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 Settings (PHONE Port)**
- **PSTN (LINE Port)**
- **User 1**
- **PSTN User**
- **DECT Line 1 - DECT Line 10**
- **DECT User**

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.

To open this page: Click **Voice** on the menu bar, and then click **Information** in the navigation tree. 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.
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.

Field	Description
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)
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 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.

Field	Description
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.
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.

Field	Description
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.
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.

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

PSTN Line Status

Field	Description
Hook State	The hook state of the LINE port: On or Off.
Line Voltage	The voltage existing on the PSTN line.
Loop Current	The current (milliamperes) existing on the local loop.
Registration State	Indicates if the line has registered with the SIP proxy.
Last Registration At	The last date and time when the line was registered.
Next Registration In	The number of seconds before the next registration renewal.

Field	Description
Last Called VoIP Number	The VoIP phone number that was most recently called through this port.
Last Called PSTN Number	The PSTN phone number that was most recently called through the LINE port
Last VoIP Caller	The originating phone number of the VoIP call that was most recently received through the LINE port.
Last PSTN Caller	The originating phone number of the PSTN call that was most recently received through the LINE port.
Last PSTN Disconnect Reason	The reason for the ATA hanging up the LINE port: ▪ PSTN Disconnect Tone ▪ PSTN Activity Timeout ▪ CPC Signal ▪ Polarity Reversal ▪ VoIP Call Failed ▪ VoIP Call Ended ▪ Invalid VoIP Destination ▪ Invalid PIN ▪ PIN Digit Timeout ▪ VoIP Dialing Timeout ▪ PSTN Gateway Call Timeout ▪ VoIP Gateway Call Timeout
PSTN Activity Timer	The time in milliseconds (ms) before the ATA disconnects the current gateway unless the PSTN side has some audio activity.
Mapped SIP Port	The port number of the SIP port mapped by NAT.

Field	Description
Call Type	The type of call: ▪ 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
VoIP State	May take one of the following values: Idle ▪ Collecting PSTN Pin ▪ Invalid PSTN PIN ▪ PSTN Caller Accepted ▪ Connected to PSTN
PSTN State	May take one of the following values: Idle ▪ Collecting PSTN Pin ▪ Invalid PSTN PIN ▪ PSTN Caller Accepted ▪ Connected to PSTN
VoIP Tone	The tone that is being played to the VoIP call leg.
PSTN Tone	The tone that is being played to the PSTN call leg.
VoIP Peer Name	The of the party at the VoIP call leg.

Field	Description
PSTN Peer Name	The name of the party at the PSTN call leg.
VoIP Peer Number	The phone number of the party at the VoIP call leg.
PSTN Peer Number	The phone number of the party at the PSTN call leg.
VoIP Call Encoder	The audio encoder being used for the VoIP call leg.
VoIP Call Decoder	The audio decoder being used for the VoIP call leg.
VoIP Call FAX	The status of the fax passthrough mode for VoIP calls.
VoIP Call Remote Hold	Indicates whether the far end has placed the VoIP call on hold.
VoIP Call Duration	The duration of the VoIP call.
VoIP Call Packets Sent	The number of packets sent for VoIP calls.
VoIP Call Packets Recv	The number of packets received for VoIP calls.
VoIP Call Bytes Sent	The number of bytes sent for VoIP calls.
VoIP Call Bytes Recv	The number of bytes received for VoIP calls.
VoIP Call Decode Latency	The number of milliseconds for decoder latency for VoIP calls.
VoIP Call Jitter	The number of milliseconds for receiver jitter for VoIP calls.
VoIP Call Round Trip Delay	The number of milliseconds for delay for VoIP calls.
VoIP Call Packets Lost	The number of packets lost for VoIP calls.
VoIP Call Packet Error	The number of invalid packets received for VoIP calls.
VoIP Call Mapped RTP Port	The port mapped for Real Time Protocol traffic for VoIP calls.

DECT 1 ~ DECT 10 Status

Field	Description
Registration State	Indicates whether or not the line has registered with the SIP proxy: Registered, Not Registered, or Failed.
Last Registration At	The last date and time when the line was registered.
Next Registration In	The number of seconds before the next registration renewal.
Message Waiting	Indicates whether or not there are new messages: **yes** or **no**. The value automatically is set to yes when a message is received. You also can clear or set the flag manually from the *User 1* page.
Call Back Active	Indicates whether a call back request is in progress: **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.
Mapped SIP Port	Port number of the SIP port mapped by NAT.
Call 1 and 2 State	The current call state: ▪ 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.

Field	Description
Call 1 and 2 FAX	The status of the fax passthrough mode.
Call 1 and 2 Type	The direction of the call: ▪ 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.
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.

Field	Description
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.
Call 1 and 2 Mapped RTP Port	The port mapped for Real Time Protocol traffic for Call 1/2.

DECT Handset 1 ~ DECT Handset 10 Status

Field	Description
Handset IPEI	The unique hardware identifier of the unit, comparable to a MAC address.
Model Number	The Cisco model number of the unit.

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 186**.)

To open this page: Click **Voice** on the menu bar, and then click **System** in the navigation tree. 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	Domain of the service provider to which the ATA is connected to. It prevents the ATA from connecting to other service providers.
IVR Admin Password	Password for the administrator to manage the ATA by using the built-in IVR through a connected phone. **NOTE** The registered HS can also manage the ATA or deregister other HS with this password.
Network Startup Delay	The number of seconds of delay between restarting the voice module and initializing network interface. Default setting: 3

Field	Description
Handset (HS) Pairing Password	Password used as self authentication for Handset registration and deregistration. Default setting: blank **NOTE** There are 3 options in Handset registration: ▪ Register: For an unregistered HS, select Register and enter the Handset Pairing Passwd for authentication. ▪ Deregister: For a registered HS, select Deregister and enter the Handset Pairing Passwd for authentication. Use select softkey to deregister the HS. ▪ Deregister Others: For a registered HS, select Deregister Others. Enter <IVR Admin Passwd> for authentication. Use select softkey to deregister others.

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
Syslog Server Transport	The Syslog messages can be printed to the server with the selected tranport method (UDP or TLS) Default setting: UDP
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 Server Transport	The Debug information can be printed to the server with the selected transport method (UDP and TLS.) Default setting: UDP
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
DevTest Password	Used for internal automation test; not for the user.
Syslog Prefix	Allows the user to prefix additional information to syslog. Default setting: 215
DECT Codec Change	Codec change between handset (HS) and base to match RTP codec. Default setting: yes

SIP

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

To open this page: Click **Voice** on the menu bar, and then click **SIP** in the navigation tree. 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
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

Field	Description
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 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

Field	Description
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

Field	Description
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
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

Field	Description
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

Field	Description
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.
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

Field	Description
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
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
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

Field	Description
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
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

Field	Description
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
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
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.

NOTE Cisco SPA122, SPA112, and SPA232D supports 302/310 redirect while provisioning.

To open this page: Click **Voice** on the menu bar, and then click **Provisioning** in the navigation tree. 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)
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

Field	Description
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
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

Field	Description
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
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

Field	Description
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

Field	Description
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.

To open this page: Click **Voice** on the menu bar, and then click **Region** in the navigation tree. 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)
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)

Field	Description
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	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)

Field	Description
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)
Call Remind Tone	When there are 2 calls on the FXS port, and if one call is held by the UUT (Unit under test) and the other is connected, a holding tone is played on the UUT to remind the presence of the held call. Default setting: blank (the feature is not enabled)

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)
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)

Field	Description
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
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

Field	Description
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
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

Field	Description
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

Field	Description
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 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

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

Miscellaneous

Field	Description
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. Default setting: -16.0.
DTMF Twist	To gain difference between the two tone frequency. Default setting: 2
DTMF Playback Length	Local DTMF playback duration in milliseconds. Default setting: .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.

Field	Description
Playback ABCD	To enable local playback of OOB DTMF ABCD, select yes. Otherwise, select no. Default setting: yes
Caller ID Method	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

Field	Description
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

Line 1 Settings (PHONE Port)

Use the *Voice > Line 1* page to configure the settings for calls through the PHONE port.

To open this page: Click **Voice** on the menu bar, and then click **Line 1** in the navigation tree. 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
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
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

Field	Description
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

Field	Description
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
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

Field	Description
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
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

Field	Description
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 **NOTE** When FXS port initiates a conference call, the caller ID on the analogue phone is updated to the Conference Bridge.

Field	Description
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
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

Field	Description
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
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

Field	Description
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
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

Field	Description
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

Field	Description
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
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

Field	Description
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, page 147**.
Referral Serv	Enable Referral Service. See the Referral Services Codes parameter For more information. Default setting: yes
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
Preferred Codec, Second Preferred Codec, Third Preferred Codec	Up to three codecs to be used for all calls from this handset, listed order of preference. The actual codec used in a call still 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	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: yes
Use Remote Pref Codec	To use the preferred codec specified by the remote peer, select yes. Otherwise, select no. Default setting:
Codec Negotiation	Specify the codecs for codec negotiation: Default or List All. Default setting: Default
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

Field	Description
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
FAX Passthru Codec	Select the codec for fax passthrough, G711u or G711a. Default setting: G711u
Echo Canc Adapt Enable	To enable the echo canceller to adapt, select yes. Otherwise, select no. Default setting: yes
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

Field	Description
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
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

Configuration Tables

G3 NSE-Based Passthrough FAX Call

Parameter	Value	Description
FAX V21 Detect Enable	Yes	
FAX CNG Detect Enable	Yes	If CNG detection is not needed, set it to **No**.
FAX Tone Detect Mode	Caller or Callee	
FAX Passthru Method	NSE	
FAX Process NSE	Yes	
FAX Passthru Codec	G.711u or G.711a	Depends on the passthrough codec used.
FAX Disable ECAN	No	
Echo Canc Enable	Yes	
FAX Enable T38	No	
Modem Line	No	

SG3/V.34 NSE-Based Passthrough FAX Call

Parameter	Value	Description
FAX V21 Detect Enable	Yes	
FAX CNG Detect Enable	Yes	If CNG detection is not needed, set it to **No**.
FAX Tone Detect Mode	Caller or Callee	

Parameter	Value	Description
FAX Passthru Method	NSE	
FAX Process NSE	Yes	
FAX Passthru Codec	G.711u or G.711a	Depends on the passthrough codec used.
FAX Disable ECAN	No	
Echo Canc Enable	Yes	
FAX Enable T38	No	
Modem Line	No	

SIP Protocol-Based passthrough FAX Call

Parameter	Value	Description
FAX V21 Detect Enable	Yes	
FAX CNG Detect Enable	Yes	If CNG detection is not needed, set it to **No**.
FAX Tone Detect Mode	Caller or Callee	
FAX Passthru Method	ReINVITE	
FAX Process NSE	No	
FAX Passthru Codec	G.711u or G.711a	Depends on the passthrough codec used.
FAX Disable ECAN	No	
Echo Canc Enable	Yes	
FAX Enable T38	No	
Modem Line	No	

SG3/V.34 SIP Protocol-Based passthrough FAX Call

Parameter	Value	Description
FAX V21 Detect Enable	Yes	
FAX CNG Detect Enable	Yes	If CNG detection is not needed, set it to **No**.
FAX Tone Detect Mode	Caller or Callee	
FAX Passthru Method	ReINVITE	
FAX Process NSE	No	
FAX Passthru Codec	G.711u or G.711a	Depends on the passthrough codec used.
FAX Disable ECAN	No	
Echo Canc Enable	Yes	
FAX Enable T38	No	
Modem Line	No	

SIP Protocol-Based T.38 Relay FAX Call

Parameter	Value	Description
FAX V21 Detect Enable	Yes	
FAX CNG Detect Enable	Yes	If CNG detection is not needed, set it to **No**.
FAX Tone Detect Mode	Caller or Callee	
FAX Passthru Method	ReINVITE	
FAX Disable ECAN	No	

Parameter	Value	Description
Echo Canc Enable	Yes	
FAX Enable T38	No	
Modem Line	No	
FAX T38 Return to Voice	No	If voice call back is needed after T38 fac call, set it to **No**.

T38 Relay FAX Failover to SIP Protocol-Based Passthrough FAX call

Parameter	Value	Description
FAX V21 Detect Enable	Yes	
FAX CNG Detect Enable	Yes	If CNG detection is not needed, set it to **No**.
FAX Tone Detect Mode	Caller or Callee	
FAX Passthru Method	ReINVITE	
FAX Passthru Codec	G.711u or G.711a	Depends on the passthrough codec used.
FAX Disable ECAN	No	If connected to SG3 fax machine, set parameter to "Yes".
Echo Canc Enable	Yes	
FAX Enable T38	No	
Modem Line	No	
FAX T38 Return to Voice	No	

G3 Passthrough FAX Call With <FAX Line Toggle Code

Parameter	Value	Description
FAX Disable ECAN	No	
Echo Canc Enable	Yes	
Modem Line	No	
FAX Line Toggle Code	#99	Set it to any preferred code. **NOTE** Pre-dial #99 before dialing. Fax call made in this method does not detect fax tone or send out any fax request messages (like NSE or Re-INVITE message) to the remote gateway, which causes interoperability issue with remote side. Other fax parameters are not relevant to the call.

SG3/V.34 Passthrough FAX Call With <FAX Line Toggle Code

Parameter	Value	Description
FAX Disable ECAN	Yes	
Echo Canc Enable	Yes	
Modem Line	No	
FAX Line Toggle Code	#99	Set it to any preferred code. **NOTE** Pre-dial #99 before dialing. Fax call made in this method does not detect fax tone or send out any fax request messages (like NSE or Re-INVITE message) to the remote gateway, which causes interoperability issue with remote side. Other fax parameters are not relevant to the call.

Passthrough Modem Call With <Modem Line Toggle Code>

Parameter	Value	Description
Modem Line	No	If Modem Line Toggle Code is used to make the modem call, do not set it to **Yes**
Modem Line Toggle Code	*99	Set it to any preferred code. **NOTE** Pre-dial *99 before dialing. Other fax parameters are not relevant to the call.
FAX Disable ECAN	Yes	

Passthrough Modem Call With <Modem Line>

Parameter	Value	Description
Modem Line	Yes	
FAX Disable ECAN	Yes	

NOTE Other fax parameters are not relevant to the call.

Passthrough Modem Call With Manual Configuration

Parameter	Value	Description
Preferred Codec	G.711u or G.711a	Depends on the passthrough codec used.
FAX V21 Detect Enable	No	
FAX CNG Detect Enable	No	

Parameter	Value	Description
Echo Canc Enable	No	
Silence Supp Enable	No	
FAX Enable T38	No	
FAX Process NSE	No	
Modem Line	No	
Jitter Buffer Adjustment	No	
Network Jitter Level	High	

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

Dial Plan Entry	Functionality
xxxxxxxxxxxx.	Everything else

Gateway Accounts

Field	Description
Gateway1/2/3/4	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
GW1/2/3/4 NAT Mapping Enable	If enabled, the ATA uses NAT mapping when contacting Gateway 1. Default setting: no
GW1/2/3/4 Auth ID	This value is the authentication user-id to be used by the ATA to authenticate itself to Gateway 1. Default setting: blank
GW1/2/3/4 Password	This value is the password to be used by the ATA to authenticate itself to Gateway 1. Default setting: blank

VoIP Fallback to PSTN section

Field	Description
Auto PSTN Fallback	If enabled, the ATA automatically routes all calls to the PSTN gateway when the Line 1 proxy is down (registration failure or network link down). Default setting: yes

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

PSTN (LINE Port)

Use the *Voice > PSTN* page to configure the settings for calls through the LINE (PSTN) port.

To open this page: Click **Voice** on the menu bar, and then click **PSTN** in the navigation tree. 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.

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

Field	Description
Incoming Handset List	The devices that ring when an incoming call is received. Default setting: fxs,1,2,3,4,5,6,7,8,9,10

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 1 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	CoS value for RTP data. Valid values are 1 through 7. Default setting: 6
Network Jitter Level	Determines how jitter buffer size is adjusted by the ATA device. 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: low
Jitter Buffer Adjustment	Controls how the jitter buffer should be adjusted. Select the appropriate setting: up and down, up only, down only, or disable. 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. The default is UDP. 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: 5061
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 device 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

Field	Description
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.
SIP Remote-Party-ID	To use the Remote-Party-ID header instead of the From header, select yes. Otherwise, select no. Default setting: no
SIP GUID	The Global Unique ID is generated for each line for each device. When it is enabled, the ATA device 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. This feature was requested by Bell Canada (Nortel) to limit the registration of SIP accounts. Default setting: no

Field	Description
SIP Debug Option	SIP messages are received at or sent from the proxy listen port. This feature controls which SIP messages to log. Choices are as follows: ▪ **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. Default setting: none
RTP Log Intvl	The interval for the RTP log. Default setting: 0

Field	Description
Restrict Source IP	If Lines 1 and 2 use the same SIP Port value and the Restrict Source IP feature is enabled, the proxy IP address for Lines 1 and 2 is treated as an acceptable IP address for both lines. To enable the Restrict Source IP feature, select yes. Otherwise, select no. If configured, the ATA will drop all packets sent to its SIP Ports originated 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	Controls when the ATA device sends BYE to terminate stale call legs upon completion of call transfers. Multiple delay settings (Referor, Refer Target, Referee, and Refer-To Target) are configured on this screen. For the Referor Bye Delay, enter the appropriate period of time in seconds. Default setting: 4
Refer Target Bye Delay	For the Refer Target Bye Delay, enter the appropriate period of time in seconds. Default setting: 0
Referee Bye Delay	For the Referee Bye Delay, enter the appropriate period of time in seconds. Default setting: 0
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 IP telephony 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

Field	Description
Use Anonymous with RPID	When set to yes, use “anonymous” in the SIP message when remote party ID is requested in the SIP message. Default setting: yes
Use Local Addr in FROM	The IP address of the local address enclosed in the FROM of the SIP message. Default setting: no

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
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. Escape sequence of %xx is also accepted. For example, %0d%0a is unescaped into \r\n (CRLF). 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 or outbound proxy. Default setting: $PROXY

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	Enable the use of *Outbound Proxy*. If set to no, the *Outbound Proxy* parameter and *Use OB Proxy in Dialog* is 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 *Use Outbound Proxy* parameter is no, or if the *Outbound Proxy* parameter is empty. Default setting: yes
Register	Enable periodic registration with the *Proxy*. This parameter is ignored if the *Proxy* parameter 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: yes
Register Expires	Allow answering inbound calls without successful (dynamic) registration by the unit. If proxy responded to REGISTER with a smaller Expires value, the ATA will renew registration based on this smaller value instead of the configured value. If registration failed with an Expires too brief error response, the ATA will retry with the value given in the Min-Expires header in the error response. Default setting: 3600

Field	Description
Ans Call Without Reg	Expires value in sec in a REGISTER request. 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: yes
Use DNS SRV	If required by your provider, check this box 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	This parameter sets the delay (sec) after which the ATA will retry from the highest priority proxy (or outbound proxy) servers after it has failed over to a lower priority server. 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 record per server name does not allow the notion of priority and so all hosts will be considered at the same priority and the ATA will not attempt to fall back after a fail over). Default setting: 3600
Proxy Redundancy Method	The ATA makes an internal list of proxies returned in DNS SRV records. In normal mode this list will contain proxies ranked by weight and priority. If the parameter *Based on SRV port* is configured, the ATA creates a list in normal mode first, and then inspects the port numbers based on the 1st proxy's port on the list. Default setting: Normal

Subscriber Information

Field	Description
Display Name	Display name for caller ID.
User ID	Extension number for this line.

Field	Description
Password	Password for this line.
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	The Authentication ID for SIP authentication.

Audio Configuration

NOTE A codec resource is considered as allocated if it has been included in the SDP codec list of an active call, even though it eventually may not be the one chosen for the connection. So, if the G.729a codec is enabled and included in the codec list, that resource is tied up until the end of the call whether or not the call actually uses G.729a. If the G729a resource is already allocated and since only one G.729a resource is allowed per device, no other low-bit-rate codec may be allocated for subsequent calls; the only choices are G711a and G711u. On the other hand, two G.723.1/G.726 resources are available per device. Therefore it is important to disable the use of G.729a in order to guarantee the support of two simultaneous G.723/G.726 codec.

Field	Description
Preferred Codec	Preferred codec for all calls. (The actual codec used in a call still depends on the outcome of the codec negotiation protocol.) Select one of the following: G711u, G711a, G726-32, or G729a. Default setting: G711u
Use Pref Codec Only	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: yes
G729a Enable	To enable the use of the G729a codec at 8 kbps, select yes. Otherwise, select no. Default setting: no

Field	Description
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 G726 codec at 32 kbps, select yes. Otherwise, select no. Default setting: no
Echo Canc Enable	To enable the use of the echo canceller, select yes. Otherwise, select no. Default setting: yes
FAX V21 Detect Enable	To enable detection of V21 fax tones, select yes. Otherwise, select no. Default setting: no
Echo Canc Adapt Enable	To enable the echo canceller to adapt, 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: no
Echo Supp Enable	To enable the use of the echo suppressor, select yes. Otherwise, select no. Default setting: no
FAX Passthru Codec	Select the codec for fax passthrough, G711u or G711a. Default setting: G711u
DTMF Process INFO	To use the DTMF process info feature, select yes. Otherwise, select no. Default setting: yes
FAX Codec Symmetric	To force the ATA device to use a symmetric codec during fax passthrough, select yes. Otherwise, select no. Default setting: yes
DTMF Process AVT	To use the DTMF process AVT feature, select yes. Otherwise, select no. When set to no, the AVT (RFC2833) payload type is not be included in outbound SDP. Default setting: yes

Field	Description
FAX Passthru Method	Select the fax passthrough method: None, NSE, or ReINVITE. Default setting: NSE
DTMF Tx Method	Select the method to transmit DTMF signals to the far end: InBand, AVT, INFO, Auto, InBand+INFO, or AVT+INFO. InBand sends DTMF 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 Process NSE	To use the fax process NSE feature, select yes. Otherwise, select no. Default setting: yes
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: yes
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

Dial Plans

Field	Description
Dial Plan 1/2/3/4/ 5/6/7/8	Enter up to eight dial plans into this dial plan pool. You can associate a dial plan with a VoIP Caller or a PSTN Caller by referencing the index number (1~8). See **VoIP-To-PSTN Gateway Setup**, **VoIP Users and Passwords (HTTP Authentication)**, and **PSTN-To-VoIP Gateway Setup**. For information about the dial plan syntax, see **Configuring Dial Plans, page 225**. Default setting: (xx.)

VoIP-To-PSTN Gateway Setup

Field	Description
VoIP-To-PSTN Gateway Enable	Choose yes to enable or choose no to disable the VoIP-To-PSTN Gateway functionality. Default setting: yes
VoIP Caller Auth Method	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	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	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	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	The index number of the dial plan to use when the VoIP Caller is not authenticated. Dial plans are configured in the **Dial Plans** section. Default setting: 1
Line 1 Fallback DP	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

Field	Description
VoIP Caller ID Pattern	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	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 1/2/3/4/5/6/7/8 PIN	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 1/2/3/4/5/6/7/8 DP	The index number of the dial plan to use upon successful entry of the corresponding VoIP Caller PIN. Dial plans are configured in the **Dial Plans** section. Default setting: 1

VoIP Users and Passwords (HTTP Authentication)

Field	Description
VoIP User 1/2/3/4/5/6/7/8 Auth ID	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 user-id equals to this ID is referred to VoIP User 1 of this ATA. **NOTE:** 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 1/2/3/4/5/6/7/8 Password	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 1/2/3/4/5/6/7/8 DP	For up to 8 VoIP users, specify the index of the dial plan to be used after successful authentication. Dial plans are configured in the **Dial Plans** section. If authentication is disabled, the default dial plan is used for all unknown VoIP users. Default setting: 1.

PSTN-To-VoIP Gateway Setup

Field	Description
PSTN-To-VoIP Gateway Enable	Select yes to enable or select no to disable PSTN-To-VoIP Gateway functionality. Default setting: yes
PSTN Caller Auth Method	The method to authenticate a PSTN Caller to access the VoIP gateway. Choose from none or PIN. Default setting: none

Field	Description
PSTN Ring Thru Line 1	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**. Default setting: yes
PSTN PIN Max Retry	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	Choose yes or no. Default setting: no
PSTN CID Number Prefix	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	The index number of the dial plan that is used when the PSTN Caller Auth Method is set to none. Dial plans are configured in the **Dial Plans** section. Default settings: 1
Line 1 Signal Hook Flash to PSTN	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	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

Field	Description
PSTN Caller ID Pattern	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	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 1/2/3/4/5/6/7/8 PIN	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 1/2/3/4/5/6/7/8 DP	The index number of the dial plan to use upon successful entry of the corresponding PSTN Caller PIN. Dial plans are configured in the **Dial Plans** section. Default setting: 1

PSTN Timer Values (sec)

Field	Description
VoIP Answer Delay	The number of seconds to wait before auto-answering an inbound VoIP call for the FXO account. The range is 0-255. Default setting: 0
VoIP PIN Digit Timeout	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	After an inbound PSTN call starts ringing, the number of seconds to wait before auto-answering the call. The range is 0-255. Default setting: 16
PSTN PIN Digit Timeout	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	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	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	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	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

Field	Description
VoIP DLG Refresh Intvl	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	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	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	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	The length of the hook flash in seconds. Default setting: .25

PSTN Disconnect Detection

Field	Description
Detect CPC	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

Field	Description
Detect Polarity Reversal	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	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	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	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	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	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	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

Field	Description
Detect Disconnect Tone	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	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.

Field	Description
Disconnect Tone (continued)	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)

International Settings

Field	Description
FXO Country Setting	The country of deployment. This setting applies the relevant regional settings for PSTN calls. Default setting: USA
Tip Ring Voltage Adjustment	Voltage adjustment. The choices are 3.1V, 3.2V, 3.35V, and 3.5V. Default setting: 3.5V.
Ring Frequency Min	The lower limit of the ring frequency used to detect the ring signal. Default setting: 10
SPA To PSTN Gain	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	The higher limit of the ring frequency used to detect the ring signal. Default setting: 100

Field	Description
PSTN To SPA Gain	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	The minimum signal duration required by the Gateway for recognition as a ring signal. Default setting: 256 ms
Ring Indication Delay	Choose from {0, 512, 768, 1024, 1280, 1536, 1792} (ms). Default setting: 512ms
Ring Timeout	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	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	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

User 1

Use the *Voice > User 1* page to set the user preferences for the calls through the PHONE port1 .

To open this page: Click **Voice** on the menu bar, and then click **User 1** in the navigation tree. 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

Field	Description
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/Forward 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

Field	Description
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

<table>
<tr><th>Field</th><th>Description</th></tr>
<tr><td>Dist Ring Setting</td><td>Distinctive Ring on or off.
Default setting: yes</td></tr>
<tr><td>Secure Call Setting</td><td>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 Settings (PHONE Port), page 99.</td></tr>
<tr><td>Message Waiting</td><td>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</td></tr>
<tr><td>Accept Media Loopback Request</td><td>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</td></tr>
</table>

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

PSTN User

Use the *Voice > PSTN User* page to set the user preferences for calls through the LINE port.

To open this page: Click **Voice** on the menu bar, and then click **PSTN** in the navigation tree. 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.

PSTN-To-VoIP Selective Call Forward Settings

Field	Description
Cfwd Sel1-8 Caller	Eight PSTN Caller Number Patterns to be blocked for VoIP gateway services or forwarded to a certain VoIP number. If the caller is blocked, the ATA will not auto-answers the call.
Cfwd Sel1-8 Dest	Eight VoIP destinations to forward a PSTN caller matching the *Cfwd Sel x Caller parameter.* If this entry is blank, the PSTN caller is blocked for VoIP service.
Cfwd Last Caller	The Caller number that is actively forwarded to Cfwd Last Dest by using the Call Forward Last activation code. Default setting: blank
Cfwd Last Dest	Forward number for the Cfwd Last Caller parameter. Default setting: blank
Block Last Caller	ID of caller blocked via the Block Last Caller service. Default setting: blank
Accept Last Caller	ID of caller accepted via the Accept Last Caller service. Default setting: blank

PSTN-To-VoIP Speed Dial Settings

Field	Description
Speed Dial 2-9	The VoIP number to call when the PSTN caller dials the specified digit. Default setting: blank

PSTN Ring Thru Line 1 Distinctive Ring Settings

Field	Description
Ring1-8 Caller	Eight PSTN Caller Number Patterns such that the corresponding ring will be used to ring through Line 1 if the PSTN caller matches this pattern. The ring patterns are configured on the *Voice* > *Regional* page. For more information, see **Distinctive Ring Patterns, page 85**. Default setting: blank

PSTN Ring Thru Line 1 Ring Settings

Field	Description
Default Ring	The default ring to be used to ring through Line 1. Choose from {1,2,3,4,5,6,7,8,Follow Line Cfg}. If Follow Line Cfg is selected, the ring is determined by the distinctive ring settings for Line 1. The ring patterns are configured on the *Voice > Regional* page. For more information, see **Distinctive Ring Patterns, page 85**. Default setting: 2

DECT Line 1 - DECT Line 10

Use the *Voice > DECT Line 1~DECT Line 10* pages to configure the settings for calls using Cisco SPA302D handsets.

To open this page: Click **Voice** on the menu bar, and then click **DECT Line 1~10** in the navigation tree. 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 When a DECT line receives an incoming from a RU and if no handset is registered to this line or all handsets attached to this line are turned off, the base responds to the RU in 9 seconds with "486 Busy Here."

General

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
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	This setting works around devices that do not play inbound RTP if the streaming audio server line declares itself as a send-only device and tells the client not to stream out audio. Enter a Fully Qualified Domain Name (FQDN) or IP address of an RTP sink; this value is used by the streaming audio server line in the SDP of its 200 response to an inbound INVITE message from a client. 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 a FQDN or IP address of a 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 and, if specified, in the m = line of the SDP. If this value is not specified or 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
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. Escape sequence of %xx is also accepted. For example, %0d%0a is unescaped into \r\n (CRLF). 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 or outbound proxy. 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

Field	Description
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

Field	Description
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
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

Field	Description
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 Lines 1 and 2 use the same SIP Port value and the Restrict Source IP feature is enabled, the proxy IP address for Lines 1 and 2 is treated as an acceptable IP address for both lines. To enable the Restrict Source IP feature, select yes. Otherwise, select no. If configured, the ATA will drop all packets sent to its SIP Ports originated 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	Controls when the ATA sends BYE to terminate stale call legs upon completion of call transfers. Multiple delay settings (Referor, Refer Target, Referee, and Refer-To Target) are configured on this screen. For the Referor Bye Delay, enter the appropriate period of time in seconds. Default setting: 4
Refer Target Bye Delay	For the Refer Target Bye Delay, enter the appropriate period of time in seconds. Default setting: 0
Referee Bye Delay	For the Referee Bye Delay, enter the appropriate period of time in seconds. Default setting: 0
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
Use Anonymous With RPID	When set to yes, use "anonymous" 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

Field	Description
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

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

Field	Description
Voice Mail Number	The phone number for the voice mail system. 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	Enable the use of *Outbound Proxy*. If set to no, the *Outbound Proxy* parameter and *Use OB Proxy in Dialog* is 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 *Use Outbound Proxy* parameter is no, or if the *Outbound Proxy* parameter is empty. Default setting: yes
Register	Enable periodic registration with the *Proxy*. This parameter is ignored if the *Proxy* parameter 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: yes

Field	Description
Register Expires	Allow answering inbound calls without successful (dynamic) registration by the unit. If proxy responded to REGISTER with a smaller Expires value, the ATA will renew registration based on this smaller value instead of the configured value. If registration failed with an Expires too brief error response, the ATA will retry with the value given in the Min-Expires header in the error response. Default setting: 3600
Ans Call Without Reg	Expires value in sec in a REGISTER request. 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: yes
Use DNS SRV	If required by your provider, check this box 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	This parameter sets the delay (sec) after which the ATA will retry from the highest priority proxy (or outbound proxy) servers after it has failed over to a lower priority server. 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 record per server name does not allow the notion of priority and so all hosts will be considered at the same priority and the ATA will not attempt to fall back after a fail over). Default setting: 3600

Field	Description
Proxy Redundancy Method	The ATA makes an internal list of proxies returned in DNS SRV records. In normal mode this list will contain proxies ranked by weight and priority. If the parameter *Based on SRV port* is configured, the ATA creates a list in normal mode first, and then inspects the port numbers based on the 1st proxy's port on the list. Default setting: Normal
Voice Mail Server	The URL or IP address of the voice mail server.
Mailbox Subscribe Expires	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

Subscriber Information

Field	Description
Display Name	Display name for caller ID.
User ID	Extension number for this line.
Password	Password for this line.
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	The Authentication ID for SIP authentication.
Directory Number	The number for this line.
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. 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

Field	Description
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
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

Field	Description
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 For more information about star code settings, see **Vertical Service Activation Codes, page 90**. To enable secure calling by default, without requiring a star code, set the user's Secure Call Setting to yes. See **User 1, page 147**.
Referral Serv	Enable Referral Service. See the Referral Services Codes parameter For more information. Default setting: yes
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

Audio Configuration

NOTE A codec resource is considered as allocated if it has been included in the SDP codec list of an active call, even though it eventually may not be the one chosen for the connection. So, if the G.729a codec is enabled and included in the codec list, that resource is tied up until the end of the call whether or not the call actually uses G.729a. If the G729a resource is already allocated and since only one G.729a resource is allowed per device, no other low-bit-rate codec may be allocated for subsequent calls; the only choices are G711a and G711u. On the other hand, two G.723.1/G.726 resources are available per device. Therefore it is important to disable the use of G.729a in order to guarantee the support of two simultaneous G.723/G.726 codec.

Field	Description
Preferred Codec, Second Preferred Codec, Third Preferred Codec	Up to three codecs to be used for all calls from this handset, listed order of preference. The actual codec used in a call still 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	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: yes
Use Remote Pref Codec	To use the preferred codec specified by the remote peer, select yes. Otherwise, select no. Default setting:
Codec Negotiation	Specify the codecs for codec negotiation: Default or List All. Default setting: Default
Silence Supp Enable	To enable silence suppression so that silent audio frames are not transmitted, select yes. Otherwise, select no. Default setting: no

Field	Description
Silence Threshold	Select the appropriate setting for the threshold: high, medium, or low. Default setting: medium
G729a Enable	To enable the use of the G729a codec at 8 kbps, select yes. Otherwise, select no. Default setting: no
Echo Canc Enable	To enable the use of the echo canceller, select yes. Otherwise, select no. Default setting: yes
G726-32 Enable	To enable the use of the G726 codec at 32 kbps, select yes. Otherwise, select no. Default setting: no
G722 Enable	To enable the use of the G722 codec at 32 kbps, select yes. Otherwise, select no. Default setting: no
DTMF Process INFO	To use the DTMF process info feature, select yes. Otherwise, select no. Default setting: yes
DTMF Process AVT	To use the DTMF process AVT feature, select yes. Otherwise, select no. When set to no, the AVT (RFC2833) payload type is not be included in outbound SDP. Default setting: yes
DTMF Tx Method	Select the method to transmit DTMF signals to the far end: InBand, AVT, INFO, Auto, InBand+INFO, or AVT+INFO. InBand sends DTMF 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

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
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
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: yes

Dial Plan

Field	Description
Dial Plan	The allowed number patterns for outbound calls. For information about the dial plan syntax, see **Configuring Dial Plans, page 225**. Default setting: (*xx|[3469]11|0|00|[2-9]xxxxxx|1xxx[2-9]xxxxxxS0|xxxxxxxxxxxx.)

Field	Description
PSTN Fallback Dial Plan	The dial plan used when PSTN fallback is enabled and in use. Default setting: (S0<:@gw0>)
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
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
Incoming Handset List	The devices that ring when an incoming call is received. Default setting: fxs,1,2,3,4,5,6,7,8,9,10

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

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

Supplementary Service Settings

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 Settings (PHONE Port), page 99**.
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

Field	Description
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
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)

DECT User

Use the *Voice > DECT User page* to set the user preferences for calls using Cisco SPA302D handsets.

To open this page: Click **Voice** on the menu bar, and then click **DECT User** in the navigation tree. 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.

General

Field	Description
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

Handset 1

Field	Description
Handset Name	Name of the handset can be configured either from the SPA232D web GUI or from the handset GUI. The name can support only alphabets and numbers and the maximum length is 10. Default setting: Handset 1. Similary for Handset 2, 3, 4, and 5, the default setting is Handset 2, 3, 4, 5, respectively. **NOTE** When multiple handset names are to be changed, they are sent to CSS at a time with a format like this: 5#1: Name1#2: Name2#3: Name3#4: Name4#5: Name5 For example: If 5 handsets are requesting name provisioning simultaneously, and their names are "Hank", "White", "Pinkman", "Gus" and "Skyler" from handset1 to handset5 respectively. Then the format as following: 5#1:Hank#2:White#3:Pinkman#4:Gus#5:Skyler If the handset name is changed from handset in Settings>Handset Settings>Handset Name, it will be sent back to base and updated on the web page.
Outgoing DECT 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.

Field	Description
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/PSTN line. Select yes to enable this feature or select no to disable it. Default setting: no **NOTE** Cisco SPA232D now supports PSTN to DECT and DECT to PSTN outgoing line failover.
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

Field	Description
Handset Auto Update	To enable handset auto update select yes, otherwise select no. Default setting: yes **NOTE** When enabled, one of the following events triggers software update notification from SPA232D to handset: ▪ SPA232D reboots ▪ New handset subscribes to SPA232D ▪ Registered handset comes in range a. Five minutes after the firmware update notification from SPA232D, the handset gets the version number from SPA232D when it is idle. b. If the version retrieved from SPA232D is different from its local version number, the HS displays a message Software Update in 30 Seconds with 2 options—Update and Delay. ▪ If Delay is selected, the firmware update window pops up again in 1 hour. ▪ If Update is selected, the handset performs the upgrade immediately. c. If no soft-key is selected, the handset upgrades automatically in 30 seconds. If the upgrade fails for some reason, a splash window displays "System busy. Will retry in 5 minutes."