4

Configuring the Voice Settings

This chapter describes how to configure the voice settings and voice services for the ATA. It includes the following sections:

- **Information**
- **System**
- **SIP**
- **Provisioning**
- **Regional**
- **Line 1 and Line 2 Settings (PHONE Port1 and PHONE2)**
- **User 1 and User 2**

NOTE For additional information, see **Appendix C, "Advanced Options for Voice Services."**

Information

Use the *Voice > Information* page to view information about the ATA voice application.

Enter the settings as described below. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

Product Information

Field	Description
Product Name	Model number/name.

Field	Description
Serial Number	Product serial number.
Software Version	Software version number.
Hardware Version	Hardware version number.
MAC Address	MAC Address. For example: 8843E1657936.
Client Certificate	Status of the client certificate, which can indicate if the ATA was authorized by your ITSP.
Customization	Used for Remote Configuration by service providers who deploy the ATA to their customers. ▪ **Open:** Not a Remote Configuration unit. This ATA can be configured by using the configuration utility. ▪ **Pending:** This Remote Configuration unit has not yet connected to the server to get its profile. ▪ **Customized:** This Remote Configuration unit has received its profile from the server.

System Status

Field	Description
Current Time	Current date and time of the system; for example, 10/3/2003 16:43:00. Set the system time by using the *Network Setup > Time Settings* page.
Elapsed Time	Total time elapsed since the last reboot of the system; for example, 25 days and 18:12:36.
RTP Packets Sent	Total number of RTP packets sent (including redundant packets)
RTP Bytes Sent	Total number of RTP bytes sent.
RTP Packets Recv	Total number of RTP packets received (including redundant packets)
RTP Bytes Recv	Total number of RTP bytes received.
SIP Messages Sent	Total number of SIP messages sent (including retransmissions)

Field	Description
SIP Bytes Sent	Total number of bytes of SIP messages sent (including retransmissions)
SIP Messages Recv	Total number of SIP messages received (including retransmissions)
SIP Bytes Recv	Total number of bytes of SIP messages received (including retransmissions)
External IP	The External IP address used for NAT mapping.

Line 1/Line 2 Status

Field	Description
Hook State	The hook state of the port: On or Off.
Registration State	Indicates if the line has registered with the SIP proxy.
Last Registration At	Last date and time the line was registered.
Next Registration In	The number of seconds before the next registration renewal. Indicates whether you have new voice mail waiting.
Message Waiting	Indicates Yes when a message is received.
Mapped SIP Port	Port number of the SIP port mapped by NAT.
Call Back Active	Indicates whether or not a call back request is in progress. Options are either **yes** or **no**.
Last Called Number	The phone number that was most recently called through this port.
Last Caller Number	The originating phone number of the call that was most recently received through this port.

Field	Description
Call 1 and 2 State	Indicates the state of calls, if any: ▪ Idle ▪ Collecting PSTN PIN ▪ Invalid PSTN PIN ▪ PSTN Caller Accepted ▪ Connected to PSTN
Call 1 and 2 Tone	The type of tone used by the call.
Call 1 and 2 Encoder	The codec used for encoding.
Call 1 and 2 Decoder	The codec used for decoding.
Call 1 and 2 FAX	The status of the fax passthrough mode.
Call 1 and 2 Type	The direction of the call. May take one of the following values: ▪ PSTN Gateway Call = VoIP-To-PSTN Call ▪ VoIP Gateway Call = PSTN-To-VoIP Call ▪ PSTN To Line 1 = PSTN call ring through and answered by Line 1 ▪ Line 1 Forward to PSTN Gateway = VoIP calls Line 1 then forwarded to PSTN GW ▪ Line 1 Forward to PSTN Number =VoIP calls Line 1 then forwarded to PSTN number ▪ Line 1 To PSTN Gateway ▪ Line 1 Fallback To PSTN Gateway
Call 1 and 2 Remote Hold	Indicates whether the far end has placed the call on hold.
Call 1 and 2 Callback	Indicates whether the call was triggered by a call back request.

Field	Description
Call 1 and 2 Peer Name	The name of the peer phone.
Call 1 and 2 Peer Phone	The phone number of the peer phone.
Call 1 and 2 Call Duration	The duration of the call.
Call 1 and 2 Packets Sent	The number of packets sent
Call 1 and 2 Packets Recv	The number of packets received.
Call 1 and 2 Bytes Sent	The number of bytes sent.
Call 1 and 2 Bytes Recv	The number of bytes received.
Call 1 and 2 Decode Latency	The number of milliseconds for decoder latency.
Call 1 and 2 Jitter	The number of milliseconds for receiver jitter
Call 1 and 2 Round Trip Delay	The number of milliseconds for delay.
Call 1 and 2 Packets Lost	The number of packets lost.
Call 1 and 2 Packet Error	The number of invalid packets received.

Custom CA Status

Field	Description
Custom CA Provisioning Status	The status of the latest custom CA (Certificate Authority) certificate download.

Field	Description
Custom CA Info	The successfully downloaded CA information, or “Not Installed” if no custom CA certificate was installed. Default setting: Not Installed

System

Use the *Voice > System* page to configure general voice system settings and to enable logging by using a syslog server. (Logging also can be configured in the *Administration > Logging* pages. For more information, see **Logging, page 114**.)

Enter the settings as described below. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

Requirements for Logging

- You need a computer that is on the same subnetwork as the ATA, to capture the log files. This computer needs to be running a syslog daemon. Enter the IP address of this computer in the Syslog Server and Debug Server fields.
- You can deploy a syslog server to receive syslog messages from the ATA, which acts as a syslog client. The syslog client device uses the syslog protocol to send messages, based on its configuration, to a syslog server. The syslog messages can be accessed by reviewing the "syslog.514.log" file which resides in the same directory as the slogsrv.exe syslog server application.
- Partners can download the Syslog Server for SPA Devices by using the link below (login required):
  www.cisco.com/en/US/partner/prod/collateral/voicesw/ps6788/phones/ps10499/syslog_server_for_spa_devices.zip

Enter the settings as described below. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

System Configuration

Field	Description
Restricted Access Domains	This feature is not currently used.
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

Miscellaneous Settings

Field	Description
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
Syslog Server	Specify the syslog server name and port. This feature specifies the server for logging ATA system information and critical events. If both Debug Server and Syslog Server are specified, Syslog messages are also logged to the Debug Server. Default setting: blank
Debug Server	The debug server name and port. This feature specifies the server for logging debug information. The level of detailed output depends on the debug level parameter setting. Default setting: blank

Field	Description
Debug Level	Determines the level of debug information that will be generated. Select 0, 1, 2, 3 or 3+Router from the drop-down list. The higher the debug level, the more debug information will be generated. Level 0 means that no information will be collected. Levels 1, 2 & 3 generate messages related to the voice ports only. Level 3+Router generates debug content for both voice and router components. Default setting: 3

SIP

Use the *Voice > SIP* page to configure SIP parameters and values.

Enter the settings as described below. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

NOTE For a deeper understanding of these fields, refer to Request for Comments (RFC) 3261.

SIP Parameters

Field	Description
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

Field	Description
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

Field	Description
Use Compact Header	Determines whether or not the ATA uses compact SIP headers in outbound SIP messages. Select yes or no from the drop-down 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 Timer Values

Field	Description
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–($2^{31}-1$) 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–($2^{31}-1$) Default setting: 30

Field	Description
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 feature.

Response Status Code Handling

Field	Description
SIT1 RSC	SIP response status code for the appropriate Special Information Tone (SIT) 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
SIT2 RSC	SIP response status code to INVITE on which to play the SIT2 Tone. Default setting: blank
SIT3 RSC	SIP response status code to INVITE on which to play the SIT3 Tone. Default setting: blank
SIT4 RSC	SIP response status code to INVITE on which to play the SIT4 Tone. 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 Parameters

Field	Description
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.

Field	Description
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/Hardware-platform-software-version. 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

Field	Description
Stats In BYE	Determines whether the ATA includes the P-RTP-Stat header or response in a BYE message. The header contains the RTP statistics of the current call. Select yes or no from the drop-down 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 received>,PL=<packets lost>,JI=<jitter in ms>,LA=<delay in ms>,DU=<call duration ins>,EN=<encoder>,DE=<decoder>.

SDP Payload Types

Field	Description
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
G729b Dynamic Payload	G.729b dynamic payload type. The valid range is 96-127. Default setting: 99
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

Field	Description
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
G729b Codec Name	G.729b codec name used in SDP. Default setting: G729ab
EncapRTP Codec Name	EncapRTP codec name used in SDP. Default setting: encaprtp

NAT Support Parameters

Field	Description
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

Field	Description
Insert VIA received	Inserts the received parameter into the VIA header of SIP responses if the received-from 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 parameter into the VIA header of SIP responses if the received-from 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 drop-down 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

Field	Description
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

Linksys Key System Parameters

Field	Description
Linksys Key System	To enable operation with the Cisco SPA9000, choose yes. Otherwise, choose no. Default setting: no
Multicast Address	The multicast address for devices in the Cisco SPA9000 voice network. Default setting: 224.168.168.168:6061
Key System Auto Discovery	To enable auto-discovery of the Cisco SPA9000 voice system, choose yes. Otherwise, choose no. Default setting: yes

Field	Description
Key System IP Address	The IP address of the Cisco SPA9000. Default setting: blank
Force LAN Codec	If needed, specify a voice codec. Default setting: none

Provisioning

Use the *Voice > Provisioning* page to configure profiles and parameters to provision the ATA from a remote server.

Enter the settings as described below. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

Configuration Profile

Field	Description
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 pseudo-random 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)

Field	Description
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 immediately retries to sync with the provisioning server following a failed attempt. Default setting: 3600 seconds

Field	Description
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 $MA, which expands to the ATA MAC address. Default setting: /spa$PSN.cfg

Field	Description
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

Firmware Upgrade

Field	Description
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.

CA Settings

Field	Description
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: null

General Purpose Parameters

Field	Description
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 $GPP_A. Default setting: blank

Regional

Use the *Voice* > *Regional* page to localize your system with the appropriate regional settings.

Enter the settings as described below. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

Defining Ring and Cadence and Tone Scripts

To define ring and tone patterns, the ATA uses the concept of scripts. Below is information about creating Cadence Scripts (CadScripts), Frequency Scripts (FreqScripts), and Tone Scripts (ToneScripts).

NOTE Total tone length is not configurable.

CadScript

A mini-script of up to 127 characters that specifies the cadence parameters of a signal.

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* and 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 "*" represents 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
```

FreqScript

A mini-script of up to 127 characters that specifics the frequency and level parameters of a tone.

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) and L_1–L_6 are corresponding levels in dBm (with up to 1 decimal places) 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
```

ToneScript

A mini-script of up to 127 characters that specifies the frequency, level and cadence parameters of a call progress tone. 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 $f_{i,j} = n_1[+n_2]+n_3[+n_4[+n_5[+n_6]]]]$ and $1 < n_k < 6$ indicates which of the frequency components given in the FreqScript 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
```

Enter the settings as described below. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

Call ProgressTones

Field	Description
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)

Field	Description
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)

Field	Description
SIT1 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)
SIT2 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: 914@-4,1371@-4,1777@-4;20(.274/0/1,.274/0/2,.380/0/3,0/4/0)
SIT3 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: 914@-4,1371@-4,1777@-4;20(.380/0/1,.380/0/2,.380/0/3,0/4/0)
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@-4,1371@-4,1777@-4;20(.380/0/1,.274/0/2,.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)

Field	Description
Feature Invocation Tone	Played when a feature is implemented. Default setting: 350@-4;*(.1/.1/1)

Distinctive Ring Patterns

Field	Description
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)

Distinctive Call Waiting Tone Patterns

Field	Description
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)

Field	Description
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)

Distinctive Ring/CWT Pattern Names

Field	Description
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

Field	Description
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 and Call Waiting Tone Spec

IMPORTANT: Ring and Call Waiting tones do not work the same way on all phones. When setting ring tones, consider the following recommendations:

- Begin with the default Ring Waveform, Ring Frequency, and Ring Voltage.
- If your ring cadence doesn't sound right, or your phone doesn't ring, change the following settings:
  - Ring Waveform: Sinusoid
  - Ring Frequency: 25
  - Ring Voltage: 80Vc

Field	Description
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

Control Timer Values (sec)

Field	Description
Hook Flash Timer Min	Minimum on-hook time before off-hook qualifies as hook flash. Less than this the on-hook 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 on-hook event is treated as on hook (no hook-flash 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. Default setting: 0.5
VMWI Refresh Intvl	Interval between VMWI refresh to the device. Default setting: 0

Field	Description
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)

Vertical Service Activation Codes

Vertical Service Activation Codes are automatically appended to the dial-plan. There is no need to include them in dial-plan, although no harm is done if they are inclu`ded.

Field	Description
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

Field	Description
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	Disables 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

Field	Description
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

Field	Description
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

Field	Description
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 the 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

<table>
<tr><th>Field</th><th>Description</th></tr>
<tr><td>Feature Dial Services Codes</td><td>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 to 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</td></tr>
</table>

Vertical Service Announcement Codes

Field	Description
Service Annc Base Number	Base number for service announcements. Default setting: blank
Service Annc Extension Codes	Extension codes for service announcements. Default setting: blank

Outbound Call Codec Selection Codes

Field	Description
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

Field	Description
Force G722 Code	Dial prefix to make G.722 the only codec that can be used for the call. Default setting: *02722

Line 1 and Line 2 Settings (PHONE Port1 and PHONE2)

Use the *Voice > Line 1* and *Voice > Line 2* pages to configure the settings for calls through the PHONE 1 and PHONE2 ports.

or Line 2 Enter the settings as described below. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

NOTE In a configuration profile, the FXS parameters must be appended with the appropriate numeral (for example, [1] or [2]) to identify the port to which the setting applies.

Line Enable

Field	Description
Line Enable	To enable this line for service, select yes. Otherwise, select no. Default setting: yes

Streaming Audio Server (SAS)

Field	Description
SAS Enable	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

Field	Description
SAS DLG Refresh Intvl	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	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 Settings

Field	Description
NAT Mapping Enable	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	To send the configured NAT keep alive message periodically, select yes. Otherwise, select no. Default setting: no

Field	Description
NAT Keep Alive Msg	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	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

Network Settings

Field	Description
SIP ToS/DiffServ Value	TOS/DiffServ field value in UDP IP packets carrying a SIP message. Default setting: 0x68
SIP CoS Value	CoS value for SIP messages. Valid values are 0 through 7. Default setting: 3
RTP ToS/DiffServ Value	ToS/DiffServ field value in UDP IP packets carrying RTP data. Default setting: 0xb8
RTP CoS Value [0-7]	CoS value for RTP data. Valid values are 0 through 7. Default setting: 6
Network Jitter Level	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	Choose yes to enable or no to disable this feature. Default setting: yes

SIP Settings

Field	Description
SIP Transport	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. Default setting: UDP
SIP Port	Port number of the SIP message listening and transmission port. Default setting: 5060
SIP 100REL Enable	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	The external SIP port number. Default setting: blank
Auth Resync-Reboot	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	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	To use the Remote-Party-ID header instead of the From header, select yes. Otherwise, select no. Default setting: yes

Field	Description
SIP GUID	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
SIP Debug Option	SIP messages are received at or sent from the proxy listen port. This feature controls which SIP messages to log. The choices are described below. Default setting: none ▪ **none—**No logging. ▪ **1-line—**Logs the start-line only for all messages. ▪ **1-line excl. OPT—**Logs the start-line only for all messages except OPTIONS requests/responses. ▪ **1-line excl. NTFY—**Logs the start-line only for all messages except NOTIFY requests/responses. ▪ **1-line excl. REG—**Logs the start-line only for all messages except REGISTER requests/responses. ▪ **1-line excl. OPT|NTFY|REG—**Logs the start-line only for all messages except OPTIONS, NOTIFY, and REGISTER requests/responses. ▪ **full—**Logs all SIP messages in full text. ▪ **full excl. OPT—**Logs all SIP messages in full text except OPTIONS requests/responses. ▪ **full excl. NTFY—**Logs all SIP messages in full text except NOTIFY requests/responses. ▪ **full excl. REG—**Logs all SIP messages in full text except REGISTER requests/responses. ▪ **full excl. OPT|NTFY|REG—**Logs all SIP messages in full text except for OPTIONS, NOTIFY, and REGISTER requests/responses.
RTP Log Intvl	The interval for the RTP log. Default setting: 0

Field	Description
Restrict Source IP	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	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	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	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	To contact the refer-to target, select yes. Otherwise, select no. Default setting: no
Sticky 183	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	When enabled, authorization is required for initial incoming INVITE requests from the SIP proxy. Default setting: no
Reply 182 On Call Waiting	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	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	Use the local ATA IP address in the SIP FROM message. Default setting: no

Call Feature Settings

Field	Description
Blind Attn-Xfer Enable	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	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	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	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	Select the maximum number of conference call participants. The range is 3 to 10. Default setting: 3
Enable IP Dialing	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) and a, b, c, d must be between 0 and 255, and 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

Field	Description
Emergency Number	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	Enter the ID number of the mailbox for this line. Default setting: blank

Proxy and Registration

Field	Description
Proxy	SIP proxy server for all outbound requests. Default setting: blank
Outbound Proxy	SIP Outbound Proxy Server where all outbound requests are sent as the first hop. Default setting: blank
Use Outbound Proxy	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	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	Enable periodic registration with the Proxy parameter. This parameter is ignored if Proxy is not specified. Default setting: yes
Make Call Without Reg	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

Field	Description
Register Expires	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 – (2^{31} – 1) sec. Default setting: 3600
Ans Call Without Reg	Allow answering inbound calls without successful (dynamic) registration by the unit. Default setting: no
Use DNS SRV	Whether to use DNS SRV lookup for Proxy and Outbound Proxy. Default setting: no
DNS SRV Auto Prefix	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	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. (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.) Default setting: 3600
Proxy Redundancy Method	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	The URL or IP address of the voicemail server. Default setting: blank

Field	Description
Mailbox Subscribe Expires	Sets 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

Subscriber Information

Field	Description
Display Name	Display name for caller ID. Default setting: blank
User ID	User ID for this line. Default setting: blank
Password	Password for this line. Default setting: blank
Use Auth ID	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	Authentication ID for SIP authentication. Default setting: blank
Resident Online Number	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

Supplementary Service Subscription

The ATA provides native support of a large set of enhanced or supplementary services. All of these services are optional. The parameters listed in the following table are used to enable or disable a specific supplementary service. A supplementary service should be disabled if a) the user has not subscribed for it, or b) the Service Provider intends to support similar service using other means than relying on the ATA.

Field	Description
Call Waiting Serv	Enable Call Waiting Service. Default setting: yes
Block CID Serv	Enable Block Caller ID Service. Default setting: yes
Block ANC Serv	Enable Block Anonymous Calls Service Default setting: yes
Dist Ring Serv	Enable Distinctive Ringing Service Default setting: yes
Cfwd All Serv	Enable Call Forward All Service Default setting: yes
Cfwd Busy Serv	Enable Call Forward Busy Service Default setting: yes
Cfwd No Ans Serv	Enable Call Forward No Answer Service Default setting: yes
Cfwd Sel Serv	Enable Call Forward Selective Service. Configure this service in the **Selective Call Forward Settings** section. Default setting: yes
Cfwd Last Serv	Enable Forward Last Call Service Default setting: yes
Block Last Serv	Enable Block Last Call Service Default setting: yes
Accept Last Serv	Enable Accept Last Call Service Default setting: yes
DND Serv	Enable Do Not Disturb Service Default setting: yes
CID–Serv	Enable Caller ID Service Default setting: yes
CWCID Serv	Enable Call Waiting Caller ID Service Default setting: yes
Call Return Serv	Enable Call Return Service Default setting: yes

Field	Description
Call Redial Serv	Enable Call Redial Service.
Call Back Serv	Enable Call Back Service.
Three Way Call Serv	Enable Three Way Calling Service. Three Way Calling is required for Three Way Conference and Attended Transfer. Default setting: yes
Three Way Conf Serv	Enable Three Way Conference Service. Three Way Conference is required for Attended Transfer. Default setting: yes
Attn Transfer Serv	Enable Attended Call Transfer Service. Three Way Conference is required for Attended Transfer. Default setting: yes
Unattn Transfer Serv	Enable Unattended (Blind) Call Transfer Service. Default setting: yes
MWI Serv	Enable MWI Service. MWI is available only if a Voice Mail Service is set-up in the deployment. Default setting: yes
VMWI Serv	Enable VMWI Service (FSK) Default setting: yes
Speed Dial Serv	Enable Speed Dial Service. Default setting: yes
Secure Call Serv	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 Star codes are set in **Vertical Service Activation Codes**. To enable secure calling by default, without requiring a star code, set the user's Secure Call Setting to yes. See **User 1 and User 2, page 100**.
Referral Serv	Enable Referral Service. See the Referral Services Codes parameter For more information. Default setting: yes

Field	Description
Feature Dial Serv	Enable Feature Dial Service. See the Feature Dial Services Codes parameter For more information. Default setting: yes
Service Announcement Serv	Enable Service Announcement Service. Default setting: no
Reuse CID Number As Name	Use the Caller ID number as the caller name. Default settings: yes

Audio Configuration

Field	Description
G729a Enable	To enable the use of the G.729a codec at 8 kbps, select yes. Otherwise, select no. Default setting: yes
Silence Supp Enable	To enable silence suppression so that silent audio frames are not transmitted, select yes. Otherwise, select no. Default setting: no
G726-32 Enable	To enable the use of the G.726 codec at 32 kbps, select yes. Otherwise, select no. Default setting: yes
Silence Threshold	Select the appropriate setting for the threshold: high, medium, or low. Default setting: medium
FAX V21 Detect Enable	To enable detection of V21 fax tones, select yes. Otherwise, select no. Default setting: yes
Echo Canc Enable	To enable the use of the echo canceller, select yes. Otherwise, select no. Default setting: yes
FAX CNG Detect Enable	To enable detection of the fax Calling Tone (CNG), select yes. Otherwise, select no. Default setting: yes

Field	Description
FAX Passthru Codec	Select the codec for fax passthrough, G711u or G711a. Default setting: G711u
FAX Codec Symmetric	To force the ATA to use a symmetric codec during fax passthrough, select yes. Otherwise, select no. Default setting: yes
DTMF Process INFO	To use the DTMF process info feature, select yes. Otherwise, select no. Default setting: yes
FAX Passthru Method	Select the fax passthrough method: None, NSE, or ReINVITE. Default setting: NSE
DTMF Process AVT	To use the DTMF process AVT feature, select yes. Otherwise, select no. Default setting: yes
FAX Process NSE	To use the fax process NSE feature, select yes. Otherwise, select no. Default setting: yes
DTMF Tx Method	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	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

Field	Description
DTMF Tx Mode	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	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	To enable the use of ITU-T T.38 standard for FAX Relay, select yes. Otherwise select no. Default setting: yes
Hook Flash Tx Method	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

Field	Description
FAX T38 Redundancy	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	Select yes to enable T.38 Error Correction Mode. Otherwise select no. Default setting: yes
FAX Tone Detect Mode	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	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	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

Dial Plan

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 semi-colon (;)

Example 1:

```
*1xxxxxxxxxx<:@fwdnat.pulver.com:5082;uid=jsmith;pwd=xy z
```

Example 2:

```
*1xxxxxxxxxx<:@fwd.pulver.com;nat;uid=jsmith;pwd=xyz
```

The syntax for a dial plan expression is described in the table below.

Dial Plan Entry	Functionality
*xx	Allow arbitrary 2 digit star code
[3469]11	Allow x11 sequences
0	Operator
00	Int'l Operator
[2-9]xxxxxx	US local number
1xxx[2-9]xxxxxx	US 1 + 10-digit long distance number
xxxxxxxxxxxx.	Everything else

FXS Port Polarity Configuration

Field	Description
Idle Polarity	Polarity before a call is connected: Forward or Reverse. Default setting: Forward
Caller Conn Polarity	Polarity after an outbound call is connected: Forward or Reverse. Default setting: Forward.
Callee Conn Polarity	Polarity after an inbound call is connected: Forward or Reverse. Default setting: Forward

User 1 and User 2

Use the *Voice > User 1* page and *Voice > User 2* pages to set the user preferences for the calls through the PHONE port1 and PHONE2 ports.

or User 2 Enter the settings as described below. After making changes, click **Submit** to save your settings, or click **Cancel** to redisplay the page with the saved settings.

Call Forward Settings

Field	Description
Cfwd All Dest	Forward number for Call Forward All Service. Default setting: blank
Cfwd Busy Dest	Forward number for Call Forward Busy Service. Same as Cfwd All Dest. Default setting: blank
Cfwd No Ans Dest	Forward number for Call Forward No Answer Service. Same as Cfwd All Dest. Default setting: blank
Cfwd No Ans Delay	Delay in sec before Call Forward No Answer triggers. Same as Cfwd All Dest. Default setting: 20

Selective Call Forward Settings

Field	Description
Cfwd Sel1-8 Caller	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 Sel1-8 Dest	The destination for the corresponding Call Forward Selective caller pattern (Cfwd Sel1-8 Caller). Default setting: blank
Cfwd Last Caller	The number of the last caller; this caller is actively forwarded to the Cfwd Last Dest via the Call Forward Last service. For more information, see **Vertical Service Activation Codes**. Default setting: blank
Cfwd Last Dest	The destination for the Cfwd Last Caller.
Block Last Caller	The number of the last caller; this caller is blocked via the Block Last Caller Service. For more information, see **Vertical Service Activation Codes**. Default setting: blank
Accept Last Caller	The number of the last caller; this caller is accepted via the Accept Last Caller Service. For more information, see **Vertical Service Activation Codes**. Default setting: blank

Speed Dial Settings

Field	Description
Speed Dial 2-9	Target phone number (or URL) assigned to speed dial 2, 3, 4, 5, 6, 7, 8, or 9. Default setting: blank

Supplementary Service Settings (User)

Field	Description
CW Setting	Call Waiting on/off for all calls. Default setting: yes
Block CID Setting	Block Caller ID on/off for all calls. Default setting: no
Block ANC Setting	Block Anonymous Calls on or off. Default setting: no
DND Setting	DND on or off. Default setting: no
CID Setting	Caller ID Generation on or off. Default setting: yes
CWCID Setting	Call Waiting Caller ID Generation on or off. Default setting: yes
Dist Ring Setting	Distinctive Ring on or off. Default setting: yes

Field	Description
Secure Call Setting	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. See **Line 1 and Line 2 Settings (PHONE Port1 and PHONE2), page 83**.
Message Waiting	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	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

Field	Description
Media Loopback Mode	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	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)

Distinctive Ring Settings

Field	Description
Ring1 - 8 Caller	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. The distinctive rings are set on the **Regional** page. Default setting: blank

Ring Settings

Field	Description
Default Ring	Default ringing pattern, 1–8, for all callers. Default setting: 1
Default CWT	Default CWT pattern, 1–8, for all callers. Default setting: 1
Hold Reminder Ring	Ring pattern for reminder of a holding call when the phone is on-hook. Default setting: 8
Call Back Ring	Ring pattern for call back notification. Default setting: 7

Field	Description
Cfwd Ring Splash Len	Duration of ring splash when a call is forwarded (0 – 10.0s) Default setting: 0
Cblk Ring Splash Len	Duration of ring splash when a call is blocked (0 – 10.0s) Default setting: 0
VMWI Ring Policy	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. Default setting: New VM 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.
VMWI Ring Splash Len	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	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