.

```
$UPGTMR gt 300 and $PRVTMR ge 600
```

Resync at a Specific Time

The Resync_At parameter allows the phone to resync at a specific time. This parameter uses the 24-hour format (hhmm) to specify the time.

The Resync_At_Random_Delay parameter allows the phone to resync at an unspecified delay in time. This parameter uses a positive integer format to specify the time.

Flooding the server with resync requests from multiple phones that are set to resync at the same time should be avoided. To do so, the phone triggers the resync up to 10 minutes after the specified time.

For example, if you set the resync time to 1000 (10 a.m.), the phone triggers the resync anytime between 10:00 a.m. and 10:10 a.m.

By default, this feature is disabled. If the Resync_At parameter is provisioned, the Resync_Periodic parameter is ignored.

Configurable Schedules

You can configure schedules for periodic resyncs, and you can specify the retry intervals for resync and upgrade failures by using these provisioning parameters:

- Resync_Periodic
- Resync_Error_Retry_Delay
- Upgrade_Error_Retry_Delay

Each parameter accepts a single delay value (seconds). The new extended syntax allows for a comma-separated list of consecutive delay elements. The last element in the sequence is implicitly repeated forever.

Optionally, you can use a plus sign to specify another numeric value that appends a random extra delay.

Example 1

In this example, the phone periodically resyncs every 2 hours. If a resync failure occurs, the device retries at these intervals: 30 minutes, 1 hour, 2 hours, 4 hours. The device continues to try at 4-hour intervals until it resyncs successfully.

```
Resync_Periodic=7200
Resync_Error_Retry_Delay=1800,3600,7200,14400
```

Example 2

In this example, the device periodically resyncs every hour (plus an extra random delay of up to 10 minutes). In the case of a resync failure, the device retries at these intervals: 30 minutes (plus up to 5 minutes). 1 hour (plus up to 10 minutes), 2 hours (plus up to 15 minutes). The device continues to try at 2-hour intervals (plus up to 15 minutes) until it successfully resyncs.

```
Resync_Periodic=3600+600
Resync_Error_Retry_Delay=1800+300,3600+600,7200+900
```

Example 3

In this example, if a remote upgrade attempt fails, the device retries the upgrade in 30 minutes, then again after one more hour, then in two hours. If the upgrade still fails, the device retries every four to five hours until the upgrade succeeds.

```
Upgrade_Error_Retry_Delay  =  1800,3600,7200,14400+3600
```

Profile Rules

The phone provides multiple remote configuration profile parameters (Profile_Rule*). Thus, each resync operation can retrieve multiple files that different servers manage.

In the simplest scenario, the device resyncs periodically to a single profile on a central server, which updates all pertinent internal parameters. Alternatively, the profile can be split between different files. One file is common for all the phones in a deployment. A separate, unique file is provided for each account. Encryption keys and certificate information can be supplied by still another profile, stored on a separate server.

Whenever a resync operation is due, the phone evaluates the four Profile_Rule* parameters in sequence:

1. Profile_Rule
2. Profile_Rule_B
3. Profile_Rule_C
4. Profile_Rule_D

Each evaluation can result in a profile retrieval from a remote provisioning server, with a possible update of some number of internal parameters. If an evaluation fails, the resync sequence is interrupted, and is retried again from the beginning specified by the Resync_Error_Retry_Delay parameter (seconds). If all evaluations succeed, the device waits for the second specified by the Resync_Periodic parameter and then performs another resync.

The contents of each Profile_Rule* parameter consist of a set of alternatives. The alternatives are separated by the | (pipe) character. Each alternative consists of a conditional expression, an assignment expression, a profile URL, and any associated URL options. All these components are optional within each alternative. The following are the valid combinations, and the order in which they must appear, if present:

```
[ conditional-expr ] [ assignment-expr ] [[ options ] URL ]
```

Within each Profile_Rule* parameter, all alternatives except the last one must provide a conditional expression. This expression is evaluated and is processed as follows:

1. Conditions are evaluated from left to right, until one is found that evaluates as true (or until one alternative is found with no conditional expression).
2. Any accompanying assignment expression is evaluated, if present.
3. If a URL is specified as part of that alternative, an attempt is made to download the profile that is located at the specified URL. The system attempts to update the internal parameters accordingly.

If all alternatives have conditional expressions and none evaluates to true (or if the whole profile rule is empty), the entire Profile_Rule* parameter is skipped. The next profile rule parameter in the sequence is evaluated.

Example 1

This example resyncs unconditionally to the profile at the specified URL, and performs an HTTP GET request to the remote provisioning server:

```
http://remote.server.com/cisco/$MA.cfg
```

Example 2

In this example, the device resyncs to two different URLs, depending on the registration state of Line 1. In case of lost registration, the device performs an HTTP POST to a CGI script. The device sends the contents of the macro expanded GPP_A, which may provide additional information on the device state:

```
($PRVTMR ge 600)? http://p.tel.com/has-reg.cfg
| [--post a] http://p.tel.com/lost-reg?
```

Example 3

In this example, the device resyncs to the same server. The device provides additional information if a certificate is not installed in the unit (for legacy pre-2.0 units):

```
("$CCERT" eq "Installed")? https://p.tel.com/config?
| https://p.tel.com/config?cisco$MAU
```

Example 4

In this example, Line 1 is disabled until GPP_A is set equal to Provisioned through the first URL. Afterwards, it resyncs to the second URL:

```
("$A" ne "Provisioned")? (Line_Enable_1_ = "No";)! https://p.tel.com/init-prov
| https://p.tel.com/configs
```

Example 5

In this example, the profile that the server returns is assumed to contain XML element tags. These tags must be remapped to proper parameter names by the aliases map stored in GPP_B:

```
[--alias b] https://p.tel.com/account/$PN$MA.xml
```

A resync is typically considered unsuccessful if a requested profile is not received from the server. The Resync_Fails_On_FNF parameter can override this default behavior. If Resync_Fails_On_FNF is set to No, the device accepts a file-not-found response from the server as a successful resync. The default value for Resync_Fails_On_FNF is Yes.

Report Rule

The ATA provides a mechanism for reporting its current internal configuration to the provisioning server. This is useful for development and debugging. The report syntax is similar to the Open format profile. All provisionable parameters are included, except for the values of passwords, keys, and the GPP_SA to GPP_SD parameters, which are not shown.

The Report_Rule parameter is evaluated like a profile rule parameter. In other words, it accepts a URL, optionally qualified with a bracketed expression. The URL specifies the target destination for the report and an encryption key can be included as an option.

The URL scheme can be TFTP, HTTP, or HTTPS. When using TFTP, the operation performed is TFTP put.

For the HTTP and HTTPS Report Method field, the operation performed is HTTP post or HTTP put.

Note The default option is HTTP post.

If an encryption key is specified, the report is encrypted using 256-bit AES in CBC mode. The encrypted report can be decrypted with the following OpenSSL (or equivalent) command:

```
openssl enc –d –aes-256-cbc –k secretphrase –in rep.xml.enc –out rep.xml
```

The following is an example of the corresponding Report_Rule configuration:

```
[ --key secretphrase ] http://prov.serv.net/spa/$MA/rep.xml.enc
```

After the report rule is configured, an actual report can be generated and transmitted by sending the device a SIP NOTIFY message, with the Event: report type. The SIP NOTIFY request is handled like other SIP notifies, with the device requiring authentication from the requesting server before honoring the request to issue a report. Each SIP NOTIFY report request generates one attempt to transmit the report. Retries are not supported.

Upgrade Rule

Upgrade rule is to tell the device to activate to a new load and from where to get the load, if necessary. If the load is already on the device, it will not try to get the load. So, validity of the load location does not matter when the desired load is in the inactive partition.

The Upgrade_Rule specifies a firmware load which, if different from the current load, will be downloaded and applied unless limited by a conditional expression or Upgrade_Enable is set to **No**.

The phone provides one configurable remote upgrade parameter, Upgrade_Rule. This parameter accepts syntax similar to the profile rule parameters. URL options are not supported for upgrades, but conditional expressions and assignment expressions can be used. If conditional expressions are used, the parameter can be populated with multiple alternatives, separated by the | character. The syntax for each alternative is as follows:

```
[ conditional-expr ] [ assignment-expr ] URL
```

As in the case of Profile_Rule* parameters, the Upgrade_Rule parameter evaluates each alternative until a conditional expression is satisfied or an alternative has no conditional expression. The accompanying assignment expression is evaluated, if specified. Then, an upgrade to the specified URL is attempted.

If the Upgrade_Rule contains a URL without a conditional expression, the device upgrades to the firmware image that the URL specifies. After macro expansion and evaluation of the rule, the device does not reattempt to upgrade until the rule is modified or the effective combination of scheme + server + port + filepath is changed.

To attempt a firmware upgrade, the device disables audio at the start of the procedure and reboots at the end of the procedure. The device automatically begins an upgrade that is driven by the contents of Upgrade_Rule only if all voice lines are currently inactive.

For example,

For the Cisco ATA 191 and 192:

```
http://p.tel.com/firmware/ATA19x.11-1-0MPP-BN.img

where BN==Build Number
```

In this example, the Upgrade_Rule upgrades the firmware to the image that is stored at the indicated URL.

Here is another example for the ATA 191 and 192:

```
("$F" ne "beta-customer")? http://p.tel.com/firmware/ATA19x.11-1-0MPP-BN.img
| http://p.tel.com/firmware/ATA19x.11-1-0MPP-BN.img

where BN==Build Number
```

This example directs the unit to load one of two images, based on the contents of a general-purpose parameter, GPP_F.

The device can enforce a downgrade limit regarding firmware revision number, which can be a useful customization option. If a valid firmware revision number is configured in the Downgrade_Rev_Limit parameter, the device rejects upgrade attempts for firmware versions earlier than the specified limit.

Data Types

These data types are used with configuration profile parameters:

- {a,b,c,…}—A choice among a, b, c, …
- Bool—Boolean value of either "yes" or "no."
- CadScript—A miniscript that specifies the cadence parameters of a signal. Up to 127 characters.

  Syntax: S_1[;S_2], where:

  - S_i=D_i($on_{i,1}$/$off_{i,1}$[,$on_{i,2}$/$off_{i,2}$[,$on_{i,3}$/$off_{i,3}$[,$on_{i,4}$/$off_{i,4}$[,$on_{i,5}$/$off_{i,5}$[,$on_{i,6}$/$off_{i,6}$]]]]]) and is known as a section.
  - $on_{i,j}$ and $off_{i,j}$ are the on/off duration in seconds of a *segment*. i = 1 or 2, and j = 1 to 6.
  - D_i is the total duration of the section in seconds.

  All durations can have up to three decimal places to provide 1 ms resolution. The wildcard character "*" stands for infinite duration. The segments within a section are played in order and repeated until the total duration is played.

  Example 1:

  ```
  60(2/4)

  Number of Cadence Sections = 1
  Cadence Section 1: Section Length = 60 s
  Number of Segments = 1
  ```

```
Segment 1: On=2s, Off=4s

Total Ring Length = 60s
```

Example 2—Distinctive ring (short,short,short,long):

```
60(.2/.2,.2/.2,.2/.2,1/4)

Number of Cadence Sections = 1
Cadence Section 1: Section Length = 60s
Number of Segments = 4
Segment 1: On=0.2s, Off=0.2s
Segment 2: On=0.2s, Off=0.2s
Segment 3: On=0.2s, Off=0.2s
Segment 4: On=1.0s, Off=4.0s

Total Ring Length = 60s
```

- DialPlanScript—Scripting syntax that is used to specify Line 1 and Line 2 dial plans.
- Float<n>—A floating point value with up to n decimal places.
- FQDN—Fully Qualified Domain Name. It can contain up to 63 characters. Examples are as follows:
  - sip.Cisco.com:5060 or 109.12.14.12:12345
  - sip.Cisco.com or 109.12.14.12
- FreqScript—A miniscript that specifics the frequency and level parameters of a tone. Contains up to 127 characters.

  Syntax: F_1@L_1[,F_2@L_2[,F_3@L_3[,F_4@L_4[,F_5@L_5[,F_6@L_6]]]]], where:
  - F_1–F_6 are frequency in Hz (unsigned integers only).
  - L_1–L_6 are corresponding levels in dBm (with up to one decimal place).

  White spaces before and after the comma are allowed but not recommended.

  Example 1—Call Waiting Tone:

```
440@-10

Number of Frequencies = 1
Frequency 1 = 440 Hz at –10 dBm
```

  Example 2—Dial Tone:

```
350@-19,440@-19

Number of Frequencies = 2
Frequency 1 = 350 Hz at –19 dBm
Frequency 2 = 440 Hz at –19 dBm
```

- IP— Valid IPv4 Address in the form of x.x.x.x, where x is between 0 and 255. Example: 10.1.2.100.
- UserID—User ID as it appears in a URL; up to 63 characters.

- Phone—A phone number string, such as 14081234567, *69, *72, 345678; or a generic URL, such as, 1234@10.10.10.100:5068 or jsmith@Cisco.com. The string can contain up to 39 characters.
- PhTmplt—A phone number template. Each template may contain one or more patterns that are separated by a comma (,). White space at the beginning of each pattern is ignored. "?" and "*" represent wildcard characters. To represent literally, use %xx. For example, %2a represents *. The template can contain up to 39 characters. Examples: "1408*, 1510*", "1408123????, 555?1.".
- Port—TCP/UDP Port number (0-65535). It can be specified in decimal or hex format.
- ProvisioningRuleSyntax—Scripting syntax that is used to define configuration resync and firmware upgrade rules.
- PwrLevel—Power level expressed in dBm with one decimal place, such as –13.5 or 1.5 (dBm).
- RscTmplt—A template of SIP Response Status Code, such as "404, 5*", "61?", "407, 408, 487, 481". It can contain up to 39 characters.
- Sig<n>—Signed n-bit value. It can be specified in decimal or hex format. A "-" sign must precede negative values. A + sign before positive values is optional.
- Star Codes—Activation code for a supplementary service, such as *69. The code can contain up to 7 characters.
- Str<n>—A generic string with up to n nonreserved characters.
- Time<n>—Time duration in seconds, with up to n decimal places. Extra specified decimal places are ignored.
- ToneScript—A miniscript that specifies the frequency, level, and cadence parameters of a call progress tone. Script may contain up to 127 characters.

  Syntax: FreqScript;Z_1[;Z_2].

  The section Z_1 is similar to the S_1 section in a CadScript, except that each on/off segment is followed by a frequency components parameter: $Z_1 = D_1(on_{i,1}/off_{i,1}/f_{i,1}[,on_{i,2}/off_{i,2}/f_{i,2} [,on_{i,3}/off_{i,3}/f_{i,3} [,on_{i,4}/off_{i,4}/f_{i,4} [,on_{i,5}/off_{i,5}/f_{i,5} [,on_{i,6}/off_{i,6}/f_{i,6}]]]]])$ where:

  - $f_{i,j} = n_1[+n_2]+n_3[+n_4[+n_5[+n_6]]]]$.
  - $1 < n_k < 6$ specifies the frequency components in the FreqScript that are used in that segment.

  If more than one frequency component is used in a segment, the components are summed together.

  Example 1—Dial tone:

  ```
  350@-19,440@-19;10(*/0/1+2)

  Number of Frequencies = 2
  Frequency 1 = 350 Hz at −19 dBm
  Frequency 2 = 440 Hz at −19 dBm
  Number of Cadence Sections = 1
  Cadence Section 1: Section Length = 10 s
  Number of Segments = 1
  Segment 1: On=forever, with Frequencies 1 and 2

  Total Tone Length = 10s
  ```

  Example 2—Stutter tone:

```
350@-19,440@-19;2(.1/.1/1+2);10(*/0/1+2)

Number of Frequencies = 2
Frequency 1 = 350 Hz at -19 dBm
Frequency 2 = 440 Hz at -19 dBm
Number of Cadence Sections = 2
Cadence Section 1: Section Length = 2s
Number of Segments = 1
Segment 1: On=0.1s, Off=0.1s with Frequencies 1 and 2
Cadence Section 2: Section Length = 10s
Number of Segments = 1
Segment 1: On=forever, with Frequencies 1 and 2

Total Tone Length = 12s
```

- Uns<n>—Unsigned n-bit value, where n = 8, 16, or 32. It can be specified in decimal or hex format, such as 12 or 0x18, as long as the value can fit into n bits.

Note Keep these under consideration:

- <Par Name> represents a configuration parameter name. In a profile, the corresponding tag is formed by replacing the space with an underscore “_”, such as **Par_Name**.
- An empty default value field implies an empty string < “” >.
- The phone continues to use the last configured values for tags that are not present in a given profile.
- Templates are compared in the order given. The first, *not the closest*, match is selected. The parameter name must match exactly.
- If more than one definition for a parameter is given in a profile, the last such definition in the file is the one that takes effect in the phone.
- A parameter specification with an empty parameter value forces the parameter back to its default value. To specify an empty string instead, use the empty string "" as the parameter value.

Profile Updates and Firmware Upgrades

The phone supports secure remote provisioning (configuration) and firmware upgrades. An unprovisioned phone can receive an encrypted profile targeted for that device. The phone does not require an explicit key due to a secure first-time provisioning mechanism that uses SSL functionality.

User intervention is not required to either start or complete a profile update, or firmware upgrade, or if intermediate upgrades are required to reach a future upgrade state from an older release. A profile resync is only attempted when the phone is idle, because a resync can trigger a software reboot and disconnect a call.

General-purpose parameters manage the provisioning process. Each phone can be configured to periodically contact a normal provisioning server (NPS). Communication with the NPS does not require the use of a secure protocol because the updated profile is encrypted by a shared secret key. The NPS can be a standard TFTP, HTTP, or HTTPS server with client certificates.

The administrator can upgrade, reboot, restart, or resync phones by using the phone web user interface. The administrator can also perform these tasks by using a SIP notify message.

Configuration profiles are generated by using common, open-source tools that integrate with service provider provisioning systems.

Allow Profile Updates

Profile updates can be allowed at specified intervals. Updated profiles are sent from a server to the phone by using TFTP, HTTP, or HTTPS.

You can also configure the parameters in the phone configuration file with XML(cfg.xml) code.

Before you begin

Access the phone administration web page. See Access the Phone Web Interface, on page 34.

Procedure

Step 1 Select **Voice** > **Provisioning**.

Step 2 In the **Configuration Profile** section, choose **Yes** from the **Provision Enable** parameter.

Step 3 Click **Submit All Changes**.

Related Topics

Profile Updates and Firmware Upgrades, on page 31

Allow and Configure Firmware Upgrades

Firmware updates can be allowed at specified intervals. Updated firmware is sent from a server to the phone by using TFTP or HTTP. Security is less of an issue with a firmware upgrade, because firmware does not contain personal information.

You can also configure the parameters in the phone configuration file with XML(cfg.xml) code.

Before you begin

Access the phone administration web page. See Access the Phone Web Interface, on page 34.

Procedure

Step 1 Select **Voice** > **Provisioning**.

Step 2 In the **Firmware Upgrade** section, choose **Yes** from the **Upgrade Enable** parameter.

You can configure this parameter in the phone configuration XML file (cfg.xml) by entering a string in this format:

```
<Upgrade_Enable ua="na">Yes</Upgrade_Enable>
```

Options: Yes and No

Default: Yes

Step 3 Set the **Upgrade Error Retry Delay** parameter in seconds.

The upgrade retry interval (in seconds) applied in case of upgrade failure. The device has a firmware upgrade error timer that activates after a failed firmware upgrade attempt. The timer is initialized with the value in this parameter. The next firmware upgrade attempt occurs when this timer counts down to zero.

You can configure this parameter in the phone configuration XML file (cfg.xml) by entering a string in this format:

```
<Upgrade_Error_Retry_Delay ua="na">3600</Upgrade_Error_Retry_Delay>
```

Default: 3600

:

```
<tftp|http|https>://<ip address>/image/<load name>
```

Step 4 Set the **Upgrade Rule** parameter by entering a firmware upgrade script that defines upgrade conditions and associated firmware URLs. It uses the same syntax as Profile Rule. Enter a script and use the following format to enter the upgrade rule:

```
<tftp|http|https>://<ipaddress>/image/<load name>
```

For example:

tftp://192.168.1.5/image/sip88xx.11-0-0MPP-BN.loads

tftp://192.168.1.5/image/sip78xx.11-0-1MPP-BN.loads

You can configure this parameter in the phone configuration XML file (cfg.xml) by entering a string in this format:

```
<Upgrade_Rule ua="na">http://10.74.10.205:6970/sip8845_65.0104-MPP-9875dev.loads
</Upgrade_Rule>
```

Step 5 Click **Submit All Changes**.

Upgrade Firmware by TFTP, HTTP, or HTTPS

The phone supports firmware upgrade by TFTP, HTTP, or HTTPS.

Note Downgrades to earlier releases may not be available for all devices. For more information, see the release notes for your phone and firmware version.

Before you begin

The firmware load file must be downloaded to an accessible server.

Procedure

Step 1 Use the **`tar -xzvf`** command to untar the tar ball.

Step 2 Copy the folder to a TFTP, HTTP, or HTTPS download directory.

Step 3 Access the phone administration web page. See Access the Phone Web Interface, on page 34.

Step 4 Select **Voice** > **Provisioning**.

Step 5 Find the load filename which ends in **.img** and append it to the valid URL.

Step 6 Click **Submit All Changes**.

Access the Phone Web Interface

If your service provider has disabled access to the configuration utility, contact the service provider before proceeding.

Procedure

Step 1 Ensure that the computer can communicate with the phone. No VPN in use.

Step 2 Start a web browser.

Step 3 Enter the IP address of the phone in your web browser address bar.

- User or Admin Access: **`https://<ip address>:<port>/`**, and then enter the username and password.

For example, `https://10.64.84.147/`

CHAPTER 4

In-House Preprovisioning and Provisioning

In-House Preprovisioning and Provisioning Servers

The service provider preprovisions phones, other than RC units, with a profile. The preprovision profile can comprise a limited set of parameters that resynchronizes the phone. The profile can also comprise a complete set of parameters that the remote server delivers. By default, the phone resynchronizes on power-up and at intervals that are configured in the profile. When the user connects the phone at the customer premises, the device downloads the updated profile and any firmware updates.

This process of preprovisioning, deployment, and remote provisioning can be accomplished in many ways.

Server Preparation and Software Tools

The examples in this chapter require the availability of one or more servers. These servers can be installed and run on a local PC:

- TFTP (UDP port 69)
- syslog (UDP port 514)
- HTTP (TCP port 80)
- HTTPS (TCP port 443).

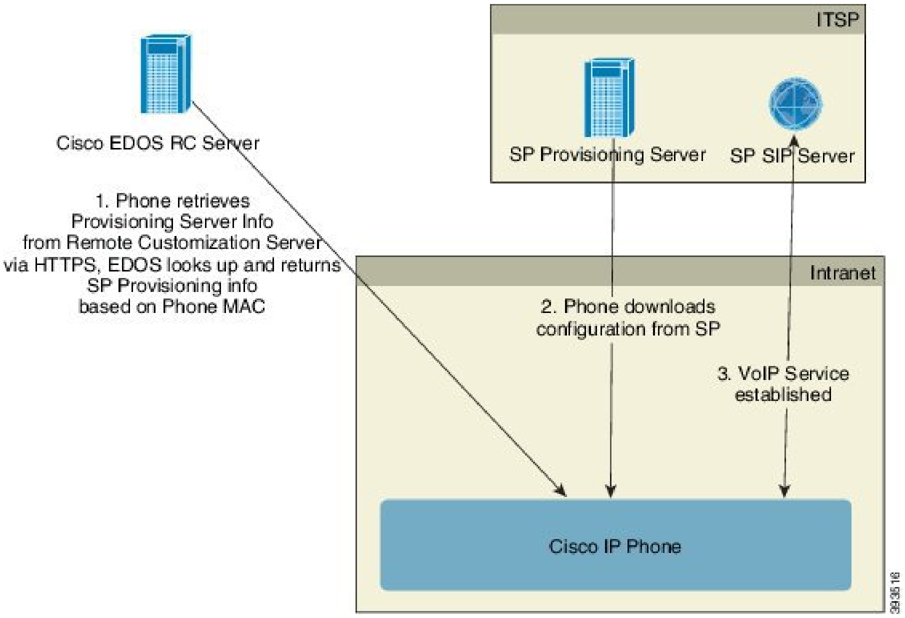
To troubleshoot server configuration, it is helpful to install clients for each type of server on a separate server machine. This practice establishes proper server operation, independent of the interaction with the phones.

We also recommend that you install these software tools:

- To generate configuration profiles, install the open source gzip compression utility.
- For profile encryption and HTTPS operations, install the open source OpenSSL software package.

- To test the dynamic profile generation and one-step remote provisioning using HTTPS, we recommend a scripting language with CGI scripting support. Open source Perl language tools is an example of such a scripting language.
- To verify secure exchanges between provisioning servers and the phones, install an Ethernet packet sniffer (such as the freely downloadable Ethereal/Wireshark). Capture an Ethernet packet trace of the interaction between the phone and the provisioning server. To do so, run the packet sniffer on a PC that is connected to a switch with port mirroring enabled. For HTTPS transactions, you can use the ssldump utility.

Remote Customization (RC) Distribution


ITSP
Cisco EDOS RC Server
SP Provisioning Server
SP SIP Server
1. Phone retrieves Provisioning Server Info from Remote Customization Server via HTTPS, EDOS looks up and returns SP Provisioning info based on Phone MAC
Intranet
2. Phone downloads configuration from SP
3. VoIP Service established
Cisco IP Phone
393518

All phones contact the Cisco EDOS RC server until they are provisioned initially.

In an RC distribution model, a customer purchases a phone that has already been associated with a specific Service Provider in the Cisco EDOS RC Server. The Internet Telephony Service Provider (ITSP) sets up and maintains a provisioning server, and registers their provisioning server information with the Cisco EDOS RC Server.

When the phone is powered on with an internet connection, the customization state for the unprovisioned phone is **Open**. The phone first queries the local DHCP server for provisioning server information and sets the customization state of the phone. If DHCP query is successful, Customization State is set to **Aborted** and RC is not attempted due to DHCP providing the needed provisioning server information.

When a phone connects to a network for the first time or after a factory reset, if there are no DHCP options setup, it contacts a device activation server for zero touch provisioning. New phones will use “activate.cisco.com” instead of “webapps.cisco.com” for provisioning. Phones with firmware release prior to 11.2(1), will continue to use webapps.cisco.com. Cisco recommends that you allow both the domain names through your firewall.

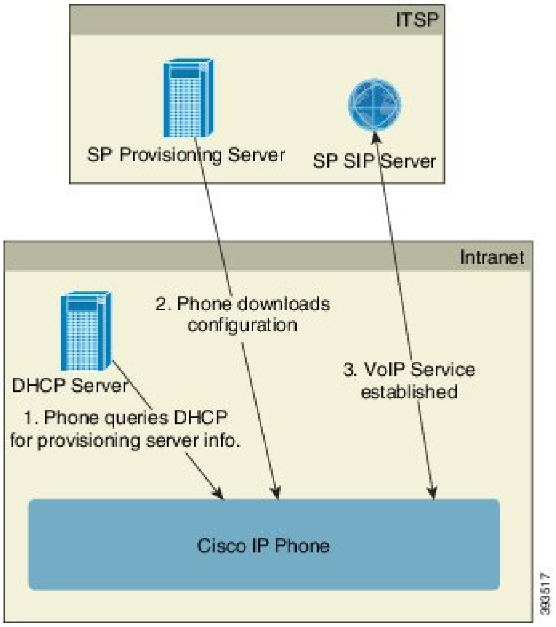
If the DHCP server provisioning fails, the phone queries the Cisco EDOS RC Server and provides its MAC address and model and the Customization State is set to **Pending**. The Cisco EDOS server responds with the associated service provider's provisioning server information including provisioning server URL and the phone's Customization State is set to **Custom Pending**. The phone then performs a resync URL command to retrieve the Service Provider's configuration and, if successful, the Customization State is set to **Acquired**. If queries either for local DHCP server or for EDOS server fails for provisioning, phone retries to onboard over DHCP and EDOS.

If the Cisco EDOS RC Server does not have a service provider associated with the phone, the customization state of the phone is set to **Unavailable**. The phone can be manually configured or an association added for the service provider of the phone to the Cisco EDOS Server.

If a phone is provisioned via either the LCD or Web Configuration Utility, prior to the Customization State becoming **Acquired**, the Customization State is set to **Aborted** and the Cisco EDOS Server will not be queried unless the phone is factory reset.

Once the phone has been provisioned, the Cisco EDOS RC Server is not utilized unless the phone is factory reset.

In-House Device Preprovisioning


ITSP
SP Provisioning Server
SP SIP Server
Intranet
2. Phone downloads configuration
DHCP Server
3. VoIP Service established
1. Phone queries DHCP for provisioning server info.
Cisco IP Phone
393517

With the Cisco factory default configuration, the phone automatically tries to resync to a profile on a TFTP server. A managed DHCP server on a LAN delivers the information about the profile and TFTP server that is configured for preprovisioning to the device. The service provider connects each new phone to the LAN. The phone automatically resyncs to the local TFTP server and initializes its internal state in preparation for deployment. This preprovisioning profile typically includes the URL of a remote provisioning server. The provisioning server keeps the device updated after the device is deployed and connected to the customer network.

The preprovisioned device bar code can be scanned to record its MAC address or serial number before the phone is shipped to the customer. This information can be used to create the profile to which the phone resynchronizes.

Upon receiving the phone, the customer connects it to the broadband link. On power-up, the phone contacts the provisioning server through the URL that is configured through preprovisioning. The phone can thus resync and update the profile and firmware, as necessary.

Related Topics

Retail Distribution, on page 3

TFTP Provisioning, on page 38

Provisioning Server Setup

This section describes setup requirements for provisioning a phone by using various servers and different scenarios. For the purposes of this document and for testing, provisioning servers are installed and run on a local PC. Also, generally available software tools are useful for provisioning the phones.

TFTP Provisioning

The phones support TFTP for both provisioning resync and firmware upgrade operations. When devices are deployed remotely, HTTPS is recommended, but HTTP and TFTP can also be used. This then requires provisioning file encryption to add security, as it offers greater reliability, given NAT and router protection mechanisms. TFTP is useful for the in-house preprovisioning of a large number of unprovisioned devices.

The phone is able to obtain a TFTP server IP address directly from the DHCP server through DHCP option 66. If a Profile_Rule is configured with the filepath of that TFTP server, the device downloads its profile from the TFTP server. The download occurs when the device is connected to a LAN and powered up.

The Profile_Rule provided with the factory default configuration is *ata$PSN*.cfg, where *$PSN* represents the product serial number.

For example, for ATA192-MPP, the filename is ata192.cfg.

For a device with the factory default profile, upon powering up, the device resyncs to this file on the local TFTP server that DHCP option 66 specifies. The filepath is relative to the TFTP server virtual root directory.

Related Topics

In-House Device Preprovisioning, on page 37

Remote Endpoint Control and NAT

The phone is compatible with network address translation (NAT) to access the Internet through a router. For enhanced security, the router might attempt to block unauthorized incoming packets by implementing symmetric NAT, a packet-filtering strategy that severely restricts the packets that are allowed to enter the protected network from the Internet. For this reason, remote provisioning by using TFTP is not recommended.

VoIP can coexist with NAT only when some form of NAT traversal is provided. Configure Simple Traversal of UDP through NAT (STUN). This option requires that the user have:

- A dynamic external (public) IP address from your service
- A computer that is running STUN server software
- An edge device with an asymmetric NAT mechanism

HTTP Provisioning

The phone behaves like a browser that requests web pages from a remote Internet site. This provides a reliable means of reaching the provisioning server, even when a customer router implements symmetric NAT or other protection mechanisms. HTTP and HTTPS work more reliably than TFTP in remote deployments, especially when the deployed units are connected behind residential firewalls or NAT-enabled routers. HTTP and HTTPs are used interchangeably in the following request type descriptions.

Basic HTTP-based provisioning relies on the HTTP GET method to retrieve configuration profiles. Typically, a configuration file is created for each deployed phone, and these files are stored within an HTTP server directory. When the server receives the GET request, it simply returns the file that is specified in the GET request header.

Rather than a static profile, the configuration profile can be generated dynamically by querying a customer database and producing the profile on-the-fly.

When the phone requests a resynch, it can use the HTTP POST method to request the resync configuration data. The device can be configured to convey certain status and identification information to the server within the body of the HTTP POST request. The server uses this information to generate a desired response configuration profile, or to store the status information for later analysis and tracking.

As part of both GET and POST requests, the phone automatically includes basic identifying information in the User-Agent field of the request header. This information conveys the manufacturer, product name, current firmware version, and product serial number of the device.

The following example is the User-Agent request field from an ATA192-MPP:

```
User-Agent: Cisco/ATA192-MPP-11-1-0MPP-16(FCH2118DGQP)
```

User Agent is configurable, and the phone uses this the value if it has not be configured (still at default).

When the phone is configured to resync to a configuration profile by using HTTP, it is recommended that HTTPS be used or the profile be encrypted to protect confidential information. Encrypted profiles that the phone downloads by using HTTP avoid the danger of exposing confidential information that is contained in the configuration profile. This resync mode produces a lower computational load on the provisioning server when compared to using HTTPS.

The phone supports 256-bit AES in CBC mode to decrypt profiles.

Note The phones support HTTP Version 1.0, HTTP Version 1.1, and Chunk Encoding when HTTP Version 1.1 is the negotiated transport protocol.

HTTP Status Code Handling on Resync and Upgrade

The phone supports HTTP response for remote provisioning (Resync). Current phone behavior is categorized in three ways:

- A—Success, where the “Resync Periodic” and “Resync Random Delay” values determine subsequent requests.
- B—Failure when File Not Found or corrupt profile. The “Resync Error Retry Delay” value determines subsequent requests.

- C—Other failure when a bad URL or IP address causes a connection error. The "Resync Error Retry Delay" value determines subsequent requests.

Table 1: Phone Behavior for HTTP Responses

HTTP Status Code	Description	Phone Behavior
301 Moved Permanently	This and future requests should be directed to a new location.	Retry request immediately with new location.
302 Found	Known as Temporarily Moved.	Retry request immediately with new location.
3xx	Other 3xx responses not processed.	C
400 Bad Request	The request cannot be fulfilled due to bad syntax.	C
401 Unauthorized	Basic or digest access authentication challenge.	Immediately retry request with authentication credentials. Maximum 2 retries. Upon failure, the phone behavior is C.
403 Forbidden	Server refuses to respond.	C
404 Not Found	Requested resource not found. Subsequent requests by client are permissible.	B
407 Proxy Authentication Required	Basic or digest access authentication challenge.	Immediately retry request with authentication credentials. Maximum two retries. Upon failure, the phone behavior is C.
4xx	Other client error status codes are not processed.	C
500 Internal Server Error	Generic error message.	Phone behavior is C.
501 Not Implemented	The server does not recognize the request method, or it lacks the ability to fulfill the request.	Phone behavior is C.
502 Bad Gateway	The server is acting as a gateway or proxy and receives an invalid response from the upstream server.	Phone behavior is C.
503 Service Unavailable	The server is currently unavailable (overloaded or down for maintenance). This is a temporary state.	Phone behavior is C.
504 Gateway Timeout	The server behaves as a gateway or proxy and does not receive timely response from the upstream server.	C

HTTP Status Code	Description	Phone Behavior
5xx	Other server error	C

HTTPS Provisioning

The phone supports HTTPS for provisioning for increased security in managing remotely deployed units. Each phone carries a unique SLL Client Certificate (and associated private key), in addition to a Sipura CA server root certificate. The latter allows the phone to recognize authorized provisioning servers, and reject non-authorized servers. On the other hand, the client certificate allows the provisioning server to identify the individual device that issues the request.

For a service provider to manage deployment by using HTTPS, a server certificate must be generated for each provisioning server to which a phone resyncs by using HTTPS. The server certificate must be signed by the Cisco Server CA Root Key, whose certificate is carried by all deployed units. To obtain a signed server certificate, the service provider must forward a certificate signing request to Cisco, which signs and returns the server certificate for installation on the provisioning server.

The provisioning server certificate must contain the Common Name (CN) field, and the FQDN of the host running the server in the subject. It might optionally contain information following the host FQDN, separated by a slash (/) character. The following examples are of CN entries that are accepted as valid by the phone:

```
CN=sprov.callme.com
CN=pv.telco.net/mailto:admin@telco.net
CN=prof.voice.com/info@voice.com
```

In addition to verifying the server certificate, the phone tests the server IP address against a DNS lookup of the server name that is specified in the server certificate.

Get a Signed Server Certificate

The OpenSSL utility can generate a certificate signing request. The following example shows the **openssl** command that produces a 1024-bit RSA public/private key pair and a certificate signing request:

```
openssl req -new -out provserver.csr
```

This command generates the server private key in **privkey.pem** and a corresponding certificate signing request in **provserver.csr**. The service provider keeps the **privkey.pem** secret and submits **provserver.csr** to Cisco for signing. Upon receiving the **provserver.csr** file, Cisco generates **provserver.crt**, the signed server certificate.

Procedure

Step 1 Navigate to https://software.cisco.com/software/cda/home and log in with your CCO credentials.

Note When a phone connects to a network for the first time or after a factory reset, and there are no DHCP options set up, it contacts a device activation server for zero touch provisioning. New phones use “activate.cisco.com” instead of “webapps.cisco.com” for provisioning. Phones with firmware release earlier than 11.2(1) continues to use “webapps.cisco.com”. We recommend that you allow both the domain names through your firewall.

Step 2 Select **Certificate Management**.

On the **Sign CSR** tab, the CSR of the previous step is uploaded for signing.

Step 3 From the **Select Product** drop-down list box, select **SPA1xx firmware 1.3.3 and newer/SPA232D firmware 1.3.3 and newer/SPA5xx firmware 7.5.6 and newer/CP-78xx-3PCC/CP-88xx-3PCC**.

Step 4 In the **CSR File** field, click **Browse** and select the CSR for signing.

Step 5 From the **Sign in Duration** drop-down list box, select the applicable duration (for example, 1 year).

Step 6 Click **Sign Certificate Request**.

Step 7 Select one of the following options to receive the signed certificate:

- **Enter Recipient's Email Address**—If you wish to receive the certificate via email, enter your email address in this field.
- **Download**—If you wish to download the signed certificate, select this option.

Step 8 Click **Submit**.

The signed server certificate is either emailed to the email address previously provided or downloaded.

Multiplatform Phone CA Client Root Certificate

Cisco also provides a Multiplatform Phone Client Root Certificate to the service provider. This root certificate certifies the authenticity of the client certificate that each phone carries. The Multiplatform Phones also support third-party signed certificates such as those provided by Verisign, Cybertrust, and so on.

The unique client certificate that each device offers during an HTTPS session carries identifying information that is embedded in its subject field. This information can be made available by the HTTPS server to a CGI script invoked to handle secure requests.

To determine if a phone carries an individualized certificate, use the $CCERT provisioning macro variable. The variable value expands to either Installed or Not Installed, according to the presence or absence of a unique client certificate. In the case of a generic certificate, it is possible to obtain the serial number of the unit from the HTTP request header in the User-Agent field.

HTTPS servers can be configured to request SSL certificates from connecting clients. If enabled, the server can use the Multiplatform Phone Client Root Certificate that Cisco supplies to verify the client certificate. The server can then provide the certificate information to a CGI for further processing.

The location for certificate storage may vary. For example, in an Apache installation, the file paths for storage of the provisioning server-signed certificate, its associated private key, and the Multiplatform Phone CA client root certificate are as follows:

```
# Server Certificate:
SSLCertificateFile /etc/httpd/conf/provserver.crt

# Server Private Key:
SSLCertificateKeyFile /etc/httpd/conf/provserver.key

# Certificate Authority (CA):
SSLCACertificateFile /etc/httpd/conf/spacroot.crt
```

For specific information, refer to the documentation for an HTTPS server.

The Cisco Client Certificate Root Authority signs each unique certificate. The corresponding root certificate is made available to service providers for client authentication purposes.

Redundant Provisioning Servers

The provisioning server can be specified as an IP address or as a Fully Qualified Domain Name (FQDN). The use of an FQDN facilitates the deployment of redundant provisioning servers. When the provisioning server is identified through an FQDN, the phone attempts to resolve the FQDN to an IP address through DNS. Only DNS A-records are supported for provisioning; DNS SRV address resolution is not available for provisioning. The phone continues to process A-records until a server responds. If no server that is associated with the A-records responds, the phone logs an error to the syslog server.

The ATA can associate with up to 10 DNS A-records for a DNS SRV record.

Syslog Server

If a syslog server is configured on the phone through use of the <Syslog Server> parameters, the resync and upgrade operations send messages to the syslog server. A message can be generated at the start of a remote file request (configuration profile or firmware load), and at the conclusion of the operation (indicating either success or failure).

The logged messages are configured in the following parameters and macro expanded into the actual syslog messages:

- Log_Resync_Request_Msg
- Log_Resync_Success_Msg
- Log_Resync_Failure_Msg

CHAPTER 5

Provisioning Examples

Provisioning Examples Overview

This chapter provides example procedures for transferring configuration profiles between the phone and the provisioning server.

For information about creating configuration profiles, refer to Provisioning Formats, on page 11.

Basic Resync

This section demonstrates the basic resync functionality of the phones.

TFTP Resync

The phone supports multiple network protocols for retrieving configuration profiles. The most basic profile transfer protocol is TFTP (RFC1350). TFTP is widely used for the provisioning of network devices within private LAN networks. Although not recommended for the deployment of remote endpoints across the Internet, TFTP can be convenient for deployment within small organizations, for in-house preprovisioning, and for development and testing. See In-House Device Preprovisioning, on page 37 for more information on in-house preprovisioning. In the following procedure, a profile is modified after downloading a file from a TFTP server.

Procedure

Step 1 Within a LAN environment, connect a PC and a phone to a hub, switch, or small router.

Step 2 Connect an analog phone to the Phone 1 port of the ATA

Step 3 On the PC, install and activate a TFTP server.

Step 4 Use a text editor to create a configuration profile that sets the value for GPP_A to 12345678 as shown in the example.

```
<flat-profile>
  <GPP_A> 12345678
  </GPP_A>
</flat-profile>
```

Step 5 Save the profile with the name `basic.txt` in the root directory of the TFTP server.

You can verify that the TFTP server is properly configured: request the `basic.txt` file by using a TFTP client other than the phone. Preferably, use a TFTP client that is running on a separate host from the provisioning server.

Step 6 Using an analog phone, obtain the IP address of the ATA (IVR menu ****** 110 #**).

If the configuration has been modified since it was manufactured, perform a factory reset on the phone by using the IVR RESET option (****** 73738#**).

Step 7 Open the PC web browser. For example, if the IP address of the device is 192.168.1.100:

```
http://192.168.1.100
```

Step 8 Select the **Voice** > **Provisioning** tab, and inspect the values of the general purpose parameters GPP_A through GPP_P. These should be empty.

Step 9 Resync the test phone to the `basic.txt` configuration profile by opening the resync URL in a web browser window.

If the IP address of the TFTP server is 192.168.1.200, the command should be similar to the following example:

```
http://192.168.1.100/admin/resync?tftp://192.168.1.200/basic.txt
```

When the phone receives this command, the device at address 192.168.1.100 requests the file `basic.txt` from the TFTP server at IP address 192.168.1.200. The phone then parses the downloaded file and updates the GPP_A parameter with the value 12345678.

Step 10 Verify that the parameter was correctly updated: Refresh the configuration page on the PC web browser and select the **Voice** > **Provisioning** tab.

The GPP_A parameter should now contain the value 12345678.

Use Syslog to Log Messages

A phone can be configured to send logging messages to a syslog server over UDP, including messages related to provisioning. This server is identified in the web server administration (**Admin Login** > **Administration** > **Log** > **Debug Log Settings**, IPv4 Address field). Configure the syslog server IP address into the device and observe the messages that are generated during the remaining procedures.

To get the information, you can access the phone Web interface, select **Info** > **Debug Info** > **Control Logs** and click **messages**.

Before you begin

Procedure

Step 1 Install and activate a syslog server on the local PC.

Step 2 Program the PC IP address into the Syslog_Server_IP parameter of the profile and submit the change:

```
<Syslog_Server_IP>192.168.1.210</Syslog_Server_IP>
```

Step 3 Click the **System** tab and enter the value of your local syslog server into the Syslog_Server parameter.

Step 4 Repeat the resync operation as described in TFTP Resync, on page 45.

The device generates two syslog messages during the resync. The first message indicates that a request is in progress. The second message marks success or failure of the resync.

Step 5 Verify that your syslog server received messages similar to the following:

```
ATA192-MPP 00:0e:08:ab:cd:ef -- Successful resync tftp://192.168.1.200/basic.txt
```

The contents of these messages can be configured by using the following parameters:

- Log_Resync_Request_Msg
- Log_Resync_Success_Msg
- Log_Resync_Failure_Msg

If any of these parameters are cleared, the corresponding syslog message is not generated.

Resync a Device Automatically

A device can resync periodically to the provisioning server to ensure that any profile changes made on the server are propagated to the endpoint device (as opposed to sending an explicit resync request to the endpoint).

To cause the phone to periodically resync to a server, a configuration profile URL is defined by using the Profile_Rule parameter, and a resync period is defined by using the Resync_Periodic parameter.

Before you begin

Access the phone administration web page. See Access the Phone Web Interface, on page 34.

Procedure

Step 1 Select **Voice** > **Provisioning**.

Step 2 Define the Profile_Rule parameter. This example assumes a TFTP server IP address of 192.168.1.200.

Step 3 In the **Resync Periodic** field, enter a small value for testing, such as **30** seconds.

Step 4 Click **Submit all Changes**.

With the new parameter settings, the phone resyncs twice a minute to the configuration file that the URL specifies.

Step 5 Observe the resulting messages in the syslog trace (as described in the Use Syslog to Log Messages, on page 46 section).

Step 6 Ensure that the **Resync On Reset** field is set to **Yes**.

```
<Resync_On_Reset>Yes</Resync_On_Reset>
```

Step 7 Power cycle the phone to force it to resync to the provisioning server.

If the resync operation fails for any reason, such as if the server is not responding, the unit waits (for the number of seconds configured in **Resync Error Retry Delay**) before it attempts to resync again. If **Resync Error Retry Delay** is zero, the phone does not try to resync after a failed resync attempt.

Step 8 (Optional) Set the value of **Resync Error Retry Delay** field to a small number, such as **30**.

```
<Resync_Error_Retry_Delay>30</Resync_Error_Retry_Delay>
```

Step 9 Disable the TFTP server, and observe the results in the syslog output.

Unique Profiles, Macro Expansion, and HTTP

In a deployment where each phone must be configured with distinct values for some parameters, such as User_ID or Display_Name, the service provider can create a unique profile for each deployed device and host those profiles on a provisioning server. Each phone, in turn, must be configured to resync to its own profile according to a predetermined profile naming convention.

The profile URL syntax can include identifying information that is specific to each phone, such as MAC address or serial number, by using the macro expansion of built-in variables. Macro expansion eliminates the need to specify these values in multiple locations within each profile.

A profile rule undergoes macro expansion before the rule is applied to the phone. The macro expansion controls a number of values, for example:

- $MA expands to the unit 12-digit MAC address (using lower case hex digits). For example, 000e08abcdef.
- $SN expands to the unit serial number. For example, 88012BA01234.

Other values can be macro expanded in this way, including all the general purpose parameters, GPP_A through GPP_P. An example of this process can be seen in TFTP Resync, on page 45. Macro expansion is not limited to the URL file name, but can also be applied to any portion of the profile rule parameter. These parameters are referenced as $A through $P. For a complete list of variables that are available for macro expansion, see Macro Expansion Variables, on page 68.

In this exercise, a profile specific to a phone is provisioned on a TFTP server.

Provision a Specific IP Phone Profile on a TFTP Server

Procedure

Step 1 Obtain the MAC address of the phone from its product label. (The MAC address is the number, using numbers and lower–case hex digits, such as 000e08aabbcc.

Step 2 Copy the `basic.txt` configuration file (described in TFTP Resync, on page 45) to a new file named `ataxxxx.cfg` (replacing `xxxx` with the `macaddress` with the MAC address of the phone).

Step 3 Move the new file in the virtual root directory of the TFTP server.

Step 4 Access the phone administration web page. See Access the Phone Web Interface, on page 34.

Step 5 Select **Voice** > **Provisioning**.

Step 6 Enter `tftp://192.168.1.200/ata$MA.cfg` in the **Profile Rule** field.

```
<Profile_Rule>
  tftp://192.168.1.200/ata$MA.cfg
</Profile_Rule>
```

Step 7 Click **Submit All Changes**. This causes an immediate reboot and resync.

When the next resync occurs, the phone retrieves the new file by expanding the $MA macro expression into its MAC address.

HTTP GET Resync

HTTP provides a more reliable resync mechanism than TFTP because HTTP establishes a TCP connection and TFTP uses the less reliable UDP. In addition, HTTP servers offer improved filtering and logging features compared to TFTP servers.

On the client side, the phone does not require any special configuration setting on the server to be able to resync by using HTTP. The Profile_Rule parameter syntax for using HTTP with the GET method is similar to the syntax that is used for TFTP. If a standard web browser can retrieve a profile from your HTTP server, the phone should be able to do so as well.

Resync with HTTP GET

Procedure

Step 1 Install an HTTP server on the local PC or other accessible host.

The open source Apache server can be downloaded from the internet.

Step 2 Copy the `basic.txt` configuration profile (described in TFTP Resync, on page 45) onto the virtual root directory of the installed server.

Step 3 To verify proper server installation and file access to `basic.txt`, access the profile with a web browser.

Step 4 Modify the Profile_Rule of the test phone to point to the HTTP server in place of the TFTP server, so as to download its profile periodically.

For example, assuming the HTTP server is at 192.168.1.300, enter the following value:

```
<Profile_Rule>
http://192.168.1.200/basic.txt
</Profile_Rule>
```

Step 5 Click **Submit All Changes**. This causes an immediate reboot and resync.

Step 6 Observe the syslog messages that the phone sends. The periodic resyncs should now be obtaining the profile from the HTTP server.

Step 7 In the HTTP server logs, observe how information that identifies the test phone appears in the log of user agents.

This information should include the manufacturer, product name, current firmware version, and serial number.

Provisioning Through Cisco XML

For each of the phones, designated as xxxx here, you can provision through Cisco XML functions.

You can send an XML object to the phone by a SIP Notify packet or an HTTP Post to the CGI interface of the phone: `http://IPAddressPhone/CGI/Execute`.

The CP-xxxx-3PCC extends the Cisco XML feature to support provisioning via an XML object:

```
<CP-xxxx-3PCCExecute>
        <ExecuteItem URL=Resync:[profile-rule]/>
</CP-xxxx-3PCCExecute>
```

After the phone receives the XML object, it downloads the provisioning file from [profile-rule]. This rule uses macros to simplify the development of the XML services application.

URL Resolution with Macro Expansion

Subdirectories with multiple profiles on the server provide a convenient method for managing a large number of deployed devices. The profile URL can contain:

- A provisioning server name or an explicit IP address. If the profile identifies the provisioning server by name, the phone performs a DNS lookup to resolve the name.
- A nonstandard server port that is specified in the URL by using the standard syntax `:port` following the server name.
- The subdirectory of the server virtual root directory where the profile is stored, specified by using standard URL notation and managed by macro expansion.

For example, the following Profile_Rule requests the profile file ($PN.cfg), in the server subdirectory `/cisco/config`, from the TFTP server that is running on host prov.telco.com listening for a connection on port 6900:

```
<Profile_Rule>
tftp://prov.telco.com:6900/cisco/config/$PN.cfg
</Profile_Rule>
```

A profile for each phone can be identified in a general purpose parameter, with its value referred within a common profile rule by using macro expansion.

For example, assume GPP_B is defined as `Dj6Lmp23Q`.

The Profile_Rule has the value:

```
tftp://prov.telco.com/cisco/$B/$MA.cfg
```

When the device resyncs and the macros are expanded, the phone with a MAC address of 000e08012345 requests the profile with the name that contains the device MAC address at the following URL:

```
tftp://prov.telco.com/cisco/Dj6Lmp23Q/000e08012345.cfg
```

Secure HTTPS Resync

These mechanisms are available on the phone for resyncing by using a secure communication process:

- Basic HTTPS Resync
- HTTPS with Client Certificate Authentication
- HTTPS Client Filtering and Dynamic Content

Basic HTTPS Resync

HTTPS adds SSL to HTTP for remote provisioning so that the:

- The phone can authenticate the provisioning server.
- Provisioning server can authenticate the phone.
- Confidentiality of information exchanged between the phone and the provisioning server is ensured.

SSL generates and exchanges secret (symmetric) keys for each connection between the phone and the server, using public/private key pairs that are pre-installed in the phone and the provisioning server.

On the client side, the phone does not require any special configuration setting on the server to be able to resync using HTTPS. The Profile_Rule parameter syntax for using HTTPS with the GET method is similar to the syntax that is used for HTTP or TFTP. If a standard web browser can retrieve a profile from a your HTTPS server, the phone should be able to do so as well.

In addition to installing a HTTPS server, a SSL server certificate that Cisco signs must be installed on the provisioning server. The devices cannot resync to a server that is using HTTPS unless the server supplies a Cisco-signed server certificate. Instructions for creating signed SSL Certificates for Voice products can be found at https://supportforums.cisco.com/docs/DOC-9852.

Related Topics

Secure HTTPS Resync, on page 51

Authenticate with Basic HTTPS Resync

Procedure

Step 1 Install an HTTPS server on a host whose IP address is known to the network DNS server through normal hostname translation.

The open source Apache server can be configured to operate as an HTTPS server when installed with the open source mod_ssl package.

Step 2 Generate a server Certificate Signing Request for the server. For this step, you might need to install the open source OpenSSL package or equivalent software. If using OpenSSL, the command to generate the basic CSR file is as follows:

```
openssl req -new -out provserver.csr
```

This command generates a public/private key pair, which is saved in the `privkey.pem` file.

Step 3 Submit the CSR file (provserver.csr) to Cisco for signing.

A signed server certificate is returned (provserver.cert) along with a Sipura CA Client Root Certificate, spacroot.cert.

See https://supportforums.cisco.com/docs/DOC-9852 for more information

Step 4 Store the signed server certificate, the private key pair file, and the client root certificate in the appropriate locations on the server.

In the case of an Apache installation on Linux, these locations are typically as follows:

```
# Server Certificate:
SSLCertificateFile /etc/httpd/conf/provserver.cert
# Server Private Key:
SSLCertificateKeyFile /etc/httpd/conf/pivkey.pem
# Certificate Authority:
SSLCACertificateFile /etc/httpd/conf/spacroot.cert
```

Step 5 Restart the server.

Step 6 Copy the `basic.txt` configuration file (described in TFTP Resync, on page 45) onto the virtual root directory of the HTTPS server.

Step 7 Verify proper server operation by downloading `basic.txt` from the HTTPS server by using a standard browser from the local PC.

Step 8 Inspect the server certificate that the server supplies.

The browser probably does not recognize the certificate as valid unless the browser has been pre-configured to accept Cisco as a root CA. However, the phones expect the certificate to be signed this way.

Modify the Profile_Rule of the test device to contain a reference to the HTTPS server, for example:

```
<Profile_Rule>
https://my.server.com/basic.txt
</Profile_Rule>
```

This example assumes the name of the HTTPS server is **`my.server.com`**.

Step 9 Click **Submit All Changes**.

Step 10 Observe the syslog trace that the phone sends.

The syslog message should indicate that the resync obtained the profile from the HTTPS server.

Step 11 (Optional) Use an Ethernet protocol analyzer on the phone subnet to verify that the packets are encrypted.

In this exercise, client certificate verification was not enabled. The connection between the phone and server is encrypted. However, the transfer is not secure because any client can connect to the server and request the file, given knowledge of the file name and directory location. For secure resync, the server must also authenticate the client, as demonstrated in the exercise described in HTTPS with Client Certificate Authentication, on page 53.

HTTPS with Client Certificate Authentication

In the factory default configuration, the server does not request an SSL client certificate from a client. Transfer of the profile is not secure because any client can connect to the server and request the profile. You can edit the configuration to enable client authentication; the server requires a client certificate to authenticate the phone before it accepts a connection request.

Because of this requirement, the resync operation cannot be independently tested by using a browser that lacks the proper credentials. The SSL key exchange within the HTTPS connection between the test phone and the server can be observed with the ssldump utility. The utility trace shows the interaction between client and server.

Related Topics

Secure HTTPS Resync, on page 51

Authenticate HTTPS with Client Certificate

Procedure

Step 1 Enable client certificate authentication on the HTTPS server.

Step 2 In Apache (v.2), set the following in the server configuration file:

```
SSLVerifyClient  require
```

Also, ensure that the spacroot.cert has been stored as shown in the Basic HTTPS Resync, on page 51 exercise.

Step 3 Restart the HTTPS server and observe the syslog trace from the phone.

Each resync to the server now performs symmetric authentication, so that both the server certificate and the client certificate are verified before the profile is transferred.

Step 4 Use ssldump to capture a resync connection between the phone and the HTTPS server.

If client certificate verification is properly enabled on the server, the ssldump trace shows the symmetric exchange of certificates (first server-to-client, then client-to-server) before the encrypted packets that contain the profile.

With client authentication enabled, only a phone with a MAC address that matches a valid client certificate can request the profile from the provisioning server. The server rejects a request from an ordinary browser or other unauthorized device.

Configure a HTTPS Server for Client Filtering and Dynamic Content

If the HTTPS server is configured to require a client certificate, the information in the certificate identifies the resyncing phone and supplies it with the correct configuration information.

The HTTPS server makes the certificate information available to CGI scripts (or compiled CGI programs) that are invoked as part of the resync request. For the purpose of illustration, this exercise uses the open source Perl scripting language, and assumes that Apache (v.2) is used as the HTTPS server.

Procedure

Step 1 Install Perl on the host that is running the HTTPS server.

Step 2 Generate the following Perl reflector script:

```
#!/usr/bin/perl -wT
use strict;
print “Content-Type: text/plain\n\n”;
print “<flat-profile><GPP_D>”;

print “OU=$ENV{‘SSL_CLIENT_I_DN_OU’},\n”;
print “L=$ENV{‘SSL_CLIENT_I_DN_L’},\n”;
print “S=$ENV{‘SSL_CLIENT_I_DN_S’}\n”;
print “</GPP_D></flat-profile>”;
```

Step 3 Save this file with the file name `reflect.pl`, with executable permission (chmod 755 on Linux), in the CGI scripts directory of the HTTPS server.

Step 4 Verify accessibility of CGI scripts on the server (that is, `/cgi-bin/`…).

Step 5 Modify the Profile_Rule on the test device to resync to the reflector script, as in the following example:

```
https://prov.server.com/cgi-bin/reflect.pl?
```

Step 6 Click **Submit All Changes**.

Step 7 Observe the syslog trace to ensure a successful resync.

Step 8 Access the phone administration web page. See Access the Phone Web Interface, on page 34.

Step 9 Select **Voice** > **Provisioning**.

Step 10 Verify that the GPP_D parameter contains the information that the script captured.

This information contains the product name, MAC address, and serial number if the test device carries a unique certificate from the manufacturer. The information contains generic strings if the unit was manufactured before firmware release 2.0.

A similar script can determine information about the resyncing device and then provide the device with appropriate configuration parameter values.

HTTPS Certificates

The phone provides a reliable and secure provisioning strategy that is based on HTTPS requests from the device to the provisioning server. Both a server certificate and a client certificate are used to authenticate the phone to the server and the server to the phone.

In addition to Cisco issued certifications, the phone also accepts server certificates from a set of commonly used SSL certificate providers.

To use HTTPS with the phone, you must generate a Certificate Signing Request (CSR) and submit it to Cisco. The phone generates a certificate for installation on the provisioning server. The phone accepts the certificate when it seeks to establish an HTTPS connection with the provisioning server.

HTTPS Methodology

HTTPS encrypts the communication between a client and a server, thus protecting the message contents from other network devices. The encryption method for the body of the communication between a client and a server is based on symmetric key cryptography. With symmetric key cryptography, a client and a server share a single secret key over a secure channel that is protected by Public/Private key encryption.

Messages encrypted by the secret key can only be decrypted by using the same key. HTTPS supports a wide range of symmetric encryption algorithms. The phone implements up to 256-bit symmetric encryption, using the American Encryption Standard (AES), in addition to 128-bit RC4.

HTTPS also provides for the authentication of a server and a client engaged in a secure transaction. This feature ensures that a provisioning server and an individual client cannot be spoofed by other devices on the network. This capability is essential in the context of remote endpoint provisioning.

Server and client authentication is performed by using public/private key encryption with a certificate that contains the public key. Text that is encrypted with a public key can be decrypted only by its corresponding private key (and vice versa). The phone supports the Rivest-Shamir-Adleman (RSA) algorithm for public/private key cryptography.

SSL Server Certificate

Each secure provisioning server is issued a secure sockets layer (SSL) server certificate that Cisco signs directly. The firmware that runs on the phone recognizes only a Cisco certificate as valid. When a client connects to a server by using HTTPS, it rejects any server certificate that is not signed by Cisco.

This mechanism protects the service provider from unauthorized access to the phone, or any attempt to spoof the provisioning server. Without such protection, an attacker might be able to reprovision the phone, to gain configuration information, or to use a different VoIP service. Without the private key that corresponds to a valid server certificate, the attacker is unable to establish communication with a phone.

Obtain a Server Certificate

Procedure

Step 1 Contact a Cisco support person who will work with you on the certificate process. If you are not working with a specific support person, email your request to ciscosb-certadmin@cisco.com.

Step 2 Generate a private key that will be used in a CSR (Certificate Signing Request). This key is private and you do not need to provide this key to Cisco support. Use open source “openssl” to generate the key. For example:

```
openssl genrsa -out <file.key> 1024
```

Step 3 Generate a CSR that contains fields that identify your organization and location. For example:

```
openssl req -new -key <file.key> -out <file.csr>
```

You must have the following information:

- Subject field—Enter the Common Name (CN) that must be an FQDN (Fully Qualified Domain Name) syntax. During SSL authentication handshake, the phone verifies that the certificate it receives is from the machine that presented it.
- Server hostname—For example, provserv.domain.com.
- Email address—Enter an email address so that customer support can contact you if needed. This email address is visible in the CSR.

Step 4 Email the CSR (in zip file format) to the Cisco support person or to ciscosb-certadmin@cisco.com. The certificate is signed by Cisco. Cisco sends the certificate to you to install on your system.

Client Certificate

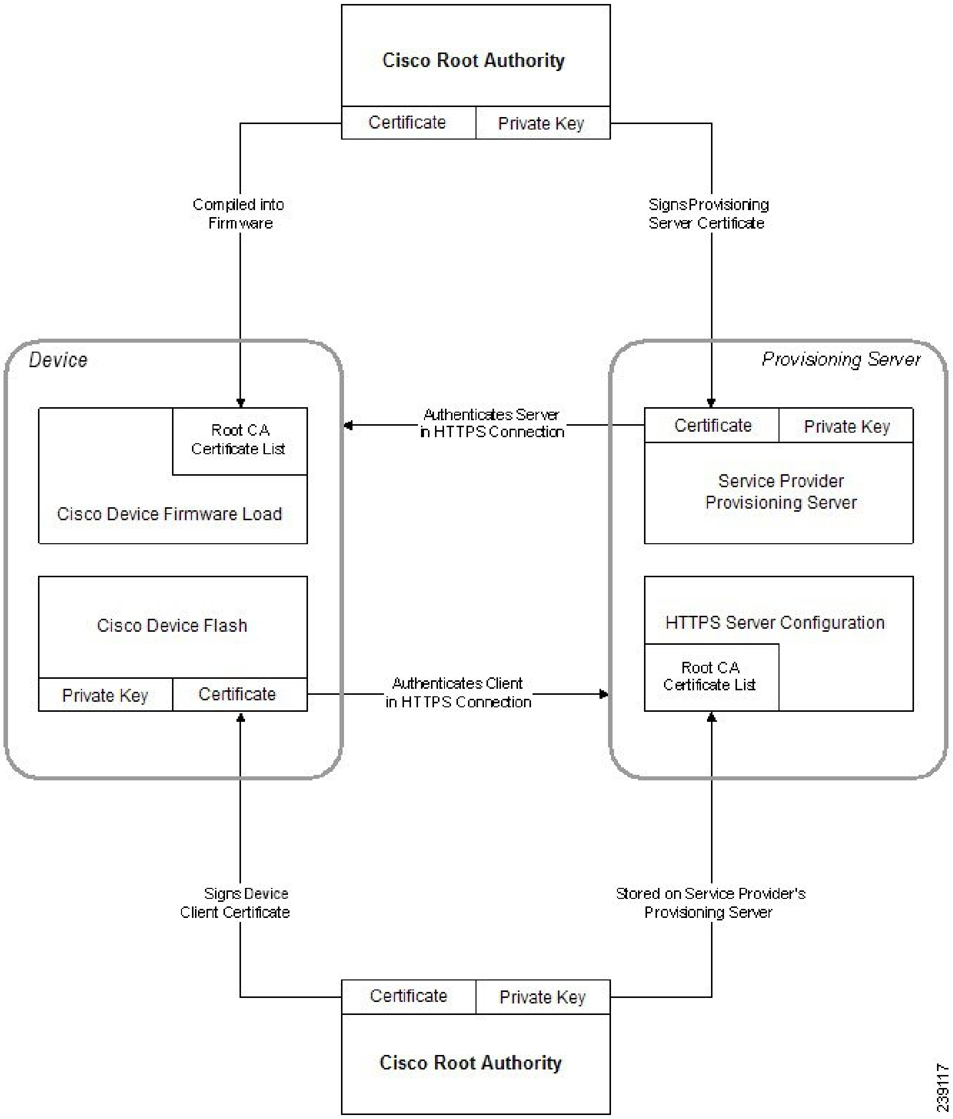
In addition to a direct attack on a phone, an attacker might attempt to contact a provisioning server through a standard web browser or another HTTPS client to obtain the configuration profile from the provisioning server. To prevent this kind of attack, each phone also carries a unique client certificate, signed by Cisco, that includes identifying information about each individual endpoint. A certificate authority root certificate that is capable of authenticating the device client certificate is given to each service provider. This authentication path allows the provisioning server to reject unauthorized requests for configuration profiles.

Certificate Structure

The combination of a server certificate and a client certificate ensures secure communication between a remote phone and its provisioning server. The figure below illustrates the relationship and placement of certificates, public/private key pairs, and signing root authorities, among the Cisco client, the provisioning server, and the certification authority.

The upper half of the diagram shows the Provisioning Server Root Authority that is used to sign the individual provisioning server certificate. The corresponding root certificate is compiled into the firmware, which allows the phone to authenticate authorized provisioning servers.

Figure 2: Certificate Authority Flow


Cisco Root Authority
Certificate
Private Key
Compiled into Firmware
Signs Provisioning Server Certificate
Device
Provisioning Server
Root CA Certificate List
Cisco Device Firmware Load
Authenticates Server in HTTPS Connection
Certificate
Private Key
Service Provider Provisioning Server
Cisco Device Flash
HTTPS Server Configuration
Root CA Certificate List
Private Key
Certificate
Authenticates Client in HTTPS Connection
Signs Device Client Certificate
Stored on Service Provider's Provisioning Server
Certificate
Private Key
Cisco Root Authority
239117

Configure a Custom Certificate Authority

Digital certificates can be used to authenticate network devices and users on the network. They can be used to negotiate IPSec sessions between network nodes.

A third party uses a Certificate Authority certificate to validate and authenticate two or more nodes that are attempting to communicate. Each node has a public and private key. The public key encrypts data. The private key decrypts data. Because the nodes have obtained their certificates from the same source, they are assured of their respective identities.

The device can use digital certificates provided by a third-party Certificate Authority (CA) to authenticate IPSec connections.

The phones support a set of preloaded Root Certificate Authority embedded in the firmware:

- Cisco Small Business CA Certificate

- CyberTrust CA Certificate
- Verisign CA certificate
- Sipura Root CA Certificate
- Linksys Root CA Certificate

Before you begin

Access the phone administration web page. See Access the Phone Web Interface, on page 34.

Procedure

Step 1 Select **Info** > **Status**.

Step 2 Scroll to **Custom CA Status** and see the following fields:

- Custom CA Provisioning Status—Indicates the provisioning status.
  - Last provisioning succeeded on mm/dd/yyyy HH:MM:SS; or
  - Last provisioning failed on mm/dd/yyyy HH:MM:SS
- Custom CA Info—Displays information about the custom CA.
  - Installed—Displays the "CN Value," where "CN Value" is the value of the CN parameter for the Subject field in the first certificate.
  - Not Installed—Displays if no custom CA certificate is installed.

Profile Management

This section demonstrates the formation of configuration profiles in preparation for downloading. To explain the functionality, TFTP from a local PC is used as the resync method, although HTTP or HTTPS can be used as well.

Compress an Open Profile with Gzip

A configuration profile in XML format can become quite large if the profile specifies all parameters individually. To reduce the load on the provisioning server, the phone supports compression of the XML file, by using the deflate compression format that the gzip utility (RFC 1951) supports.

Note Compression must precede encryption for the phone to recognize a compressed and encrypted XML profile.

For integration into customized back-end provisioning server solutions, the open source zlib compression library can be used in place of the standalone gzip utility to perform the profile compression. However, the phone expects the file to contain a valid gzip header.

Procedure

Step 1 Install gzip on the local PC.

Step 2 Compress the `basic.txt` configuration profile (described in TFTP Resync, on page 45) by invoking gzip from the command line:

```
gzip basic.txt
```

This generates the deflated file `basic.txt.gz`.

Step 3 Save the `basic.txt.gz` file in the TFTP server virtual root directory.

Step 4 Modify the Profile_Rule on the test device to resync to the deflated file in place of the original XML file, as shown in the following example:

```
tftp://192.168.1.200/basic.txt.gz
```

Step 5 Click **Submit All Changes.**

Step 6 Observe the syslog trace from the phone.

Upon resync, the phone downloads the new file and uses it to update its parameters.

Related Topics

Open Profile Compression, on page 15

Encrypt a Profile with OpenSSL

A compressed or uncompressed profile can be encrypted (however, a file must be compressed before it is encrypted). Encryption is useful when the confidentiality of the profile information is of particular concern, such as when TFTP or HTTP is used for communication between the phone and the provisioning server.

The phone supports symmetric key encryption by using the 256-bit AES algorithm. This encryption can be performed by using the open source OpenSSL package.

Procedure

Step 1 Install OpenSSL on a local PC. This might require that the OpenSSL application be recompiled to enable AES.

Step 2 Using the `basic.txt` configuration file (described in TFTP Resync, on page 45), generate an encrypted file with the following command:

```
>openssl enc -aes-256-cbc -k MyOwnSecret -in basic.txt -out basic.cfg
```

The compressed basic.txt.gz file that was created in Compress an Open Profile with Gzip, on page 58 also can be used, because the XML profile can be both compressed and encrypted.

Step 3 Store the encrypted basic.cfg file in the TFTP server virtual root directory.

Step 4 Modify the Profile_Rule on the test device to resync to the encrypted file in place of the original XML file. The encryption key is made known to the phone with the following URL option:

```
[--key MyOwnSecret ] tftp://192.168.1.200/basic.cfg
```

Step 5 Click **Submit All Changes**.

Step 6 Observe the syslog trace from the phone.

Upon resync, the phone downloads the new file and uses it to update its parameters.

Related Topics

AES-256-CBC Encryption, on page 16

Create Partitioned Profiles

A phone downloads multiple separate profiles during each resync. This practice allows management of different kinds of profile information on separate servers and maintenance of common configuration parameter values that are separate from account specific values.

Procedure

Step 1 Create a new XML profile, basic2.txt, that specifies a value for a parameter that makes it distinct from the earlier exercises. For instance, to the basic.txt profile, add the following:

```
<GPP_B>ABCD</GPP_B>
```

Step 2 Store the basic2.txt profile in the virtual root directory of the TFTP server.

Step 3 Leave the first profile rule from the earlier exercises in the folder, but configure the second profile rule (Profile_Rule_B) to point to the new file:

```
<Profile_Rule_B>tftp://192.168.1.200/basic2.txt
</Profile_Rule_B>
```

Step 4 Click **Submit All Changes**.

The phone now resyncs to both the first and second profiles, in that order, whenever a resync operation is due.

Step 5 Observe the syslog trace to confirm the expected behavior.

Manage Provisioning with Parameter Name Aliases

When generating an XML profile for the ATA, it might be convenient to assign names to certain configuration parameters that are different from the canonical names recognized by the ATA. For example, a customer account database might generate XML element tags for a customer telephone number and SIP registration password with names, such as SIP-number and SIP-password. These names can be mapped to the canonical names (User_ID_1_ and Password_1_) before being applied to Line1.

In many instances, the back-end provisioning solution used by the service provider can perform this mapping. However, the ATA itself can remap the parameter names internally. To do this, an alias map is defined and stored in one of the general purpose provisioning parameters. Then, the profile rule which invokes the resync is directed to remap the non-canonical XML elements as specified by the alias map.

Procedure

Step 1 Generate a profile named customer.XML containing the proprietary customeraccount XML form indicated in the following example:

```
<customer-account>
<SIP-number> 17775551234</SIP-number>
<SIP-password> 512835907884</SIP-password>
</customer-account>
```

Step 2 Store the profile in the TFTP server virtual root directory.

Step 3 Open the web interface on the device to **Voice** > **Provisioning**, and edit GPP_A to contain the alias map. Do not enter new lines through the web interface, instead simply enter each alias consecutively:

```
/customer-account/SIP-number = /flat-profile/User_ID_1_ ;
/customer-account/SIP-password = /flat-profile/Password_1_ ;
```

Step 4 Edit the Profile_Rule to point to the new XML profile, and specify the alias map as a URL option, as follows:

```
[--alias a ] tftp://192.168.1.200/customer.xml
```

Step 5 Click **Submit All Changes**.

When the ATA resyncs, it receives the XML profile, remaps the elements, as indicated by the alias map, and populates the User_ID_1_ and Password_1_ parameters.

Step 6 View the Line 1 tab to verify the new configuration.

Note The ATA supports alias remapping of a limited number of parameters. It is not meant to rename all parameters in its configuration.

CHAPTER 6

Provisioning Parameters

Configuration Parameters Overview

This chapter describes the provisioning parameters that can be used in configuration profile scripts.

Configuration Profile Parameters

The following table defines the function and usage of each parameter in the Configuration Profile Parameters section under the Provisioning tab.

Parameter Name	Description and Default Value
Provision_Enable	Controls all resync actions independently of firmware upgrade actions. Set to yes to enable remote provisioning. The default value is Yes.
Resync_On_Reset	Triggers a resync after every reboot except for reboots caused by parameter updates and firmware upgrades. The default value is Yes.

Parameter Name	Description and Default Value
Resync_Random_Delay	Prevents an overload of the provisioning server when a large number of devices power-on simultaneously and attempt initial configuration. This delay is effective only on the initial configuration attempt, following a device power-on or reset. The parameter is the maximum time interval that the device waits before making contact with the provisioning server. The actual delay is a pseudo-random number between zero and this value. This parameter is in units of 20 seconds; the default value of 3 represents 60 seconds. This feature is disabled when this parameter is set to zero. The default value is 2 (40 seconds).
Resync At	The hour and minutes (HHmm) that the device resyncs with the provisioning server. The default value is empty. If the value is invalid, the parameter is ignored. If this parameter is set with a valid value, the Resync_Periodic parameter is ignored.
Resync_At_Random_Delay	Prevents an overload of the provisioning server when a large number of devices power-on simultaneously. To avoid flooding resync requests to the server from multiple phones, the phone resyncs in the range between the hours and minutes, and the hours and minutes plus the random delay (hhmm, hhmm+random_delay). For example, if the random delay = (Resync_At_Random_Delay + 30)/60 minutes. The input value in seconds is converted to minutes, rounding up to the next minute to calculate the final random_delay interval. This feature is disabled when this parameter is set to zero. The default value is 600 seconds (10 minutes). If the parameter value is set to less than 600, the default value is used.
Resync_Periodic	The time interval between periodic resyncs with the provisioning server. The associated resync timer is active only after the first successful sync with the server. Set this parameter to zero to disable periodic resyncing. The default value is 3600 seconds.
Resync_Error_Retry_Delay	Resync retry interval (in seconds) applied in case of resync failure. The device has an error retry timer that activates if the previous attempt to sync with the provisioning server fails. The device waits to contact the server again until the timer counts down to zero. This parameter is the value that is initially loaded into the error retry timer. If this parameter is set to zero, the device does not try to resync with the provisioning server following a failed attempt. The default value is 3600 seconds.

Parameter Name	Description and Default Value
Forced_Resync_Delay	Maximum delay (in seconds) the ATA waits before performing a resync. The device does not resync while one of its phone lines is active. Because a resync can take several seconds, it is desirable to wait until the device has been idle for an extended period before resyncing. This allows a user to make calls in succession without interruption. The device has a timer that begins counting down when all of its lines become idle. This parameter is the initial value of the counter. Resync events are delayed until this counter decrements to zero. The default value is 14,400 seconds.
Resync_From_SIP	Enables a resync to be triggered via a SIP NOTIFY message. The default value is Yes.
Resync_After_Upgrade_Attempt	Triggers a resync after every firmware upgrade attempt. The default value is Yes.
Resync_Trigger_1, Resync_Trigger_2	Configurable resync trigger conditions. A resync is triggered when the logic equation in these parameters evaluates to TRUE. The default value is (empty).
Resync_Fails_On_FNF	Determines whether a file-not-found response from the provisioning server constitutes a successful or a failed resync. A failed resync activates the error resync timer. The default value is Yes.
HTTPS_Name_Validate	Determines whether to check the Subject Alternative Name (SAN) for the HTTPS provisioning. Set to yes to enable the SAN check. The default value is Yes.
Profile_Rule	This parameter is a profile script that evaluates to the provisioning resync command. The command specifies the protocol (TFTP, HTTP, or HTTPS) and an associated URL. If the command is not specified, TFTP is assumed, and the address of the TFTP server is obtained through DHCP option 66. In the URL, either the IP address or the FQDN of the server can be specified. The file name can have macros, such as $MA, which expands to the device MAC address. The default value is /ata$PSN.cfg.

Parameter Name	Description and Default Value
Profile_Rule_B, Profile_Rule_C, Profile_Rule_D	Defines second, third, and fourth resync commands and associated profile URLs. These profile scripts are executed sequentially after the primary Profile Rule resync operation has completed. If a resync is triggered and Profile Rule is blank, Profile Rule B, C, and D are still evaluated and executed. The default value is (empty).
Log_Resync_Request_Msg	This parameter contains the message that is sent to the syslog server at the start of a resync attempt. The default value is $PN $MAC –Requesting resync $SCHEME://$SERVIP:$PORT$PATH.
Log_Resync_Success_Msg	The syslog message that is issued upon successful completion of a resync attempt. The default value is $PN $MAC –Successful resync $SCHEME://$SERVIP:$PORT$PATH -- $ERR.
Log_Resync_Failure_Msg	The syslog message that is issued after a failed resync attempt. The default value is $PN $MAC – Resyncfailed: $ERR.
Report_Rule	The target URL to which configuration reports are sent. This parameter has the same syntax as the Profile_Rule parameter, and resolves to a TCP/IP command with an associated URL. A configuration report is generated in response to an authenticated SIP NOTIFY message, with Event: report. The report is an XML file containing the name and value of all the device parameters. This parameter may optionally contain an encryption key. For example: [--key $K] tftp://ps.callhome.net/$MA/rep.xml.enc

Firmware Upgrade Parameters

The following table defines the function and usage of each parameter in the Firmware Upgrade section of the Provisioning tab.

Parameter Name	Description and Default Value
Upgrade_Enable	Enables firmware upgrade operations independently of resync actions. The default value is Yes.

Parameter Name	Description and Default Value
Upgrade_Error_Retry_Delay	The upgrade retry interval (in seconds) applied in case of upgrade failure. The device has a firmware upgrade error timer that activates after a failed firmware upgrade attempt. The timer is initialized with the value in this parameter. The next firmware upgrade attempt occurs when this timer counts down to zero. The default value is 3600 seconds.
Downgrade_Rev_Limit	Enforces a lower limit on the acceptable version number during a firmware upgrade or downgrade. The device does not complete a firmware upgrade operation unless the firmware version is greater than or equal to this parameter. The default value is (empty).
Upgrade_Rule	This parameter is a firmware upgrade script with the same syntax as Profile_Rule. Defines upgrade conditions and associated firmware URLs. The default value is (empty).
Log_Upgrade_Request_Msg	The syslog message that is issued at the start of a firmware upgrade attempt. The default value is $PN $MAC – Requesting upgrade $SCHEME://$SERVIP:$PORT$PATH.
Log_Upgrade_Success_Msg	The syslog message that is issued after a firmware upgrade attempt completes successfully. The default value is $PN $MAC – Successful upgrade $SCHEME://$SERVIP:$PORT$PATH --$ERR.
Log_Upgrade_Failure_Msg	The syslog message that is issued after a failed firmware upgrade attempt. The default value is $PN $MAC – Upgrade failed: $ERR.

General Purpose Parameters

The following table defines the function and usage of each parameter in the General Purpose Parameters section of the Provisioning tab.

Parameter Name	Description and Default Value
GPP_SA, GPP_SB, GPP_SC, GPP_SD	Special purpose provisioning parameters, designed to hold encryption keys and passwords. To ensure the integrity of the encryption mechanism, these parameters must be kept secret. Therefore these parameters are not displayed on the device configuration web page, and they are not included in the configuration report sent in response to a SIP NOTIFY command. The default value is (empty).

Parameter Name	Description and Default Value
GPP_A through GPP_P	General purpose provisioning parameters. These parameters can be used as variables in provisioning and upgrade rules. They are referenced by prepending the variable name with a '$' character, such as $A for GPP_A. The default value is (empty).

Macro Expansion Variables

Certain macro variables are recognized within the following provisioning parameters:

- Profile_Rule
- Profile_Rule_*
- Resync_Trigger_*
- Upgrade_Rule
- Log_*
- GPP_* (under specific conditions)

Within these parameters, syntax types, such as $NAME or $(NAME), are recognized and expanded.

Macro variable substrings can be specified with the notation $(NAME:p) and $(NAME:p:q), where p and q are non-negative integers (available in revision 2.0.11 and above). The resulting macro expansion is the substring starting at character offset p, with length q (or else till end-of-string if q is not specified). For example, if GPP_A contains ABCDEF, then $(A:2) expands to CDEF, and $(A:2:3) expands to CDE.

An unrecognized name is not translated, and the $NAME or $(NAME) form remains unchanged in the parameter value after expansion.

Parameter Name	Description and Default Value
$	The form $$ expands to a single $ character.
A through P	Replaced by the contents of the general purpose parameters GPP_A through GPP_P.
SA through SD	Replaced by special purpose parameters GPP_SA through GPP_SD. These parameters hold keys or passwords used in provisioning. **Note** $SA through $SD are recognized as arguments to the optional resync URL qualifier, --key.
MA	MAC address using lower case hex digits, for example, 000e08aabbcc.
MAU	MAC address using upper case hex digits, for example 000E08AABBCC.

Parameter Name	Description and Default Value
MAC	MAC address using lower case hex digits, and colons to separate hex digit pairs. For example 00:0e:08:aa:bb:cc.
PN	Product name. For example, ATA191-MPP.
PSN	Product Series Number. For example, 191
SN	Serial Number string. for example 88012BA01234.
CCERT	SSL Client Certificate status: Installed or Not Installed.
IP	IP address of the phone within its local subnet. For example 192.168.1.100.
EXTIP	External IP of the phone, as seen on the Internet. For example 66.43.16.52.
SWVER	Software version string. For example, Software version string. For example, 11-1-0MPP-19
HWVER	Hardware version string. For example, 4
PRVST	Provisioning State (a numeric string): -1 = explicit resync request 0 = power-up resync 1 = periodic resync 2 = resync failed, retry attempt
UPGST	Upgrade State (a numeric string): 1 = first upgrade attempt 2 = upgrade failed, retry attempt
UPGERR	Result message (ERR) of previous upgrade attempt; for example http_get failed.
PRVTMR	Seconds since last resync attempt.
UPGTMR	Seconds since last upgrade attempt.
REGTMR1	Seconds since Line 1 lost registration with SIP server.
REGTMR2	Seconds since Line 2 lost registration with SIP server.
UPGCOND	Legacy macro name.
SCHEME	File access scheme, one of TFTP, HTTP, or HTTPS, as obtained after parsing resync or upgrade URL.

Parameter Name	Description and Default Value
SERV	Request target server host name, as obtained after parsing resync or upgrade URL.
SERVIP	Request target server IP address, as obtained after parsing resync or upgrade URL, possibly following DNS lookup.
PORT	Request target UDP/TCP port, as obtained after parsing resync or upgrade URL.
PATH	Request target file path, as obtained after parsing resync or upgrade URL.
ERR	Result message of resync or upgrade attempt. Only useful in generating result syslog messages. The value is preserved in the UPGERR variable in the case of upgrade attempts.
UIDn	The contents of the Line n UserID configuration parameter.

ORIGTYPE
AUTHSTATUS
	Controls whether the phone needs to request for a license.
	Values for
	`ORIGTYPE`
	are: orig_ent, orig_mpp, none
	Values for
	`AUTHSTATUS`
	are: classic, wxc, none
	Add the variables in:
	• profile rule or upgrade rule macro expansion and conditional expression
	• transition authorization rule macro expansion

Internal Error Codes

The ATA defines a number of internal error codes (X00–X99) to facilitate configuration in providing finer control over the behavior of the unit under certain error conditions.

Parameter Name	Description and Default Value
X00	Transport layer (or ICMP) error when sending a SIP request.
X20	SIP request times out while waiting for a response.
X40	General SIP protocol error (for example, unacceptable codec in SDP in 200 and ACK messages, or times out while waiting for ACK).
X60	Dialed number invalid according to given dial plan.

CHAPTER 7

Voice Parameters

Voice Parameter Numbering

Certain types of parameters apply to multiple elements, such as users and lines. In the configuration file, the parameter name is appended with a number, such as <Line_Enable_1> and <Line_Enable_2>. To understand this numbering system, use the following key:

- 1—User 1 or Line1 (PHONE1 port)
- 2—User 2 or Line 2 (PHONE2 port)

FXS port 1 uses <Proxy_1_>

FXS port 2 used <Proxy_2_>

Voice Parameters

<Restricted_Access_Domains>	Domain of the service provider to which the ATA is connected to. It prevents the ATA from connecting to other service providers.
<Enable_Web_Admin_Access>	This feature is not available in ATA web voice.
<IVR_Admin_Password>	Password for the administrator to manage the ATA by using the built-in IVR through a connected phone.
<Network_Startup_Delay>	The number of seconds of delay between restarting the voice module and initializing network interface. Default setting—3

<DNS_Query_TTL_Ignore>	In DNS packages, the server will suggest a TTL value to the client; if this parameter is set to yes, the value from the server will be ignored. Default setting—Yes

<Provision_Enable>	Controls all resync actions independently of firmware upgrade actions. Set to yes to enable remote provisioning. Default setting—Yes
<Resync_On_Reset>	Triggers a resync after every reboot except for reboots caused by parameter updates and firmware upgrades. Default setting—Yes
<Resync_Random_Delay>	The maximum value for a random time interval that the ATA waits before making its initial contact with the provisioning server. This delay is effective only on the initial configuration attempt following power-on or reset. The delay is a pseudorandom number between zero and this value. This parameter is in units of 20 seconds; the default value of 2 represents 40 seconds. This feature is disabled when this parameter is set to zero. This feature can be used to prevent an overload of the provisioning server when a large number of devices power-on simultaneously. Default setting—2 (40 seconds)
<Resync_At_HHmm>	The time of day when the device tries to resync. The resync is performed each day. Used in conjunction with the Resync At Random Delay. Default setting—blank
<Resync_At_Random_Delay>	Used in conjunction with the Resync At (HHmm) setting, this parameter sets a range of possible values for the resync delay. The system randomly chooses a value from this range and waits the specified number of seconds before attempting to resync. This feature is intended to prevent the network jam that would occur if all resynchronizing devices began the resync at the exact same time of day. Default setting—600
<Resync_Periodic>	The time interval between periodic resyncs with the provisioning server. The associated resync timer is active only after the first successful synchronization with the server. Setting this parameter to zero disables periodic resynchronization. Default setting—3600 seconds
<Resync_Error_Retry_Delay>	Resync retry interval (in seconds) applied in case of resync failure. The ATA has an error retry timer that activates if the previous attempt to sync with the provisioning server fails. The ATA waits to contact the server again until the timer counts down to zero. This parameter is the value that is initially loaded into the error retry timer. If this parameter is set to zero, the ATA does not try to resync with the provisioning server following a failed attempt. Default setting—3600 seconds
<Forced_Resync_Delay>	Maximum delay (in seconds) that the ATA waits before performing a resync. The ATA does not resync while one of its lines is active. Because a resync can take several seconds, it is desirable to wait until the ATA has been idle for an extended period before resynchronizing. This allows a user to make calls in succession without interruption. The ATA has a timer that begins counting down when all of its lines become idle. This parameter is the initial value of the counter. Resync events are delayed until this counter decrements to zero. Default setting—14400 seconds

<Resync_From_SIP>	Enables a resync to be triggered via a SIP NOTIFY message. Default setting—yes
<Resync_After_Upgrade_Attempt>	Triggers a resync after every firmware upgrade attempt. Default setting—Yes
<Resync_Trigger_1> <Resync_Trigger_2>	Configurable resync trigger conditions. A resync is triggered when the logic equation in these parameters evaluates to TRUE. Default setting—blank
<Resync_Fails_On_FNF>	Determines whether a file-not-found response from the provisioning server constitutes a successful or a failed resync. A failed resync activates the error resync timer. Default setting—Yes
<Profile_Rule>	This parameter is a profile script that evaluates to the provisioning resync command. The command is a TCP/IP operation and an associated URL. The TCP/IP operation can be TFTP, HTTP, or HTTPS. If the command is not specified, TFTP is assumed, and the address of the TFTP server is obtained through DHCP option 66. In the URL, either the IP address or the FQDN of the server can be specified. The file name can have macros, such as $MA, which expands to the ATA MAC address. Default setting—/ata$PSN.cfg
<Profile_Rule_B> <Profile_Rule_C> <Profile_Rule_D>	Defines second, third, and fourth resync commands and associated profile URLs. These profile scripts are executed sequentially after the primary Profile Rule resync operation has completed. If a resync is triggered and Profile Rule is blank, Profile Rule B, C, and D are still evaluated and executed. Default setting—blank
<Log_Resync_Request_Msg>	This parameter contains the message that is sent to the Syslog server at the start of a resync attempt. Default setting—$PN $MAC – Requesting resync $SCHEME://$SERVIP:$PORT$PATH
<Log_Resync_Success_Msg>	Syslog message issued upon successful completion of a resync attempt. Default setting—$PN $MAC – Successful resync $SCHEME://$SERVIP:$PORT$PATH
<Log_Resync_Failure_Msg>	Syslog message issued after a failed resync attempt. Default setting—$PN $MAC -- Resync failed: $ERR

<Report_Rule>	The target URL to which configuration reports are sent. This parameter has the same syntax as the Profile_Rule parameter, and resolves to a TCP/IP command with an associated URL. A configuration report is generated in response to an authenticated SIP NOTIFY message, with Event: report. The report is an XML file containing the name and value of all the device parameters. This parameter may optionally contain an encryption key. For example: [--key $K] tftp://ps.callhome.net/$MA/rep.xml.enc Default setting—blank
<Upgrade_Enable>	Determines whether or not firmware upgrade operations can occur independently of resync actions. Default setting—Yes
<Upgrade_Error_Retry_Delay>	The upgrade retry interval (in seconds) applied in case of upgrade failure. The ATA has a firmware upgrade error timer that activates after a failed firmware upgrade attempt. The timer is initialized with the value in this parameter. The next firmware upgrade attempt occurs when this timer counts down to zero. Default setting—3600 seconds
<Downgrade_Rev_Limit>	Enforces a lower limit on the acceptable version number during a firmware upgrade or downgrade. The ATA does not complete a firmware upgrade operation unless the firmware version is greater than or equal to this parameter. Default setting—blank
<Upgrade_Rule>	This parameter is a firmware upgrade script with the same syntax as Profile_Rule. Defines upgrade conditions and associated firmware URLs. Default setting—blank
<Log_Upgrade_Request_Msg>	Syslog message issued at the start of a firmware upgrade attempt. Default setting—$PN $MAC – Requesting upgrade $SCHEME://$SERVIP:$PORT$PATH
<Log_Upgrade_Success_Msg>	Syslog message issued after a firmware upgrade attempt completes successfully. Default setting—$PN $MAC – Successful upgrade $SCHEME://$SERVIP:$PORT$PATH -- $ERR
<Log_Upgrade_Failure_Msg>	Syslog message issued after a failed firmware upgrade attempt. Default setting—$PN $MAC – Upgrade failed: $ERR
<License_Keys>	This field is not currently used.

<Custom_CA_URL>	The URL of a file location for a custom Certificate Authority (CA) certificate. Either the IP address or the FQDN of the server can be specified. The file name can have macros, such as $MA, which expands to the ATA MAC address. Default setting—blank

<GPP_A> to <GPP_P>	General purpose provisioning parameters. These parameters can be used as variables in provisioning and upgrade rules. They are referenced by prepending the variable name with a '$' character, such as $A for GPP_A. Default setting—blank
<GPP_SA> to <GPP_SD>	The two-letter upper-case macro names SA through SD identify GPP_SA through GPP_SD as a special case when used as arguments of the key URL option.

<Max_Forward>	The maximum times a call can be forwarded. The valid range is from 1 to 255. Default setting—70
<Max_Redirection>	Number of times an invite can be redirected to avoid an infinite loop. Default setting—5.
<Max_Auth>	The maximum number of times (from 0 to 255) a request may be challenged. Default setting—2
<SIP_User_Agent_Name>	The User-Agent header used in outbound requests. If empty, the header is not included. Macro expansion of $A to $D corresponding to GPP_A to GPP_D allowed. Default setting—$VERSION
<SIP_Server_Name>	The server header used in responses to inbound responses. Default setting—$VERSION
<SIP_Reg_User_Agent_Name>	The User-Agent name to be used in a REGISTER request. If this value is not specified, the SIP User Agent Name parameter is also used for the REGISTER request. Default setting—blank
<SIP_Accept_Language>	Accept-Language header used. There is no default (this indicates that the ATA does not include this header) If empty, the header is not included. Default setting—blank
<DTMF_Relay_MIME_Type>	The MIME Type used in a SIP INFO message to signal a DTMF event. Default setting—application/dtmf-relay.
<Hook_Flash_MIME_Type>	The MIME Type used in a SIP INFO message to signal a hook flash event. Default setting—application/hook-flash
<Remove_Last_Reg>	Determines whether or not the ATA removes the last registration before submitting a new one, if the value is different. Select yes to remove the last registration, or select no to omit this step. Default setting—no

<Use_Compact_Header>	Determines whether or not the ATA uses compact SIP headers in outbound SIP messages. Select yes or no from the dropdown list. Select yes to use compact SIP headers in outbound SIP messages. Select no to use normal SIP headers. If inbound SIP requests contain compact headers, the ATA reuses the same compact headers when generating the response regardless the settings of the Use Compact Header parameter. If inbound SIP requests contain normal headers, the ATA substitutes those headers with compact headers (if defined by RFC 261) if Use Compact Header parameter is set to yes. Default setting—no
<Escape_Display_Name>	Determines whether or not the Display Name is private. Select yes if you want the ATA to enclose the string (configured in the Display Name) in a pair of double quotes for outbound SIP messages. If the display name includes " or \, these will be escaped to \" and \\ within the double quotes. Otherwise, select no. Default setting—no
<RFC_2543_Call_Hold>	Configures the type of call hold: a:sendonly or 0.0.0.0. Do not use the 0.0.0.0 syntax in a HOLD SDP; use the a:sendonly syntax. Default setting—no
<Mark_all_AVT_Packets>	Select yes if you want all AVT tone packets (encoded for redundancy) to have the marker bit set for each DTMF event. Select no to have the marker bit set only for the first packet. Default setting—yes
<SIP_TCP_Port_Min>	The lowest TCP port number that can be used for SIP sessions. Default setting—5060
<SIP_TCP_Port_Max>	The highest TCP port number that can be used for SIP sessions. Default setting—5080
<CTI_Enable>	Enables or disables the Computer Telephone Interface feature provided by some servers. Default setting—no

<SIP_T1>	RFC 3261 T1 value (round-trip time estimate), which can range from 0 to 64 seconds. Default setting—0.5
<SIP_T2>	RFC 3261 T2 value (maximum retransmit interval for non-INVITE requests and INVITE responses), which can range from 0 to 64 seconds. Default setting—4
<SIP_T4>	RFC 3261 T4 value (maximum duration a message remains in the network), which can range from 0 to 64 seconds. Default setting—5
<SIP_Timer_B>	INVITE time-out value, which can range from 0 to 64 seconds. Default setting—32

<SIP_Timer_F>	Non-INVITE time-out value, which can range from 0 to 64 seconds. Default setting—32
<SIP_Timer_H>	H INVITE final response, time-out value, which can range from 0 to 64 seconds. Default setting—32
<SIP_Timer_D>	ACK hang-around time, which can range from 0 to 64 seconds. Default setting—32
<SIP_Timer_J>	Non-INVITE response hang-around time, which can range from 0 to 64 seconds. Default setting—32
<INVITE_Expires>	INVITE request Expires header value. If you enter 0, the Expires header is not included in the request. Range: 0–(231–1) Default setting—240
<ReINVITE_Expires>	ReINVITE request Expires header value. If you enter 0, the Expires header is not included in the request. Range: 0–(231–1) Default setting—30
<Reg_Min_Expires>	Minimum registration expiration time allowed from the proxy in the Expires header or as a Contact header parameter. If the proxy returns a value less than this setting, the minimum value is used. Default setting—1
<Reg_Max_Expires>	Maximum registration expiration time allowed from the proxy in the Min-Expires header. If the value is larger than this setting, the maximum value is used. Default setting—7200
<Reg_Retry_Intvl>	Interval to wait before the ATA retries registration after failing during the last registration. Default setting—30
<Reg_Retry_Long_Intvl>	When registration fails with a SIP response code that does not match Retry Reg RSC, the ATA waits for the specified length of time before retrying. If this interval is 0, the ATA stops trying. This value should be much larger than the Reg Retry Intvl value, which should not be 0. Default setting—1200
<Reg_Retry_Random_Delay>	Random delay range (in seconds) to add to Register Retry Intvl when retrying REGISTER after a failure. Default setting—0 (disabled)
<Reg_Retry_Long_Random_Delay>	Random delay range (in seconds) to add to Register Retry Long Intvl when retrying REGISTER after a failure. Default setting—0 (disabled)

<Reg_Retry_Intvl_Cap>	The maximum value to cap the exponential back-off retry delay (which starts at Register Retry Intvl and doubles on every REGISTER retry after a failure) In other words, the retry interval is always at Register Retry Intvl seconds after a failure. If this feature is enabled, Reg Retry Random Delay is added on top of the exponential back-off adjusted delay value. Default setting—0, which disables the exponential backoff
<SIT1_RSC> <SIT2_RSC> <SIT3_RSC> <SIT4_RSC>	SIP response status code for the corresponding Special Information Tone (SIT), SIT1 through SIT4. For example, if you set the SIT1 RSC to 404, when the user makes a call and a failure code of 404 is returned, the SIT1 tone is played. Reorder or Busy tone is played by default for all unsuccessful response status code for SIT 1 RSC through SIT 4 RSC. Default setting—blank
<Try_Backup_RSC>	SIP response code that retries a backup server for the current request. Default setting—blank
<Retry_Reg_RSC>	Interval to wait before the ATA retries registration after failing during the last registration. Default setting—blank

<RTP_Port_Min>	Minimum port number for RTP transmission and reception. The RTP Port Min and RTP Port Max parameters should define a range that contains at least 4 even number ports, such as 100 –106. Default setting—16384.
<RTP_Port_Max>	Maximum port number for RTP transmission and reception. Default setting—16482.
<RTP_Packet_Size>	Packet size in seconds, which can range from 0.01 to 0.16. Valid values must be a multiple of 0.01 seconds. Default setting—0.030
<Max_RTP_ICMP_Err>	Number of successive ICMP errors allowed when transmitting RTP packets to the peer before the ATA terminates the call. If value is set to 0, the ATA ignores the limit on ICMP errors. Default setting—0
<RTCP_Tx_Interval>	Interval for sending out RTCP sender reports on an active connection. It can range from 0 to 255 seconds. During an active connection, the ATA can be programmed to send out compound RTCP packet on the connection. Each compound RTP packet except the last one contains a SR (Sender Report) and a SDES (Source Description) The last RTCP packet contains an additional BYE packet. Each SR except the last one contains exactly 1 RR (Receiver Report); the last SR carries no RR. The SDES contains CNAME, NAME, and TOOL identifiers. The CNAME is set to <User ID>@<Proxy>, NAME is set to <Display Name> (or Anonymous if user blocks caller ID), and TOOL is set to the Vendor/Hardwareplatform-softwareversion. The NTP timestamp used in the SR is a snapshot of the local time for the ATA, not the time reported by an NTP server. If the ATA receives a RR from the peer, it attempts to compute the round trip delay and show it as the Call Round Trip Delay value (ms) on the Information page. Default setting—0

<No_UDP_Checksum>	Select yes if you want the ATA to calculate the UDP header checksum for SIP messages. Otherwise, select no. Default setting—no
<Stats_In_BYE>	Determines whether the ATA includes the PRTP-Stat header or response in a BYE message. The header contains the RTP statistics of the current call. Select yes or no from the dropdown list. Default setting—yes The format of the P-RTP-Stat header is: P-RTP-State: PS=<packets sent>,OS=<octets sent>,PR=<packets received>,OR=<octets eceived>,PL=<packets lost>,JI=<jitter in ms>,LA=<delay in ms>,DU=<call durationins>,EN=<encoder>,DE=<decoder>.

<NSE_Dynamic_Payload>	NSE dynamic payload type. The valid range is 96-127. Default setting—100
<AVT_Dynamic_Payload>	AVT dynamic payload type. The valid range is 96-127. Default setting—101
<INFOREQ_Dynamic_Payload>	INFOREQ dynamic payload type. Default setting—blank
<G726r32_Dynamic_Payload>	G726r32 dynamic payload type. Default setting—2
<EncapRTP_Dynamic_Payload>	EncapRTP Dynamic Payload type. Default setting—112
<RTP-Start-Loopback_Dynamic_Payload>	RTP-Start-Loopback Dynamic Payload type. Default setting—113
<RTP-Start-Loopback_Codec>	RTP-Start-Loopback Codec. Select one of the following: G711u, G711a, G726-32, G729a. Default setting—G711u
<NSE_Codec_Name>	NSE codec name used in SDP. Default setting—NSE
<AVT_Codec_Name>	AVT codec name used in SDP. Default setting—telephone-event
<G711u_Codec_Name>	G.711u codec name used in SDP. Default setting—PCMU
<G711a_Codec_Name>	G.711a codec name used in SDP. Default setting—PCMA

<G726r32_Codec_Name>	G.726-32 codec name used in SDP. Default setting—G726-32
<G729a_Codec_Name>	G.729a codec name used in SDP. Default setting—G729a
<G722_Codec_Name> G.722 codec name used in SDP.	Default setting—G722
<EncapRTP_Codec_Name>	EncapRTP codec name used in SDP. Default setting—encaprtp

<Handle_VIA_received>	If you select yes, the ATA processes the received parameter in the VIA header (this value is inserted by the server in a response to any one of its requests) If you select no, the parameter is ignored. Select yes or no from the drop-down menu. Default setting—no
<Handle_VIA_rport>	If you select yes, the ATA processes the rport parameter in the VIA header (this value is inserted by the server in a response to any one of its requests) If you select no, the parameter is ignored. Select yes or no from the drop-down menu. Default setting—no
<Insert_VIA_received>	Inserts the received parameter into the VIA header of SIP responses if the receivedfrom IP and VIA sent-by IP values differ. Select yes or no from the drop-down menu. Default setting—no
<Insert_VIA_rport>	Inserts the rport parameter into the VIA header of SIP responses if the receivedfrom IP and VIA sent-by IP values differ. Select yes or no from the drop-down menu. Default setting—no
<Substitute_VIA_Addr>	Lets you use NAT-mapped IP:port values in the VIA header. Select yes or no from the drop-down menu. Default setting—no
<Send_Resp_To_Src_Port>	Sends responses to the request source port instead of the VIA sent-by port. Select yes or no from the drop-down menu. Default setting—no
<STUN_Enable>	Enables the use of STUN to discover NAT mapping. Select yes or no from the dropdown menu. Default setting—no
<STUN_Test_Enable>	If the STUN Enable feature is enabled and a valid STUN server is available, the ATA can perform a NAT-type discovery operation when it powers on. It contacts the configured STUN server, and the result of the discovery is reported in a Warning header in all subsequent REGISTER requests. If the ATA detects symmetric NAT or a symmetric firewall, NAT mapping is disabled. Default setting—no

<STUN_Server>	IP address or fully-qualified domain name of the STUN server to contact for NAT mapping discovery. Default setting—blank
<EXT_IP>	External IP address to substitute for the actual IP address of the ATA in all outgoing SIP messages. If 0.0.0.0 is specified, no IP address substitution is performed. If this parameter is specified, the ATA assumes this IP address when generating SIP messages and SDP (if NAT Mapping is enabled for that line) However, the results of STUN and VIA received parameter processing, if available, supersede this statically configured value. This option requires that you have (1) a static IP address from your Internet Service Provider and (2) an edge device with a symmetric NAT mechanism. If the ATA is the edge device, the second requirement is met. Default setting—blank
<EXT_RTP_Port_Min>	External port mapping number of the RTP Port Min. number. If this value is not zero, the RTP port number in all outgoing SIP messages is substituted for the corresponding port value in the external RTP port range. Default setting—blank
<NAT_Keep_Alive_Intvl>	Interval between NAT-mapping keep alive messages. Default setting—15
<Redirect_Keep_Alive>	Interval between NAT Redirect keep alive messages. Default setting—15

<Line_Enable_1> <Line_Enable_2>	To enable this line for service, select yes. Otherwise, select no. Default setting—yes

<SAS_Enable_1> <SAS_Enable_2>	To enable the use of the line as a streaming audio source, select yes. Otherwise, select no. If enabled, the line cannot be used for outgoing calls. Instead, it auto-answers incoming calls and streams audio RTP packets to the caller. Default setting—no
<SAS_DLG_Refresh_Intvl_1> <SAS_DLG_Refresh_Intvl_2>	If this value is not zero, it is the interval at which the streaming audio server sends out session refresh (SIP re-INVITE) messages to determine whether the connection to the caller is still active. If the caller does not respond to the refresh message, the ATA ends this call with a SIP BYE message. The range is 0 to 255 seconds (0 means that the session refresh is disabled.) Default setting—30

<SAS_Inbound_RTP_Sink_1> <SAS_Inbound_RTP_Sink_2>	The purpose of this parameter is to work around devices that do not play inbound RTP if the SAS line declares itself as a send-only device and tells the client not to stream out audio. This parameter is an FQDN or IP address of an RTP sink to be used by the SAS line in the SDP of its 200 response to inbound INVITE from a client. It will appear in the c = line and the port number, if specified, will appear in the m = line of the SDP. If this value is not specified or is equal to 0, then c =0.0.0.0 and a=sendonly will be used in the SDP to tell the SAS client to not to send any RTP to this SAS line. If a non-zero value is specified, then a=sendrecv and the SAS client will stream audio to the given address. Special case: If the value is $IP, then the SAS line's own IP address is used in the c = line and a=sendrecv. In that case the SAS client will stream RTP packets to the SAS line. Default setting—blank

<NAT_Mapping_Enable_1> <NAT_Mapping_Enable_2>	To use externally mapped IP addresses and SIP/RTP ports in SIP messages, select yes. Otherwise, select no. Default setting—no
<NAT_Keep_Alive_Enable_1> <NAT_Keep_Alive_Enable_2>	To send the configured NAT keep alive message periodically, select yes. Otherwise, select no. Default setting—no
<NAT_Keep_Alive_Msg_1> <NAT_Keep_Alive_Msg_2>	Enter the keep alive message that should be sent periodically to maintain the current NAT mapping. If the value is $NOTIFY, a NOTIFY message is sent. If the value is $REGISTER, a REGISTER message without contact is sent. Default setting—$NOTIFY
<NAT_Keep_Alive_Dest_1> <NAT_Keep_Alive_Dest_2>	Destination that should receive NAT keep alive messages. If the value is $PROXY, the messages are sent to the current proxy server or outbound proxy server. Default setting—$PROXY
<Blind_Attn-Xfer_Enable_1> <Blind_Attn-Xfer_Enable_2>	Enables the ATA to perform an attended transfer operation by ending the current call leg and performing a blind transfer of the other call leg. If this feature is disabled, the ATA performs an attended transfer operation by referring the other call leg to the current call leg while maintaining both call legs. To use this feature, select yes. Otherwise, select no. Default setting—no
<MOH_Server_1> <MOH_Server_2>	User ID or URL of the auto-answering streaming audio server. When only a user ID is specified, the current or outbound proxy is contacted. Music-on-hold is disabled if the MOH Server is not specified. Default setting—blank
<Xfer_When_Hangup_Conf_1> <Xfer_When_Hangup_Conf_2>	Makes the ATA perform a transfer when a conference call has ended. Select yes or no from the drop-down menu. Default setting—yes
<Conference_Bridge_URL_1> <Conference_Bridge_URL_2>	This feature supports external conference bridging for n-way conference calls (n>2), instead of mixing audio locally. To use this feature, set this parameter to that of the server's name. For example: conf@mysefver.com:12345 or conf (which uses the Proxy value as the domain). Default setting—blank

<Conference_Bridge_Ports_1> <Conference_Bridge_Ports_2>	Select the maximum number of conference call participants. The range is 3 to 10. Default setting—3
<Enable_IP_Dialing_1> <Enable_IP_Dialing_2>	Enable or disable IP dialing. If IP dialing is enabled, one can dial [userid@] a.b.c.d[:port], where • '@', '.', and ':' are dialed by entering * • user-id must be numeric (like a phone number) • a, b, c, d must be between 0 and 255 • port must be larger than 255. If port is not given, 5060 is used. Port and User-Id are optional. If the user-id portion matches a pattern in the dial plan, then it is interpreted as a regular phone number according to the dial plan. The INVITE message, however, is still sent to the outbound proxy if it is enabled. Default setting—no
<Emergency_Number_1> <Emergency_Number_2>	Comma separated list of emergency number patterns. If outbound call matches one of the pattern, the ATA will disable hook flash event handling. The condition is restored to normal after the call ends. Blank signifies that there is no emergency number. Maximum number length is 63 characters. Default setting—blank
<Mailbox_ID_1> <Mailbox_ID_2>	Enter the ID number of the mailbox for this line. Default setting—blank
<Feature_Key_Sync_1>	Allows the phone to synchronize with the call server. If Do Not Disturb or Call Forwarding settings are changed on the phone, the changes are also made on the server. If changes are made on the server, they are propagated to the phone. Default setting: no
<Secure_Call_Option_1> <Secure_Call_Option_2>	Configures a line to only accept secure calls. Options are: Optional: Retains the current secure call option for the phone adapter. Strict: Allows SRTP only when SIP transport is set to TLS and if the ATA receives an unsecure call, the call fails. Allows RTP only when SIP transport is UDP/TCP and if the ATA receives an unsecure call, the call fails. Default setting: Optional

<Company_UUID_1> <Company_UUID_2>	The Universally Unique Identifier (UUID) assigned to the customer by the emergency call services provider. For example: `19c8168c-a366-44b5-853c-960fcaa19592` Allowed values: Maximum identifier length is 128 characters. Default setting—blank

<Primary_Request_URL_1> <Primary_Request_URL_2>	URL of the primary location server that provides the emergency call services. The location server returns an HELD response to the phone with the requested location URI that is tied to the user phone IP address. This parameter must be in the form of a valid HTTP or HTTPS URL. Allowed values: A valid URL not exceeding 255 characters. Default setting—blank
<Secondary_Request_URL_1> <Secondary_Request_URL_2>	URL of the backup server to obtain the user's phone location. If the primary request URL fails, ATA tries to send the secondary request URL to the emergency call services provider. This parameter must be in the form of a valid HTTP or HTTPS URL. Allowed values: A valid URL not exceeding 255 characters. Default setting—blank

<Proxy_1> <Proxy_2>	SIP proxy server for all outbound requests. Default setting—blank
<Outbound_Proxy_1> <Outbound_Proxy_2>	SIP Outbound Proxy Server where all outbound requests are sent as the first hop. Default setting—blank
<Use_Outbound_Proxy_1> <Use_Outbound_Proxy_2>	Enables the use of an Outbound Proxy. If set to no, the Outbound Proxy and Use OB Proxy in Dialog parameters are ignored. Default setting—no
<Use_OB_Proxy_In_Dialog_1> through <Use_OB_Proxy_In_Dialog_2>	Whether to force SIP requests to be sent to the outbound proxy within a dialog. Ignored if the parameter Use Outbound Proxy is no, or the Outbound Proxy parameter is empty. Default setting—yes
<Register_1> <Register_2>	Enable periodic registration with the Proxy parameter. This parameter is ignored if Proxy is not specified. Default setting—yes
<Make_Call_Without_Reg_1> <Make_Call_Without_Reg_2>	Allow making outbound calls without successful (dynamic) registration by the unit. If No, dial tone will not play unless registration is successful. Default setting—no
<Register_Expires_1> <Register_Expires_2>	Expires value in sec in a REGISTER request. The ATA will periodically renew registration shortly before the current registration expired. This parameter is ignored if the Register parameter is no. Range: 0 – (231 – 1) sec. Default setting—3600
<Ans_Call_Without_Reg_1> <Ans_Call_Without_Reg_2>	Allow answering inbound calls without successful (dynamic) registration by the unit. Default setting—no

<Use_DNS_SRV_1> <Use_DNS_SRV_2>	Whether to use DNS SRV lookup for Proxy and Outbound Proxy. Default setting—no
<DNS_SRV_Auto_Prefix_1> <DNS_SRV_Auto_Prefix_2>	If enabled, the ATA will automatically prepend the Proxy or Outbound Proxy name with _sip._udp when performing a DNS SRV lookup on that name. Default setting—no
<Proxy_Fallback_Intvl_1> <Proxy_Fallback_Intvl_2>	After failing over to a lower priority server, the ATA waits for the specified Proxy Fallback Interval, in seconds, before retrying the highest priority proxy (or outbound proxy) servers. This parameter is useful only if the primary and backup proxy server list is provided to the ATA via DNS SRV record lookup on the server name. The ATA can contain up to 10 A records for an SRV record. Using multiple DNS A records per server name does not allow the notion of priority, so all hosts will be considered at the same priority and the ATA will not attempt to fall back after a failover. If the value is 0, the SIP proxy fallback feature is disabled. Default setting—3600
<Proxy_Redundancy_Method_1> <Proxy_Redundancy_Method_2>	The method that the ATA uses to create a list of proxies returned in the DNS SRV records. If you select Normal, the list will contain proxies ranked by weight and priority. If you select Based on SRV port, the ATA also inspects the port number based on 1st proxy's port. Default setting—Normal
<Mailbox_Subscribe_URL_1> <Mailbox_Subscribe_URL_2>	The URL or IP address of the voicemail server. Default setting—blank
<Mailbox_Subscribe_Expires_1> <Mailbox_Subscribe_Expires_2>	The subscription interval for voicemail message waiting indication. When this time period expires, the ATA sends another subscribe message to the voice mail server. Default: 2147483647
<Display_Name_1> <Display_Name_2>	Display name for caller ID. Default setting—blank
<User_ID_1> <User_ID_2>	User ID for this line. Default setting—blank
<Password_1> <Password_2>	Password for this line. Default setting—blank
<Use_Auth_ID_1> <Use_Auth_ID_2>	To use the authentication ID and password for SIP authentication, select yes. Otherwise, select no to use the user ID and password. Default setting—no
<Auth_ID_1> <Auth_ID_2>	Authentication ID for SIP authentication. Default setting—blank

<Resident_Online_Number_1> <Resident_Online_Number_2>	This setting allows you to associate a "local" telephone number with this line using a valid Skype Online Number from Skype. Calls made to that number will ring your phone. Enter the number without spaces or special characters. Default setting—blank
<SIP URI>	The SIP URI, in the following format: sip:<username>@<WAN_IP>:<port> or sip:<username>@<domain>:<port>

<Call_Waiting_Serv_1> <Call_Waiting_Serv_2>	Enable Call Waiting Service. Default setting—yes
<Block_CID_Serv_1> <Block_CID_Serv_2>	Enable Block Caller ID Service. Default setting—yes
<Block_ANC_Serv_1> <Block,_ANC_Serv_2>	Enable Block Anonymous Calls Service. Default setting—yes
<Dist_Ring_Serv_1> <Dist_Ring_Serv_2>	Enable Distinctive Ringing Service. Default setting—yes
<Cfwd_All_Serv_1> <Cfwd_All_Serv_2>	Enable Call Forward All Service. Default setting—yes
<Cfwd_Busy_Serv_1> <Cfwd_Busy_Serv_2>	Enable Call Forward Busy Service. Default setting—yes
<Cfwd_No_Ans_Serv_1> <Cfwd_No_Ans_Serv_2>	Enable Call Forward No Answer Service. Default setting—yes
<Cfwd_Sel_Serv_1> <Cfwd_Sel_Serv_2>	Enable Call Forward Selective Service. Default setting—yes
<Cfwd_Last_Serv_1> <Cfwd_Last_Serv_2>	Enable Forward Last Call Service Default setting—yes
<Block_Last_Serv_1> <Block_Last_Serv_2>	Enable Block Last Call Service. Default setting—yes
<Accept_Last_Serv_1> <Accept_Last_Serv_2>	Enable Accept Last Call Service. Default setting—yes
<DND_Serv_1> <DND_Serv_2>	Enable Do Not Disturb Service. Default setting—yes

<CID–Serv_1> <CID–Serv_2>	Enable Caller ID Service. Default setting—yes
<CWCID_Serv_1> <CWCID_Serv_2>	Enable Call Waiting Caller ID Service. Default setting—yes
<Call_Return_Serv_1> <Call_Return_Serv_2>	Enable Call Return Service. Default setting—yes
<Call_Redial_Serv_1> <Call_Redial_Serv_2>	Enable Call Redial Service.
<Call_Back_Serv_1> <Call_Back_Serv_2>	Enable Call Back Service.
<Three_Way_Call_Serv_1> <Three_Way_Call_Serv_2>	Enable Three Way Calling Service. Three Way Calling is required for Three Way Conference and Attended Transfer. Default setting—yes
<Three_Way_Conf_Serv_1> <Three_Way_Conf_Serv_2>	Enable Three Way Conference Service. Three Way Conference is required for Attended Transfer. Default setting—yes
<Attn_Transfer_Serv_1> <Attn_Transfer_Serv_2>	Enable Attended Call Transfer Service. Three Way Conference is required for Attended Transfer. Default setting—yes
<Unattn_Transfer_Serv_1> <Unattn_Transfer_Serv_2>	Enable Unattended (Blind) Call Transfer Service. Default setting—yes
<MWI_Serv_1> <MWI_Serv_2>	Enable MWI Service. MWI is available only if a Voice Mail Service is set-up in the deployment. Default setting—yes
<VMWI_Serv_1> <VMWI_Serv_2>	Enable VMWI Service (FSK) Default setting—yes
<Speed_Dial_Serv_1> <Speed_Dial_Serv_2>	Enable Speed Dial Service. Default setting—yes
<Secure_Call_Serv_1> <Secure_Call_Serv_2>	Secure Call Service. If this feature is enabled, a user can make a secure call by entering an activation code (*18 by default) before dialing the target number. Then audio traffic in both directions is encrypted for the duration of the call. Default setting—yes
<Referral_Serv_1> <Referral_Serv_2>	Enable Referral Service. See the Referral Services Codes parameter for more information. Default setting—yes

<Feature_Dial_Serv_1> <Feature_Dial_Serv_2>	Enable Feature Dial Service. See the Feature Dial Services Codes parameter for more information. Default setting—yes
<Service_Announcement_Serv_1> <Service_Announcement_Serv_2>	Enable Service Announcement Service. Default setting—no
<Reuse_CID_Number_As_Name_1> <Reuse_CID_Number_As_Name_2>	Use the Caller ID number as the caller name. Default settings: yes

<Preferred_Codec_1>, <Preferred_Codec_2> <Second_Preferred_Codec_1>, <Second_Preferred_Codec_2> <Third_Preferred_Codec_1>, <Third_Preferred_Codec_2>	Up to three codecs to be used for all calls from the specified line/handset, listed order of preference. The actual codec used in a call depends on the outcome of the codec negotiation protocol. Select one of the following: G711u, G711a, G726-32, G729a, or G722. Default setting for Preferred Codec: G711u Default setting for Second and Third Preferred Codec: Unspecified
<Use_Pref_Codec_Only_1> <Use_Pref_Codec_Only_2>	To use only the preferred codec for all calls, select yes. (The call fails if the far end does not support this codec.) Otherwise, select no. Default setting—no
<Use_Remote_Pref_Codec_1> <Use_Remote_Pref_Codec_2>	To use the preferred codec specified by the remote peer, select yes. Otherwise, select no. Default setting:
<Codec_Negotiation_1> <Codec_Negotiation_2>	Specify the codecs for codec negotiation: Default or List All. Default setting—Default
<G729a_Enable_1> <G729a_Enable_2>	To enable the use of the G.729a codec at 8 kbps, select yes. Otherwise, select no. Default setting—yes
<Silence_Supp_Enable_1> <Silence_Supp_Enable_2>	To enable silence suppression so that silent audio frames are not transmitted, select yes. Otherwise, select no. Default setting—no
<G726-32_Enable_1> <G726-32_Enable_2>	To enable the use of the G.726 codec at 32 kbps, select yes. Otherwise, select no. Default setting—yes
<Silence_Threshold_1> <Silence_Threshold_2>	Select the appropriate setting for the threshold: high, medium, or low. Default setting—medium
<FAX_V21_Detect_Enable_1> <FAX_V21_Detect_Enable_2>	To enable detection of V21 fax tones, select yes. Otherwise, select no. Default setting—yes
<Echo_Canc_Enable_1> <Echo_Canc_Enable_2>	To enable the use of the echo canceller, select yes. Otherwise, select no. Default setting—yes

<FAX_CNG_Detect_Enable_1> <FAX_CNG_Detect_Enable_2>	To enable detection of the fax Calling Tone (CNG), select yes. Otherwise, select no. Default setting—yes
<FAX_Passthru_Codec_1> <FAX_Passthru_Codec_2>	Select the codec for fax passthrough, G711u or G711a. Default setting—G711u
<FAX_Codec_Symmetric_1> through <FAX_Codec_Symmetric_2>	To force the ATA to use a symmetric codec during fax passthrough, select yes. Otherwise, select no. Default setting—yes
<DTMF_Process_INFO_1> <DTMF_Process_INFO_2>	To use the DTMF process info feature, select yes. Otherwise, select no. Default setting—yes
<FAX_Passthru_Method_1> <FAX_Passthru_Method_2>	Select the fax passthrough method: None, NSE, or ReINVITE. Default setting—NSE
<DTMF_Process_AVT_1> <DTMF_Process_AVT_2>	To use the DTMF process AVT feature, select yes. Otherwise, select no. Default setting—yes
<FAX_Process_NSE_1> <FAX_Process_NSE_2>	To use the fax process NSE feature, select yes. Otherwise, select no. Default setting—yes
<DTMF_Tx_Method_1> <DTMF_Tx_Method_2>	Select the method to transmit DTMF signals to the far end: InBand, AVT, INFO, or Auto. InBand sends DTMF by using the audio path. AVT sends DTMF as AVT events. INFO uses the SIP INFO method. Auto uses InBand or AVT based on the outcome of codec negotiation. Default setting—Auto
<FAX_Disable_ECAN_1> <FAX_Disable_ECAN_2>	If enabled, this feature automatically disables the echo canceller when a fax tone is detected. To use this feature, select yes. Otherwise, select no. Default setting—no
<DTMF_Tx_Mode_1> <DTMF_Tx_Mode_2>	DTMF Detection Tx Mode is available for SIP information and AVT. Options are: Strict or Normal. Default setting—Strict for which the following are true: • A DTMF digit requires an extra hold time after detection. • The DTMF level threshold is raised to -20 dBm. The minimum and maximum duration thresholds are: • strict mode for AVT: 70 ms • normal mode for AVT: 40 ms • strict mode for SIP info: 90 ms • normal mode for SIP info: 50 ms

<DTMF_Tx_Strict_Hold_Off_Time_1> <DTMF_Tx_Strict_Hold_Off_Time_2>	This parameter is in effect only when DTMF Tx Mode is set to strict, and when DTMF Tx Method is set to out-ofband; i.e. either AVT or SIP-INFO. The value can be set as low as 40 ms. There is no maximum limit. A larger value will reduce the chance of talk-off (beeping) during conversation, at the expense of reduced performance of DTMF detection, which is needed for interactive voice response systems (IVR). Default: 70 ms
<FAX_Enable_T38_1> <FAX_Enable_T38_2>	To enable the use of ITU-T T.38 standard for FAX Relay, select yes. Otherwise select no. Default setting—no
<Hook_Flash_Tx_Method_1> <Hook_Flash_Tx_Method_2>	Select the method for signaling hook flash events: None, AVT, or INFO. None does not signal hook flash events. AVT uses RFC2833 AVT (event = 16) INFO uses SIP INFO with the single line signal=hf in the message body. The MIME type for this message body is taken from the Hook Flash MIME Type setting. Default setting—None
<FAX_T38_Redundancy_1> <FAX_T38_Redundancy_2>	Select the appropriate number to indicate the number of previous packet payloads to repeat with each packet. Choose 0 for no payload redundancy. The higher the number, the larger the packet size and the more bandwidth consumed. Default setting—1
<FAX_T38_ECM_Enable_1> <FAX_T38_ECM_Enable_2>	Select yes to enable T.38 Error Correction Mode. Otherwise select no. Default setting—yes
<FAX_Tone_Detect_Mode_1> <FAX_Tone_Detect_Mode_2>	This parameter has three possible values: • caller or callee: The ATA will detect FAX tone whether it is callee or caller • caller only: The ATA will detect FAX tone only if it is the caller • callee only: The ATA will detect FAX tone only if it is the callee Default setting—caller or callee.
<Symmetric_RTP_1> <Symmetric_RTP_2>	Enable symmetric RTP operation. If enabled, the ATA sends RTP packets to the source address and port of the last received valid inbound RTP packet. If disabled (or before the first RTP packet arrives) the ATA sends RTP to the destination as indicated in the inbound SDP. Default setting—no
<FAX_T38_Return_to_Voice_1> <FAX_T38_Return_to_Voice_2>	When this feature is enabled, upon completion of the fax image transfer, the connection remains established and reverts to a voice call using the previously designated codec. Select yes to enable this feature, or select no to disable it. Default setting—no

<table>
<tr><td>&lt;Dial_Plan_1>
<Dial_Plan_2&gt;</td><td>The allowed number patterns for outbound calls. The default dial plan script for the line is as follows: (*xx|[3469]11|0|00|[2-9]xxxxxx|1xxx[2-9]xxxxxx|xxxxxxxxxxxx.)
Each parameter is separated by a semicolon (;)
<b>Examples</b> of Dial Plan Entry and Functionality
<b>(*xx</b>
Allow arbitrary 2 digit star code
<b>[3469]11</b>
Allow x11 sequences
<b>0</b>
Operator
<b>00</b>
International Operator
<b>[2-9]xxxxxx</b>
US local number
<b>1xxx[2-9]xxxxxx</b>
US 1 + 10-digit long distance
<b>xxxxxxxxxxxx.</b>
Everything else</td></tr>
</table>

<Gateway_1_1 <Gateway_4_1>	The first of 4 gateways that can be specified to be used in the <Dial Plan> to facilitate call routing specification (that overrides the given proxy information). This gateway is represented by gw1 in the <Dial Plan>. For example, the rule 1408xxxxxxx<:@gw1> can be added to the dial plan such that when the user dials 1408+7digits, the call will be routed to Gateway 1. Without the <:@gw1> syntax, all calls are routed to the given proxy by default (except IP dialing). Default setting—blank
<GW1_NAT_Mapping_Enable_1> <GW4_NAT_Mapping_Enable_1>	If enabled, the ATA uses NAT mapping when contacting Gateway 1. Default setting—no
<GW1_Auth_ID_1_ > <GW4_Auth_ID_1_ >	This value is the authentication user-id to be used by the ATA to authenticate itself to Gateway 1. Default setting—blank
<GW1_Password_1> <GW4_Password_1>	This value is the password to be used by the ATA to authenticate itself to Gateway 1. Default setting—blank
<Auto_PSTN_Fallback_1> <Auto_PSTN_Fallback_2>	If enabled, the ATA automatically routes all calls to the PSTN gateway when the SIP proxy is down (registration failure or network link down). Default setting—yes

<Cfwd_No_Ans_Dest_1> <Cfwd_No_Ans_Dest_2>	Forward number for Call Forward No Answer Service. Same as Cfwd All Dest. Default setting—blank
<Cfwd_No_Ans_Delay_1> <Cfwd_No_Ans_Delay_2>	Delay in sec before Call Forward No Answer triggers. Same as Cfwd All Dest. Default setting—20

<Idle_Polarity_1> <Idle_Polarity_2>	Polarity before a call is connected: Forward or Reverse. Default setting—Forward
<Caller_Conn_Polarity_1> <Caller_Conn_Polarity_2>	Polarity after an outbound call is connected: Forward orReverse. Default setting—Forward.
<Callee_Conn_Polarity_1> <Callee_Conn_Polarity_2>	Polarity after an inbound call is connected: Forward or Reverse. Default setting—Forward

<Cfwd_All_Dest_1> <Cfwd_All_Dest_2>	Forward number for Call Forward All Service. Default setting—blank
<Cfwd_Busy_Dest_1> <Cfwd_Busy_Dest_2>	Forward number for Call Forward Busy Service. Same as Cfwd All Dest. Default setting—blank
<Cfwd_Sel1_Caller_1>, <Cfwd_Sel8_Caller_1> <Cfwd_Sel1_Caller_1>, <Cfwd_Sel8_Caller_2>	Caller number pattern to trigger Call Forward Selective service. When the caller's phone number matches the entry, the call is forwarded to the corresponding Cfwd Selective Destination (Cfwd Sel1-8 Dest). • Use ? to match any single digit. • Use * to match any number of digits **Example**: 1408*, 1512???1234 In the above example, a call is forwarded to the corresponding destination if the caller ID either starts with 1408 or is an 11-digit numbering starting with 1512 and ending with 1234. Default setting—blank
<Cfwd_Sel1_Dest_1>, <Cfwd_Sel8_Dest_1> <Cfwd_Sel1_Dest_2>, <Cfwd_Sel8_Dest_2>	The destination for the corresponding Call Forward Selective caller pattern (Cfwd Sel1-8 Caller). Default setting—blank
<Cfwd_Last_Caller_1> <Cfwd_Last_Caller_2>	The number of the last caller; this caller is actively forwarded to the Cfwd Last Dest via the Call Forward Last service. Default setting—blank
<Cfwd_Last_Dest_1> <Cfwd_Last_Dest_2>	The destination for the Cfwd Last Caller.
<Block_Last_Caller_1> <Block_Last_Caller_2>	The number of the last caller; this caller is blocked via the Block Last Caller Service. Default setting—blank
<Accept_Last_Caller_1> <Accept_Last_Caller_2>	The number of the last caller; this caller is accepted via the Accept Last Caller Service. Default setting—blank

<Speed_Dial_2_1>, <Speed_Dial_9_1> <Speed_Dial_2_2>, <Speed_Dial_9_2>	Target phone number (or URL) assigned to speed dial 2, 3, 4, 5, 6, 7, 8, or 9. Default setting—blank

<CW_Setting_1> <CW_Setting_2>	Call Waiting on/off for all calls. Default setting—yes
<Block_CID_Setting_1> <Block_CID_Setting_2>	Block Caller ID on/off for all calls. Default setting—no
<Block_ANC_Setting_1> <Block_ANC_Setting_2>	Block Anonymous Calls on or off. Default setting—no
<DND_Setting_1> <DND_Setting_2>	DND on or off. Default setting—no
<CID_Setting_1> <CID_Setting_2>	Caller ID Generation on or off. Default setting—yes
<CWCID_Setting_1> <CWCID_Setting_2>	Call Waiting Caller ID Generation on or off. Default setting—yes
<Dist_Ring_Setting_1> <Dist_Ring_Setting_2>	Distinctive Ring on or off. Default setting—yes
<Secure_Call_Setting_1> <Secure_Call_Setting_2>	If yes, all outbound calls are secure calls by default, without requiring the user to dial a star code first. Default setting—no • If Secure Call Setting is set to yes, all outbound calls are secure. However, a user can disable security for a call by dialing *19 before dialing the target number. • If Secure Call Setting is set to No, the user can make a secure outbound call by dialing *18 before dialing the target number. • A user cannot force inbound calls to be secure or not secure; that depends on whether the caller has security enabled or not. **Note** This setting is applicable only if Secure Call Serv is set to yes on the line interface.
<Message_Waiting_1> <Message_Waiting_2>	Setting this value to yes can activate stutter tone and VMWI signal. This parameter is stored in long term memory and will survive after reboot or power cycle. Default setting—no

<Accept_Media_Loopback_Request_1> <Accept_Media_Loopback_Request_2>	Controls how to handle incoming requests for loopback operation. Default setting—automatic • never: Never accepts loopback calls; replies 486 to the caller. • automatic: Automatically accepts the call without ringing. • manual: Rings the phone first, and the call must be picked up manually before loopback starts. Default setting—Automatic
<Media_Loopback_Mode_1> <Media_Loopback_Mode_2>	The loopback mode to assume locally when making call to request media loopback. Choices are: Source and Mirror. Default setting—source **Note** If the ATA answers the call, the mode is determined by the caller.
<Media_Loopback_Type_1> <Media_Loopback_Type_2>	The loopback type to use when making call to request media loopback operation. Choices are Media and Packet. Default setting—media Note that if the ATA answers the call, then the loopback type is determined by the caller (the ATA always picks the first loopback type in the offer if it contains multiple type.)
<Ring1_Caller_1_ > through <Ring8_Caller_1_ > <Ring1_Caller_2_ > through <Ring8_Caller_2_ > <FAX_CNG_Detect_Enable_1>	Caller number pattern to play Distinctive Ring/CWT 1, 2, 3, 4, 5, 6, 7, or 8. Caller number patterns are matched from Ring 1 to Ring 8. The first match (not the closest match) will be used for alerting the subscriber. Default setting—blank

<Default_Ring_1> <Default_Ring_2>	Default ringing pattern, 1–8, for all callers. Default setting—1
<Default_CWT_1> <Default_CWT_2>	Default CWT pattern, 1–8, for all callers. Default setting—1
<Hold_Reminder_Ring_1> <Hold_Reminder_Ring_2>	Ring pattern for reminder of a holding call when the phone is on-hook. Default setting—8
<Hold_Reminder_Ring_1> <Hold_Reminder_Ring_2>	Ring pattern for reminder of a holding call when the phone is on-hook. Default setting—8
<Call_Back_Ring_1> <Call_Back_Ring_2>	Ring pattern for call back notification. Default setting—7
<Cfwd_Ring_Splash_Len_1> <Cfwd_Ring_Splash_Len_2>	Duration of ring splash when a call is forwarded (0 – 10.0s) Default setting—0

<Cblk_Ring_Splash_Len_1> <Cblk_Ring_Splash_Len_2>	Duration of ring splash when a call is blocked (0 – 10.0s) Default setting—0
<VMWI_Ring_Policy_1> <VMWI_Ring_Policy_2>	Duration of ring splash when new messages arrive before the VMWI signal is applied (0 – 10.0s) Default setting: 0
<Ring_On_No_New_VM_1> <Ring_On_No_New_VM_2>	The parameter controls when a ring splash is played when a the VM server sends a SIP NOTIFY message to the ATA indicating the status of the subscriber's mail box. Three settings are available. • New VM Available: Ring as long as there new voicemail messages. • New VM Becomes Available: Ring at the point when the first new voicemail message is received. • New VM Arrives: Ring when the number of new voicemail messages increases. Default setting—New VM Available
<VMWI_Ring_Splash_Len_1> <VMWI_Ring_Splash_Len_2>	Duration of ring splash when new messages arrive before the VMWI signal is applied (0 – 10.0s) Default setting—0
<Ring_On_No_New_VM_1> <Ring_On_No_New_VM_2>	If enabled, the ATA plays a ring splash when the voicemail server sends SIP NOTIFY message to the ATA indicating that there are no more unread voice mails. Some equipment requires a short ring to precede the FSK signal to turn off VMWI lamp. Default setting—no

<PSTN_Line_Enable_3>	To enable this line for service, select yes. Otherwise, select no. Default setting—yes
<Incoming_Handset_List_3> <Incoming_Handset_List_2>	The devices that ring when an incoming call is received on the specified line. Default setting—fxs,1,2,3,4,5,6,7,8,9,10

<SIP_ToS/DiffServ_Value_1> <SIP_ToS/DiffServ_Value_2>	TOS/DiffServ field value in UDP IP packets carrying a SIP message. Default setting—0x68
<SIP_CoS_Value_1> <SIP_CoS_Value_2>	CoS value for SIP messages. Valid values are 0 through 7. Default setting—3
<RTP_ToS/DiffServ_Value_1> <RTP_ToSDiffServ_Value_2>	ToS/DiffServ field value in UDP IP packets carrying RTP data. Default setting—0xb8
<RTP_CoS_Value_1> <RTP_CoS_Value_2>	CoS value for RTP data. Valid values are 0 through 7. Default setting—6

<Network_Jitter_Level_1> <Network_Jitter_Level_2>	Determines how jitter buffer size is adjusted by the ATA. Jitter buffer size is adjusted dynamically. The minimum jitter buffer size is 30 milliseconds or (10 milliseconds + current RTP frame size), whichever is larger, for all jitter level settings. However, the starting jitter buffer size value is larger for higher jitter levels. This setting controls the rate at which the jitter buffer size is adjusted to reach the minimum. Select the appropriate setting: low, medium, high, very high, or extremely high. Default setting—high
<Jitter_Buffer_Adjustment_1> <Jitter_Buffer_Adjustment_2>	Choose yes to enable or no to disable this feature. Default setting—yes

<SIP_Transport_1> <SIP_Transport_2>	The TCP choice provides “guaranteed delivery”, which assures that lost packets are retransmitted. TCP also guarantees that the SIP packages are received in the same order that they were sent. As a result, TCP overcomes the main disadvantages of UDP. In addition, for security reasons, most corporate firewalls block UDP ports. With TCP, new ports do not need to be opened or packets dropped, because TCP is already in use for basic activities such as Internet browsing or e-commerce. Options are: UDP, TCP, TLS, AUTO. AUTO allows the ATA to select the appropriate protocol automatically, based on the NAPTR records on the DNS server. Default setting—UDP
<SIP_Port_1> <SIP_Port_2>	Port number of the SIP message listening and transmission port. Default setting—5060
<SIP_100REL_Enable_1> <SIP_100REL_Enable_2>	To enable the support of 100REL SIP extension for reliable transmission of provisional responses (18x) and use of PRACK requests, select yes. Otherwise, select no. Default setting—no
<EXT_SIP_Port_1> <EXT_SIP_Port_2>	The external SIP port number. Default setting—blank
<Auth_Resync-Reboot_1> <Auth_Resync-Reboot_2>	If this feature is enabled, the ATA authenticates the sender when it receives the NOTIFY resync reboot (RFC 2617) message. To use this feature, select yes. Otherwise, select no. Default setting—yes
<SIP_Proxy-Require_1> <SIP_Proxy-Require_2>	The SIP proxy can support a specific extension or behavior when it sees this header from the user agent. If this field is configured and the proxy does not support it, it responds with the message, unsupported. Enter the appropriate header in the field provided. Default setting—blank
<SIP_Remote-Party-ID_1> <SIP_Remote-Party-ID_2>	To use the Remote-Party-ID header instead of the From header, select yes. Otherwise, select no. Default setting—yes

<SIP_GUID_1> <SIP_GUID_2>	This feature limits the registration of SIP accounts. The Global Unique ID is generated for each line for each ATA. When it is enabled, the ATA adds a GUID header in the SIP request. The GUID is generated the first time the unit boots up and stays with the unit through rebooting and even factory reset. Default setting—no
<RTP_Log_Intvl_1> <RTP_Log_Intvl_2>	The interval for the RTP log. Default setting—0
<Restrict_Source_IP_1> <Restrict_Source_IP_2>	If configured, the ATA drops all packets sent to its SIP Ports from an untrusted IP address. A source IP address is untrusted if it does not match any of the IP addresses resolved from the configured Proxy (or Outbound Proxy if Use Outbound Proxy is yes) Default setting—no
<Referor_Bye_Delay_1> <Referor_Bye_Delay_2>	The number of seconds to wait before sending a BYE to the referrer to terminate a stale call leg after a call transfer.
<Refer_Target_Bye_Delay_1> <Refer_Target_Bye_Delay_2>	The number of seconds to wait before sending a BYE to the refer target to terminate a stale call leg after a call transfer.
<Referee_Bye_Delay_1> <Referee_Bye_Delay_2>	The number of seconds to wait before sending a BYE to the referee to terminate a stale call leg after a call transfer.
<Refer-To_Target_Contact_1> <Refer-To_Target_Contact_2>	To contact the refer-to target, select yes. Otherwise, select no. Default setting—no
<Sticky_183_1> <Sticky_183_2>	If this feature is enabled, the ATA ignores further 180 SIP responses after receiving the first 183 SIP response for an outbound INVITE. To enable this feature, select yes. Otherwise, select no. Default setting—no
<Auth_INVITE_1> <Auth_INVITE_2>	When enabled, authorization is required for initial incoming INVITE requests from the SIP proxy. Default setting—no
<Reply_182_On_Call_Waiting_1> <Reply_182_On_Call_Waiting_2>	When enabled, the ATA replies with a SIP182 response to the caller if it is already in a call and the line is off-hook. To use this feature select yes. Default setting—no
<Use_Anonymous_With_RPID_1> <Use_Anonymous_With_RPID_2>	Determines whether or not the ATA uses “Anonymous” when Remote Party ID is requested in the SIP message. Default setting—yes
<Use_Local_Addr_In_From_1> <Use_Local_Addr_In_From_2>	Use the local ATA IP address in the SIP FROM message. Default setting—no
<Broadsoft_ALTC_1> <Broadsoft_ALTC_2>	Use Broadsoft ALTC SDP negotiation. Default setting—No

<Dial_Plan_1_3> through <Dial_Plan_8_3>	The PSTN dial plan pool. You can associate a dial plan with a VoIP Caller or a PSTN Caller by referencing the index number (1~8). Default setting—(xx.)
<VoIP-To-PSTN_Gateway_Enable_3>	Choose yes to enable or choose no to disable the VoIP-To-PSTN Gateway functionality. Default setting—yes
<VoIP_Caller_Auth_Method_3>	The method to authenticate a VoIP Caller to access the PSTN gateway. Choose from none, PIN, or HTTP Digest. Default setting—none
<VoIP_PIN_Max_Retry_3>	The number of times that a VoIP caller can attempt to enter a PIN, if the VoIP Caller Auth Method is set to PIN. Default setting—3
<One_Stage_Dialing_3>	Choose yes to enable or choose no to disable one-stage dialing. This setting applies if the VoIP Caller Auth Method is none or HTTP Digest, or if caller is in the Access List. Default setting—yes
<Line_1_VoIP_Caller_DP_3>	The index number of the dial plan to use when the VoIP Caller is calling from Line 1 of the same ATA during normal operation (in other words, not due to fallback to PSTN service when Line 1 VoIP service is down). The Authentication is skipped for Line 1 VoIP caller. Default setting—1
<VoIP_Caller_Default_DP_3>	The index number of the dial plan to use when the VoIP Caller is not authenticated. Default setting—1
<Line_1_Fallback_DP_3>	The index number of the dial plan to use when the VoIP Caller is calling from Line 1 of the same ATA due to fallback to PSTN service when Line 1 VoIP service is down. Default setting—none
<VoIP_Caller_ID_Pattern_3>	A comma-separated list of caller phone number patterns that is used to allow or block access to the PSTN gateway based on the caller ID. If the caller ID does not match a specified pattern, access is rejected, regardless of the authentication method. This comparison is applied before the access list is applied. If this parameter is blank (not specified), all callers are considered for VoIP service. Use ? to match any single digit. Use * to match any number of digits. **Example**: 1408*, 1512???1234 In the above example, the caller ID either must start with 1408 or must be an 11-digit numbering starting with 1512 and ending with 1234. Default setting—blank

<VoIP_Access_List_3>	A comma-separated list of number patterns that is used to allow or block access to the PSTN gateway based on the source IP address. If the IP address matches a specified pattern, service is allowed without further authentication. **Example**: 192.168.*.*, 66.43.12.1??. In the above example, the source IP address either must begin with 192.168 or must be in the range of 66.43.12.100-199. Default setting—blank
<VoIP_Caller_1_PIN_3> through <VoIP_Caller_8_PIN_3>	A PIN number that a VoIP caller can use to access the PSTN gateway, when the VoIP Caller Auth Method is set to PIN. Default setting—blank
<VoIP_Caller_1_DP_3> through <VoIP_Caller_8_DP_3>	The index number of the dial plan to use upon successful entry of the corresponding VoIP Caller PIN. Default setting—1
<VoIP_User_1_Auth_ID_3> through <VoIP_User_8_Auth_ID_3>	A user ID that a VoIP Caller can use for authentication by using the HTTP Digest method (in other words, by embedding an Authorization header in the SIP INVITE message sent to the ATA. If the credentials are missing or incorrect, the ATA will challenge the caller with a 401 response). The VoIP caller whose authentication userid equals to this ID is referred to VoIP User 1 of this ATA. If the caller specifies an authentication user-id that does not match any of the VoIP User Auth ID's, the INVITE will be rejected with a 403 response. Default setting—blank.
<VoIP_User_1_Password_3> through <VoIP_User_8_Password_3>	The password to be used with VoIP User 1. The user assumes the identity of VoIP User 1 must therefore compute the credentials using this password, or the INVITE will be challenged with a 401 response Default setting—blank.
<VoIP_User_1_DP_3> through <VoIP_User_8_DP_3>	For up to 8 VoIP users, specify the index of the dial plan to be used after successful authentication. If authentication is disabled, the default dial plan is used for all unknown VoIP users. Default setting—1.

<PSTN-To-VoIP_Gateway_Enable_3>	Select yes to enable or select no to disable PSTN-To-VoIP Gateway functionality. Default setting—yes
<PSTN_Caller_Auth_Method_3>	The method to authenticate a PSTN Caller to access the VoIP gateway. Choose from none or PIN. Default setting—none
<PSTN_Ring_Thru_1_3>	To enable ring through to Line 1 based on caller number patterns, choose yes. Otherwise choose no. **Note** For more information about PSTN Caller number patterns, see <PSTN_Caller_ID_Pattern_3>. Default setting—yes

<PSTN_PIN_Max_Retry_3>	The number of times that a PSTN caller can attempt to enter a PIN number, if the authentication method is set to PIN. Default setting—3
<PSTN_CID_for_VoIP_CID_3>	Choose yes or no. Default setting—no
<PSTN_CID_Number_Prefix_3>	A dialing prefix, if needed, to add to the caller ID number on the PBX to ensure that a callback goes to the correct number. Default setting—blank
<PSTN_Caller_Default_DP_3>	The index number of the dial plan that is used when the PSTN Caller Auth Method is set to none. Default settings: 1
<Line_1_Signal_Hook_Flash_to_PSTN_3>	Specify the operation of the hook flash on the analog phone when a PSTN-to-VoIP call is active. Choose Disabled or Double Hook Flash. Default setting—Disabled
<PSTN_CID_Name_Prefix_3>	The prefix to add to the caller ID name that is sent to the PBX. Enter the characters to add to the caller ID name. Default setting—blank
<PSTN_Caller_ID_Pattern_3>	A comma-separated list of phone number patterns that is used to allow or block access to the VoIP gateway based on the caller ID. If the caller ID does not match a specified pattern, access is rejected, regardless of the authentication method. This comparison is applied before the access list is applied. If this parameter is blank (not specified), all callers are considered for VoIP service. • Use ? to match any single digit. • Use * to match any number of digits. **Example**: 1408*, 1512???1234 In the above example, the caller ID either must start with 1408 or must be an 11-digit numbering starting with 1512 and ending with 1234. Default setting—blank
<PSTN_Access_List_3>	A comma-separated list of number patterns that is used to allow or block access to the VoIP gateway based on the destination IP address. If the destination IP address matches a specified pattern, service is allowed without further authentication. **Example**: 192.168.*.*, 66.43.12.1??. In the above example, the IP address either must begin with 192.168 or must be in the range of 66.43.12.100-199. The default is blank.
<PSTN_Caller_1_PIN_3> through <PSTN_Caller_8_PIN_3>	A PIN number that allows a PSTN caller to access to the VoIP gateway. Calls will be subject to the dial plan specified by the corresponding PSTN Caller DP setting (see below). These settings apply when the PSTN Caller Authentication Method parameter is set to PIN. Default setting—blank

<PSTN_Caller_1_DP_3> through <PSTN_Caller_8_DP_3>	The index number of the dial plan to use upon successful entry of the corresponding PSTN Caller PIN. Default setting—1

<VoIP_Answer_Delay_3>	The number of seconds to wait before autoanswering an inbound VoIP call for the FXO account. The range is 0-255. Default setting—0
<VoIP_PIN_Digit_Timeout_3>	After a VoIP caller is prompted for a PIN or enters a digit, the number of seconds to wait for an entry. The range is 0-255. Default setting—10
<PSTN_Answer_Delay_3>	After an inbound PSTN call starts ringing, the number of seconds to wait before autoanswering the call. The range is 0-255. Default setting—16
<PSTN_PIN_Digit_Timeout_3>	After a PSTN caller is prompted for a PIN or enters a digit, the number of seconds to wait for an entry. The range is 0-255. Default setting—10
<PSTN-To-VoIP_Call_Max_Dur_3>	The limit on the duration of a PSTN-To-VoIP Gateway Call. Unit is in seconds. 0 means unlimited. The range is 0-2147483647. Default setting—0
<PSTN_Ring_Thru_Delay_3>	After a PSTN call starts ringing, the number of seconds to wait before ring through to Line 1. In order for Line 1 to have the caller ID information, this value must be greater than the time required to complete the PSTN caller ID delivery. The range is 0-255. Default setting—1
<VoIP-To-PSTN_Call_Max_Dur_3>	The limit on the duration of a VoIP-To-PSTN Gateway Call. Unit is in seconds. 0 means unlimited. The range is 0-2147483647. Default setting—0
<PSTN_Ring_Thru_CWT_Delay_3>	When a call is active and a new PSTN call starts ringing, the number of seconds to wait before ring through to Line 1 with a Call Waiting Tone. Default setting—3
<VoIP_DLG_Refresh_Intvl_3>	The interval between (SIP) Dialog refresh messages sent by the ATA to detect if the VoIP call-leg is still up. If this value is set to 0, the VoIP call-leg status will not be checked by the ATA. The refresh message is a SIP ReINVITE, and the VoIP peer must response with a 2xx response. If the VoIP peer does not reply or the response is not greater than 2xx, the ATA will disconnect both call legs automatically. The range is 0-255. Default setting—0
<PSTN_Ring_Timeout_3>	After a ring burst, the number of seconds to wait before concluding that PSTN ring has ceased. The range is 0-255. Default setting—5

<PSTN_Dialing_Delay_3>	After hook, the number of seconds to wait before dialing a PSTN number. The range is 0-255. Default setting—1
<PSTN_Dial_Digit_Len_3>	The on/off time when the Gateway transmits digits through the Line (FXO) port. The syntax is on-time/off-time, expressed in seconds. The permitted range is 0.05 to 3.00 (up to two decimal places only). Default setting—.1/.1
<PSTN_Hook_Flash_Len_3>	The length of the hook flash in seconds. Default setting—.25
<Detect_CPC_3>	Choose yes to enable or choose no to disable this feature. CPC is a brief removal of tip-and-ring voltage. If enabled, the ATA will disconnect both call legs when this signal is detected during a gateway call. Default setting—yes
<Detect_Polarity_Reversal_3>	Choose yes to enable or choose no to disable this feature. If enabled, the ATA will disconnect both call legs when this signal is detected during a gateway call. If it is a PSTN gateway call, the first polarity reversal is ignored and the second one triggers the disconnection. For VoIP gateway call, the first polarity reversal triggers the disconnection. Default setting—yes
<Detect_PSTN_Long_Silence_3>	Choose yes to enable or choose no to disable this feature. If enabled, the ATA will disconnect both call legs when the PSTN side has no voice activity for a duration longer than the length specified in the Long Silence Duration parameter during a gateway call. Default setting—no
<Detect_VoIP_Long_Silence_3>	Choose yes to enable or choose no to disable this feature. If enabled, the ATA will disconnect both call legs when the VoIP side has no voice activity for a duration longer than the length specified in the Long Silence Duration parameter during a gateway call. Default setting—no
<PSTN_Long_Silence_Duration_3>	This value is minimum length of PSTN silence (or inactivity) in seconds to trigger a gateway call disconnection if Detect Long Silence is enabled. Default setting—30
<VoIP_Long_Silence_Duration_3>	This value is minimum length of VoIP silence (or inactivity) in seconds to trigger a gateway call disconnection if Detect Long Silence is enabled. Default setting—30
<PSTN_Silence_Threshold_3>	This parameter adjusts the sensitivity of PSTN silence detection. Choose from {very low, low, medium, high, very high}. The higher the setting, the easier to detect silence and hence easier to trigger a disconnection. Default setting—medium
<Min_CPC_Duration_3>	Specify the minimum duration of a low tip and ring voltage (below 1V) for the Gateway to recognize it as a CPC signal or PSTN line removal. Default setting—0.2

<Detect_Disconnect_Tone_3>	Choose yes to enable or choose no to disable this feature. If enabled, the ATA will disconnect both call legs when it detects the disconnect tone from the PSTN side during a gateway call. Disconnect tone is specified in the Disconnect Tone parameter, which depends on the region of the PSTN service. Default setting—yes
<Disconnect_Tone_3>	This value is the tone script which describes to the ATA the tone to detect as a disconnect tone. The syntax follows a standard Tone Script with some restrictions. Default value is standard US reorder (fast busy) tone, for 4 seconds. Default setting—480@-30,620@-30;4(.25/.25/1+2) Restrictions: • Two frequency components must be given. If single frequency is desired, the same frequency is used for both. • The tone level value is not used. –30 (dBm) should be used for now. • Only 1 segment set is allowed. • Total duration of the segment set is interpreted as the minimum duration of the tone to trigger detection. • 6 segments of on/off time (seconds) can be specified. A 10% margin is used to validated cadence characteristics of the tone.
<Disconnect_Tone_3>	Disconnect Tone Script values: • US—480@-30,620@-30;4(.25/.25/1+2) • UK—400@-30,400@-30; 2(3/0/1+2) • France—440@-30,440@-30; 2(0.5/0.5/1+2) • Germany—440@-30,440@-30; 2(0.5/0.5/1+2) • Netherlands—425@-30,425@-30; 2(0.5/0.5/1+2) • Sweden—425@-10; 10(0.25/0.25/1) • • Norway—425@-10; 10(0.5/0.5/1) • Italy—425@-30,425@-30; 2(0.2/0.2/1+2) • Spain—425@-10; 10(0.2/0.2/1,0.2/0.2/1,0.2/0.6/1) • Portugal—425@-10; 10(0.5/0.5/1) • Poland—425@-10; 10(0.5/0.5/1) • Denmark—425@-10; 10(0.25/0.25/1) • New Zealand—400@-15; 10(0.25/0.25/1) • Australia—425@-13; 10(0.375/0.375/1)

<FXO_Country_Setting_3>	The country of deployment. This setting applies the relevant regional settings for PSTN calls. Default setting—USA
<Tip_Ring_Voltage_Adjust_3>	Voltage adjustment. The choices are 3.1V, 3.2V, 3.35V, and 3.5V. Default setting—3.5V.
<Ring_Frequency_Min_3>	The lower limit of the ring frequency used to detect the ring signal. Default setting—10
<SPA_To_PSTN_Gain_3>	dB of digital gain (or attenuation if negative) to be applied to the signal sent from the ATA to the PSTN side. The range is -15 to 12. Default setting—0
<Ring_Frequency_Max_3>	The higher limit of the ring frequency used to detect the ring signal. Default setting—100
<PSTN_To_SPA_Gain_3>	dB of digital gain (or attenuation if negative) to be applied to the signal sent from the PSTN side to the ATA. The range is -15 to 12. Default setting—0
<Ring_Validation_Time_3>	The minimum signal duration required by the Gateway for recognition as a ring signal. Default setting—256 ms
<Ring_Indication_Delay_3>	Choose from {0, 512, 768, 1024, 1280, 1536, 1792} (ms). Default setting—512ms
<Ring_Timeout_3>	Choose from {0, 128, 256, 384, 512, 640, 768, 896, 1024, 1152, 1280, 1408, 1536, 1664, 1792, 1920} (ms). Default setting—640 ms
<Ring_Threshold_3>	Choose from {13.5–16.5, 19.35–2.65, 40.5–49.5} (Vrms). Default setting—13.5-16.5 Vrms
<Line-In-Use_Voltage_3>	The voltage threshold at which the ATA assumes the PSTN is in use by another handset sharing the same line (and will declare PSTN gateway service not available to incoming VoIP callers). Default setting—30
<Dial_Tone>	Prompts the user to enter a phone number. Reorder Tone is played automatically when Dial Tone or any of its alternatives times out. Default setting—350@-5,440@-5;10(*/0/1+2)
<Second_Dial_Tone>	Alternative to the Dial Tone when the user dials a three-way call. Default setting—420@-5,520@-5;10(*/0/1+2)
<Outside_Dial_Tone>	Alternative to the Dial Tone. It prompts the user to enter an external phone number, as opposed to an internal extension. It is triggered by a comma character encountered in the dial plan. Default setting—420@-4;10(*/0/1)

<Prompt_Tone>	Prompts the user to enter a call forwarding phone number. Default setting—520@-5,620@-5;10(*/0/1+2)
<Busy_Tone>	Played when a 486 RSC is received for an outbound call. Default setting—480@-5,620@-5;10(.5/.5/1+2)
<Reorder_Tone>	Played when an outbound call has failed, or after the far end hangs up during an established call. Reorder Tone is played automatically when Dial Tone or any of its alternatives times out. Default setting—480@-5,620@-5;10(.25/.25/1+2)
<Off_Hook_Warning_Tone>	Played when the caller has not properly placed the handset on the cradle. Off Hook Warning Tone is played when the Reorder Tone times out. Default setting—480@-3,620@3;10(.125/.125/1+2)
<Ring_Back_Tone>	Played during an outbound call when the far end is ringing. Default setting—440@-5,480@-5;*(2/4/1+2)
<Ring_Back_2_Tone>	Your ATA plays this ringback tone instead of Ring Back Tone if the called party replies with a SIP 182 response without SDP to its outbound INVITE request. Default setting—the same as Ring Back Tone, except the cadence is 1s on and 1s off. Default setting—440@-5,480@-5;*(1/1/1+2)
<Confirm_Tone>	Brief tone to notify the user that the last input value has been accepted. Default setting—600@-4;1(.25/.25/1)
<SIT1_Tone> through <SIT4_Tone>	Alternative to the Reorder Tone played when an error occurs as a caller makes an outbound call. The RSC to trigger this tone is configurable on the SIP screen. Default setting—985@-4,1428@-4,1777@-4;20(.380/0/1,.380/0/2,.380/0/3,0/4/0)
<MWI_Dial_Tone>	Played instead of the Dial Tone when there are unheard messages in the caller’s mailbox. Default setting—350@-5,440@-5;2(.1/.1/1+2);10(*/0/1+2)
<Cfwd_Dial_Tone>	Played when all calls are forwarded. Default setting—350@-5,440@-5;2(.2/.2/1+2);10(*/0/1+2)
<Holding_Tone>	Informs the local caller that the far end has placed the call on hold. Default setting—600@-5;*(.1/.1/1,.1/.1/1,.1/9.5/1)
<Conference_Tone>	Played to all parties when a three-way conference call is in progress. Default setting—350@-5;20(.1/.1/1,.1/9.7/1)
<Secure_Call_Indication_Tone>	Played when a call has been successfully switched to secure mode. It should be played only for a short while (less than 30 seconds) and at a reduced level (less than -19 dBm) so it does not interfere with the conversation. Default setting—397@-5,507@-5;15(0/2/0,.2/.1/1,.1/2.1/2)
<VoIP_PIN_Tone>	This tone is played to prompt a VoIP caller to enter a PIN number.

<PSTN_PIN_Tone>	This tone is played to prompt a PSTN caller to enter a PIN number.
<Feature_Invocation_Tone>	Played when a feature is implemented. Default setting—350@-4;*(.1/.1/1)

<Ring1_Cadence>	Cadence script for distinctive ring 1. Default setting—60(2/4)
<Ring2_Cadence>	Cadence script for distinctive ring 2. Default setting—60(.8/.4,.8/4)
<Ring3_Cadence>	Cadence script for distinctive ring 3. Default setting—60(.4/.2,.4/.2,.8/4)
<Ring4_Cadence>	Cadence script for distinctive ring 4. Default setting—60(.3/.2,1/.2,.3/4)
<Ring5_Cadence>	Cadence script for distinctive ring 5. Default setting—1(.5/.5)
<Ring6_Cadence>	Cadence script for distinctive ring 6. Default setting—60(.2/.4,.2/.4,.2/4)
<Ring7_Cadence>	Cadence script for distinctive ring 7. Default setting—60(.4/.2,.4/.2,.4/4)
<Ring8_Cadence>	Cadence script for distinctive ring 8. Default setting—60(0.25/9.75)

<CWT1_Cadence>	Cadence script for distinctive CWT 1. Default setting—30(.3/9.7)
<CWT2_Cadence>	Cadence script for distinctive CWT 2. Default setting—30(.1/.1, .1/9.7)
<CWT3_Cadence>	Cadence script for distinctive CWT 3. Default setting—30(.1/.1, .1/.1, .1/9.7)
<CWT4_Cadence>	Cadence script for distinctive CWT 4. Default setting—30(.1/.1, .3/.1, .1/9.3)
<CWT5_Cadence>	Cadence script for distinctive CWT 5. Default setting—1(.5/.5)

<CWT6_Cadence>	Cadence script for distinctive CWT 6. Default setting—30(.3/.1,.3/.1,.1/9.1)
<CWT7_Cadence>	Cadence script for distinctive CWT 7. Default setting—30(.3/.1,.3/.1,.1/9.1)
<CWT8_Cadence>	Cadence script for distinctive CWT 8. Default setting—2.3(.3/2)
<Ring1_Name>	Name in an INVITE’s Alert-Info Header to pick distinctive ring/CWT 1 for the inbound call. Default setting—Bellcore-r1
<Ring2_Name>	Name in an INVITE’s Alert-Info Header to pick distinctive ring/CWT 2 for the inbound call. Default setting—Bellcore-r2
<Ring3_Name>	Name in an INVITE’s Alert-Info Header to pick distinctive ring/CWT 3 for the inbound call. Default setting—Bellcore-r3
<Ring4_Name>	Name in an INVITE’s Alert-Info Header to pick distinctive ring/CWT 4 for the inbound call. Default setting—Bellcore-r4
<Ring5_Name>	Name in an INVITE’s Alert-Info Header to pick distinctive ring/CWT 5 for the inbound call. Default setting—Bellcore-r5
<Ring6_Name>	Name in an INVITE’s Alert-Info Header to pick distinctive ring/CWT 6 for the inbound call. Default setting—Bellcore-r6
<Ring7_Name>	Name in an INVITE’s Alert-Info Header to pick distinctive ring/CWT 7 for the inbound call. Default setting—Bellcore-r7
<Ring8_Name>	Name in an INVITE’s Alert-Info Header to pick distinctive ring/CWT 8 for the inbound call. Default setting—Bellcore-r8

<Ring_Waveform>	Waveform for the ringing signal. Choices are Sinusoid or Trapezoid. Default setting—Sinusoid
<Ring_Frequency>	Frequency of the ringing signal. Valid values are 10–100 (Hz) Default setting—20
<Ring_Voltage>	Ringing voltage. Choices are 60–90 (V) Default setting—85
<CWT_Frequency>	Frequency script of the call waiting tone. All distinctive CWTs are based on this tone. Default setting—440@-10

<Synchronized_Ring>	If this is set to yes, when the ATA is called, all lines ring at the same time (similar to a regular PSTN line) After one line answers, the others stop ringing. Default setting—no

<Hook_Flash_Timer_Min>	Minimum on-hook time before off-hook qualifies as hook flash. Less than this the onhook event is ignored. Range: 0.1–0.4 seconds. Default setting—0.1
<Hook_Flash_Timer_Max>	Maximum on-hook time before off-hook qualifies as hook flash. More than this the onhook event is treated as on hook (no hookflash event) Range: 0.4–1.6 seconds. Default setting—0.9
<Callee_On_<Hook_Delay>	Phone must be on-hook for at this time in sec. before the ATA will tear down the current inbound call. It does not apply to outbound calls. Range: 0–255 seconds. Default setting—0
<Reorder_Delay>	Delay after far end hangs up before reorder tone is played. 0 =plays immediately, inf =never plays. Range: 0–255 seconds. Default setting—5.
<Call_Back_Expires>	Expiration time in seconds of a call back activation. Range: 0–65535 seconds. Default setting—1800
<Call_Back_Retry_Intvl>	Call back retry interval in seconds. Range: 0–255 seconds. Default setting—30
<Call_Back_Delay>	Delay after receiving the first SIP 18x response before declaring the remote end is ringing. If a busy response is received during this time, the ATA still considers the call as failed and keeps on retrying. Range: 0–65 seconds Default setting—0.5
<VMWI_Refresh_Intvl>	Interval between VMWI refresh to the device. Range: 0–65535 seconds Default setting—0
<Interdigit_Long_Timer>	Long timeout between entering digits when dialing. The interdigit timer values are used as defaults when dialing. The Interdigit_Long_Timer is used after any one digit, if all valid matching sequences in the dial plan are incomplete as dialed. Range: 0–64 seconds. Default setting—10
<Interdigit_Short_Timer>	Short timeout between entering digits when dialing. The Interdigit_Short_Timer is used after any one digit, if at least one matching sequence is complete as dialed, but more dialed digits would match other as yet incomplete sequences. Range: 0–64 seconds. Default setting—3

<CPC_Delay>	Delay in seconds after caller hangs up when the ATA starts removing the tip-and-ring voltage to the attached equipment of the called party. The range is 0–255 seconds. This feature is generally used for answer supervision on the caller side to signal to the attached equipment when the call has been connected (remote end has answered) or disconnected (remote end has hung up) This feature should be disabled for the called party (in other words, by using the same polarity for connected and idle state) and the CPC feature should be used instead. Without CPC enabled, reorder tone will is played after a configurable delay. If CPC is enabled, dial tone will be played when tip-to-ring voltage is restored. Resolution is 1 second. Default setting—2
<CPC_Duration>	Duration in seconds for which the tip-to-ring voltage is removed after the caller hangs up. After that, tip-to-ring voltage is restored and the dial tone applies if the attached equipment is still off-hook. CPC is disabled if this value is set to 0. Range: 0 to 1.000 second. Resolution is 0.001 second. Default setting—0 (CPC disabled)

<Call_Return_Code>	Call Return Code This code calls the last caller. Default setting—*69
<Call_Redial_Code>	Redials the last number called. Default setting—*07
<Blind_Transfer_Code>	Begins a blind transfer of the current call to the extension specified after the activation code. Default setting—*98
<Call_Back_Act_Code>	Starts a callback when the last outbound call is not busy. Default setting—*66
<Call_Back_Deact_Code>	Cancels a callback. Default setting—*86
<Call_Back_Busy_Act_Code>	Starts a callback when the last outbound call is busy. Default setting—*05
<Cfwd_All_Act_Code>	Forwards all calls to the extension specified after the activation code. Default setting—*72
<Cfwd_All_Deact_Code>	Cancels call forwarding of all calls. Default setting—*73
<Cfwd_Busy_Act_Code>	Forwards busy calls to the extension specified after the activation code. Default setting—*90
<Cfwd_Busy_Deact_Code>	Cancels call forwarding of busy calls. Default setting—*91

<Cfwd_No_Ans_Act_Code>	Forwards no-answer calls to the extension specified after the activation code. Default setting—*92
<Cfwd_No_Ans_Deact_Code>	Cancels call forwarding of no-answer calls. Default setting—*93
<Cfwd_Last_Act_Code>	Forwards the last inbound or outbound call to the number that the user specifies after entering the activation code. Default setting—*63
<Cfwd_Last_Deact_Code>	Cancels call forwarding of the last inbound or outbound call. Default setting—*83
<Block_Last_Act_Code>	Blocks the last inbound call. Default setting—*60
<Block_Last_Deact_Code>	Cancels blocking of the last inbound call. Default setting—*80
<Accept_Last_Act_Code>	Accepts the last outbound call. It lets the call ring through when do not disturb or call forwarding of all calls are enabled. Default setting—*64
<Accept_Last_Deact_Code>	Cancels the code to accept the last outbound call. Default setting—*84
<CW_Act_Code>	Enables call waiting on all calls. Default setting—*56
<CW_Deact_Code>	
	Enables call waiting on all calls. Default setting—*57
<CW_Per_Call_Act_Code>	Enables call waiting for the next call. Default setting—*71
<CW_Per_Call_Deact_Code>	Disables call waiting for the next call. Default setting—*70
<Block_CID_Act_Code>	Blocks caller ID on all outbound calls. Default setting—*67
<Block_CID_Deact_Code>	Removes caller ID blocking on all outbound calls. Default setting—*68

<Block_CID_Per_Call_Act_Code>	Blocks caller ID on the next outbound call. Default setting—*81
<Block_CID_Per_Call_Deact_Code>	Removes caller ID blocking on the next inbound call. Default setting—*82
<Block_ANC_Act_Code>	Blocks all anonymous calls. Default setting—*77
<Block_ANC_Deact_Code>	Removes blocking of all anonymous calls. Default setting—*87
<DND_Act_Code>	Enables the do not disturb feature. Default setting—*78
<DND_Deact_Code>	Disables the do not disturb feature. Default setting—*79
<CID_Act_Code>	Enables caller ID generation. Default setting—*65
<CID_Deact_Code>	Disables caller ID generation. Default setting—*85
<CWCID_Act_Code>	Enables call waiting, caller ID generation. Default setting—*25
<CWCID_Deact_Code>	Disables call waiting, caller ID generation. Default setting—*45
<Dist_Ring_Act_Code>	Enables the distinctive ringing feature. Default setting—*26
<Dist_Ring_Deact_Code>	Disables the distinctive ringing feature. Default setting—*46
<Speed_Dial_Act_Code>	Assigns a speed dial number. Default setting—*74
<Paging_Code>	Used for paging other clients in the group. Default setting—*96
<Secure_All_Call_Act_Code>	Makes all outbound calls secure. Default setting—*16

<Secure_No_Call_Act_Code>	Makes all outbound calls not secure. Default setting—*17
<Secure_One_Call_Act_Code>	Makes the next outbound call secure. (It is redundant if all outbound calls are secure by default.) Default setting—*18
<Secure_One_Call_Deact_Code>	Makes the next outbound call not secure. (It is redundant if all outbound calls are not secure by default.) Default setting—*19
<Conference_Act_Code>	If this code is specified, the user must enter it before dialing the third party for a conference call. Enter the code for a conference call. Default setting—blank
<Attn-Xfer_Act_Code>	If the code is specified, the user must enter it before dialing the third party for a call transfer. Enter the code for a call transfer. Default setting—blank
<Modem_Line_Toggle_Code>	Toggles the line to a modem. Modem passthrough mode can be triggered only by pre-dialing this code. Default setting—*99
<FAX_Line_Toggle_Code>	Toggles the line to a fax machine. Default setting—#99
<Media_Loopback_Code>	Use for media loopback. Default setting—*03
<Referral_Services_Codes>	These codes tell the ATA what to do when the user places the current call on hold and is listening to the second dial tone. One or more *codes can be configured into this parameter, such as *98, or *97|*98|*123, etc. The maximum length is 79 characters. This parameter applies when the user places the current call on hold by pressing the hook flash button. Each *code (and the following valid target number according to current dial plan) triggers the ATA to perform a blind transfer to a target number that is prepended by the service *code. For example, after the user dials *98, the ATA plays a special dial tone called the Prompt Tone while waiting for the user to enter a target number (which is checked according to dial plan as in normal dialing). When a complete number is entered, the ATA sends a blind REFER to the holding party with the Refer-To target equal to *98 target_number. This feature allows the ATA to hand off a call to an application server to perform further processing, such as call park. The *codes should not conflict with any of the other vertical service codes internally processed by the ATA. You can empty the corresponding *code that you do not want the ATA to process. Default setting—blank

<Feature_Dial_Services_Codes>	These codes tell the ATA what to do when the user is listening to the first or second dial tone. One or more *codes can be configured into this parameter, such as *72, or *72|*74|*67|*82, etc. The maximum length is 79 characters. This parameter applies when the user has a dial tone (first or second dial tone.) After receiving dial tone, a user enters the *code and the target number according to current dial plan. For example, after user dials *72, the ATA plays a special tone called a Prompt tone while awaiting the user to enter a valid target number. When a complete number is entered, the ATA sends a INVITE to *72 target_number as in a normal call. This feature allows the proxy to process features like call forward (*72) or Block Caller ID (*67.) The *codes should not conflict with any of the other vertical service codes internally processed by the ATA. You can remove a corresponding *code that you do not want the ATA to process. You can add a parameter to indicate which tone plays after the *code is entered, such as *72‘c‘|*67‘p‘. Below is a list of allowed tone parameters. (Note the use of open quotes surrounding the parameter, without spaces.) • ‘c‘ = <Cfwd Dial Tone> • ‘d‘ = <Dial Tone> • ‘m‘ = <MWI Dial Tone> • ‘o‘ = <Outside Dial Tone> • ‘p‘ = <Prompt Dial Tone> • ‘s‘ = <Second Dial Tone> • ‘x‘ = No tones are place, x is any digit not used above If no tone parameter is specified, the ATA plays Prompt tone by default. If the *code is not to be followed by a phone number, such as *73 to cancel call forwarding, do not include this parameter. Instead, add the code in the dial plan and the ATA send INVITE *73@..... as usual when user dials *73. Default setting—blank

<Service_Annc_Base_Number>	Base number for service announcements. Default setting—blank
<Service_Annc_Extension_Codes>	Extension codes for service announcements. Default setting—blank

<Prefer_G711u_Code>	Dial prefix to make G.711u the preferred codec for the call. Default setting—*017110
<Force_G711u_Code>	Dial prefix to make G.711u the only codec that can be used for the call. Default setting—*027110

<Prefer_G711a_Code>	Dial prefix to make G.711a the preferred codec for the call. Default setting—*017111
<Force_G711a_Code>	Dial prefix to make G.711a the only codec that can be used for the call. Default setting—*027111
<Prefer_G726r32_Code>	Dial prefix to make G.726r32 the preferred codec for the call. Default setting—*0172632
<Force_G726r32_Code>	Dial prefix to make G.726r32 the only codec that can be used for the call. Default setting—*0272632
<Prefer_G729a_Code>	Dial prefix to make G.729a the preferred codec for the call. Default setting—*01729
<Force_G729a_Code>	Dial prefix to make G.729a the only codec that can be used for the call. Default setting—*02729
<Prefer_G722_Code>	Dial prefix to make G.722 the preferred codec for the call. Default setting—*01722
<Force_G722_Code>	Dial prefix to make G.722 the only codec that can be used for the call. Default setting—*02722

<FXS_Port_Impedance>	Sets the electrical impedance of the PHONE port. Choices are: 600, 900, 600+2.16uF, 900+2.16uF, 270+750||150nF, 220+850||120nF, 220+820||115nF, or 200+600||100nF. Default setting—600 **Note** For New Zealand impedance (370+620||310nF), use 270+750||150nF.
<FXS_Port_Input_Gain>	Input gain in dB, up to three decimal places. The range is 6.000 to -12.000. Default setting—-3
<FXS_Port_Output_Gain>	Output gain in dB, up to three decimal places. The range is 6.000 to -12.000. The Call Progress Tones and DTMF playback level are not affected by the FXS Port Output Gain parameter. Default setting—-3
<DTMF_Playback_Level>	Local DTMF playback level in dBm, up to one decimal place. Range: -30–0. Default setting—-16.0
<DTMF_Playback_Length>	Local DTMF playback duration in milliseconds. Range: 0–65 seconds. Default setting—0.1
<Detect_ABCD>	To enable local detection of DTMF ABCD, select yes. Otherwise, select no. Default setting—yes This setting has no effect if DTMF Tx Method is INFO; ABCD is always sent OOB regardless in this setting.

<Playback_ABCD>	To enable local playback of OOB DTMF ABCD, select yes. Otherwise, select no. Default setting—yes
<Caller_ID_Method> Default setting—Bellcore(N.Amer, China)	The choices are described below. Default setting—Bellcore (N.Amer, China) • Bellcore(N.Amer,China): CID, CIDCW, and VMWI. FSK sent after first ring (same as ETSI FSK sent after first ring) (no polarity reversal or DTAS) • DTMF(Finland, Sweden): CID only. DTMF sent after polarity reversal (and no DTAS) and before first ring. • DTMF(Denmark): CID only. DTMF sent before first ring with no polarity reversal and no DTAS. • ETSI DTMF: CID only. DTMF sent after DTAS (and no polarity reversal) and before first ring. • ETSI DTMF With PR: CID only. DTMF sent after polarity reversal and DTAS and before first ring. • ETSI DTMF After Ring: CID only. DTMF sent after first ring (no polarity reversal or DTAS) • ETSI FSK: CID, CIDCW, and VMWI. FSK sent after DTAS (but no polarity reversal) and before first ring. Waits for ACK from a device after DTAS for CIDCW. • ETSI FSK With PR (UK): CID, CIDCW, and VMWI. FSK is sent after polarity reversal and DTAS and before first ring. Waits for ACK from a device after DTAS for CIDCW. Polarity reversal is applied only if equipment is on hook. • DTMF (Denmark) with PR: CID only. DTMF sent after polarity reversal (and no DTAS) and before first ring. Default setting—Bellcore(N.Amer, China)
<FXS_Port_Power_Limit>	The choices are from 1 to 8. Default setting—3
<Caller_ID_FSK_Standard>	The ATA supports bell 202 and v.23 standards for caller ID generation. Default setting—bell 202
<Feature_Invocation_Method>	Select the method you want to use, Default or Sweden default. Default setting—Default
<DECT_Enable>	To enable this handset for service, select yes. Otherwise, select no. Default setting—yes
<Call_Park_Enable>	Enables or disables Call Park. Default setting—No
<Call_Pickup_Enable>	Enables or disables Call Pickup. Default setting—No

<Call_Group_Pickup_Enable>	Enables or disables Group Pickup. Default setting—No

<Outgoing_Lines>	A comma-separated list of the index numbers (1~10) for the lines that are available from this handset for an outgoing call. These lines will be listed on the phone screen when the user displays the call options or holds down the green call button. **Example**: 1,2,8 In this example, a user can select DECT line 1, 2, or 8 for an outbound call. Default setting—1 **Note** You also can choose these lines from the DECT Handset Outgoing Line Selection section of the Quick Setup page.
<Failover>	When this feature is enabled and a call fails through the selected line, the ATA automatically attempts to place the call over another enabled DECT line. Select yes to enable this feature or select no to disable it. Default setting—no
<Deregister>	To deregister a handset, select yes. After you submit the settings and the voice module reboots, then the handset is deregistered. At that point, this parameter is reset to the default value. Default setting—no
<Bound_IPEI>	Enter the device's IPEI number (a unique hardware identifier comparable to a MAC address) if you want to bind this device to the specified handset ID, such as Handset 3. The IPEI can be found in the **Settings** > **Phone Info** menu on the handset. Default setting—blank

CHAPTER 8

Router Configuration Parameters

Nested Structure

All items in the <router_configuration> section of the XML file need to be nested under <router-configuration> and the section headings as shown below.

• The </router-configuration> tag must appear at the end of the section.

• In the XML file, each section can be opened or closed by clicking the section heading. A + symbol indicates that a section is closed, and a -symbol indicates that it is open.

• To enter a null value, enter a backslash at the end of the parameter name, as show in this example: <MAC_Address_Clone_Address />

Nested Sections

```
- <flat-profile>
   ...
   ...
- <router-configuration>
   + <WAN_Basic_Setting>
   + <WAN_Interface>
   + <WAN_IP6_Setting>
   + <PHY_Port_Setting>
   + <MAC_Address_Clone>
   + <Internet_Option>
   + <DHCP_Server_Pool>
   + <LAN_IP6_Setting>
   + <WAN_VLAN_Setting>
   + <CLDP_Setting>
   + <Single_Port_Forwarding>
   + <Port_Rang_Forwarding>
   + <SNMP>
   + <Time_Setup>
    <QoS_Bandwidth_Control>
   + <Software_DMZ>
     <Bonjour_Enable>1</Bonjour_Enable>
     <Reset_Button_Enable>1</Reset_Button_Enable>
     <Router_Mode>1</Router_Mode>
     <Monitor_WAN_Port_Only>0</Monitor_WAN_Port_Only>
   + <VPN_Passthrough>
   + <Web_Management>
   + <TR-069>
   + <Log_Configuration>
     <Web_Login_Admin_Name>admin</Web_Login_Admin_Name>
     <!--  <Web_Login_Admin_Password></Web_Login_Admin_Password  -->
     <Web_Login_Guest_Name>cisco</Web_Login_Guest_Name>
     <!--  <Web_Login_Guest_Password></Web_Login_Guest_Password  -->
   + <SSH>
   </router-configuration>
</flat-profile>
```

WAN_Basic_Setting Parameters

This section describes the parameters in the <x> section of the config.xml file.

TIP: You can click the <x> heading in the XML file to expand or collapse the nested parameters in this section.

Parameter	Details
WAN_Stack_Mode	**Description**—IP stack mode **User Interface**—**Network Setup** > **Basic Settings** page, **Stack Mode** field **Values** • 0: IPv4 Only • 1: IPv6 Only • 2: Dual Stack **Default**—0
WAN_Signal_Preference	**Description**—Preference IP mode for SIP Signaling. **User Interface**—**Network Setup** > **Basic Settings** page, **Signaling Preference** field. **Values** • 0: Prefer IPv4 • 1: Prefer IPv6 **Default**—0
WAN_Media_Preference	**Description**—Preference IP mode for RTP stream. **User Interface**—**Network Setup** > **Basic Settings** page, **Media Preference** field. **Values** • 0: Prefer IPv4 • 1: Prefer IPv6 **Default**—0

WAN_Interface Parameters

This section describes the parameters in the <WAN_Interface> section of the config.xml file.

TIP: You can click the <WAN_Interface> heading in the XML file to expand or collapse the nested parameters in this section.

Parameter	Details
<WAN_Connection_Type>	**Description**—Defines the connection/addressing mode used for the INTERNET (WAN) port. **User Interface**—**Network Setup** > **Basic Setup** > **IPv4 Settings** page, **Connection Type** field **Values** • dh: DHCP • st: Static • pp: PPPoE **Default**—dh **Example**—Static connection type <WAN_Connection_Type>st</WAN_Connection_Type>
<WAN_DHCP_MTU_Mode> <WAN_Static_MTU_Mode> <WAN_PPPoE_MTU_Mode>	**Description**—MTU mode. Use the parameter corresponding to the configured connection type. **User Interface**—**Network Setup** > **Basic Setup** > **IPv4 Settings** page, **MTU** drop-down list **Values** • 0: Auto • 1: Manual **Default**—0 **Example**—Manual MTU mode for a static connection <WAN_Static_MTU_Mode>1</WAN_Static_MTU_Mode>
<WAN_DHCP_MTU_Size> <WAN_Static_MTU_Size> <WAN_PPPoE_MTU_Size>	**Description**—MTU size. Use the parameter corresponding to the configured connection type. **User Interface**—**Network Setup** > **Basic Setup** > **IPv4 Settings** page, **MTU** text box **Values**—576 to1492 **Default**—0 **Example**—Customized MTU size for PPPoE <WAN_PPPoE_MTU_Size>1492</WAN_PPPoE_MTU_Size>

Parameter	Details
<WAN_Static_IP_NET>	**Description**—Specifies the IPv4 address for the Static IP connection. **User Interface**—**Network Setup** > **Basic Setup** > **IPv4 Settings** page, **Internet IPv4**, **Subnet Mark**,and **Default Gateway** text boxes (available when Static IP is the Connection Type) **Parameters**—Internet_IP:Subnet_Mask:Default_Gateway [:DNS1[:DNS2[:DNS3]]] **Values** • Internet_IP: IPv4 address • Subnet_Mask: IPv4 mask address • Default_Gateway: IPv4 address • DNS_1: IPv4 address • DNS_2: IPv4 address • DNS_3: IPv4 address **Default**—0.0.0.0:0.0.0.0:0.0.0.0:0.0.0.0:0.0.0.0:0.0.0.0 **Example** <WAN_Static_IP_NET>10.1.1.1:255.255.255.0:10.1.1.254:10.1.1.2:10.1.1.3</WAN_Static_IP_NET>
<WAN_PPPoE_User_Name>	**Description**—Username for PPTP session through the INTERNET (WAN) port. **User Interface**—**Network Setup** > **Basic Setup** > **IPv4 Settings** page, **User Name** field (available when PPPoE is the Connection Type) **Values**—(up to 64 characters), Printable ASCII characters **Default**—null **Example** <WAN_PPPoE_User_Name>test@example.net</WAN_PPPoE_User_Name>
<WAN_PPPoE_Password>	**Description**—Configures the interface settings for defined VLAN sub interfaces. VLAN ID n must be previously defined in the VLAN_ID_Index tag. This tag defines the password for PPPoE session configured over the sub interface. Note: the value of this field is hidden when reading the config.xml file from the device. **User Interface**—**Network Setup** > **Basic Setup** > **IPv4 Settings** page, **Password** field (available when PPPoE is the Connection Type) **Values**—password (up to 64 characters) **Default**—commented out, <!-- <WAN_PPPoE_Password></WAN_PPPoE_Password>--> **Example** <WAN_PPPoE_Password>my-password</WAN_PPPoE_Password>

Parameter	Details
<WAN_PPPoE_Service_Name>	**Description**—Descriptive service name (provided by the ISP), for a PPPoE session. **User Interface**—**Network Setup** > **Basic Setup** > **IPv4 Settings** page, **Service Name** field (available when PPPoE is the Connection Type) **Parameter**—Service name **Values**—name (up to 64 characters) **Default**—null **Example** <WAN_PPPoE_Service_Name>ServiceX_PPP</WAN_PPPoE_Service_Name>
<WAN_PPPoE_Keep_Alive>	**User Interface**—**Network Setup** > **Basic Setup** > **IPv4 Settings** page, **Keep Alive** field, **Connect on Demand**, and **Max Idle** fields (available when PPPoE is the Connection Type) **Description**—Keep Alive or Connect on Demand settings for a PPPoE session configured. **Parameter**—Type:Max_Idle_Time:30 **Values** • Type: • 0 (Keep Alive) • 1 (Connect on Demand) • Max_Idle_Time (Minutes)=1...9999 (for Connect on Demand) • 30 is a static value **Default**—0:0:30 **Example** <WAN_PPPoE_Keep_Alive>1:5:30</WAN_PPPoE_Keep_Alive>

WAN Example 1: DHCP with automatic MTU mode

```
<router-configuration>
<WAN_Interface>
<WAN_Connection_Type>dh</WAN_Connection_Type>
<WAN_DHCP_MTU_Mode>0</WAN_DHCP_MTU_Mode>
<WAN_DHCP_MTU_Size>0</WAN_DHCP_MTU_Size>
<WAN_Static_IP_NET>0.0.0.0:0.0.0.0:0.0.0.0</WAN_Static_IP_NET>
<WAN_Static_MTU_Mode>0</WAN_Static_MTU_Mode>
<WAN_Static_MTU_Size>0</WAN_Static_MTU_Size>
<WAN_PPPoE_User_Name />
<WAN_PPPoE_Service_Name />
<WAN_PPPoE_Password />
<WAN_PPPoE_Keep_Alive>0:0:30</WAN_PPPoE_Keep_Alive>
<WAN_PPPoE_MTU_Mode>0</WAN_PPPoE_MTU_Mode>
<WAN_PPPoE_MTU_Size>0</WAN_PPPoE_MTU_Size>
</WAN_Interface>
```

```
...
</router-configuration>
```

WAN Example 2: Static IP with manual MTU mode

```
<router-configuration>
...
<WAN_Interface>
<WAN_Connection_Type>st</WAN_Connection_Type>
<WAN_DHCP_MTU_Mode>0</WAN_DHCP_MTU_Mode>
<WAN_DHCP_MTU_Size>0</WAN_DHCP_MTU_Size>
<WAN_Static_IP_NET>10.1.1.1:255.255.255.0:10.1.1.254:10.1.1.2:10.1.1.3</
WAN_Static_IP_NET>
<WAN_Static_MTU_Mode>1</WAN_Static_MTU_Mode>
<WAN_Static_MTU_Size>1492</WAN_Static_MTU_Size>
</WAN_Interface>
...
</router-configuration>
```

WAN Example 3: PPPoE with Connect on Demand

```
<router-configuration>
...
<WAN_Interface>
<WAN_Connection_Type>pppoe</WAN_Connection_Type>
<WAN_DHCP_MTU_Mode>0</WAN_DHCP_MTU_Mode>
<WAN_DHCP_MTU_Size>0</WAN_DHCP_MTU_Size>
<WAN_Static_IP_NET>0.0.0.0:0.0.0.0:0.0.0.0</WAN_Static_IP_NET>
<WAN_Static_MTU_Mode>0</WAN_Static_MTU_Mode>
<WAN_Static_MTU_Size>0</WAN_Static_MTU_Size>
<WAN_PPPoE_User_Name>test@example.net</WAN_PPPoE_User_Name>
<WAN_PPPoE_Password>my-password</WAN_PPPoE_Password>
<WAN_PPPoE_Service_Name>ServiceX_PPP</WAN_PPPoE_Service_Name>
<WAN_PPPoE_Keep_Alive>1:5:30</WAN_PPPoE_Keep_Alive>
<WAN_PPPoE_MTU_Mode>0</WAN_PPPoE_MTU_Mode>
<WAN_PPPoE_MTU_Size>0</WAN_PPPoE_MTU_Size>
</WAN_Interface>
...
</router-configuration>
```

WAN_IP6_Setting Parameters

This section describes the parameters in the <WAN_IP6_Setting> section of the config.xml file.

TIP: You can click the <WAN_IP6_Setting> heading in the XML file to expand or collapse the nested parameters in this section.

Parameter	Details
WAN_IP6_Allow_AutoConfig	**Description**—Set enabled to allow stateless IPv6 adddress generation on receiving RA. **User Interface**—**Network Setup** > **IPv6 Settings** page, **Allow Auto Configuration** field. **Values** • 0: Disabled • 1: Enabled **Default**—1
WAN_IP6_Connection_Type	**Description**—IPv6 connection type **User Interface**—**Network Setup** > **IPv6 Settings** page, **Connection Type** field. **Values** • 0: DHCPv6 • 1: Static • 2: PPPoE **Default**—0
WAN_Static_IP6_Address	**Description**—Manually configured IP v6 address. **User Interface**—**Network Setup** > **IPv6 Settings** page, **Internet IPv6 Address** field. **Values**—address (up to 64 characters) **Default**—null
WAN_Static_IP6_Prefix_Length	**Description**—Manually configured IP v6 prefix length. **User Interface**—**Network Setup** > **IPv6 Settings** page, **Prefix Length** field. **Values**—0 to 64 **Default**—64
WAN_Static_IP6_Gateway	**Description**—Manually configured IPv6 router address **User Interface**—**Network Setup** > **IPv6 Settings** page, **Default Gateway** field. **Values**—0-64 **Default**—null

PHY_Port_Setting Parameters

This section describes the parameters in the <PHY_Port_Setting> section of the config.xml file.

TIP: You can click the <PHY_Port_Setting> heading in the XML file to expand or collapse the nested parameters in this section.

Parameter	Details
<Flow_Control>	**Description**—Enables or disables flow control **User Interface**—**Interface Setup** > **Advanced Settings** > **Port Setting** page, **Flow Control** field **Values** • 0: Disabled • 1: Enabled **Default**—1 **Example**—Flow control enabled <Flow_Control>1</Flow_Control>
<Speed_Duplex>	**Description**—The port speed and duplex mode **User Interface**—**Interface Setup** > **Advanced Settings** > **Port Setting** page, **Speed Duplex** field **Values** • auto • 10h • 10f • 100h • 100f **Default**—auto **Example**—100 Mbps, half-duplex mode <Speed_Duplex>100h</Speed_Duplex>

<PHY_Port_Setting> Example: Flow control enabled with auto-negotiated duplex mode

```
<router-configuration>
...
<PHY_Port_Setting>
<Flow_Control>1</Flow_Control>
<Speed_Duplex>auto</Speed_Duplex>
</PHY_Port_Setting>
...
</router-configuration>
```

MAC_Address_Clone Parameters

This section describes the parameters in the <MAC_Address_Clone> section of the config.xml file.

TIP: You can click the <MAC_Address_Clone> heading in the XML file to expand or collapse the nested parameters in this section.

Parameter	Details
<MAC_Address_Clone_Enabled>	**Description**—Enables or disables MAC address cloning. **User Interface**—**Interface Setup** > **Advanced Settings** > **MAC Address Clone** page, **MAC Clone** field **Values** • 0: Disabled • 1: Enabled **Default**—0 **Example**—MAC clone enabled <MAC_Address_Clone_Enabled>1</MAC_Address_Clone_Enabled>
<MAC_Address_Clone_Address>	**Description**—MAC address to assign (clone) to this ATA **User Interface**—**Interface Setup** > **Advanced Settings** > **MAC Address Clone** page, **MAC Address** field (available when **MAC Clone** is enabled) **Values**—MAC address **Default**—null **Example** <MAC_Address_Clone_Address>00:22:68:19:EF:83</MAC_Address_Clone_Address>

<MAC_Address_Clone> Example: MAC Address Clone enabled

```
<router-configuration>
...
<MAC_Address_Clone>
<MAC_Address_Clone_Enabled>1</MAC_Address_Clone_Enabled>
<MAC_Address_Clone_Address>00:22:68:19:EF:83</MAC_Address_Clone_Address>
</MAC_Address_Clone>
...
</router-configuration>
```

Internet_Option Parameters

This section describes the parameters in the <Internet_Option> section of the config.xml file.

TIP: You can click the <Internet_Option> heading in the XML file to expand or collapse the nested parameters in this section.

Parameter	Details
<Host_Name>	**Description**—The name of the ATA **User Interface**—**Network Setup** > **Basic Setup** > **Basic Settings** page, **Host Name** field **Values**—name **Default**—model number **Example** <Host_Name>ATA-192-MPP</Host_Name>
<Domain_Name>	**Description**—A domain name specified by the ISP, if applicable **User Interface**—**Network Setup** > **Basic Setup** > **Basic Settings** page, **Domain Name** field **Values**—name **Default**—null **Example** <Domain_Name>My ISP</Domain_Name>
<DNS_Order>	**Description**—Method for choosing a DNS server **User Interface**—**Network Setup** > **Basic Setup** > **IPv4 Settings** page, **DNS Server Order** field **Values** • 0:Manual • 1:Manual-DHCP • 2:DHCP-Manual **Default**—2 **Example**—Manual-DHCP order <DNS_Order>2</DNS_Order>
<DNS>	**Description**—For manual DNS server order, the IPv4 address of a DNS server; optionally, a secondary server can be specified **User Interface**—**Network Setup** > **Basic Setup** > **IPv4 Settings** page, **Primary DNS** and **Secondary DNS** fields **Values**—DNS1[:DNS2] **Default**—null **Example**—Primary and secondary DNS server <DNS>209.165.201.1:209.165.201.2</DNS>

Parameter	Details
DNS6_Order	**Description**—IPv6 DNS server order **User Interface**—**Network Setup** > **IPv6 Settings** page, **DNS Server Order** field **Values** • 0: only use manual DNS server • 1: manual DNS server first, then dhcpv6 DNS server • 2: dhcpv6 DNS serer first, then manual DNS server **Default**—2
DNS6	**Description**—manual configured IPv6 DNS server, optionally a secondary server can be specified **User Interface**—**Network Setup** > **IPv6 Settings** page, **Primary DNS** and **Secondary DNS** fields **Values**—DNS6_1[:DNS6_2] **Default**—null

<Internet_Option> Example

```
<router-configuration>
...
<Internet_Option>
<Host_Name>ATA192-MPP</Host_Name>
<Domain_Name>My ISP</Domain_Name>
<DNS_Order>2</DNS_Order>
<DNS>209.165.201.1:209.165.201.2</DNS>
</Internet_Option>
...
</router-configuration>
```

DHCP_Server_Pool Parameters

This section describes the parameters in the <DHCP_Server_Pool> section of the config.xml file.

Rule

All parameters in the <DHCP_Server> section of the XML file are nested between <Rule> and </Rule>.

Parameter	Details
<DHCP_Server>	**Description**—Enables or disables the DHCP server **User Interface**—**Network Setup** > **Basic Setup** > **IPv4 LAN Settings** page, **DHCP Server** field **Values** • 0: Disabled • 1: Enabled **Default**—1 **Example**—DHCP server enabled <DHCP_Server>1</DHCP_Server>
<Local_IP>	**Description**—The IPv4 address of the LAN interface **User Interface**—**Network Setup** > **Basic Setup** > **IPv4 LAN Settings** page, **Local IP address** field **Values**—IPv4 address **Default**—192.168.15.1 **Example:** <Local_IP>192.168.15.1</Local_IP>
<Subnet_Mask>	**Description**—The subnet mask for the local network **User Interface**—**Network Setup** > **Basic Setup** > **IPv4 LAN Settings** page, **Subnet Mask** field **Values**—Class C subnet mask • 255.255.255.0 • 255.255.255.128 • 255.255.255.192 • 255.255.255.224 • 255.255.255.240 • 255.255.255.248 • 255.255.255.252 **Default**—255.255.255.0 **Example:** <Subnet_Mask>255.255.255.0</Subnet_Mask>

Parameter	Details
<DHCP_Client_Table>	**Description**—Clients with reserved IPv4 addresses **User Interface**—**Network Setup** > **Basic Setup** > **IPv4 LAN Settings** page, **IP Reservation** list (available after clicking the **Show DHCP Reservation** button) **Values**—Semi-colon separated list of client information in the following order: <MAC address> <ip_address> on <client_name> **Default**—null **Example:** <DHCP_Client_Table>58:8D:09:72:73:DA 192.168.15.100 on Computer-1;00:22:68:19:EF:83 192.168.15.101 on Computer-2;</DHCP_Client_Table>
<Option_66>	**Description**—Method for specifying a TFTP server for remote configuration of the ATA **User Interface**—**Network Setup** > **Basic Setup** > **IPv4 LAN Settings** page, **Option 66** field **Values** • 0: None • 2: Remote TFTP Server • 3: Manual TFTP Server **Default**—0 **Example**—Remote TFTP server <Option_66>2</Option_66>
<TFTP_IP>	**Description**—IPv4 address of a TFTP server, if Option 66 is set to Manual **User Interface**—**Network Setup** > **Basic Setup** > **IPv4 LAN Settings** page, **TFTP Server** field **Values**—IPv4 address **Default**—0.0.0.0 **Example** <TFTP_IP>209.165.202.129</TFTP_IP>

Parameter	Details
<Option_67>	**Description**—Provides a configuration/bootstrap filename to hosts that request this option **User Interface**—**Network Setup** > **Basic Setup** > **IPv4 LAN Settings** page, **Option 67** field **Values**—filename **Default**—null **Example** <Option_67>MyDirectory/MyFile.cfg</Option_67>
<Option_159 >	**Description**—Provides a configuration URL to hosts that request this option **User Interface**—**Network Setup** > **Basic Setup** > **IPv4 LAN Settings** page, **Option 159** field **Values**—URL **Default**—null **Example** <Option_159>http://MyDomain.com/MyDirectory/MyFile.cfg></Option_159>
<Option_160 >	**Description**—Provides a configuration URL to hosts that request this option **User Interface**—**Network Setup** > **Basic Setup** > **IPv4 LAN Settings** page, **Option 160** field **Values**—filename **Default**—null **Example** <Option_67>MyDirectory/MyFile.cfg</Option_67>
<DNS_Proxy>	**Description**—Enables or disables the DNS proxy, which relays DNS requests to the current public network DNS server for the proxy, and replies as a DNS resolver to the client device on the network **User Interface**—**Network Setup** > **Basic Setup** > **IPv4 LAN Settings** page, **DNS Proxy** field **Values** • 0: Disabled • 1: Enabled **Default**—1 **Example**—DNS proxy enabled <DNS_Proxy>1</DNS_Proxy>

Parameter	Details
<Starting_IP>	**Description**—The first IPv4 address in the range of IPv4 addresses that are assigned by the DHCP server **User Interface**—**Network Setup** > **Basic Setup** > **IPv4 LAN Settings** page, **Starting IP address** field **Values**—IPv4 address **Default**—192.168.15.100 **Example** <Starting_IP>192.168.15.110</Starting_IP>
<Max_DHCP_User>	**Description**—The maximum number of devices that can receive DHCP addresses from the DHCP server **User Interface**—**Network Setup** > **Basic Setup** > **IPv4 LAN Settings** page, **Maximum DHCP Users** field **Values**—number **Default**—50 **Example**—10-device maximum <Max_DHCP_User>10</Max_DHCP_User>
<Client_Lease_Time>	**Description**—The number of minutes that a dynamically assigned IPv4 address can be in use, or “leased” **User Interface**—**Network Setup** > **Basic Setup** > **IPv4 LAN Settings** page, **Client Lease Time** field **Values**—number. Enter the number of minutes. Enter 0 to represent 1 day. Enter 9999 to never expire. **Default**—0 (1 day) **Example**—No expiration <Client_Lease_Time>9999</Client_Lease_Time>
<Static_DNS>	**Description**—Defines a DNS server address that will be provided to DHCP clients. If DNS Proxy is enabled, clients will automatically be issued the Local IPv4 address to use for DNS. **User Interface**—**Network Setup** > **Basic Setup** > **IPv4 LAN Settings** page, **Static DNS** field **Values**—IPv4 address **Default**—0.0.0.0 **Example** <Static_DNS>209.165.202.129</Static_DNS>

Parameter	Details
<Default_Gateway>	**Description**—Enter the IPv4 address of the default gateway to be used by the DHCP clients. **User Interface**—**Network Setup** > **Basic Setup** > **IPv4 LAN Settings** page, **Default Gateway** field **Default**—192.168.15.1 **Example** <Default_Gateway>192.168.15.1</Default_Gateway>

<DHCP_Server_Pool> Example: DHCP enabled with two DHCP reservations <router-configuration>

```
...
<DHCP_Server_Pool>
<Rule>
<DHCP_Server>1</DHCP_Server>
<Local_IP>192.168.15.1</Local_IP>
<Subnet_Mask>255.255.255.0</Subnet_Mask>
<DHCP_Client_Table>58:8D:09:72:73:DA 192.168.15.100 on Computer-1;00:22:68:19:EF:83
192.168.15.101 on Computer-2;</DHCP_Client_Table>
<TFTP_IP>0.0.0.0</TFTP_IP>
<Starting_IP>192.168.15.100</Starting_IP>
<Max_DHCP_User>50</Max_DHCP_User>
<Client_Lease_Time>0</Client_Lease_Time>
<Default_Gateway>192.168.15.1</Default_Gateway>
</Rule>
</DHCP_Server_Pool>
...
</router-configuration>
```

LAN_IP6_Setting Parameters

This section describes the parameters in the <LAN_IP6_Setting> section of the config.xml file.

TIP: You can click the <LAN_IP6_Setting> heading in the XML file to expand or collapse the nested parameters in this section.

Parameter	Details
LAN_IP6_Address_Assign_Type	**Description**—Method for IPv6 assignment to LAN device.. **User Interface**—**Network Setup** > **IPv6 LAN Settings** page, **Address Assign Type** field **Values** • 0: SLACC • 1: DHCP6s **Default**—0

Parameter	Details
LAN_DHCP6_Delegation_Enable	**Description**—Set enabled to support DHCPv6 delegation which support to obtain LAN prefix via DHCPv6 client **User Interface**—**Network Setup** > **IPv6 LAN Settings** page, **DHCPv6 Delegation** field. **Values** • 0: Disabled • 1: Enabled **Default**—0
LAN_IP6_Prefix	**Description**—Manual LAN prefix, editable only when DHCP delegation is disabled. **User Interface**—**Network Setup** > **IPv6 LAN Settings** page, **IPv6 Address Prefix** field. **Values**—0-64 **Default**—null

WAN_VLAN_Setting Parameters

This section describes the parameters in the <WAN_VLAN_Setting> section of the config.xml file.

Parameter	Details
<WAN_VLAN_Enable>	**Description**—Enables or disables a VLAN on your network **User Interface**—**Network Setup** > **Advanced Settings** > **VLAN** page, **Enable VLAN** field **Valid inputs** • 0: Disabled • 1: Enabled **Default**—0 **Example**—VLAN enabled <WAN_VLAN_Enable>1</WAN_VLAN_Enable>

Parameter	Details
<WAN_VLAN_ID>	**Description**—A number that identifies the VLAN **User Interface**—**Network Setup** > **Advanced Settings** > **VLAN** page, **VLAN ID** field **Valid inputs**—1~4094 **Default**—1 **Example**—VLAN ID 100 <WAN_VALN_ID>100</WAN_VALN_ID>

<WAN_VLAN_Setting> Example: VLAN Enabled with ID 10

```
<router-configuration>
...
<WAN_VLAN_Setting>
<WAN_VLAN_Enable>1</WAN_VLAN_Enable>
<WAN_VALN_ID>100</WAN_VALN_ID>
</WAN_VLAN_Setting>
...
</router-configuration>
```

CLDP_Setting Parameters

This section describes the parameters in the <CLDP_Setting> section of the config.xml file.

Parameter	Details
<CDP_ENABLE>	**Description**—Enables or disables Cisco Discovery Protocol (CDP) **User Interface**—**Network Setup** > **Advanced Settings** > **CDP & LLDP** page, **Enable CDP** field **Valid inputs** • 0 • 1 0 means that the CDP is disabled. 1 means that the CDP is enabled. **Default**—1 **Example**—CDP enabled <CDP_ENABLE>1</CDP_ENABLE>

Parameter	Details
<LLDP_ENABLE>	**Description**—Enables or disables Link Layer Discovery Protocol (LLDP) **User Interface**—**Network Setup** > **Advanced Settings** > **CDP & LLDP** page, **Enable LLDP-MED** field **Valid inputs** • 0 • 1 `0` means that the LLDP is disabled. `1` means that the LLDP is enabled. **Default**—1 **Example**—LLDP enabled <LLDP_ENABLE>1</LLDP_ENABLE>
<LAYER2_LOGGING_ENABLE>	**Description**—Enables Layer 2 logging, which is used by CDP and LLDP for debugging purposes **User Interface**—**Network Setup** > **Advanced Settings** > **CDP & LLDP** page, **Layer 2 Logging** field **Valid inputs** • 0: Disabled • 1: Enabled **Default**—0 **Example**—Layer 2 logging enabled <LAYER2_LOGGING_ENABLE>1</LAYER2_LOGGING_ENABLE>

<CLDP_Setting> Example: CDP, LLDP, and Layer 2 logging enabled

```
<router-configuration>
...
<CLDP_Setting>
<CDP_ENABLE>1</CDP_ENABLE>
<LLDP_ENABLE>1</LLDP_ENABLE>
<LAYER2_LOGGING_ENABLE>1</LAYER2_LOGGING_ENABLE>
</CLDP_Setting>
...
</router-configuration>
```

Single_Port_Forwarding Parameters

This section describes the parameters in the <Single_Port_Forwarding> section of the config.xml file.

TIP: You can click the <Single_Port_Forwarding> heading in the XML file to expand or collapse the nested parameters in this section.

Parameter	Details

Single_Port_Forwarding_Index	**Description**—Index for single port forwarding. Should be listed in order with colon depending on the amount of entry added (Rule<index>). Index in order 0-9
	User Interface—**Network Setup** > **Application** > **Port Forwarding** > **Add Entry** > **Port Forwarding Tupe** page, **Single Port Forwarding** field
	Values:
	- 0: Disabled
	- 1: Enabled
	Default—null
	Example
	`<Single_Port_Forwarding>`
	`<Single_Port_Forwarding_Index>0:1:2</Single_Port_Forwarding_Index>`
	`<Rule0>1:SNMP:br1:161:161:udp:192.168.15.30</Rule0>`
	`<Rule1>0:Finger:br1:79:79:tcp:192.168.15.30</Rule1>`
	`<Rule2>1:forward_rule:br1:25:27:both:192.168.15.15</Rule2>`
	`</Single_Port_Forwarding>`

Rule<index>	**Description**—Forwards traffic for a specified port to the same or an alternative port on the target server in the LAN. <index> can be 0-9
	User Interface—**Network Setup** > **Application** > **Port Forwarding** > **Add Entry** > **Port Forwarding Tupe** page, **Single Port Forwarding** field.
	Format: <Enabled>:<Name>:<Interface>:<External Port>:<Internal Port>:<Protocol>:<Target server IP>
	Values
	- <Enabled>: 0-1
	- <Name>: String
	- <Interface>: br1
	- <External Port>: 1-65535
	- <Internal Port>: 1-65535
	- <Protocol>: tcp,udp,both
	- <Target server IP>: ipv4 address
	Default—null
	Example
	`<Rule0>1:SNMP:br1:161:161:udp:192.168.15.30</Rule0>`
	`<Rule1>0:Finger:br1:79:79:tcp:192.168.15.30</Rule1>`
	`<Rule2>1:forward_rule:br1:25:27:both:192.168.15.15</Rule2>`

Port_Range_Forwarding Parameters

This section describes the parameters in the <Port_Range_Forwarding> section of the config.xml file.

TIP: You can click the <Port_Range_Forwarding> heading in the XML file to expand or collapse the nested parameters in this section.

Parameter	Details

Port_Range_Forwarding_Index	**Description**—Index for port range forwarding. Should be listed in order with colon depending on the amount of entry added (Rule<index>).
	User Interface—**Network Setup** > **Application** > **Port Forwarding** > **Add Entry** > **Port Forwarding Type** page, **Port Range Forwarding** field
	Values: index in order: 0-9
	Default—Null
	Example
	`<Port_Range_Forwarding>`
	`<Port_Range_Forwarding_Index>0:1</Port_Range_Forwarding_Index>`
	`<Rule0>1:Rule_0:br1:50:60:tcp:192.198.15.22</Rule0>`
	`<Rule1>0:Rule_1:br1:11:13:both:192.168.15.12</Rule1>`
	`</Port_Range_Forwarding>`

Rule<index>	**Description**—Forwards traffic to a range of ports to the same ports on the target server in the LAN. <index> can be 0-9.
	User Interface—**Network Setup** > **Application** > **Port Forwarding** > **Add Entry** > **Port Forwarding Type** page, **Port Range Forwarding** field.
	Format: <Enabled>:<Name>:<Interface>:<Start Port>:<End Port>:<Protocol>:<Target server IP>
	Values
	• <Enabled>: 0-1
	• <Name>: String
	• <Interface>: br1
	• <Start Port>: 1-65535
	• <End Port>: 1-65535
	• <Protocol>: tcp,udp,both
	• <Target server IP>: ipv4 address
	Default—null
	Example
	`<Rule0>1:Rule_0:br1:50:60:tcp:192.198.15.22</Rule0>`
	`<Rule1>0:Rule_1:br1:11:13:both:192.168.15.12</Rule1>`

SNMP Parameters

This section describes the parameters in the <SNMP> section of the config.xml file.

Parameter	Details
<SNMP_Enabled>	**Description**—Enables or disables SNMP **User Interface**—**Administration** > **Management** > **SNMP** page, **SNMP** section, **Enabled** and **Disabled** options **Valid inputs** • 0: Disabled • 1: Enabled **Default**—0 **Example**—SNMP enabled <SNMP_Enabled>1</SNMP_Enabled>
<SNMP_Trusted_IP>	**Description**—trusted v4 IP address that can access the ATA through SNMP **User Interface**—**Administration** > **Management** > **SNMP** page, **SNMP** section, **Trusted IP** field **Valid inputs**—IPv4 address and subnet mask in this order: 0.0.0.0/0.0.0.0 **Default**—0.0.0.0/0.0.0.0 (Any IP address) **Example** <SNMP_Trusted_IP>209.165.202.129/255.255.255.0</SNMP_Trusted_IP>
SNMP_Trusted_IP6	**Description**—trusted v4 IP address that can access the ATA through SNMP **User Interface**—**Administration** > **SNMP** page, **Trusted IPv6** field **Valid inputs**—IPv6 address **Default**—::
SNMP_Trusted_IP6_Prefix_Length	**Description**—prefix of the trusted v6 IP that can access the ATA through SNMP **User Interface**—**Administration** > **SNMP** page, **Trusted IPv6** field **Valid inputs**—0-128 **Default**—0

Parameter	Details
<Get_Community>	**Description**—A community string for authentication for SNMP GET commands. **User Interface**——**Administration** > **Management** > **SNMP** page, **SNMP** section, **Get/Trap Community** field **Valid inputs**—string **Default**—public **Example** <Get_Community>MyGet</Get_Community>
<Set_Community>	**Description**—A community string for authentication for SNMP GET commands. **User Interface**—**Administration** > **Management** > **SNMP** page, **SNMP** section, **Set Community** field **Valid inputs**—string **Default**—private **Example** <Set_Community>MySet</Set_Community>
<SNMPV3>	**User Interface**—**Administration** > **Management** > **SNMP** page, **SNMPV3** section, **Enable** and **Disable** fields **Valid inputs** • 0: Disabled • 1: Enabled **Default**—0 **Example**—SNMPv3 enabled <SNMPV3>1</SNMPV3>
<RW_User>	**Description**—A username for SNMP authentication **User Interface**—**Administration** > **Management** > **SNMP** page, **SNMPV3** section, **R/W User** field **Valid inputs**—username **Default**—v3rwuser **Example** <RW_User>MyUsername</RW_User>

Parameter	Details
<Auth_Protocol>	**Description**—SNMPv3 authentication protocol **User Interface**—**Administration** > **Management** > **SNMP** page, **SNMPV3** section, Auth-Protocol field **Valid inputs** • MD5 • SHA **Default**—MD5 Example—SHA enabled <Auth_Protocol>SHA</Auth_Protocol>
<Auth_Password>	**Description**—Password for SNMPv3 authentication **User Interface**—**Administration** > **Management** > **SNMP** page, **Auth-Password** field for **SNMPv3** **Valid inputs**—string **Default**—1111111111 **Example** <Auth_Password>MyPassword</Auth_Password>
<Privacy_Protocol>	**Description**—Privacy authentication protocol for SNMPv3 **User Interface**—**Administration** > **Management** > **SNMP** page, **SNMPV3** section, **privprotocol** field **Valid inputs** • None • DES **Default**—DES **Example**—DES enabled <Privacy_Protocol>DES</Privacy_Protocol>
<Privacy_Password>	**Description**—Privacy authentication password for SNMPv3 **User Interface**—**Administration** > **Management** > **SNMP** page, **SNMPV3** section, **Privacy Password** field **Valid inputs**—string **Default**—1111111111 **Example** <Privacy_Password>MyPrivacyPassword</Privacy_Password>

Parameter	Details
<TRAP_IP_Address>	**Description**—The IP Address of the SNMP manager or trap agent **User Interface**—**Administration** > **Management** > **SNMP** page, **Trap Configuration** section, **IP Address** field **Valid inputs**—IPv4 address **Default**—192.168.15.100 **Example** <TRAP_IP_Address>209.165.202.129</TRAP_IP_Address>
<TRAP_Port>	**Description**—The SNMP trap port used by the SNMP manager or trap agent to receive the trap messages **User Interface**—**Administration** > **Management** > **SNMP** page, **Trap Configuration** section, **Port** field **Valid inputs**—162 or 1025~65535 **Default**—162 **Example** <TRAP_Port>162</TRAP_Port>
<TRAP_SNMP_Version>	**Description**—The SNMP version in use by the SNMP manager or trap agent **User Interface**—**Administration** > **Management** > **SNMP** page, **Trap Configuration** section, **SNMP Version** field **Valid inputs**—One of the SNMP version number listed below • v1 • v2c • v3 **Default**—v1 **Example** <TRAP_SNMP_Version>v3</TRAP_SNMP_Version>

<SNMP> Example 1: SNMP Enabled from Any IP Address

```
<router-configuration>
...
<SNMP>
<SNMP_Enabled>1</SNMP_Enabled>
<SNMP_Trusted_IP>0.0.0.0/0.0.0.0</SNMP_Trusted_IP>
<Get_Community>MyGet</Get_Community>
<Set_Community>MySet</Set_Community>
<TRAP_IP_Address>209.165.202.129</TRAP_IP_Address>
<TRAP_Port>162</TRAP_Port>
<TRAP_SNMP_Version>v3</TRAP_SNMP_Version>
</SNMP>
```

```
...
</router-configuration>
```

<SNMP> Example 2: SNMPv3 Enabled from Trusted IP Address

```
<router-configuration>
...
<SNMP>
<SNMP_Enabled>1</SNMP_Enabled>
<SNMP_Trusted_IP>209.165.202.129/255.255.255.0</SNMP_Trusted_IP>
<Get_Community>MyGet</Get_Community>
<Set_Community>MySet</Set_Community>
<SNMPV3>1</SNMPV3>
<RW_User>MyUsername</RW_User>
<Auth_Protocol>SHA</Auth_Protocol>
<Auth_Password>MyPassword</Auth_Password>
<Privacy_Protocol>DES</Privacy_Protocol>
<Privacy_Password>MyPrivacyPassword</Privacy_Password>
<TRAP_IP_Address>209.165.201.1</TRAP_IP_Address>
<TRAP_Port>162</TRAP_Port>
<TRAP_SNMP_Version>v3</TRAP_SNMP_Version>
</SNMP>
...
<router-configuration>
```

Time_Setup Parameters

Parameter	Details
<Time_Zone>	**Description**—The time zone for the site where the ATA is in operation **User Interface**—**Network Setup** > **Basic Setup** > **Time Settings** page, **Time Zone** field **Valid inputs**—number identifying the time zone. See Time Zone Settings, on page 187 **Default**—08 1 1 **Example**—Germany <Time_Zone>+01 2 2</Time_Zone>
<Auto_Adjust_Clock>	**Description**—Enables or disables automatic time adjustments for daylight savings time **User Interface**—**Network Setup** > **Basic Setup** > **Time Settings** page, **Adjust Clock for Daylight Saving Changes** field **Valid inputs** • 0: Disabled • 1: Enabled **Default**—1 **Example**—Automatic Daylight Saving adjustment enabled <Auto_Adjust_Clock>1</Auto_Adjust_Clock>

Parameter	Details
<Time_Server_Mode>	**Description**—The method for specifying an NTP time server Time Server Address **User Interface**—**Network Setup** > **Basic Setup** > **Time Settings** page, **Time Server** field **Valid inputs** • manual • auto **Default**—auto **Example**—Manual mode <Time_Server_Mode>manual</Time_Server_Mode>
<Time_Server>	**Description**—IPv4 address or domain name of an NTP server User Interface—**Network Setup** > **Basic Setup** > **Time Settings** page, **Time Server Address** field **Valid inputs**—IPv4 address or domain name **Default**—0.ciscosb.pool.ntp.org **Example**—European pool <Time_Server>server 0.europe.pool.ntp.org </Time_Server>
<Resync_Timer>	**Description**—The interval, in seconds, at which the ATA resynchronizes with the NTP server **User Interface**—**Network Setup** > **Basic Setup** > **Time Settings** page, Resync Timer field **Valid inputs**—number **Default**—3600 **Example** <Resync_Timer>3600</Resync_Timer>

Parameter	Details
<Auto_Recovery_System_Time>	**Description**—When enabled, allows the ATA to automatically reconnect to the time server after a system reboot **User Interface**—**Network Setup** > **Basic Setup** > **Time Settings** page, **Auto Recovery After System Reboot** field **Valid inputs** • 0: Disabled • 1: Enabled **Default**—0 **Example**—Auto Recovery enabled <Auto_Recovery_System_Time>1</Auto_Recovery_System_Time>
<Time_Mode>	**Description**—The method of specifying a time server **User Interface**—**Network Setup** > **Basic Setup** > **Time Settings** page, **Time Server** field **Valid inputs** • 0: Manual • 1: Auto **Default**—1 **Example**—Automatic mode <Time_Mode>1</Time_Mode>

<Time_Setup> Example: Germany Time Zone with Daylight Savings and Auto-Recovery Enabled

```
<router-configuration>
...
<Time_Setup>
<Time_Zone>+01 2 2</Time_Zone>
<Auto_Adjust_Clock>1</Auto_Adjust_Clock>
<Time_Server_Mode>auto</Time_Server_Mode>
<Time_Server>0.ciscosb.pool.ntp.org</Time_Server>
<Resync_Timer>3600</Resync_Timer>
<Auto_Recovery_System_Time>1</Auto_Recovery_System_Time>
<Time_Mode>1</Time_Mode>
</Time_Setup>
...
<router-configuration>
```

QoS_Bandwidth_Control Parameters

This section describes the parameters in the <QoS_Bandwidth_Control> section of the config.xml file.

WAN

All parameters in the <Qos_Bandwidth_Control> section are nested between <WAN> and </WAN>.

Parameter	Details
<QoS_Always_ON>	**Description**—Determines whether QoS settings are enabled at all times or only when there is voice traffic **User Interface**—**Network Setup** > **Application** > **QoS** page, **QoS Policy** field **Valid inputs** • 0: On When Phone In Use • 1: Always On **Default**—0 **Example**—On when phone is in use <QoS_Always_ON>0</QoS_Always_ON>
<Upstream_Bandwidth>	**Description**—The maximum available upstream bandwidth, in kbps, as specified by the Internet Service Provider **User Interface**—**Network Setup** > **Application** > **QoS** page, **Upstream Bandwidth** field **Valid inputs**—number **Default**—10000 **Example** <Upstream_Bandwidth>20000</Upstream_Bandwidth>

<QoS_Bandwidth_Control> Example: QoS always on, maximum bandwidth of 20,000 kbps

```
<router-configuration>
...
<QoS_Bandwidth_Control>
<WAN>
<QoS_Always_ON>1</QoS_Always_ON>
<Upstream_Bandwidth>20000</Upstream_Bandwidth>
</WAN>
</QoS_Bandwidth_Control>
...
</router-configuration>
```

HTTP_Proxy Parameters

This section describes the parameters in the <HTTP_Proxy> section of the config.xml file.

Parameter	Details
<Proxy_Mode>	**Description**—Specifies the HTTP proxy mode that the ATA uses, or disables the HTTP proxy feature. **User Interface**—**Administration** > **Network Setup** > **Applications** > **HTTP Proxy** page, **Proxy Mode** field. **Valid inputs** • Off • Auto • Manual **Default**—Off **Example**—Auto proxy mode <Proxy_Mode>Auto</Proxy_Mode>
<Use_Auto_Discovery__WPAD_>	**Description**—Determines whether the ATA uses the Web Proxy Auto-Discovery (WPAD) protocol to retrieve a PAC file. If the parameter is set to No, you must configure the parameter <PAC_URL>. The parameter configuration takes effect only when the <Proxy_Mode> is set to Auto. **User Interface**—**Administration** > **Network Setup** > **Applications** > **HTTP Proxy** page, **Use Auto Discovery** field. **Valid inputs** • No • Yes **Default**—Yes **Example**—The WPAD protocol is not used. <Use_Auto_Discovery__WPAD_>No</Use_Auto_Discovery__WPAD_>
<PAC_URL>	**Description**—The URL of a Proxy Auto-Configuration (PAC) file. This parameter configuration takes effect when the <Proxy_Mode> is set to Auto and <Use_Auto_Discovery__WPAD_> is set to No. **User Interface**—**Administration** > **Network Setup** > **Applications** > **HTTP Proxy** page, **PAC URL** field. **Valid inputs—**URL **Default**—null **Example** http://proxy.department.branch.example.com

Parameter	Details
<Proxy_Server_Requires_Authentication>	**Description**—Select the option according to the actual behaviour of the proxy server. If the proxy server requires the user to provide authentication credentials, set it to Yes. Otherwise, select it to No. If the parameter is set to Yes, you must further configure the parameters <Proxy_Username> and <Proxy_Password>. **User Interface**—**Administration** > **Network Setup** > **Applications** > **HTTP Proxy** page, **Proxy Server Requires Authentication** field. **Valid inputs** • No • Yes **Default**—No **Example**—The proxy server requires the user authentication. <Proxy_Server_Requires_Authentication>1</Proxy_Server_Requires_Authentication>
<Proxy_Host>	**Description**—Specifies an IP address or hostname of the proxy host server that the ATA uses. The parameter configuration is required if the <Proxy_Mode> is set to Manual. **User Interface**—**Administration** > **Network Setup** > **Applications** > **HTTP Proxy** page, **Proxy Host** field. **Valid inputs**—A valid IP address or hostname of the proxy host server **Default**—null **Example** <Proxy_Host>proxy.example.com</Proxy_Host>
<Proxy_Port>	**Description**—Specifies a port number of the proxy host server that the ATA uses. The parameter configuration is required if the <Proxy_Mode> is set to Manual. **User Interface**—**Administration** > **Network Setup** > **Applications** > **HTTP Proxy** page, **Proxy Port** field. **Valid inputs**—A valid port number from 2 to 65535. **Default**—3128 **Example** <Proxy_Port>3128</Proxy_Port>

Parameter	Details
<Proxy_Username>	**Description**—Enter a username for the authentication purpose of the proxy server. The parameter configuration is required when <Proxy_Mode> is set to Manual and <Proxy_Server_Requires_Authentication> is set to Yes. **User Interface**—**Administration** > **Network Setup** > **Applications** > **HTTP Proxy** page, **Username** field. **Valid inputs**—string **Default**—null **Example** <Proxy_Username>Example</Proxy_Username>
<Proxy_Password>	**Description**—Enter the password of the specified username that the proxy server requires. The parameter configuration is required when <Proxy_Mode> is set to Manual and <Proxy_Server_Requires_Authentication> is set to Yes. **User Interface**—**Administration** > **Network Setup** > **Applications** > **HTTP Proxy** page, **Password** field. **Valid inputs**—string **Default**—null **Example** <Proxy_Password>Example</Proxy_Password>

<HTTP_Proxy> Example: Auto proxy mode with WPAD enabled

```
<router-configuration>
...
<HTTP_Proxy>
<Proxy_Mode>Auto</Proxy_Mode>
<Use_Auto_Discovery__WPAD_>Yes</Use_Auto_Discovery__WPAD_>
</HTTP_Proxy>
...
</router-configuration>
```

<HTTP_Proxy> Example: Manual proxy mode with proxy authentication required

```
<router-configuration>
...
<HTTP_Proxy>
<Proxy_Mode>Manual</Proxy_Mode>
<Proxy_Host>proxy.example.com</Proxy_Host>
<Proxy_Host>3128</Proxy_Host>
<Proxy_Server_Requires_Authentication>Yes</Proxy_Server_Requires_Authentication>
<Proxy_Username>Username_Example</Proxy_Username>
<Proxy_Password>Password_Example</Proxy_Password>
</HTTP_Proxy>
...
</router-configuration>
```

Software_DMZ Parameters

This section describes the parameters in the <Software_DMZ> section of the config.xml file.

Rule1

All parameters in the <Software_DMZ> section are nested between <Rule1> and </Rule1>. Only one DMZ rule is allowed on this device.

Parameter	Details
<Status>	**Description**—Enables or disables exposing a local device to the Internet for a special purpose service **User Interface**—**Network Setup** > **Application** > **DMZ** page, **Status** field **Valid inputs** • 0: Disabled • 1: Enabled **Default**—0 **Example**—DMZ enabled <Status>1</Status>
<Private_IP>	**Description**—The local IPv4 address of the device that can be accessed through the DMZ **User Interface**—**Network Setup** > **Application** > **DMZ** page, **Private IP** field **Valid inputs**—IPv4 address **Default**—0.0.0.0 **Example** <Private_IP>192.168.15.1</Private_IP>
<Rule_Number>	**Description**—A static setting used to define the DMZ rule **User Interface**—not applicable **Valid inputs**—1 (do not change this number) **Default**—1

<Software_DMZ> Example: DMZ allowing Internet traffic to access

```
192.168.15.101
<router-configuration>
...
<Software_DMZ>
<Rule1>
<Status>1</Status>
```

```
<Private_IP>192.168.15.1</Private_IP>
</Rule1>
<Rule_Number>1</Rule_Number>
</Software_DMZ>
...
</router-configuration>
```

Bonjour_Enable

Parameter	Details
<Bonjour_Enable>	**Description**—Enables or disables the Bonjour service discovery protocol, which may be required by network management systems that you use **User Interface**—**Administration** > **Management** > **Bonjour** page, **Enabled** and **Disabled** fields **Valid inputs** • 0: Disabled • 1: Enabled **Default**—1 **Example**—Bonjour enabled <Bonjour_Enable>1</Bonjour_Enable>

Reset_Button_Enable

Note No other settings are nested below <Reset_Button_Enable>.

Parameter	Details
<Reset_Button_Enable>	**Description**—Enables or disables the RESET button **User Interface** **Valid inputs** • 0: Disabled (button) • 1: Enabled (button can be pressed for 1-2 seconds for reboot and 5-6 seconds for a factory reset) **Default**—1 **Example**—Button disabled <Reset_Button_Enable>0</<Reset_Button_Enable>

Router_Mode

Parameter	Details
<Router_Mode>	**Description**—The operating mode of the router **User Interface**—**Network Setup** > **Basic Setup** > **Network Service** page, **Networking Service** field **Valid inputs** • 0: Bridge • 1: NAT **Default**—1 **Example**—Bridge mode enabled <Router_Mode>0<Router_Mode>

Monitor_WAN_Port_Only Parameters

This section describes the parameters in the <Monitor_WAN_Port_Only> section of the config.xml file.

TIP: You can click the <Monitor_WAN_Port_Only> heading in the XML file to expand or collapse the nested parameters in this section.

Parameter	Details
Monitor_WAN_Port_Only	**Description**—To monitor device link status base on wan port only. This configuration is only valid when <Router_Mode> is set to 0 (bridge). **User Interface**—**Network Setup** > **Basic Setup** > **Network Service** page, **Monitor Network Drop on Internet Port only** field **Values** • 0: Off • 1: On **Default**—0

VPN_Passthrough

This section describes the parameters in the <VPN_Passthrough> section of the config.xml file.

Parameter	Details
<IPSec_Passthrough>	**Description**—Enables or disables VPN passthrough for Internet Protocol Security (IPsec) **User Interface**—**Network Setup** > **Advanced Settings** > **VPN Passthrough** page, **IPsec Passthrough** field **Valid inputs** • 0: Disabled • 1: Enabled **Default**—1 **Example** <IPSec_Passthrough>1</IPSec_Passthrough>
<PPTP_Passthrough>	**Description**—Enables or disables VPN passthrough for Point-to-Point Tunneling Protocol (PPTP) **User Interface**—**Network Setup** > **Advanced Settings** > **VPN Passthrough** page, **PPTP Passthrough** field **Valid inputs** • 0: Disabled • 1: Enabled **Default**—1 **Example** <PPTP_Passthrough>1</PPTP_Passthrough>
<L2TP_Passthrough>	**Description**—Enables or disables VPN passthrough for Layer 2 Tunneling Protocol (L2TP) **User Interface**—**Network Setup** > **Advanced Settings** > **VPN Passthrough** page, **L2TP Passthrough** field **Valid inputs** • 0: Disabled • 1: Enabled **Default**—1 **Example** <L2TP_Passthrough>1</L2TP_Passthrough>

<VPN_Passthrough> Example: All passthrough options enabled

```
<router-configuration>
...
```

```
<VPN_Passthrough>
<IPSec_Passthrough>1</IPSec_Passthrough>
<PPTP_Passthrough>1</PPTP_Passthrough>
<L2TP_Passthrough>1</L2TP_Passthrough>
</VPN_Passthrough>
...
</router-configuration>
```

Web_Management

This section describes the parameters in the <Web_Management> section of the config.xml file.

Parameter	Details
<Web_Utility_Access_HTTP>	**Description**—Enables or disables access to the web-based configuration utility via HTTP, from a computer on the LAN **User Interface**—**Administration** > **Management** > **Web Access Management** page, **Web Utility Access** field, **HTTP** option **Valid inputs** • 0: Disabled • 1: Enabled **Default**—0 **Example** <Web_Utility_Access_HTTP>1</Web_Utility_Access_HTTP>
<Web_Utility_Access_HTTPS>	**Description**—Enables or disables access to the web-based configuration utility via HTTPS, from a computer on the LAN **User Interface**—**Administration** > **Management** > **Web Access Management** page, **Web Utility Access** field, HTTPS option **Valid inputs** • 0: Disabled • 1: Enabled **Default**—1 **Example** <Web_Utility_Access_HTTPS>1</Web_Utility_Access_HTTPS>

Parameter	Details
<Web_Remote_Management>	**Description**—Enables or disables access to the web-based configuration utility through the WAN interface (INTERNET port) **User Interface**—**Administration** > **Management** > **Web Access Management** page, **Remote Management** field **Valid inputs** • 0: Disabled • 1: Enabled **Default**—0 **Example** <Web_Remote_Management>1</Web_Remote_Management>
<Remote_Web_Utility_Access>	**Description**—Specifies the protocol that can be used to access the web-based configuration utility through the WAN interface (INTERNET port), when Remote Management is enabled **User Interface**—**Administration** > **Management** > **Web Access Management** page, **Web Utility Access** field **Valid inputs** • 0: HTTP • 1: HTTPS **Default**—1 **Example** <Remote_Web_Utility_Access>1</Remote_Web_Utility_Access>
<Web_Remote_Upgrade>	**Description**—Enables or disables upgrading the firmware from a computer on the WAN, when Remote Management is enabled **User Interface**—**Administration** > **Management** > **Web Access Management** page, Remote Upgrade field **Valid inputs** • 0: Disabled • 1: Enabled **Default**—0 **Example** <Web_Remote_Upgrade>1</Web_Remote_Upgrade>

Parameter	Details
<Allowed_Remote_IP_Type>	**Description**—Specifies a method for identifying remote devices that are allowed access to the web-based configuration utility, when Remote Management is enabled **User Interface**—**Administration** > **Management** > **Web Access Management** page, **Allowed Remote IPv4 Address** field, **Any IP Address** option **Valid inputs**: • 0: Specified IP Address • 1: Any IP Address **Default**—1 **Example** <Allowed_Remote_IP_Type>0</Allowed_Remote_IP_Type>
<Allowed_Remote_IP_Address>	**Description**—Specifies a remote IPv4 address that is allowed access to the web-based configuration utility, when Remote Management is enabled **User Interface**—**Administration** > **Management** > **Web Access Management** page, **Allowed Remote IPv4 Address** field, unlabeled text box **Valid inputs**—IPv4 address **Default**—0.0.0.0 **Example** <Allowed_Remote_IP_Address>209.165.201.129</Allowed_Remote_IP_Address>
<Remote_Management_Port>	**Description**—Specifies the port to use for access to the web-based configuration utility through the WAN interface (INTERNET port) **User Interface**—**Administration** > **Management** > **Web Access Management** page, **Remote Management Port** field **Valid inputs**—port number **Default**—443 **Example** <Remote_Management_Port>443</Remote_Management_Port>

<Web_Management> Example: Remote Management and Remote Upgrade enabled

```
<router-configuration>
...
<Web_Management>
<Web_Utility_Access_HTTP>0</Web_Utility_Access_HTTP>
<Web_Utility_Access_HTTPS>1</Web_Utility_Access_HTTPS>
<Web_Remote_Management>1</Web_Remote_Management>
<Remote_Web_Utility_Access>1</Remote_Web_Utility_Access>
<Web_Remote_Upgrade>1</Web_Remote_Upgrade>
<Allowed_Remote_IP_Type>0</Allowed_Remote_IP_Type>
<Allowed_Remote_IP_Address>209.165.201.129 129</Allowed_Remote_IP_Address>
<Remote_Management_Port>443</Remote_Management_Port>
```

```
</Web_Management>
...
</router-configuration>
```

TR-069 Parameters

This section describes the parameters in the <TR_069> section of the config.xml file.

Parameter	**Details**
<TR_069_Status>	**Description**—Enables or disables remote provisioning via TR-069 CPE WAN Management Protocol **User Interface**—Administration > Management > TR-069 page, Status field **Valid inputs** • 0: Disabled • 1: Enabled **Default**—0 **Example** <TR_069_Status>1</TR_069_Status>
<TR_069_ACS_URL>	**Description**—The URL of the Auto-Configuration Server (ACS) **User Interface**—Administration > Management > TR-069 page, ACS URL field **Valid inputs**—Domain name or IP address, starting with http:// or https://, and optionally ending with a port number **Default**—null **Example** <TR_069_ACS_URL>http://ACS-example.com</TR_069_ACS_URL>
<TR_069_ACS_Username>	**Description**—The username for HTTP-based authentication to the ACS **User Interface**—Administration > Management > TR-069 page, ACS Username field **Valid inputs**—username **Default**—null **Example** <TR_069_ACS_Username>MyUsername</TR_069_ACS_Username>

Parameter	Details
<TR_069_ACS_Password>	**Description**—The password for HTTP-based authentication to the ACS **User Interface**—Administration > Management > TR-069 page, ACS Password field **Valid inputs**—password **Default**—commented out: <!-- <TR_069_ACS_Password></TR_069_ACS_Password> --> **Example** <TR_069_ACS_Password>MyACSPassword</TR_069_ACS_Password>
<TR_069_Connection_Request_URL>	**Description**—This field will be autofilled and does not need to be entered manually **User Interface**—Administration > Management > TR-069 page, Connection Request URL field **Valid inputs**—URL **Default**—null **Example**—not applicable, value is autofilled
<TR_069_Connection_Request_Username>	**Description**—This field will be autofilled and does not need to be entered manually **User Interface**—Administration > Management > TR-069 page, Connection Request Username field **Valid inputs**—username **Default**—null **Example**—not applicable, value is autofilled
<TR_069_Connection_Request_Password>	**Description**—This field will be autofilled and does not need to be entered manually **User Interface**—Administration > Management > TR-069 page, Connection Request Password field **Valid inputs**—password **Default**—commented out, <!--<TR_069_Connection_Request_Password></TR_069_Connection_Request_Password>--> **Example** <TR_069_Connection_Request_Password> MyPassword</TR_069_Connection_Request_Password>

Parameter	Details
<TR_069_Periodic_Inform_Interval>	**Description**—When Periodic Information is enabled, the duration, in seconds, between CPE attempts to connect to the ACS **User Interface**—Administration > Management > TR-069 page, Periodic Inform Interval field **Valid inputs**—number **Default**—86400 **Example**—Interval of 36000 seconds (10 minutes) <TR_069_Periodic_Inform_Interval>36000</TR_069_Periodic_Inform_Interval>
<TR_069_Periodic_Inform_Enable>	**Description**—Enables or disables CPE connection requests to the ACS **User Interface**—Administration > Management > TR-069 page, Periodic Inform Enable field **Valid inputs**— • 0: Disabled • 1: Enabled **Default**—1 **Example**—Periodic Inform enabled <TR_069_Periodic_Inform_Enable>1</TR_069_Periodic_Inform_Enable>

Log_Configuration Parameters

This section describes the parameters in the <Log_Configuration> section of the config.xml file.

Parameter	Details
<Log_Module>	**Description**—Value that indicates the debug flag of modules: • 0: Default • 1: Preset • 2: Telephony • 3: SIP • 4: UI • 5: Network • 6: Media • 7: System • 8: Web • 9: NTP • 10: CDP/LLDP • 11: Security • 12: CSSD_RTP • 13: CSSD_FAX • 14: CSSD_ANY **User Interface**—**Administration** > **Debug Log Module** page, **Debug Log Module** field **Valid inputs**—0-14 **Default**—0
<RAM_Log_Size>	**Description**—The maximum size of the log file in kilobytes **User Interface**—**Administration** > **Debug Log Setting** page, **Debug Log Size** field **Valid inputs**—number from 128~1024 **Default**—200 **Example** `<RAM_Log_Size>200</RAM_Log_Size>`
Syslog_Server_IP	**Description**—IPv4 address of debug log server **User Interface**—**Administration** > **Debug Log Setting** page, **Pv4 Address** field **Valid inputs**—Valid IPv4 address format **Default**—null

Parameter	Details
Syslog_Server_IP6	**Description**—IPv6 address of debug log server **User Interface**—**Administration** > **Debug Log Setting** page, **Pv6 Address** field **Valid inputs**—Valid IPv6 address format **Default**—null
Syslog_Server_Port	**Description**—debug log server port **User Interface**—**Administration** > **Debug Log Setting** page, **Port** field **Valid inputs**—0-65535 **Default**—514
Event_Log_Server	**Description**—Address of event log server, supports IPv4, IPv6, and FQDN **User Interface**—**Administration** > **Debug Log Setting** page, **Address** field **Valid inputs**—Valid IPv4, IPv6, or FQDN address format. Maximum length is 128 characters **Default**—null
Event_Log_Port	**Description**—Port of event log server **User Interface**—**Administration** > **Debug Log Setting** page, **Port** field **Valid inputs**—0-65535 **Default**—514
Event_Log_Flag	**Description**—An bitwise value to turn on/off report of each event category (<DEV>: 1, <SYS>: 2 <CFG>: 4, <REG>: 8) **User Interface**—**Administration** > **Debug Log Setting** page, **Flag** field **Valid inputs**—0-65535 The available options are: • 0: Disable • 1: DEV • 2: SYS • 4: CFG • 8: REG • 15: DEV+SYS+CFG+REG **Default**—15

Parameter	Details
PRT_Upload_Url	**Description**—Address of PRT upload server **User Interface**—**Administration** > **Debug Log Setting** page, **PRT Upload URL** field **Valid inputs**—Valid URL format. Maximum length is 256 characters. **Default**—null
PRT_Upload_Method	**Description**—HTTP method to upload PRT **User Interface**—**Administration** > **Debug Log Setting** page, **Debug Log Size** field **Valid inputs** • 0: POST • 1: PUT) **Default**—0
PRT_Max_Timer	**Description**—Value in minutes to specify interval of periodical PRT report. **User Interface**—**Administration** > **Debug Log Setting** page, **PRT Max Timer** field **Valid inputs** • 0: Disable • 15-1440 **Default**—0

Web_Login_Admin_Name

Parameter	Details
<Web_Login_Admin_Name>	**Description**—The username for the administrator login, which has full read-write access to all parameters **User Interface**—**Administration** > **Management** > **User List** page, **Username** field **Valid inputs**—username **Default**—admin

Web_Login_Admin_Password

Parameter	Details
<Web_Login_Admin_Password>	Description—The password for the administrator login **User Interface**—**Administration** > **Management** > **User List** page **Valid inputs**—password (the minimum length of the characters is 8) **Default**—commented out <!--<Web_Login_Admin_Password></Web_Login_Admin_Password>--> **Example** <Web_Login_Admin_Password>MyPassword</Web_Login_Admin_Password>

Web_Login_Guest_Name

Parameter	Details
<Web_Login_Guest_Name>	**Description**—The username for the guest login, which has limited access to view or change parameters **User Interface**—**Administration** > **Management** > **User List** page **Valid inputs**—username **Default**—cisco **Example** **`<Web_Login_Guest_Name>MyUsername</Web_Login_Guest_Name>`**

Web_Login_Guest_Password

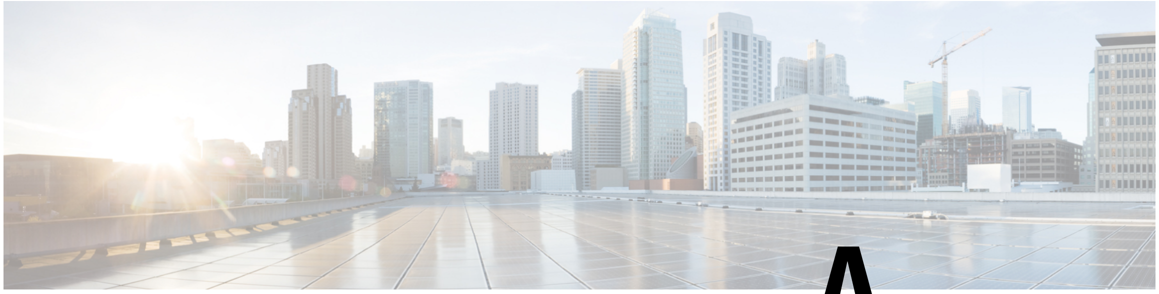
Parameter	Details
<Web_Login_Guest_Password>	**User Interface**—**Administration** > **Management** > **User List Page** **Valid inputs**—password (the minimum length of the characters is 8) **Default**—commented out, **`<!--<Web_Login_Guest_Password></Web_Login_Guest_Password>-->`** **Example**— **`<Web_Login_Guest_Password>MyPassword</Web_Login_Guest_Password>`**

SSH Parameters

This section describes the parameters in the <SSH> section of the config.xml file.

TIP: You can click the <SSH> heading in the XML file to expand or collapse the nested parameters in this section.

Parameter	Details
SSH_ACCESS	**Description**—Set enabled to allow access to SSH service. **User Interface**—**Administration** > **SSH** page, **Access** field **Values** • 0: Disabled • 1: Enabled **Default**—0
SSH_User_ID	**Description**—User name of SSH **User Interface**—**Administration** > **SSH** page, **User Name** field **Values**—0-50 **Default**—null
SSH_Password	**Description**—Password of SSH. **User Interface**—**Administration** > **SSH** page, **Password** field **Values**—0-50 **Default**—null



APPENDIX A

Sample Configuration Profiles

XML Open Format Sample

```
<?xml version="1.0" encoding="UTF-8"?>
<flat-profile>
<!--  Parameters for System Tab  -->
<!--  System Configuration  -->
<Restricted_Access_Domains ua="na"/>
<IVR_Admin_Passwd ua="na"/>
<Network_Startup_Delay ua="na">3</Network_Startup_Delay>
<!--  Miscellaneous Settings  -->
<DNS_Query_TTL_Ignore ua="na">No</DNS_Query_TTL_Ignore>
<DevTest_Password ua="na"/>
<!--  Parameters for Provisioning Tab  -->
<!--  Account Profile  -->
<User_ID ua="rw"/>
<Password ua="rw"/>
<!--  Configuration Profile  -->
<Provision_Enable ua="na">Yes</Provision_Enable>
<Resync_On_Reset ua="na">Yes</Resync_On_Reset>
<Resync_Random_Delay ua="na">2</Resync_Random_Delay>
<Resync_At__HHmm_ ua="na"/>
<Resync_At_Random_Delay ua="na">600</Resync_At_Random_Delay>
<Resync_Periodic ua="na">3600</Resync_Periodic>
<Resync_Error_Retry_Delay ua="na">3600</Resync_Error_Retry_Delay>
<Forced_Resync_Delay ua="na">14400</Forced_Resync_Delay>
<Resync_From_SIP ua="na">Yes</Resync_From_SIP>
<Resync_After_Upgrade_Attempt ua="na">Yes</Resync_After_Upgrade_Attempt>
<Resync_Trigger_1 ua="na"/>
<Resync_Trigger_2 ua="na"/>
<Resync_Fails_On_FNF ua="na">Yes</Resync_Fails_On_FNF>
<HTTPS_Name_Validate ua="na">Yes</HTTPS_Name_Validate>
<Profile_Rule ua="na">/ata$PSN.cfg</Profile_Rule>
<Profile_Rule_B ua="na"/>
<Profile_Rule_C ua="na"/>
<Profile_Rule_D ua="na"/>
<DHCP_Option_To_Use ua="na">66,160,159,150</DHCP_Option_To_Use>
<Transport_Protocol ua="na">https</Transport_Protocol>
<Log_Resync_Request_Msg ua="na">$PN $MAC -- Requesting resync
$SCHEME://$SERVIP:$PORT$PATH</Log_Resync_Request_Msg>
<Log_Resync_Success_Msg ua="na">$PN $MAC -- Successful resync
$SCHEME://$SERVIP:$PORT$PATH</Log_Resync_Success_Msg>
<Log_Resync_Failure_Msg ua="na">$PN $MAC -- Resync failed: $ERR</Log_Resync_Failure_Msg>
```

```
<Report_Rule ua="na"/>
<SP_Default ua="na"/>
<!--  Firmware Upgrade  -->
<Upgrade_Enable ua="na">Yes</Upgrade_Enable>
<Upgrade_Error_Retry_Delay ua="na">3600</Upgrade_Error_Retry_Delay>
<Downgrade_Rev_Limit ua="na"/>
<Upgrade_Rule ua="na"/>
<Log_Upgrade_Request_Msg ua="na">$PN $MAC -- Requesting upgrade
$SCHEME://$SERVIP:$PORT$PATH</Log_Upgrade_Request_Msg>
<Log_Upgrade_Success_Msg ua="na">$PN $MAC -- Successful upgrade $SCHEME://$SERVIP:$PORT$PATH
 -- $ERR</Log_Upgrade_Success_Msg>
<Log_Upgrade_Failure_Msg ua="na">$PN $MAC -- Upgrade failed: $ERR</Log_Upgrade_Failure_Msg>
<License_Keys ua="na"/>
<Recovery_URL ua="na"/>
<!--  CA Settings  -->
<Custom_CA_URL ua="na">http://10.74.51.176/cisco/certificate.pem</Custom_CA_URL>
<!--  General Purpose Parameters  -->
<GPP_A ua="na"/>
<GPP_B ua="na"/>
<GPP_C ua="na"/>
<GPP_D ua="na"/>
<GPP_E ua="na"/>
<GPP_F ua="na"/>
<GPP_G ua="na"/>
<GPP_H ua="na"/>
<GPP_I ua="na"/>
<GPP_J ua="na"/>
<GPP_K ua="na"/>
<GPP_L ua="na"/>
<GPP_M ua="na"/>
<GPP_N ua="na"/>
<GPP_O ua="na"/>
<GPP_P ua="na"/>
<GPP_SA ua="na"/>
<GPP_SB ua="na"/>
<GPP_SC ua="na"/>
<GPP_SD ua="na"/>
<!--  Parameters for SIP Tab  -->
<!--  SIP Parameters  -->
<Max_Forward ua="na">70</Max_Forward>
<Max_Redirection ua="na">5</Max_Redirection>
<Max_Auth ua="na">2</Max_Auth>
<SIP_User_Agent_Name ua="na">$VERSION</SIP_User_Agent_Name>
<SIP_Server_Name ua="na">$VERSION</SIP_Server_Name>
<SIP_Reg_User_Agent_Name ua="na"/>
<SIP_Reg_Starting_Sequence_Number ua="na"/>
<SIP_Accept_Language ua="na"/>
<DTMF_Relay_MIME_Type ua="na">application/dtmf-relay</DTMF_Relay_MIME_Type>
<Hook_Flash_MIME_Type ua="na">application/hook-flash</Hook_Flash_MIME_Type>
<Remove_Last_Reg ua="na">No</Remove_Last_Reg>
<Use_Compact_Header ua="na">No</Use_Compact_Header>
<Escape_Display_Name ua="na">No</Escape_Display_Name>
<RFC_2543_Call_Hold ua="na">Yes</RFC_2543_Call_Hold>
<Mark_All_AVT_Packets ua="na">Yes</Mark_All_AVT_Packets>
<AVT_Packet_Size ua="na">ptime</AVT_Packet_Size>
<SIP_TCP_Port_Min ua="na">5060</SIP_TCP_Port_Min>
<SIP_TCP_Port_Max ua="na">5080</SIP_TCP_Port_Max>
<CTI_Enable ua="na">No</CTI_Enable>
<Keep_Referee_When_REFER_Failed ua="na">No</Keep_Referee_When_REFER_Failed>
<Caller_ID_Header ua="na">PAID-RPID-FROM</Caller_ID_Header>
<!--  SIP Timer Values (sec)  -->
<SIP_T1 ua="na">.5</SIP_T1>
<SIP_T2 ua="na">4</SIP_T2>
<SIP_T4 ua="na">5</SIP_T4>
```

```
<SIP_Timer_B ua="na">32</SIP_Timer_B>
<SIP_Timer_F ua="na">16</SIP_Timer_F>
<SIP_Timer_H ua="na">32</SIP_Timer_H>
<SIP_Timer_D ua="na">32</SIP_Timer_D>
<SIP_Timer_J ua="na">32</SIP_Timer_J>
<INVITE_Expires ua="na">240</INVITE_Expires>
<ReINVITE_Expires ua="na">30</ReINVITE_Expires>
<Reg_Min_Expires ua="na">1</Reg_Min_Expires>
<Reg_Max_Expires ua="na">7200</Reg_Max_Expires>
<Reg_Retry_Intvl ua="na">30</Reg_Retry_Intvl>
<Reg_Retry_Long_Intvl ua="na">1200</Reg_Retry_Long_Intvl>
<Reg_Retry_Random_Delay ua="na">0</Reg_Retry_Random_Delay>
<Reg_Retry_Long_Random_Delay ua="na">0</Reg_Retry_Long_Random_Delay>
<Reg_Retry_Intvl_Cap ua="na">0</Reg_Retry_Intvl_Cap>
<!--  Response Status Code Handling  -->
<SIT1_RSC ua="na"/>
<SIT2_RSC ua="na"/>
<SIT3_RSC ua="na"/>
<SIT4_RSC ua="na"/>
<Try_Backup_RSC ua="na"/>
<Retry_Reg_RSC ua="na"/>
<!--  RTP Parameters  -->
<RTP_Port_Min ua="na">16384</RTP_Port_Min>
<RTP_Port_Max ua="na">16482</RTP_Port_Max>
<RTP_Packet_Size ua="na">0.030</RTP_Packet_Size>
<RTP_Tx_Packet_Size_Follows_Remote_SDP ua="na">Yes</RTP_Tx_Packet_Size_Follows_Remote_SDP>
<Max_RTP_ICMP_Err ua="na">0</Max_RTP_ICMP_Err>
<RTCP_Tx_Interval ua="na">0</RTCP_Tx_Interval>
<No_UDP_Checksum ua="na">No</No_UDP_Checksum>
<Stats_In_BYE ua="na">Yes</Stats_In_BYE>
<Call_Statistics ua="na">Yes</Call_Statistics>
<!--  SDP Payload Types  -->
<NSE_Dynamic_Payload ua="na">100</NSE_Dynamic_Payload>
<AVT_Dynamic_Payload ua="na">101</AVT_Dynamic_Payload>
<INFOREQ_Dynamic_Payload ua="na"/>
<G726r32_Dynamic_Payload ua="na">2</G726r32_Dynamic_Payload>
<EncapRTP_Dynamic_Payload ua="na">112</EncapRTP_Dynamic_Payload>
<RTP-Start-Loopback_Dynamic_Payload ua="na">113</RTP-Start-Loopback_Dynamic_Payload>
<RTP-Start-Loopback_Codec ua="na">G711u</RTP-Start-Loopback_Codec>
<NSE_Codec_Name ua="na">NSE</NSE_Codec_Name>
<AVT_Codec_Name ua="na">telephone-event</AVT_Codec_Name>
<G711u_Codec_Name ua="na">PCMU</G711u_Codec_Name>
<G711a_Codec_Name ua="na">PCMA</G711a_Codec_Name>
<G726r32_Codec_Name ua="na">G726-32</G726r32_Codec_Name>
<G729a_Codec_Name ua="na">G729a</G729a_Codec_Name>
<EncapRTP_Codec_Name ua="na">encaprtp</EncapRTP_Codec_Name>
<!--  NAT Support Parameters  -->
<Handle_VIA_received ua="na">No</Handle_VIA_received>
<Handle_VIA_rport ua="na">No</Handle_VIA_rport>
<Insert_VIA_received ua="na">No</Insert_VIA_received>
<Insert_VIA_rport ua="na">No</Insert_VIA_rport>
<Substitute_VIA_Addr ua="na">No</Substitute_VIA_Addr>
<Send_Resp_To_Src_Port ua="na">No</Send_Resp_To_Src_Port>
<STUN_Enable ua="na">No</STUN_Enable>
<STUN_Test_Enable ua="na">No</STUN_Test_Enable>
<STUN_Server ua="na"/>
<EXT_IP ua="na"/>
<EXT_RTP_Port_Min ua="na"/>
<NAT_Keep_Alive_Intvl ua="na">15</NAT_Keep_Alive_Intvl>
<Redirect_Keep_Alive ua="na">No</Redirect_Keep_Alive>
<!--  Parameters for Line 1 Tab  -->
<!--  General  -->
<Line_Enable_1_ ua="na">Yes</Line_Enable_1_>
<!--  Streaming Audio Server (SAS)  -->
```

```
<SAS_Enable_1_ ua="na">No</SAS_Enable_1_>
<SAS_DLG_Refresh_Intvl_1_ ua="na">30</SAS_DLG_Refresh_Intvl_1_>
<SAS_Inbound_RTP_Sink_1_ ua="na"/>
<!--  NAT Settings  -->
<NAT_Mapping_Enable_1_ ua="na">No</NAT_Mapping_Enable_1_>
<NAT_Keep_Alive_Enable_1_ ua="na">No</NAT_Keep_Alive_Enable_1_>
<NAT_Keep_Alive_Msg_1_ ua="na">$OPTIONS</NAT_Keep_Alive_Msg_1_>
<NAT_Keep_Alive_Dest_1_ ua="na">$PROXY</NAT_Keep_Alive_Dest_1_>
<!--  Network Settings  -->
<SIP_ToS_DiffServ_Value_1_ ua="na">0x68</SIP_ToS_DiffServ_Value_1_>
<SIP_CoS_Value_1_ ua="na">3</SIP_CoS_Value_1_>
<RTP_ToS_DiffServ_Value_1_ ua="na">0xb8</RTP_ToS_DiffServ_Value_1_>
<RTP_CoS_Value_1_ ua="na">6</RTP_CoS_Value_1_>
<Network_Jitter_Level_1_ ua="na">high</Network_Jitter_Level_1_>
<Jitter_Buffer_Adjustment_1_ ua="na">Yes</Jitter_Buffer_Adjustment_1_>
<!--  SIP Settings  -->
<SIP_Transport_1_ ua="na">UDP</SIP_Transport_1_>
<SIP_Port_1_ ua="na">5060</SIP_Port_1_>
<SIP_100REL_Enable_1_ ua="na">No</SIP_100REL_Enable_1_>
<EXT_SIP_Port_1_ ua="na"/>
<Auth_Resync-Reboot_1_ ua="na">Yes</Auth_Resync-Reboot_1_>
<SIP_Proxy-Require_1_ ua="na"/>
<SIP_Remote-Party-ID_1_ ua="na">Yes</SIP_Remote-Party-ID_1_>
<SIP_GUID_1_ ua="na">No</SIP_GUID_1_>
<RTP_Log_Intvl_1_ ua="na">0</RTP_Log_Intvl_1_>
<Restrict_Source_IP_1_ ua="na">No</Restrict_Source_IP_1_>
<Referor_Bye_Delay_1_ ua="na">4</Referor_Bye_Delay_1_>
<Refer_Target_Bye_Delay_1_ ua="na">0</Refer_Target_Bye_Delay_1_>
<Referee_Bye_Delay_1_ ua="na">0</Referee_Bye_Delay_1_>
<Refer-To_Target_Contact_1_ ua="na">No</Refer-To_Target_Contact_1_>
<Sticky_183_1_ ua="na">No</Sticky_183_1_>
<Auth_INVITE_1_ ua="na">No</Auth_INVITE_1_>
<Reply_182_On_Call_Waiting_1_ ua="na">No</Reply_182_On_Call_Waiting_1_>
<Use_Anonymous_With_RPID_1_ ua="na">Yes</Use_Anonymous_With_RPID_1_>
<Use_Local_Addr_In_FROM_1_ ua="na">No</Use_Local_Addr_In_FROM_1_>
<Broadsoft_ALTC_1_ ua="na">No</Broadsoft_ALTC_1_>
<TLS_Name_Validate_1_ ua="na">Yes</TLS_Name_Validate_1_>
<!--  Call Feature Settings  -->
<Blind_Attn-Xfer_Enable_1_ ua="na">No</Blind_Attn-Xfer_Enable_1_>
<MOH_Server_1_ ua="na"/>
<Xfer_When_Hangup_Conf_1_ ua="na">Yes</Xfer_When_Hangup_Conf_1_>
<Conference_Bridge_URL_1_ ua="na"/>
<Conference_Bridge_Ports_1_ ua="na">3</Conference_Bridge_Ports_1_>
<Enable_IP_Dialing_1_ ua="na">No</Enable_IP_Dialing_1_>
<Emergency_Number_1_ ua="na"/>
<Mailbox_ID_1_ ua="na"/>
<Feature_Key_Sync_1_ ua="na">No</Feature_Key_Sync_1_>
<Secure_Call_Option_1_ ua="na">Strict</Secure_Call_Option_1_>
<!--  E911 Geolocation Configuration  -->
<Company_UUID_1_ ua="na"/>
<Primary_Request_URL_1_ ua="na"/>
<Secondary_Request_URL_1_ ua="na"/>
<!--  Proxy and Registration  -->
<Proxy_1_ ua="na">10.74.51.158</Proxy_1_>
<Outbound_Proxy_1_ ua="na"/>
<Use_Outbound_Proxy_1_ ua="na">No</Use_Outbound_Proxy_1_>
<Use_OB_Proxy_In_Dialog_1_ ua="na">Yes</Use_OB_Proxy_In_Dialog_1_>
<Registrar_Server_1_ ua="na"/>
<Register_1_ ua="na">Yes</Register_1_>
<Make_Call_Without_Reg_1_ ua="na">No</Make_Call_Without_Reg_1_>
<Register_Expires_1_ ua="na">3600</Register_Expires_1_>
<Ans_Call_Without_Reg_1_ ua="na">No</Ans_Call_Without_Reg_1_>
<Use_DNS_SRV_1_ ua="na">No</Use_DNS_SRV_1_>
<DNS_SRV_Auto_Prefix_1_ ua="na">No</DNS_SRV_Auto_Prefix_1_>
```

```
<Proxy_Fallback_Intvl_1_ ua="na">3600</Proxy_Fallback_Intvl_1_>
<Proxy_Redundancy_Method_1_ ua="na">Normal</Proxy_Redundancy_Method_1_>
<Mailbox_Subscribe_URL_1_ ua="na"/>
<Mailbox_Subscribe_Expires_1_ ua="na">2147483647</Mailbox_Subscribe_Expires_1_>
<Auto_Register_When_Failover_1_ ua="na">No</Auto_Register_When_Failover_1_>
<!--  Subscriber Information  -->
<Display_Name_1_ ua="na">11422</Display_Name_1_>
<User_ID_1_ ua="na">11422</User_ID_1_>
<Password_1_ ua="na"/>
<Use_Auth_ID_1_ ua="na">No</Use_Auth_ID_1_>
<Auth_ID_1_ ua="na"/>
<Resident_Online_Number_1_ ua="na"/>
<SIP_URI_1_ ua="na"/>
<!--  Supplementary Service Subscription  -->
<Call_Waiting_Serv_1_ ua="na">Yes</Call_Waiting_Serv_1_>
<Block_CID_Serv_1_ ua="na">Yes</Block_CID_Serv_1_>
<Block_ANC_Serv_1_ ua="na">Yes</Block_ANC_Serv_1_>
<Dist_Ring_Serv_1_ ua="na">Yes</Dist_Ring_Serv_1_>
<Cfwd_All_Serv_1_ ua="na">Yes</Cfwd_All_Serv_1_>
<Cfwd_Busy_Serv_1_ ua="na">Yes</Cfwd_Busy_Serv_1_>
<Cfwd_No_Ans_Serv_1_ ua="na">Yes</Cfwd_No_Ans_Serv_1_>
<Cfwd_Sel_Serv_1_ ua="na">Yes</Cfwd_Sel_Serv_1_>
<Cfwd_Last_Serv_1_ ua="na">Yes</Cfwd_Last_Serv_1_>
<Block_Last_Serv_1_ ua="na">Yes</Block_Last_Serv_1_>
<Accept_Last_Serv_1_ ua="na">Yes</Accept_Last_Serv_1_>
<DND_Serv_1_ ua="na">Yes</DND_Serv_1_>
<CID_Serv_1_ ua="na">Yes</CID_Serv_1_>
<CWCID_Serv_1_ ua="na">Yes</CWCID_Serv_1_>
<Call_Return_Serv_1_ ua="na">Yes</Call_Return_Serv_1_>
<Call_Redial_Serv_1_ ua="na">Yes</Call_Redial_Serv_1_>
<Call_Back_Serv_1_ ua="na">Yes</Call_Back_Serv_1_>
<Three_Way_Call_Serv_1_ ua="na">Yes</Three_Way_Call_Serv_1_>
<Three_Way_Conf_Serv_1_ ua="na">Yes</Three_Way_Conf_Serv_1_>
<Attn_Transfer_Serv_1_ ua="na">Yes</Attn_Transfer_Serv_1_>
<Unattn_Transfer_Serv_1_ ua="na">Yes</Unattn_Transfer_Serv_1_>
<MWI_Serv_1_ ua="na">Yes</MWI_Serv_1_>
<VMWI_Serv_1_ ua="na">Yes</VMWI_Serv_1_>
<Speed_Dial_Serv_1_ ua="na">Yes</Speed_Dial_Serv_1_>
<Secure_Call_Serv_1_ ua="na">Yes</Secure_Call_Serv_1_>
<Referral_Serv_1_ ua="na">Yes</Referral_Serv_1_>
<Feature_Dial_Serv_1_ ua="na">Yes</Feature_Dial_Serv_1_>
<Service_Announcement_Serv_1_ ua="na">No</Service_Announcement_Serv_1_>
<Reuse_CID_Number_As_Name_1_ ua="na">Yes</Reuse_CID_Number_As_Name_1_>
<CONFCID_Serv_1_ ua="na">Yes</CONFCID_Serv_1_>
<!--  Audio Configuration  -->
<Preferred_Codec_1_ ua="na">G711u</Preferred_Codec_1_>
<Second_Preferred_Codec_1_ ua="na">Unspecified</Second_Preferred_Codec_1_>
<Third_Preferred_Codec_1_ ua="na">Unspecified</Third_Preferred_Codec_1_>
<Use_Pref_Codec_Only_1_ ua="na">No</Use_Pref_Codec_Only_1_>
<Codec_Negotiation_1_ ua="na">Default</Codec_Negotiation_1_>
<G729a_Enable_1_ ua="na">Yes</G729a_Enable_1_>
<Silence_Supp_Enable_1_ ua="na">No</Silence_Supp_Enable_1_>
<G726-32_Enable_1_ ua="na">Yes</G726-32_Enable_1_>
<Silence_Threshold_1_ ua="na">medium</Silence_Threshold_1_>
<FAX_V21_Detect_Enable_1_ ua="na">Yes</FAX_V21_Detect_Enable_1_>
<Echo_Canc_Enable_1_ ua="na">Yes</Echo_Canc_Enable_1_>
<FAX_CNG_Detect_Enable_1_ ua="na">Yes</FAX_CNG_Detect_Enable_1_>
<FAX_Passthru_Codec_1_ ua="na">G711u</FAX_Passthru_Codec_1_>
<FAX_Codec_Symmetric_1_ ua="na">Yes</FAX_Codec_Symmetric_1_>
<DTMF_Process_INFO_1_ ua="na">Yes</DTMF_Process_INFO_1_>
<FAX_Passthru_Method_1_ ua="na">ReINVITE</FAX_Passthru_Method_1_>
<DTMF_Process_AVT_1_ ua="na">Yes</DTMF_Process_AVT_1_>
<FAX_Process_NSE_1_ ua="na">Yes</FAX_Process_NSE_1_>
<DTMF_Tx_Method_1_ ua="na">Auto</DTMF_Tx_Method_1_>
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<FAX_Disable_ECAN_1_ ua="na">No</FAX_Disable_ECAN_1_>
<DTMF_Tx_Mode_1_ ua="na">Strict</DTMF_Tx_Mode_1_>
<DTMF_Tx_Strict_Hold_Off_Time_1_ ua="na">70</DTMF_Tx_Strict_Hold_Off_Time_1_>
<FAX_Enable_T38_1_ ua="na">Yes</FAX_Enable_T38_1_>
<Hook_Flash_Tx_Method_1_ ua="na">None</Hook_Flash_Tx_Method_1_>
<FAX_T38_Redundancy_1_ ua="na">1</FAX_T38_Redundancy_1_>
<FAX_T38_ECM_Enable_1_ ua="na">Yes</FAX_T38_ECM_Enable_1_>
<FAX_Tone_Detect_Mode_1_ ua="na">caller or callee</FAX_Tone_Detect_Mode_1_>
<Symmetric_RTP_1_ ua="na">No</Symmetric_RTP_1_>
<FAX_T38_Return_to_Voice_1_ ua="na">No</FAX_T38_Return_to_Voice_1_>
<Modem_Line_1_ ua="na">No</Modem_Line_1_>
<RTP_to_Proxy_in_Remote_Hold_1_ ua="na">No</RTP_to_Proxy_in_Remote_Hold_1_>
<!--  Dial Plan  -->
<Dial_Plan_1_
ua="na">(*xx|[3469]11|0|00|[2-9]xxxxxx|1xxx[2-9]xxxxxxS0|xxxxxxxxxxxx.)</Dial_Plan_1_>
<!--  FXS Port Polarity Configuration  -->
<Idle_Polarity_1_ ua="na">Forward</Idle_Polarity_1_>
<Caller_Conn_Polarity_1_ ua="na">Forward</Caller_Conn_Polarity_1_>
<Callee_Conn_Polarity_1_ ua="na">Forward</Callee_Conn_Polarity_1_>
<Profile_Stamp_1_ ua="na"/>
<!--  Parameters for Line 2 Tab  -->
<!--  General  -->
<Line_Enable_2_ ua="na">Yes</Line_Enable_2_>
<!--  Streaming Audio Server (SAS)  -->
<SAS_Enable_2_ ua="na">No</SAS_Enable_2_>
<SAS_DLG_Refresh_Intvl_2_ ua="na">30</SAS_DLG_Refresh_Intvl_2_>
<SAS_Inbound_RTP_Sink_2_ ua="na"/>
<!--  NAT Settings  -->
<NAT_Mapping_Enable_2_ ua="na">No</NAT_Mapping_Enable_2_>
<NAT_Keep_Alive_Enable_2_ ua="na">No</NAT_Keep_Alive_Enable_2_>
<NAT_Keep_Alive_Msg_2_ ua="na">$OPTIONS</NAT_Keep_Alive_Msg_2_>
<NAT_Keep_Alive_Dest_2_ ua="na">$PROXY</NAT_Keep_Alive_Dest_2_>
<!--  Network Settings  -->
<SIP_ToS_DiffServ_Value_2_ ua="na">0x68</SIP_ToS_DiffServ_Value_2_>
<SIP_CoS_Value_2_ ua="na">3</SIP_CoS_Value_2_>
<RTP_ToS_DiffServ_Value_2_ ua="na">0xb8</RTP_ToS_DiffServ_Value_2_>
<RTP_CoS_Value_2_ ua="na">6</RTP_CoS_Value_2_>
<Network_Jitter_Level_2_ ua="na">high</Network_Jitter_Level_2_>
<Jitter_Buffer_Adjustment_2_ ua="na">Yes</Jitter_Buffer_Adjustment_2_>
<!--  SIP Settings  -->
<SIP_Transport_2_ ua="na">TLS</SIP_Transport_2_>
<SIP_Port_2_ ua="na">5061</SIP_Port_2_>
<SIP_100REL_Enable_2_ ua="na">No</SIP_100REL_Enable_2_>
<EXT_SIP_Port_2_ ua="na"/>
<Auth_Resync-Reboot_2_ ua="na">Yes</Auth_Resync-Reboot_2_>
<SIP_Proxy-Require_2_ ua="na"/>
<SIP_Remote-Party-ID_2_ ua="na">Yes</SIP_Remote-Party-ID_2_>
<SIP_GUID_2_ ua="na">No</SIP_GUID_2_>
<RTP_Log_Intvl_2_ ua="na">0</RTP_Log_Intvl_2_>
<Restrict_Source_IP_2_ ua="na">No</Restrict_Source_IP_2_>
<Referor_Bye_Delay_2_ ua="na">4</Referor_Bye_Delay_2_>
<Refer_Target_Bye_Delay_2_ ua="na">0</Refer_Target_Bye_Delay_2_>
<Referee_Bye_Delay_2_ ua="na">0</Referee_Bye_Delay_2_>
<Refer-To_Target_Contact_2_ ua="na">No</Refer-To_Target_Contact_2_>
<Sticky_183_2_ ua="na">No</Sticky_183_2_>
<Auth_INVITE_2_ ua="na">No</Auth_INVITE_2_>
<Reply_182_On_Call_Waiting_2_ ua="na">No</Reply_182_On_Call_Waiting_2_>
<Use_Anonymous_With_RPID_2_ ua="na">Yes</Use_Anonymous_With_RPID_2_>
<Use_Local_Addr_In_FROM_2_ ua="na">No</Use_Local_Addr_In_FROM_2_>
<Broadsoft_ALTC_2_ ua="na">No</Broadsoft_ALTC_2_>
<TLS_Name_Validate_2_ ua="na">Yes</TLS_Name_Validate_2_>
<!--  Call Feature Settings  -->
<Blind_Attn-Xfer_Enable_2_ ua="na">No</Blind_Attn-Xfer_Enable_2_>
<MOH_Server_2_ ua="na"/>
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<Xfer_When_Hangup_Conf_2_ ua="na">Yes</Xfer_When_Hangup_Conf_2_>
<Conference_Bridge_URL_2_ ua="na"/>
<Conference_Bridge_Ports_2_ ua="na">3</Conference_Bridge_Ports_2_>
<Enable_IP_Dialing_2_ ua="na">No</Enable_IP_Dialing_2_>
<Emergency_Number_2_ ua="na"/>
<Mailbox_ID_2_ ua="na"/>
<Feature_Key_Sync_2_ ua="na">No</Feature_Key_Sync_2_>
<Secure_Call_Option_2_ ua="na">Strict</Secure_Call_Option_2_>
<!--  E911 Geolocation Configuration  -->
<Company_UUID_2_ ua="na"/>
<Primary_Request_URL_2_ ua="na"/>
<Secondary_Request_URL_2_ ua="na"/>
<!--  Proxy and Registration  -->
<Proxy_2_ ua="na">asterisk11.sipurash.com</Proxy_2_>
<Outbound_Proxy_2_ ua="na"/>
<Use_Outbound_Proxy_2_ ua="na">No</Use_Outbound_Proxy_2_>
<Use_OB_Proxy_In_Dialog_2_ ua="na">Yes</Use_OB_Proxy_In_Dialog_2_>
<Registrar_Server_2_ ua="na"/>
<Register_2_ ua="na">Yes</Register_2_>
<Make_Call_Without_Reg_2_ ua="na">No</Make_Call_Without_Reg_2_>
<Register_Expires_2_ ua="na">3600</Register_Expires_2_>
<Ans_Call_Without_Reg_2_ ua="na">No</Ans_Call_Without_Reg_2_>
<Use_DNS_SRV_2_ ua="na">Yes</Use_DNS_SRV_2_>
<DNS_SRV_Auto_Prefix_2_ ua="na">Yes</DNS_SRV_Auto_Prefix_2_>
<Proxy_Fallback_Intvl_2_ ua="na">3600</Proxy_Fallback_Intvl_2_>
<Proxy_Redundancy_Method_2_ ua="na">Normal</Proxy_Redundancy_Method_2_>
<Mailbox_Subscribe_URL_2_ ua="na"/>
<Mailbox_Subscribe_Expires_2_ ua="na">2147483647</Mailbox_Subscribe_Expires_2_>
<Auto_Register_When_Failover_2_ ua="na">No</Auto_Register_When_Failover_2_>
<!--  Subscriber Information  -->
<Display_Name_2_ ua="na"/>
<User_ID_2_ ua="na">11422</User_ID_2_>
<Password_2_ ua="na"/>
<Use_Auth_ID_2_ ua="na">No</Use_Auth_ID_2_>
<Auth_ID_2_ ua="na"/>
<Resident_Online_Number_2_ ua="na"/>
<SIP_URI_2_ ua="na"/>
<!--  Supplementary Service Subscription  -->
<Call_Waiting_Serv_2_ ua="na">Yes</Call_Waiting_Serv_2_>
<Block_CID_Serv_2_ ua="na">Yes</Block_CID_Serv_2_>
<Block_ANC_Serv_2_ ua="na">Yes</Block_ANC_Serv_2_>
<Dist_Ring_Serv_2_ ua="na">Yes</Dist_Ring_Serv_2_>
<Cfwd_All_Serv_2_ ua="na">Yes</Cfwd_All_Serv_2_>
<Cfwd_Busy_Serv_2_ ua="na">Yes</Cfwd_Busy_Serv_2_>
<Cfwd_No_Ans_Serv_2_ ua="na">Yes</Cfwd_No_Ans_Serv_2_>
<Cfwd_Sel_Serv_2_ ua="na">Yes</Cfwd_Sel_Serv_2_>
<Cfwd_Last_Serv_2_ ua="na">Yes</Cfwd_Last_Serv_2_>
<Block_Last_Serv_2_ ua="na">Yes</Block_Last_Serv_2_>
<Accept_Last_Serv_2_ ua="na">Yes</Accept_Last_Serv_2_>
<DND_Serv_2_ ua="na">Yes</DND_Serv_2_>
<CID_Serv_2_ ua="na">Yes</CID_Serv_2_>
<CWCID_Serv_2_ ua="na">Yes</CWCID_Serv_2_>
<Call_Return_Serv_2_ ua="na">Yes</Call_Return_Serv_2_>
<Call_Redial_Serv_2_ ua="na">Yes</Call_Redial_Serv_2_>
<Call_Back_Serv_2_ ua="na">Yes</Call_Back_Serv_2_>
<Three_Way_Call_Serv_2_ ua="na">Yes</Three_Way_Call_Serv_2_>
<Three_Way_Conf_Serv_2_ ua="na">Yes</Three_Way_Conf_Serv_2_>
<Attn_Transfer_Serv_2_ ua="na">Yes</Attn_Transfer_Serv_2_>
<Unattn_Transfer_Serv_2_ ua="na">Yes</Unattn_Transfer_Serv_2_>
<MWI_Serv_2_ ua="na">Yes</MWI_Serv_2_>
<VMWI_Serv_2_ ua="na">Yes</VMWI_Serv_2_>
<Speed_Dial_Serv_2_ ua="na">Yes</Speed_Dial_Serv_2_>
<Secure_Call_Serv_2_ ua="na">Yes</Secure_Call_Serv_2_>
<Referral_Serv_2_ ua="na">Yes</Referral_Serv_2_>
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<Feature_Dial_Serv_2_ ua="na">Yes</Feature_Dial_Serv_2_>
<Service_Announcement_Serv_2_ ua="na">No</Service_Announcement_Serv_2_>
<Reuse_CID_Number_As_Name_2_ ua="na">Yes</Reuse_CID_Number_As_Name_2_>
<CONFCID_Serv_2_ ua="na">Yes</CONFCID_Serv_2_>
<!--  Audio Configuration  -->
<Preferred_Codec_2_ ua="na">G711u</Preferred_Codec_2_>
<Second_Preferred_Codec_2_ ua="na">Unspecified</Second_Preferred_Codec_2_>
<Third_Preferred_Codec_2_ ua="na">Unspecified</Third_Preferred_Codec_2_>
<Use_Pref_Codec_Only_2_ ua="na">No</Use_Pref_Codec_Only_2_>
<Codec_Negotiation_2_ ua="na">Default</Codec_Negotiation_2_>
<G729a_Enable_2_ ua="na">Yes</G729a_Enable_2_>
<Silence_Supp_Enable_2_ ua="na">No</Silence_Supp_Enable_2_>
<G726-32_Enable_2_ ua="na">Yes</G726-32_Enable_2_>
<Silence_Threshold_2_ ua="na">medium</Silence_Threshold_2_>
<FAX_V21_Detect_Enable_2_ ua="na">Yes</FAX_V21_Detect_Enable_2_>
<Echo_Canc_Enable_2_ ua="na">Yes</Echo_Canc_Enable_2_>
<FAX_CNG_Detect_Enable_2_ ua="na">Yes</FAX_CNG_Detect_Enable_2_>
<FAX_Passthru_Codec_2_ ua="na">G711u</FAX_Passthru_Codec_2_>
<FAX_Codec_Symmetric_2_ ua="na">Yes</FAX_Codec_Symmetric_2_>
<DTMF_Process_INFO_2_ ua="na">Yes</DTMF_Process_INFO_2_>
<FAX_Passthru_Method_2_ ua="na">ReINVITE</FAX_Passthru_Method_2_>
<DTMF_Process_AVT_2_ ua="na">Yes</DTMF_Process_AVT_2_>
<FAX_Process_NSE_2_ ua="na">Yes</FAX_Process_NSE_2_>
<DTMF_Tx_Method_2_ ua="na">Auto</DTMF_Tx_Method_2_>
<FAX_Disable_ECAN_2_ ua="na">No</FAX_Disable_ECAN_2_>
<DTMF_Tx_Mode_2_ ua="na">Strict</DTMF_Tx_Mode_2_>
<DTMF_Tx_Strict_Hold_Off_Time_2_ ua="na">70</DTMF_Tx_Strict_Hold_Off_Time_2_>
<FAX_Enable_T38_2_ ua="na">Yes</FAX_Enable_T38_2_>
<Hook_Flash_Tx_Method_2_ ua="na">None</Hook_Flash_Tx_Method_2_>
<FAX_T38_Redundancy_2_ ua="na">1</FAX_T38_Redundancy_2_>
<FAX_T38_ECM_Enable_2_ ua="na">Yes</FAX_T38_ECM_Enable_2_>
<FAX_Tone_Detect_Mode_2_ ua="na">caller or callee</FAX_Tone_Detect_Mode_2_>
<Symmetric_RTP_2_ ua="na">No</Symmetric_RTP_2_>
<FAX_T38_Return_to_Voice_2_ ua="na">No</FAX_T38_Return_to_Voice_2_>
<Modem_Line_2_ ua="na">No</Modem_Line_2_>
<RTP_to_Proxy_in_Remote_Hold_2_ ua="na">No</RTP_to_Proxy_in_Remote_Hold_2_>
<!--  Dial Plan  -->
<Dial_Plan_2_
ua="na">(*xx|[3469]11|0|00|[2-9]xxxxxx|1xxx[2-9]xxxxxxS0|xxxxxxxxxxxx.)</Dial_Plan_2_>
<!--  FXS Port Polarity Configuration  -->
<Idle_Polarity_2_ ua="na">Forward</Idle_Polarity_2_>
<Caller_Conn_Polarity_2_ ua="na">Forward</Caller_Conn_Polarity_2_>
<Callee_Conn_Polarity_2_ ua="na">Forward</Callee_Conn_Polarity_2_>
<Profile_Stamp_2_ ua="na"/>
<!--  Parameters for User 1 Tab  -->
<!--  Call Forward Settings  -->
<Cfwd_All_Dest_1_ ua="rw"/>
<Cfwd_Busy_Dest_1_ ua="rw"/>
<Cfwd_No_Ans_Dest_1_ ua="rw"/>
<Cfwd_No_Ans_Delay_1_ ua="rw">20</Cfwd_No_Ans_Delay_1_>
<!--  Selective Call Forward Settings  -->
<Cfwd_Sel1_Caller_1_ ua="rw"/>
<Cfwd_Sel1_Dest_1_ ua="rw"/>
<Cfwd_Sel2_Caller_1_ ua="rw"/>
<Cfwd_Sel2_Dest_1_ ua="rw"/>
<Cfwd_Sel3_Caller_1_ ua="rw"/>
<Cfwd_Sel3_Dest_1_ ua="rw"/>
<Cfwd_Sel4_Caller_1_ ua="rw"/>
<Cfwd_Sel4_Dest_1_ ua="rw"/>
<Cfwd_Sel5_Caller_1_ ua="rw"/>
<Cfwd_Sel5_Dest_1_ ua="rw"/>
<Cfwd_Sel6_Caller_1_ ua="rw"/>
<Cfwd_Sel6_Dest_1_ ua="rw"/>
<Cfwd_Sel7_Caller_1_ ua="rw"/>
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<Cfwd_Sel7_Dest_1_ ua="rw"/>
<Cfwd_Sel8_Caller_1_ ua="rw"/>
<Cfwd_Sel8_Dest_1_ ua="rw"/>
<Cfwd_Last_Caller_1_ ua="rw"/>
<Cfwd_Last_Dest_1_ ua="rw"/>
<Block_Last_Caller_1_ ua="rw"/>
<Accept_Last_Caller_1_ ua="rw"/>
<!--  Speed Dial Settings  -->
<Speed_Dial_2_1_ ua="rw"/>
<Speed_Dial_3_1_ ua="rw"/>
<Speed_Dial_4_1_ ua="rw"/>
<Speed_Dial_5_1_ ua="rw"/>
<Speed_Dial_6_1_ ua="rw"/>
<Speed_Dial_7_1_ ua="rw"/>
<Speed_Dial_8_1_ ua="rw"/>
<Speed_Dial_9_1_ ua="rw"/>
<!--  Supplementary Service Settings  -->
<CW_Setting_1_ ua="rw">Yes</CW_Setting_1_>
<Block_CID_Setting_1_ ua="rw">No</Block_CID_Setting_1_>
<Block_ANC_Setting_1_ ua="rw">No</Block_ANC_Setting_1_>
<DND_Setting_1_ ua="rw">No</DND_Setting_1_>
<CID_Setting_1_ ua="rw">Yes</CID_Setting_1_>
<CWCID_Setting_1_ ua="rw">Yes</CWCID_Setting_1_>
<Dist_Ring_Setting_1_ ua="rw">Yes</Dist_Ring_Setting_1_>
<Secure_Call_Setting_1_ ua="na">No</Secure_Call_Setting_1_>
<Message_Waiting_1_ ua="rw">No</Message_Waiting_1_>
<Accept_Media_Loopback_Request_1_ ua="na">automatic</Accept_Media_Loopback_Request_1_>
<Media_Loopback_Mode_1_ ua="na">source</Media_Loopback_Mode_1_>
<Media_Loopback_Type_1_ ua="na">media</Media_Loopback_Type_1_>
<CONFCID_Setting_1_ ua="rw">Yes</CONFCID_Setting_1_>
<!--  Distinctive Ring Settings  -->
<Ring1_Caller_1_ ua="rw"/>
<Ring2_Caller_1_ ua="rw"/>
<Ring3_Caller_1_ ua="rw"/>
<Ring4_Caller_1_ ua="rw"/>
<Ring5_Caller_1_ ua="rw"/>
<Ring6_Caller_1_ ua="rw"/>
<Ring7_Caller_1_ ua="rw"/>
<Ring8_Caller_1_ ua="rw"/>
<!--  Ring Settings  -->
<Default_Ring_1_ ua="rw">1</Default_Ring_1_>
<Default_CWT_1_ ua="rw">1</Default_CWT_1_>
<Hold_Reminder_Ring_1_ ua="rw">8</Hold_Reminder_Ring_1_>
<Call_Back_Ring_1_ ua="rw">7</Call_Back_Ring_1_>
<Cfwd_Ring_Splash_Len_1_ ua="rw">0</Cfwd_Ring_Splash_Len_1_>
<Cblk_Ring_Splash_Len_1_ ua="rw">0</Cblk_Ring_Splash_Len_1_>
<VMWI_Ring_Policy_1_ ua="na">New VM Available</VMWI_Ring_Policy_1_>
<VMWI_Ring_Splash_Len_1_ ua="rw">0</VMWI_Ring_Splash_Len_1_>
<Ring_On_No_New_VM_1_ ua="na">No</Ring_On_No_New_VM_1_>
<!--  Parameters for User 2 Tab  -->
<!--  Call Forward Settings  -->
<Cfwd_All_Dest_2_ ua="rw"/>
<Cfwd_Busy_Dest_2_ ua="rw"/>
<Cfwd_No_Ans_Dest_2_ ua="rw"/>
<Cfwd_No_Ans_Delay_2_ ua="rw">20</Cfwd_No_Ans_Delay_2_>
<!--  Selective Call Forward Settings  -->
<Cfwd_Sel1_Caller_2_ ua="rw"/>
<Cfwd_Sel1_Dest_2_ ua="rw"/>
<Cfwd_Sel2_Caller_2_ ua="rw"/>
<Cfwd_Sel2_Dest_2_ ua="rw"/>
<Cfwd_Sel3_Caller_2_ ua="rw"/>
<Cfwd_Sel3_Dest_2_ ua="rw"/>
<Cfwd_Sel4_Caller_2_ ua="rw"/>
<Cfwd_Sel4_Dest_2_ ua="rw"/>
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<Cfwd_Sel5_Caller_2_ ua="rw"/>
<Cfwd_Sel5_Dest_2_ ua="rw"/>
<Cfwd_Sel6_Caller_2_ ua="rw"/>
<Cfwd_Sel6_Dest_2_ ua="rw"/>
<Cfwd_Sel7_Caller_2_ ua="rw"/>
<Cfwd_Sel7_Dest_2_ ua="rw"/>
<Cfwd_Sel8_Caller_2_ ua="rw"/>
<Cfwd_Sel8_Dest_2_ ua="rw"/>
<Cfwd_Last_Caller_2_ ua="rw"/>
<Cfwd_Last_Dest_2_ ua="rw"/>
<Block_Last_Caller_2_ ua="rw"/>
<Accept_Last_Caller_2_ ua="rw"/>
<!--  Speed Dial Settings  -->
<Speed_Dial_2_2_ ua="rw"/>
<Speed_Dial_3_2_ ua="rw"/>
<Speed_Dial_4_2_ ua="rw"/>
<Speed_Dial_5_2_ ua="rw"/>
<Speed_Dial_6_2_ ua="rw"/>
<Speed_Dial_7_2_ ua="rw"/>
<Speed_Dial_8_2_ ua="rw"/>
<Speed_Dial_9_2_ ua="rw"/>
<!--  Supplementary Service Settings  -->
<CW_Setting_2_ ua="rw">Yes</CW_Setting_2_>
<Block_CID_Setting_2_ ua="rw">No</Block_CID_Setting_2_>
<Block_ANC_Setting_2_ ua="rw">No</Block_ANC_Setting_2_>
<DND_Setting_2_ ua="rw">No</DND_Setting_2_>
<CID_Setting_2_ ua="rw">Yes</CID_Setting_2_>
<CWCID_Setting_2_ ua="rw">Yes</CWCID_Setting_2_>
<Dist_Ring_Setting_2_ ua="rw">Yes</Dist_Ring_Setting_2_>
<Secure_Call_Setting_2_ ua="na">No</Secure_Call_Setting_2_>
<Message_Waiting_2_ ua="rw">Yes</Message_Waiting_2_>
<Accept_Media_Loopback_Request_2_ ua="na">automatic</Accept_Media_Loopback_Request_2_>
<Media_Loopback_Mode_2_ ua="na">source</Media_Loopback_Mode_2_>
<Media_Loopback_Type_2_ ua="na">media</Media_Loopback_Type_2_>
<CONFCID_Setting_2_ ua="rw">Yes</CONFCID_Setting_2_>
<!--  Distinctive Ring Settings  -->
<Ring1_Caller_2_ ua="rw"/>
<Ring2_Caller_2_ ua="rw"/>
<Ring3_Caller_2_ ua="rw"/>
<Ring4_Caller_2_ ua="rw"/>
<Ring5_Caller_2_ ua="rw"/>
<Ring6_Caller_2_ ua="rw"/>
<Ring7_Caller_2_ ua="rw"/>
<Ring8_Caller_2_ ua="rw"/>
<!--  Ring Settings  -->
<Default_Ring_2_ ua="rw">1</Default_Ring_2_>
<Default_CWT_2_ ua="rw">1</Default_CWT_2_>
<Hold_Reminder_Ring_2_ ua="rw">8</Hold_Reminder_Ring_2_>
<Call_Back_Ring_2_ ua="rw">7</Call_Back_Ring_2_>
<Cfwd_Ring_Splash_Len_2_ ua="rw">0</Cfwd_Ring_Splash_Len_2_>
<Cblk_Ring_Splash_Len_2_ ua="rw">0</Cblk_Ring_Splash_Len_2_>
<VMWI_Ring_Policy_2_ ua="na">New VM Available</VMWI_Ring_Policy_2_>
<VMWI_Ring_Splash_Len_2_ ua="rw">0</VMWI_Ring_Splash_Len_2_>
<Ring_On_No_New_VM_2_ ua="na">No</Ring_On_No_New_VM_2_>
<!--  Parameters for Regional Tab  -->
<!--  Call Progress Tones  -->
<Dial_Tone ua="na">350@-19,440@-19;10(*/0/1+2)</Dial_Tone>
<Second_Dial_Tone ua="na">420@-19,520@-19;10(*/0/1+2)</Second_Dial_Tone>
<Outside_Dial_Tone ua="na">420@-16;10(*/0/1)</Outside_Dial_Tone>
<Prompt_Tone ua="na">520@-19,620@-19;10(*/0/1+2)</Prompt_Tone>
<Busy_Tone ua="na">480@-19,620@-19;10(.5/.5/1+2)</Busy_Tone>
<Reorder_Tone ua="na">480@-19,620@-19;10(.25/.25/1+2)</Reorder_Tone>
<Off_Hook_Warning_Tone ua="na">480@-10,620@0;10(.125/.125/1+2)</Off_Hook_Warning_Tone>
<Ring_Back_Tone ua="na">440@-19,480@-19;*(2/4/1+2)</Ring_Back_Tone>
```

```
<Ring_Back_2_Tone ua="na">440@-19,480@-19;*(1/1/1+2)</Ring_Back_2_Tone>
<Confirm_Tone ua="na">600@-16;1(.25/.25/1)</Confirm_Tone>
<SIT1_Tone ua="na">985@-16,1428@-16,1777@-16;20(.380/0/1,.380/0/2,.380/0/3,0/4/0)</SIT1_Tone>
<SIT2_Tone ua="na">914@-16,1371@-16,1777@-16;20(.274/0/1,.274/0/2,.380/0/3,0/4/0)</SIT2_Tone>
<SIT3_Tone ua="na">914@-16,1371@-16,1777@-16;20(.380/0/1,.380/0/2,.380/0/3,0/4/0)</SIT3_Tone>
<SIT4_Tone ua="na">985@-16,1371@-16,1777@-16;20(.380/0/1,.274/0/2,.380/0/3,0/4/0)</SIT4_Tone>
<MWI_Dial_Tone ua="na">350@-19,440@-19;2(.1/.1/1+2);10(*/0/1+2)</MWI_Dial_Tone>
<Cfwd_Dial_Tone ua="na">350@-19,440@-19;2(.2/.2/1+2);10(*/0/1+2)</Cfwd_Dial_Tone>
<Holding_Tone ua="na">600@-19;*(.1/.1/1,.1/.1/1,.1/9.5/1)</Holding_Tone>
<Conference_Tone ua="na">350@-19;20(.1/.1/1,.1/9.7/1)</Conference_Tone>
<Secure_Call_Indication_Tone
ua="na">397@-19,507@-19;15(0/2/0,.2/.1/1,.1/2.1/2)</Secure_Call_Indication_Tone>
<Feature_Invocation_Tone ua="na">350@-16;*(.1/.1/1)</Feature_Invocation_Tone>
<Call_Remind_Tone ua="na"/>
<!--  Distinctive Ring Patterns  -->
<Ring1_Cadence ua="na">60(2/4)</Ring1_Cadence>
<Ring2_Cadence ua="na">60(.8/.4,.8/4)</Ring2_Cadence>
<Ring3_Cadence ua="na">60(.4/.2,.4/.2,.8/4)</Ring3_Cadence>
<Ring4_Cadence ua="na">60(.3/.2,1/.2,.3/4)</Ring4_Cadence>
<Ring5_Cadence ua="na">1(.5/.5)</Ring5_Cadence>
<Ring6_Cadence ua="na">60(.2/.4,.2/.4,.2/4)</Ring6_Cadence>
<Ring7_Cadence ua="na">60(.4/.2,.4/.2,.4/4)</Ring7_Cadence>
<Ring8_Cadence ua="na">60(0.25/9.75)</Ring8_Cadence>
<!--  Distinctive Call Waiting Tone Patterns  -->
<CWT1_Cadence ua="na">*(.3/9.7)</CWT1_Cadence>
<CWT2_Cadence ua="na">30(.1/.1, .1/9.7)</CWT2_Cadence>
<CWT3_Cadence ua="na">30(.1/.1, .1/.1, .1/9.7)</CWT3_Cadence>
<CWT4_Cadence ua="na">30(.1/.1,.3/.1,.1/9.3)</CWT4_Cadence>
<CWT5_Cadence ua="na">1(.5/.5)</CWT5_Cadence>
<CWT6_Cadence ua="na">30(.1/.1,.3/.2,.3/9.1)</CWT6_Cadence>
<CWT7_Cadence ua="na">30(.3/.1,.3/.1,.1/9.1)</CWT7_Cadence>
<CWT8_Cadence ua="na">2.3(.3/2)</CWT8_Cadence>
<!--  Distinctive Ring/CWT Pattern Names  -->
<Ring1_Name ua="na">Bellcore-r1</Ring1_Name>
<Ring2_Name ua="na">Bellcore-r2</Ring2_Name>
<Ring3_Name ua="na">Bellcore-r3</Ring3_Name>
<Ring4_Name ua="na">Bellcore-r4</Ring4_Name>
<Ring5_Name ua="na">Bellcore-r5</Ring5_Name>
<Ring6_Name ua="na">Bellcore-r6</Ring6_Name>
<Ring7_Name ua="na">Bellcore-r7</Ring7_Name>
<Ring8_Name ua="na">Bellcore-r8</Ring8_Name>
<!--  Ring and Call Waiting Tone Spec  -->
<Ring_Waveform ua="na">Trapezoid</Ring_Waveform>
<Ring_Frequency ua="na">20</Ring_Frequency>
<Ring_Voltage ua="na">85</Ring_Voltage>
<CWT_Frequency ua="na">440@-10</CWT_Frequency>
<Synchronized_Ring ua="na">No</Synchronized_Ring>
<!--  Control Timer Values (sec)  -->
<Hook_Flash_Timer_Min ua="na">.1</Hook_Flash_Timer_Min>
<Hook_Flash_Timer_Max ua="na">.9</Hook_Flash_Timer_Max>
<Callee_On_Hook_Delay ua="na">0</Callee_On_Hook_Delay>
<Reorder_Delay ua="na">5</Reorder_Delay>
<Call_Back_Expires ua="na">1800</Call_Back_Expires>
<Call_Back_Retry_Intvl ua="na">30</Call_Back_Retry_Intvl>
<Call_Back_Delay ua="na">.5</Call_Back_Delay>
<VMWI_Refresh_Intvl ua="na">0</VMWI_Refresh_Intvl>
<Interdigit_Long_Timer ua="na">10</Interdigit_Long_Timer>
<Interdigit_Short_Timer ua="na">3</Interdigit_Short_Timer>
<CPC_Delay ua="na">2</CPC_Delay>
<CPC_Duration ua="na">.5</CPC_Duration>
<!--  Vertical Service Activation Codes  -->
<Call_Return_Code ua="na">*69</Call_Return_Code>
<Call_Redial_Code ua="na">*07</Call_Redial_Code>
<Blind_Transfer_Code ua="na">*98</Blind_Transfer_Code>
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<Call_Back_Deact_Code ua="na">*86</Call_Back_Deact_Code>
<Call_Back_Busy_Act_Code ua="na">*05</Call_Back_Busy_Act_Code>
<Cfwd_All_Act_Code ua="na">*72</Cfwd_All_Act_Code>
<Cfwd_All_Deact_Code ua="na">*73</Cfwd_All_Deact_Code>
<Cfwd_Busy_Act_Code ua="na">*90</Cfwd_Busy_Act_Code>
<Cfwd_Busy_Deact_Code ua="na">*91</Cfwd_Busy_Deact_Code>
<Cfwd_No_Ans_Act_Code ua="na">*92</Cfwd_No_Ans_Act_Code>
<Cfwd_No_Ans_Deact_Code ua="na">*93</Cfwd_No_Ans_Deact_Code>
<Cfwd_Last_Act_Code ua="na">*63</Cfwd_Last_Act_Code>
<Cfwd_Last_Deact_Code ua="na">*83</Cfwd_Last_Deact_Code>
<Block_Last_Act_Code ua="na">*60</Block_Last_Act_Code>
<Block_Last_Deact_Code ua="na">*80</Block_Last_Deact_Code>
<Accept_Last_Act_Code ua="na">*64</Accept_Last_Act_Code>
<Accept_Last_Deact_Code ua="na">*84</Accept_Last_Deact_Code>
<CW_Act_Code ua="na">*56</CW_Act_Code>
<CW_Deact_Code ua="na">*57</CW_Deact_Code>
<CW_Per_Call_Act_Code ua="na">*71</CW_Per_Call_Act_Code>
<CW_Per_Call_Deact_Code ua="na">*70</CW_Per_Call_Deact_Code>
<Block_CID_Act_Code ua="na">*67</Block_CID_Act_Code>
<Block_CID_Deact_Code ua="na">*68</Block_CID_Deact_Code>
<Block_CID_Per_Call_Act_Code ua="na">*81</Block_CID_Per_Call_Act_Code>
<Block_CID_Per_Call_Deact_Code ua="na">*82</Block_CID_Per_Call_Deact_Code>
<Block_ANC_Act_Code ua="na">*77</Block_ANC_Act_Code>
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<DND_Deact_Code ua="na">*79</DND_Deact_Code>
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<CID_Deact_Code ua="na">*85</CID_Deact_Code>
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<CWCID_Deact_Code ua="na">*45</CWCID_Deact_Code>
<Dist_Ring_Act_Code ua="na">*26</Dist_Ring_Act_Code>
<Dist_Ring_Deact_Code ua="na">*46</Dist_Ring_Deact_Code>
<Speed_Dial_Act_Code ua="na">*74</Speed_Dial_Act_Code>
<Paging_Code ua="na">*96</Paging_Code>
<Secure_All_Call_Act_Code ua="na">*16</Secure_All_Call_Act_Code>
<Secure_No_Call_Act_Code ua="na">*17</Secure_No_Call_Act_Code>
<Secure_One_Call_Act_Code ua="na">*18</Secure_One_Call_Act_Code>
<Secure_One_Call_Deact_Code ua="na">*19</Secure_One_Call_Deact_Code>
<Conference_Act_Code ua="na"/>
<Attn-Xfer_Act_Code ua="na"/>
<Modem_Line_Toggle_Code ua="na">*99</Modem_Line_Toggle_Code>
<FAX_Line_Toggle_Code ua="na">#99</FAX_Line_Toggle_Code>
<Media_Loopback_Code ua="na">*03</Media_Loopback_Code>
<Referral_Services_Codes ua="na"/>
<Feature_Dial_Services_Codes ua="na"/>
<!--  Vertical Service Announcement Codes  -->
<Service_Annc_Base_Number ua="na"/>
<Service_Annc_Extension_Codes ua="na"/>
<!--  Outbound Call Codec Selection Codes  -->
<Prefer_G711u_Code ua="na">*017110</Prefer_G711u_Code>
<Force_G711u_Code ua="na">*027110</Force_G711u_Code>
<Prefer_G711a_Code ua="na">*017111</Prefer_G711a_Code>
<Force_G711a_Code ua="na">*027111</Force_G711a_Code>
<Prefer_G726r32_Code ua="na">*0172632</Prefer_G726r32_Code>
<Force_G726r32_Code ua="na">*0272632</Force_G726r32_Code>
<Prefer_G729a_Code ua="na">*01729</Prefer_G729a_Code>
<Force_G729a_Code ua="na">*02729</Force_G729a_Code>
<!--  Miscellaneous  -->
<FXS_Port_Impedance ua="na">600</FXS_Port_Impedance>
<FXS_Port_Input_Gain ua="na">-3</FXS_Port_Input_Gain>
<FXS_Port_Output_Gain ua="na">-3</FXS_Port_Output_Gain>
<DTMF_Playback_Level ua="na">-16</DTMF_Playback_Level>
<DTMF_Twist ua="na">2</DTMF_Twist>
```

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<DTMF_Playback_Length ua="na">.1</DTMF_Playback_Length>
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<Caller_ID_FSK_Standard ua="na">bell 202</Caller_ID_FSK_Standard>
<Feature_Invocation_Method ua="na">Default</Feature_Invocation_Method>
<!--  DMZ Settings  -->
<!--  Miscellaneous Settings  -->
<!--  System Reserved Ports Range  -->
<Protect_IVR_FactoryReset ua="na">No</Protect_IVR_FactoryReset>
<Max_Session ua="na">2</Max_Session>
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<WAN_Stack_Mode>0</WAN_Stack_Mode>
<!--  options: 0:IPv4 Only, 1:IPv6 Only, 2:Dual  -->
<WAN_Signal_Preference>0</WAN_Signal_Preference>
<!--  options: 0:IPv4, 1:IPv6  -->
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<!--  options: 0:IPv4, 1:IPv6  -->
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<WAN_DHCP_MTU_Size>0</WAN_DHCP_MTU_Size>
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<!--  <WAN_PPPoE_Password></WAN_PPPoE_Password>  -->
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<WAN_IP6_Allow_AutoConfig>1</WAN_IP6_Allow_AutoConfig>
<WAN_IP6_Connection_Type>0</WAN_IP6_Connection_Type>
<!--  options: 0:DHCPv6, 1:Static, 2:PPPoEv6  -->
<WAN_Static_IP6_Address/>
<WAN_Static_IP6_Prefix_Length>64</WAN_Static_IP6_Prefix_Length>
<WAN_Static_IP6_Gatway/>
</WAN_IP6_Setting>
<PHY_Port_Setting>
<Flow_Control>1</Flow_Control>
<Speed_Duplex>auto</Speed_Duplex>
<!--  options: auto/10h/10f/100h/100f  -->
</PHY_Port_Setting>
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</MAC_Address_Clone>
<Internet_Option>
<Host_Name>ATA191-MPP</Host_Name>
<Domain_Name/>
<DNS_Order>2</DNS_Order>
<!--  options: 0:Manual, 1:Manual-DHCP, 2:DHCP-Manual  -->
<DNS/>
<DNS6_Order>2</DNS6_Order>
<!--  options: 0:Manual, 1:Manual-DHCP, 2:DHCP-Manual  -->
<DNS6/>
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<DHCP_Server_Pool>
<Rule>
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<DHCP_Server>1</DHCP_Server>
<Local_IP>192.168.15.1</Local_IP>
<Subnet_Mask>255.255.255.0</Subnet_Mask>
<!--  options: 255.255.255.0/128/192/224/240/248/252  -->
<DHCP_Client_Table/>
<Option_66>0</Option_66>
<!--  options: 0:None, 2:Remote TFTP Server, 3:Manual TFTP Server  -->
<TFTP_IP>0.0.0.0</TFTP_IP>
<Option_67/>
<Option_159/>
<Option_160/>
<DNS_Proxy>1</DNS_Proxy>
<Starting_IP>192.168.15.100</Starting_IP>
<Max_DHCP_User>50</Max_DHCP_User>
<Client_Lease_Time>0</Client_Lease_Time>
<Static_DNS>0.0.0.0</Static_DNS>
<Default_Gateway>192.168.15.1</Default_Gateway>
</Rule>
</DHCP_Server_Pool>
<LAN_IP6_Setting>
<LAN_IP6_Address_Assign_Type>0</LAN_IP6_Address_Assign_Type>
<!--  options: 0:SLAAC, 1:DHCPV6  -->
<LAN_DHCP6_Delegation_Enable>0</LAN_DHCP6_Delegation_Enable>
<LAN_IP6_Prefix>2001::</LAN_IP6_Prefix>
</LAN_IP6_Setting>
<WAN_VLAN_Setting>
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<WAN_VLAN_ID>3</WAN_VLAN_ID>
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<CDP_ENABLE>1</CDP_ENABLE>
<LLDP_ENABLE>1</LLDP_ENABLE>
<LAYER2_LOGGING_ENABLE>0</LAYER2_LOGGING_ENABLE>
</CLDP_Setting>
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<SNMP_Trusted_IP6>::</SNMP_Trusted_IP6>
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<Get_Community>public</Get_Community>
<Set_Community>private</Set_Community>
<SNMPV3>0</SNMPV3>
<RW_User>v3rwuser</RW_User>
<Auth_Protocol>MD5</Auth_Protocol>
<!--  options: MD5/SHA  -->
<!--  <Auth_Password></Auth_Password>  -->
<Privacy_Protocol>DES</Privacy_Protocol>
<!--  options: None/DES/AES  -->
<!--  <Privacy_Password></Privacy_Password>  -->
<TRAP_IP_Address>192.168.15.100</TRAP_IP_Address>
<TRAP_Port>162</TRAP_Port>
<TRAP_SNMP_Version>v1</TRAP_SNMP_Version>
<!--  options: v1/v2c/v3  -->
</SNMP>
<Time_Setup>
<Time_Zone>-08 1 1</Time_Zone>
<Auto_Adjust_Clock>1</Auto_Adjust_Clock>
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<!--  options: auto/manual  -->
<Time_Server>0.ciscosb.pool.ntp.org</Time_Server>
<Resync_Timer>3600</Resync_Timer>
<Auto_Recovery_System_Time>0</Auto_Recovery_System_Time>
<Time_Mode>1</Time_Mode>
<!--  options: 0:Manual, 1:Auto  -->
</Time_Setup>
<QoS_Bandwidth_Control/>
<HTTP_Proxy>
<Proxy_Mode>Off</Proxy_Mode>
<!--  options: Off, Auto, Manual  -->
</HTTP_Proxy>
<Software_DMZ>
<Rule1>
<Status>0</Status>
<Private_IP>0.0.0.0</Private_IP>
</Rule1>
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</Software_DMZ>
<Bonjour_Enable>1</Bonjour_Enable>
<Reset_Button_Enable>1</Reset_Button_Enable>
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<Monitor_WAN_Port_Only>0</Monitor_WAN_Port_Only>
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<L2TP_Passthrough>1</L2TP_Passthrough>
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<Admin_Access>1</Admin_Access>
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<Web_Remote_Management>0</Web_Remote_Management>
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<Web_Remote_Upgrade>0</Web_Remote_Upgrade>
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</Web_Management>
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<TR_069_Connection_Request_URL/>
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<TR_069_Periodic_Inform_Enable>1</TR_069_Periodic_Inform_Enable>
<TR_069_Loopback_Binding>0</TR_069_Loopback_Binding>
</TR_069>
<Log_Configuration>
<Log_Module>2,3,4,5,6,7,8,9,10,11,12,13,14</Log_Module>
<!--  options: 0:Default, 1:Preset, 2:Telephony, 3:SIP, 4:UI, 5:Network, 6:Media, 7:System,
 8:Web, 9:NTP, 10:CDP/LLDP, 11:Security, 12:CSSD_RTP, 13:CSSD_FAX, 14:CSSD_ANY  -->
<RAM_Log_Size>512</RAM_Log_Size>
<Syslog_Server_IP/>
<Syslog_Server_IP6/>
<Syslog_Server_Port>514</Syslog_Server_Port>
<Event_Log_Server/>
<Event_Log_Port>514</Event_Log_Port>
<Event_Log_Flag>15</Event_Log_Flag>
<!--  options: 0:Disable, 1:DEV, 2:SYS, 4:CFG, 8:REG, Default:15(DEV+SYS+CFG+REG)  -->
<PRT_Upload_Url/>
```

```
<PRT_Upload_Method>0</PRT_Upload_Method>
<!--  options: 0:POST, 1:PUT  -->
<PRT_Max_Timer>0</PRT_Max_Timer>
<PRT_Name/>
</Log_Configuration>
<Web_Login_Admin_Name>admin</Web_Login_Admin_Name>
<!--  <Web_Login_Admin_Password></Web_Login_Admin_Password>  -->
<Web_Login_Guest_Name>cisco</Web_Login_Guest_Name>
<!--  <Web_Login_Guest_Password></Web_Login_Guest_Password>  -->
<SSH>
<SSH_ACCESS>0</SSH_ACCESS>
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</SSH>
</router-configuration>
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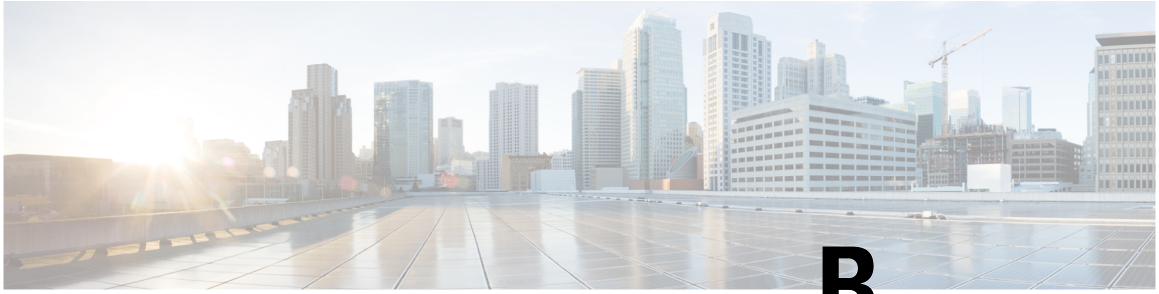

APPENDIX B

Acronyms

Acronyms

AC	Alternating Current
ACS	Access Control Server
A/D	Analog To Digital Converter
AES	Advanced Encryption Standard
ANC	Anonymous Call
AP	Access Point
ASCII	American Standard Code for Information Interchange
B2BUA	Back to Back User Agent
BLF	Busy Lamp Field
Bool	Boolean Values. Specified as yes and no, or 1 and 0 in the profile
BootP	Bootstrap Protocol
CA	Certificate Authority
CAS	CPE Alert Signal
CDP	Cisco Discovery Protocol
CDR	Call Detail Record
CGI	Computer-Generated Mmagery
CID	Caller ID
CIDCW	Call Waiting Caller ID

CNG	Comfort Noise Generation
CPC	Calling Party Control
CPE	Customer Premises Equipment
CSV	Comma separated value
CWCID	Call Waiting Caller ID
CWT	Call Waiting Tone
D/A	Digital to Analog Converter
dB	decibel
dBm	dB with respect to 1 milliwatt
DHCP	Dynamic Host Configuration Protocol
DND	Do not disturb
DNS	Domain Name System
DoS	Denial of service
DRAM	Dynamic Random Access Memory
DSL	Digital Subscriber Loop
DSP	Digital Signal Processor
DST	Daylight Saving Time
DTAS	Data Terminal Alert Signal (same as CAS)
DTMF	Dual Tone Multiple Frequency
FQDN	Fully Qualified Domain Name
FSK	Frequency Shift Keying
FW	Firmware
FXS	Foreign eXchange Station
GMT	Greenwich Mean Time
GW	Gateway
HTML	Hypertext Markup Language
HTTP	Hypertext Transfer Protocol
HTTPS	HTTP over SSL
ICMP	Internet Control Message Protocol

IGMP	Internet Group Management Protocol
ILEC	Incumbent Local Exchange Carrier
IP	Internet Protocol
IPv4	Internet Protocol version 4
IPv6	Internet Protocol version 6
ISP	Internet Service Provider
ITSP	Internet Telephony Service Provider
ITU	International Telecommunication Union
IVR	Interactive Voice Response
LAN	Local Area Network
LBR	Low Bit Rate
LBRC	Low Bit Rate Codec
LCD	Liquid Crystal Display; also known as a screen
LDAP	Lightweight Directory Access Protocol
LED	Light-Emiting Diode
MAC address	Media Access Control Address
MC	Mini-Certificate
MGCP	Media Gateway Control Protocol
MOH	Music On Hold
MOS	Mean Opinion Score (1-5, the higher the better)
MPP	Multiplatform Phones
ms	Millisecond
MSA	Music Source Adaptor
MWI	Message Waiting Indication
NAT	Network Address Translation
NPS	Normal Provisioning Server
NTP	Network Time Protocol
OOB	Out-of-band
OSI	Open Switching Interval

PBX	Private branch exchange
PCB	Printed Circuit Board
PoE	Power over Ethernet
PR	Polarity Reversal
PS	Provisioning Server
PSQM	Perceptual Speech Quality Measurement (1-5, the lower the better)
PSTN	Public Switched Telephone Network
QoS	Quality of service
RC	Remove Customization
REQT	(SIP) Request Message
RESP	(SIP) Response Message
RSC	(SIP) Response Status Code, such as 404, 302, 600
RTP	Real Time Protocol
RTT	Round Trip Time
SAS	Streaming Audio Server
SDP	Session Description Protocol
SDRAM	Synchronous DRAM
sec	seconds
SIP	Session Initiation Protocol
SLA	Shared line appearance
SLIC	Subscriber Line Interface Circuit
SP	Service Provider
SSL	Secure Socket Layer
STUN	Session Traversal UDP for NAT
TCP	Transmission Control Protocol
TFTP	Trivial File Transfer Protocol
TLS	Transport Layer Security
TTL	Time to live
ToS	Type of service

UA	User Agent
uC	Micro-controller
UDP	User Datagram Protocol
URI	Uniform Resource Identifier
URL	Uniform Resource Locator
UTC	Coordinated Universal Time
VAR	Value Added Reseller
VLAN	Voice LAN
VM	Voicemail
VMWI	Visual Message Waiting Indication/Indicator
VoIP	Voice over Internet Protocol
VQ	Voice Quality
WAN	Wide Area Network
XML	Extensible Markup Language

APPENDIX C

Time Zone Settings

Time Zone Settings

Time Zone	Setting	Example
(GMT) England	+00 2 2	<Time_Zone>+00 2 2</Time_Zone>
(GMT) Gambia, Liberia, Morocco	+00 1 0	<Time_Zone>+00 1 0</Time_Zone>
(GMT+01:00) France, Germany, Italy	+01 2 2	<Time_Zone>+01 2 2</Time_Zone>
(GMT+01:00) Tunisia	+01 1 6	<Time_Zone>+01 1 6</Time_Zone>
(GMT+02:00) Greece, Ukraine, Romania, Turkey	+02 2 2	<Time_Zone>+02 2 2</Time_Zone>
(GMT+02:00) South Africa	+02 1 0	<Time_Zone>+02 1 0</Time_Zone>
(GMT+03:00) Iraq, Kuwait	+03 1 8	<Time_Zone>+03 1 8</Time_Zone>
(GMT+03:00) Jordan	+03 2 9	<Time_Zone>+03 2 9</Time_Zone>
(GMT+04:00) ABU Dhabi, Muscat, Armenia	+04 1 0	<Time_Zone>+04 1 0</Time_Zone>
(GMT+05:00) Pakistan, Russia	+05 1 7	<Time_Zone>+05 1 7</Time_Zone>
(GMT+05:30) Bombay, Calcutta, Madras, New Delhi	+05.5 1 0	<Time_Zone>+05.5 1 0</Time_Zone>
(GMT+06:00) Bangladesh, Russia	+06 1 7	<Time_Zone>+06 1 7</Time_Zone>
(GMT+07:00) Thailand, Russia	+07 1 7	<Time_Zone>+07 1 7</Time_Zone>
(GMT+08:00) Australia Western	+08 1 4	<Time_Zone>+08 1 4</Time_Zone>
(GMT+08:00) China, Hong Kong	+08 3 0	<Time_Zone>+08 3 0</Time_Zone>

Time Zone	Setting	Example
(GMT+08:00) Russia	+08 2 7	<Time_Zone>+08 2 7</Time_Zone>
(GMT+08:00) Singapore, Taiwan	+08 4 0	<Time_Zone>+08 4 0</Time_Zone>
(GMT+09:00) Japan, Korea	+09 1 0	<Time_Zone>+09 1 0</Time_Zone>
(GMT+09:30) South Australia	+09.5 1 10	<Time_Zone>+09.5 1 10</Time_Zone>
(GMT+10:00) Australia	+10 2 4	<Time_Zone>+10 2 4</Time_Zone>
(GMT+10:00) Guam, Russia	+10 1 7	<Time_Zone>+10 1 7</Time_Zone>
(GMT+11:00) Soloman Islands	+11 1 0	<Time_Zone>+11 1 0</Time_Zone>
(GMT+12:00) Fiji	+12 1 0	<Time_Zone>+12 1 0</Time_Zone>
(GMT+12:00) Kwajalein	+12 3 0	<Time_Zone>+12 3 0</Time_Zone>
(GMT+12:00) New Zealand	+12 2 4	<Time_Zone>+12 2 4</Time_Zone>
(GMT-01:00) Azores	-01 1 2	<Time_Zone>-01 1 2</Time_Zone>
(GMT-02:00) Mid-Atlantic	-02 1 0	<Time_Zone>-02 1 0</Time_Zone>
(GMT-03:00) Brazil East, Greenland	-03 1 1	<Time_Zone>-03 1 1</Time_Zone>
(GMT-03:30) Newfoundland	–03.5 1 1	<Time_Zone>-03.5 1 1</Time_Zone>
(GMT-04:00) Atlantic Time (Canada), Brazil West	-04 2 1	<Time_Zone>-04 2 1</Time_Zone>
(GMT-04:00) Bolivia, Venezuela	-04 1 0	<Time_Zone>-04 1 0</Time_Zone>
(GMT-04:00) Guyana	-04 3 0	<Time_Zone>-04 3 0</Time_Zone>
(GMT-05:00) Eastern Time (USA & Canada)	-05 2 1	<Time_Zone>-05 2 1</Time_Zone>
(GMT-05:00) Indiana East, Columbia, Panama	-05 1 0	<Time_Zone>-05 1 0</Time_Zone>
(GMT-06:00) Central Time (USA & Canada)	-06 2 1	<Time_Zone>-06 2 1</Time_Zone>
(GMT-06:00) Mexico	-06 1 5	<Time_Zone>-06 1 5</Time_Zone>
(GMT-07:00) Arizona	-07 1 0	<Time_Zone>-07 1 0</Time_Zone>
(GMT-07:00) Mountain Time (USA & Canada)	-07 2 1	<Time_Zone>-07 2 1</Time_Zone>
(GMT-08:00) Pacific Time (USA & Canada)	-08 1 1	<Time_Zone>-08 1 1</Time_Zone>

Time Zone	Setting	Example
(GMT-09:00) Alaska	-09 1 1	<Time_Zone>-09 1 1</Time_Zone>
(GMT-10:00) Hawaii	-10 1 0	<Time_Zone>-10 1 0</Time_Zone>
(GMT-11:00) Midway Island, Samoa	-11 1 0	<Time_Zone>-11 1 0</Time_Zone>