Advanced Voice Fields

This appendix describes the Advanced settings that are available after you log in as administrator.

After you click the *Voice* tab, you can choose the following pages:

Info page

You can use the *Voice tab* > *Info* page to view information about the WRP500. This page includes the following sections:

Note The fields on the Info page are read-only and cannot be edited.

Product Information section

This table describes the fields in the Product Information section of the Voice tab > Info page.

Field	Description
Product Name	Model number/name.
Serial Number	Serial number.
Software Version	Software version number.
Hardware Version	Hardware version number.

Field	Description
MAC Address	MAC address.
Client Certificate	Status of the client certificate, which can indicate whether the WRP500 has been authorized by your ITSP.
Customization	For a Remote Configuration (RC) unit, this field indicates whether the unit has been customized or not. Pending indicates a new RC unit that is ready for provisioning. If the unit has already retrieved its customized profile, this field displays the name of the company that provisioned the unit.
Voice Module Version	Voice module number.

System Status section

This table describes the fields in the System Status section of the Voice tab > Info page.

Field	Description
Current Time	Current date and time of the system; for example, 10/3/2003 16:43:00.
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	External IP address used for NAT mapping.

Line Status section

This table describes the fields in the Line Status section of the Voice tab > Info page.

Field	Description
Hook State	Hook state of the FXS port. Options are either 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	Number of seconds before the next registration renewal.

Field	Description
Message Waiting	Indicates whether you have new voice mail waiting. Options are either Yes or No. The value automatically is set to Yes when a message is received. You also can clear or set the flag manually. Setting this value to Yes can activate stutter tone and VMWI signal. This parameter is stored in long term memory and survives after reboot or power cycle.
Call Back Active	Indicates whether a call back request is in progress. Options are either Yes or No.
Last Called Number	The last number called from the FXS line.
Last Caller Number	Number of the last caller.
Mapped SIP Port	Port number of the SIP port mapped by NAT.
Call 1 and 2 State	May take one of the following values: • Idle • Dialing • Stunning • Calling • Proceeding • Ringing • Invalid • Connected • Hold • Holding • Resuming • Transit
Call 1 and 2 Tone	Type of tone used by the call.
Call 1 and 2 Encoder	Codec used for encoding.
Call 1 and 2 Decoder	Codec used for decoding.
Call 1 and 2 FAX	Status of the fax mode.
Call 1 and 2 Type	Direction of the call. May take one of the following values: • Inbound • Outbound • Transferred
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	Name of the internal phone.
Call 1 and 2 Peer Phone	Phone number of the internal phone.
Call 1 and 2 Call Duration	Duration of the call.

Field	Description
Call 1 and 2 Packets Sent	Number of packets sent.
Call 1 and 2 Packets Recv	Number of packets received.
Call 1 and 2 Bytes Sent	Number of bytes sent.
Call 1 and 2 Bytes Recv	Number of bytes received.
Call 1 and 2 Decode Latency	Number of milliseconds for decoder latency.
Call 1 and 2 Jitter	Number of milliseconds for receiver jitter.
Call 1 and 2 Packets Lost	Number of packets lost.
Call 1 and 2 Packet Error	Number of invalid packets received.
Call 1 and 2 Mapped RTP Port	The port mapped for Real Time Protocol traffic for Call 1/2.
Call 1 and 2 Media Loopback	Media loopback is used to quantitatively and qualitatively measure the voice quality that the end user experiences.

System page

You can use the *Voice tab* > *System* page to configure your system and network connections. This page includes the following sections:

- System Configuration section, page A-4
- Miscellaneous Settings section, page A-5

System Configuration section

This table describes the fields in the System Configuration section of the Voice tab > System page.

Field	Description
Restricted Access Domains	This feature is used when implementing software customization.
IVR Admin Passwd	Password for entering IVR menu.

Miscellaneous Settings section

This table describes the fields in the Miscellaneous section of the Voice tab > System page.

Field	Description
Syslog Server	Specifies the IP address of the syslog server.
Debug Server	Specifies the IP address of the debug server, which logs debug information. The level of detailed output depends on the debug level parameter setting.
Debug Level	Determines the level of debug information that is generated. Select 0, 1, 2, or 3 from the drop-down menu. The higher the debug level, the more debug information is generated. The default is 0, which indicates that no debug information is generated.
Debug Option	Specifies what debug information is expected. Generally can be set to *dbg_all*.

SIP page

You can use the *Voice tab* > *SIP* page to configure the SIP settings. This page includes the following sections:

- SIP Parameters section, page A-5
- SIP Timer Values (sec) section, page A-7
- Response Status Code Handling section, page A-8
- RTP Parameters section, page A-8
- SDP Payload Types section, page A-9
- NAT Support Parameters section, page A-10

SIP Parameters section

This table describes the fields in the SIP Parameters section of the Voice tab > SIP page.

Field	Description
Max Forward	SIP Max Forward value, which can range from 1 to 255. The default is **70.**
Max Redirection	Number of times an invite can be redirected to avoid an infinite loop. The default is **5.**
Max Auth	Maximum number of times (from 0 to 255) a request may be challenged. The default is **2.**
SIP User Agent Name	User-Agent header used in outbound requests. The default is **$VERSION**. If empty, the header is not included. Macro expansion of $A to $D corresponding to GPP_A to GPP_D allowed.

Field	Description
SIP Server Name	Server header used in responses to inbound responses. The default is **$VERSION**.
SIP Reg User Agent Name	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. The default is blank.
SIP Accept Language	Accept-Language header used. There is no default (this indicates the WRP500 does not include this header). If empty, the header is not included.
DTMF Relay MIME Type	MIME Type used in a SIP INFO message to signal a DTMF event. The default is **application/dtmf-relay.**
Remove Last Reg	Lets you remove the last registration before registering a new one if the value is different. Select yes or no from the drop-down menu. The default is **no.**
Use Compact Header	Lets you use compact SIP headers in outbound SIP messages. Select yes or no from the drop-down menu. If set to yes, the WRP500 uses compact SIP headers in outbound SIP messages. If set to no, the WRP500 uses normal SIP headers. If inbound SIP requests contain compact headers, the WRP500 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 WRP500 substitutes those headers with compact headers (if defined by RFC 261) if *Use Compact Header* parameter is set to yes. The default is **no.**
Escape Display Name	Lets you keep the Display Name private. Select yes if you want the WRP500 to enclose the string (configured in the Display Name) in a pair of double quotes for outbound SIP messages. Any occurrences of or \ in the string is escaped with \ and \\ inside the pair of double quotes. Otherwise, select no. The default is **no.**
RFC 2543 Call Hold	Configures the type of call hold: a:sendonly or 0.0.0.0. The default is **no;** do not use the 0.0.0.0 syntax in a HOLD SDP; use the a:sendonly syntax.
Mark All AVT Packets	If set to yes, all AVT tone packets (encoded for redundancy) have the marker bit set. If set to no, only the first packet has the marker bit set for each DTMF event. The default is **yes.**
SIP TCP Port Min	Specifies the lowest TCP port number that can be used for SIP sessions. The default Port Min is 5060.
SIP TCP Port Max	Specifies the highest TCP port number that can be used for SIP sessions. The default Port Max is 5080.

SIP Timer Values (sec) section

This table describes the fields in the SIP Timer Values section of the Voice tab > SIP page.

Field	Description
SIP T1	RFC 3261 T1 value (RTT estimate), which can range from 0 to 64 seconds. The default is 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. The default is 4.
SIP T4	RFC 3261 T4 value (maximum duration a message remains in the network), which can range from 0 to 64 seconds. The default is 5.
SIP Timer B	INVITE time-out value, which can range from 0 to 64 seconds. The default is 32.
SIP Timer F	Non-INVITE time-out value, which can range from 0 to 64 seconds. The default is 32.
SIP Timer H	INVITE final response, time-out value, which can range from 0 to 64 seconds. The default is 32.
SIP Timer D	ACK hang-around time, which can range from 0 to 64 seconds. The default is 32.
SIP Timer J	Non-INVITE response hang-around time, which can range from 0 to 64 seconds. The default is 32.
INVITE Expires	INVITE request Expires header value. If you enter 0, the Expires header is not included in the request. The default is 240. Range: 0–(2^{31}–1).
ReINVITE Expires	ReINVITE request Expires header value. If you enter 0, the Expires header is not included in the request. The default is 30. Range: 0–(2^{31}–1).
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. The default is 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. The default is 7200.

Field	Description
Reg Retry Intvl	Interval to wait before the WRP500 retries registration after failing during the last registration. The default is 30.
Reg Retry Long Intvl	When registration fails with a SIP response code that does not match *Retry Reg RSC*, the WRP500 waits for the specified length of time before retrying. If this interval is 0, the WRP500 stops trying. This value should be much larger than the Reg Retry Intvl value, which should not be 0. The default is 1200.

Response Status Code Handling section

This table describes the fields in the Response Status Code Handling section of the Voice tab > SIP page.

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.
SIT2 RSC	SIP response status code to INVITE on which to play the SIT2 Tone.
SIT3 RSC	SIP response status code to INVITE on which to play the SIT3 Tone.
SIT4 RSC	SIP response status code to INVITE on which to play the SIT4 Tone.
Try Backup RSC	SIP response code that retries a backup server for the current request.
Retry Reg RSC	Interval to wait before the WRP500 retries registration after failing during the last registration. The default is 30.

RTP Parameters section

This table describes the fields in the RTP Parameters section of the Voice tab > SIP page.

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. The default is 16384.
RTP Port Max	Maximum port number for RTP transmission and reception. The default is 16482.

Field	Description
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. The default is 0.030.
Stats In BYE	Determines whether the WRP500 includes the P-RTP-Stat header or response to a BYE message. The header contains the RTP statistics of the current call. Select yes or no from the drop-down menu. 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 in s>,EN=<encoder>,DE=<decoder>. The default is **no.**

SDP Payload Types section

This table describes the fields in the SDP Payload Types section of the Voice tab > SIP page.

Field	Description
NSE Dynamic Payload	NSE dynamic payload type. The valid range is 96-127. The default is 100.
AVT Dynamic Payload	AVT dynamic payload type. The valid range is 96-127. The default is 101.
INFOREQ Dynamic Payload	INFOREQ dynamic payload type. There is no default.
NSE Codec Name	NSE codec name used in SDP. The default is NSE.
AVT Codec Name	AVT codec name used in SDP. The default is telephone-event.
G711u Codec Name	G.711u codec name used in SDP. The default is PCMU.
G711a Codec Name	G.711a codec name used in SDP. The default is PCMA.
G729a Codec Name	G.729a codec name used in SDP. The default is G729a.
G729b Codec Name	G.729b codec name used in SDP. The default is G729ab.

Field	Description
EncapRTP Codec Name	EncapRTP codec name used in SDP. The default is EncapRTP.
EncapRTP Dynamic Payload	EncapRTP dynamic payload type.

NAT Support Parameters section

This table describes the fields in the NAT Support Parameters section of the Voice tab > SIP page.

Field	Description
Handle VIA received	If you select yes, the WRP500 processes the received parameter in the VIA header (this value is inserted by the server in a response to anyone of its requests). If you select no, the parameter is ignored. Select **yes** or **no** from the drop-down menu. The default is **no.**
Handle VIA rport	If you select yes, the WRP500 processes the rport parameter in the VIA header (this value is inserted by the server in a response to anyone of its requests). If you select no, the parameter is ignored. Select **yes** or **no** from the drop-down menu. The default is **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. The default is **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. The default is **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. The default is **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. The default is **no.**
STUN Enable	Enables the use of STUN to discover NAT mapping. Select yes or no from the drop-down menu. The default is **no.**

<table>
<tr><th>Field</th><th>Description</th></tr>
<tr><td>STUN Test Enable</td><td>If the STUN Enable feature is enabled and a valid STUN server is available, the WRP500 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 WRP500 detects symmetric NAT or a symmetric firewall, NAT mapping is disabled.

The default is no.</td></tr>
<tr><td>STUN Server</td><td>IP address or fully-qualified domain name of the STUN server to contact for NAT mapping discovery.</td></tr>
<tr><td>EXT IP</td><td>External IP address to substitute for the actual IP address of the WRP500 in all outgoing SIP messages. If 0.0.0.0 is specified, no IP address substitution is performed.

If this parameter is specified, the WRP500 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.

Note 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 WRP500 is the edge device, the second requirement is met.

The default is 0.0.0.0.</td></tr>
<tr><td>EXT RTP Port Min</td><td>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.

The default is 0.</td></tr>
<tr><td>NAT Keep Alive Intvl</td><td>Interval between NAT-mapping keep alive messages.

The default is 15.</td></tr>
</table>

Regional page

You can use the *Voice tab* > *Regional* page to localize your system with the appropriate regional settings. This page includes the following sections:

- Call Progress Tones section, page A-12
- Distinctive Ring Patterns section, page A-13
- Distinctive Call Waiting Tone Patterns section, page A-14
- Distinctive Ring/CWT Pattern Names section, page A-15
- Control Timer Values (sec) section, page A-16
- Vertical Service Activation Codes section, page A-17
- Outbound Call Codec Selection Codes section, page A-22
- Miscellaneous section, page A-23

Call Progress Tones section

This table describes the fields in the Call Progress Tones section of the Voice tab > Regional page.

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. The default is 350@-19,440@-19;10(*/0/1+2).
Second Dial Tone	Alternative to the Dial Tone when the user dials a three-way call. The default is 420@-19,520@-19;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. The default is 420@-19;10(*/0/1).
Prompt Tone	Prompts the user to enter a call forwarding phone number. The default is 520@-19,620@-19;10(*/0/1+2).
Busy Tone	Played when a 486 RSC is received for an outbound call. The default is 480@-19,620@-19;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. The default is 480@-19,620@-19;10(.25/.25/1+2).
Off Hook Warning Tone	Played when the caller has not properly placed the handset on the cradle. Off Hook Warning Tone is played when Reorder Tone times out. The default is 480@10,620@0;10(.125/.125/1+2)
Ring Back Tone	Played during an outbound call when the far end is ringing. The default is 440@-19,480@-19;*(2/4/1+2).
Ring Back 2 Tone	Your WRP500 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. The default value is the same as *Ring Back Tone*, except the cadence is 1s on and 1s off. The default is 440@-19,480@-19;*(1/1/1+2).
Confirm Tone	Brief tone to notify the user that the last input value has been accepted. The default is 600@-16; 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. The default is 985@-16,1428@-16,1777@-16;20(.380/0/1,.380/0/2,.380/0/3,0/4/0).

Field	Description
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. The default is 914@-16,1371@-16,1777@-16;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. The default is 914@-16,1371@-16,1777@-16;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. The default is 985@-16,1371@-16,1777@-16;20(.380/0/1,.274/0/2,.380/0/3,0/4/0).
MWI Dial Tone	Played instead of the Dial Tone when there are unheard messages in the caller's mailbox. The default is 350@-19,440@-19;2(.1/.1/1+2);10(*/0/1+2).
Cfwd Dial Tone	Played when all calls are forwarded. The default is 350@-19,440@-19;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. The default is 600@-19*(.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. The default is 350@-19;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. The default is 397@-19,507@-19;15(0/2/0,.2/.1/1,.1/2.1/2).
Feature Invocation Tone	Played when a feature is implemented. The default is 350@-16;*(.1/.1/1).

Distinctive Ring Patterns section

This table describes the fields in the Distinctive Ring Patterns section of the Voice tab > Regional page.

Field	Description
Ring1 Cadence	Cadence script for distinctive ring 1. The default is 60(2/4).
Ring2 Cadence	Cadence script for distinctive ring 2. The default is 60(.8/.4,.8/4).

Field	Description
Ring3 Cadence	Cadence script for distinctive ring 3. The default is 60(.4/.2,.4/.2,.8/4).
Ring4 Cadence	Cadence script for distinctive ring 4. The default is 60(.3/.2,1/.2,.3/4).
Ring5 Cadence	Cadence script for distinctive ring 5. The default is 1(.5/.5).
Ring6 Cadence	Cadence script for distinctive ring 6. The default is 60(.2/.4,.2/.4,.2/4).
Ring7 Cadence	Cadence script for distinctive ring 7. The default is 60(.4/.2,.4/.2,.4/4).
Ring8 Cadence	Cadence script for distinctive ring 8. The default is 60(0.25/9.75).

Distinctive Call Waiting Tone Patterns section

This table describes the fields in the Distinctive Call Waiting Tone Patterns section of the Voice tab > Regional page.

Field	Description
CWT1 Cadence	Cadence script for distinctive CWT 1. The default is 30(.3/9.7).
CWT2 Cadence	Cadence script for distinctive CWT 2. The default is 30(.1/.1,.1/9.7).
CWT3 Cadence	Cadence script for distinctive CWT 3. The default is 30(.1/.1,.1/.1,.1/9.7).
CWT4 Cadence	Cadence script for distinctive CWT 4. The default is 30(.1/.1,.3/.1,.1/9.3).
CWT5 Cadence	Cadence script for distinctive CWT 5. The default is 1(.5/.5).
CWT6 Cadence	Cadence script for distinctive CWT 6. The default is 30(.1/.1,.3/.2,.3/9.1).
CWT7 Cadence	Cadence script for distinctive CWT 7. The default is 30(.3/.1,.3/.1,.1/9.1).
CWT8 Cadence	Cadence script for distinctive CWT 8. The default is 2.3(.3/2).

Distinctive Ring/CWT Pattern Names section

This table describes the fields in the Distinctive Ring/CWT Pattern Names section of the Voice tab > Regional page.

Field	Description
Ring1 Name	Name in an INVITE Alert-Info Header to pick distinctive ring/CWT 1 for the inbound call. The default is Bellcore-r1.
Ring2 Name	Name in an INVITE Alert-Info Header to pick distinctive ring/CWT 2 for the inbound call. The default is Bellcore-r2.
Ring3 Name	Name in an INVITE Alert-Info Header to pick distinctive ring/CWT 3 for the inbound call. The default is Bellcore-r3.
Ring4 Name	Name in an INVITE Alert-Info Header to pick distinctive ring/CWT 4 for the inbound call. The default is Bellcore-r4.
Ring5 Name	Name in an INVITE Alert-Info Header to pick distinctive ring/CWT 5 for the inbound call. The default is Bellcore-r5.
Ring6 Name	Name in an INVITE Alert-Info Header to pick distinctive ring/CWT 6 for the inbound call. The default is Bellcore-r6.
Ring7 Name	Name in an INVITE Alert-Info Header to pick distinctive ring/CWT 7 for the inbound call. The default is Bellcore-r7.
Ring8 Name	Name in an INVITE Alert-Info Header to pick distinctive ring/CWT 8 for the inbound call. The default is Bellcore-r8.

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 does not sound right, or your phone does not ring, change your Ring Waveform, Ring Frequency, and Ring Voltage to the following:
  - Ring Waveform: Sinusoid
  - Ring Frequency: 25
  - Ring Voltage: 80V

Field	Description
Ring Waveform	Waveform for the ringing signal. Choices are **Sinusoid** or **Trapezoid**. The default is **Trapezoid**.
Ring Frequency	Frequency of the ringing signal. Valid values are 10–100 (Hz). The default is **20**.
Ring Voltage	Ringing voltage. Choices are **60–90** (V). The default is **85**.
CWT Frequency	Frequency script of the call waiting tone. All distinctive CWTs are based on this tone. The default is **440@-10**.

Control Timer Values (sec) section

This table describes the fields in the Control Timer Values (sec) section of the Voice tab > Regional page.

Field	Description
Hook Flash Timer Min	Minimum on-hook time before off-hook qualifies as hook-flash. For values, less than this, the on-hook event is ignored. Range: 0.1–0.4 seconds. The default is **0.1**.
Hook Flash Timer Max	Maximum on-hook time before off-hook qualifies as hook-flash. For values greater than this, the on-hook event is treated as on-hook (no hook-flash event). Range: 0.4–1.6 seconds. The default is **0.9**.
Callee On Hook Delay	Phone must be on-hook for at least this length of time in sec before the WRP500 tears down the current inbound call. This does not apply to outbound calls. Range: 0–255 seconds. The default is **0**.
Reorder Delay	Delay after far end hangs up before reorder tone is played. 0 = plays immediately, inf = never plays. Range: 0–255 seconds. The default is **5**.
Call Back Expires	Expiration time in seconds of a call back activation. Range: 0–65535 seconds. The default is **1800**.
Call Back Retry Intvl	Call back retry interval in seconds. Range: 0–255 seconds. The default is **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 WRP500 still considers the call as failed and keeps on retrying. The default is **0.5**.
VMWI Refresh Intvl	Interval between VMWI refresh to the CPE. The default is **0.**

Field	Description
Interdigit Long Timer	Long timeout between entering digits when dialing. The interdigit timer values are used as defaults when dialing. The Interdigit_Long_Timer is used after any one digit, if all valid matching sequences in the dial plan are incomplete as dialed. Range: 0–64 seconds. The default is **10**.
Interdigit Short Timer	Short timeout between entering digits when dialing. The Interdigit_Short_Timer is used after any one digit, if at least one matching sequence is complete as dialed, but more dialed digits would match other as yet incomplete sequences. Range: 0–64 seconds. The default is **3**.
CPC Delay	Delay in seconds after caller hangs up when the WRP500 starts removing the tip-and-ring voltage to the attached equipment of the called party. Range: 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. The default is **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 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. The default is **0** (CPC disabled).

Vertical Service Activation Codes section

Vertical Service Activation Codes are automatically appended to the dial plan. There is no need to include them in the dial plan, but no harm is done if they are included.

This table describes the fields in the Vertical Service Activation Codes section of the Voice tab > Regional page.

Field	Description
Call Return Code	This code calls the last caller. The default is *69.
Call Redial Code	Redials the last number called. The default is *07.
Blind Transfer Code	Begins a blind transfer of the current call to the extension specified after the activation code. The default is *98.

Field	Description
Call Back Act Code	Starts a callback when the last outbound call is not busy. The default is *66.
Call Back Deact Code	Cancels a callback. The default is *86.
Call Back Busy Act Code	Starts a callback when the last outbound call is busy. The default is *05
Cfwd All Act Code	Forwards all calls to the extension specified after the activation code. The default is *72.
Cfwd All Deact Code	Cancels call forwarding of all calls. The default is *73.
Cfwd Busy Act Code	Forwards busy calls to the extension specified after the activation code. The default is *90.
Cfwd Busy Deact Code	Cancels call forwarding of busy calls. The default is *91.
Cfwd No Ans Act Code	Forwards no-answer calls to the extension specified after the activation code. The default is *92.
Cfwd No Ans Deact Code	Cancels call forwarding of no-answer calls. The default is *93.
Cfwd Last Act Code	Forwards the last inbound or outbound calls to the extension specified after the activation code. The default is *63.
Cfwd Last Deact Code	Cancels call forwarding of the last inbound or outbound calls. The default is *83.
Block Last Act Code	Blocks the last inbound call. The default is *60.
Block Last Deact Code	Cancels blocking of the last inbound call. The default is *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. The default is *64.
Accept Last Deact Code	Cancels the code to accept the last outbound call. The default is *84.
CW Act Code	Enables call waiting on all calls. The default is *56.
CW Deact Code	Disables call waiting on all calls. The default is *57.

Field	Description
CW Per Call Act Code	Enables call waiting for the next call. The default is *71.
CW Per Call Deact Code	Disables call waiting for the next call. The default is *70.
Block CID Act Code	Blocks caller ID on all outbound calls. The default is *67.
Block CID Deact Code	Removes caller ID blocking on all outbound calls. The default is *68.
Block CID Per Call Act Code	Blocks caller ID on the next outbound call. The default is *81.
Block CID Per Call Deact Code	Removes caller ID blocking on the next inbound call. The default is *82.
Block ANC Act Code	Blocks all anonymous calls. The default is *77.
Block ANC Deact Code	Removes blocking of all anonymous calls. The default is *87.
DND Act Code	Enables the do not disturb feature. The default is *78.
DND Deact Code	Disables the do not disturb feature. The default is *79.
CID Act Code	Enables caller ID generation. The default is *65.
CID Deact Code	Disables caller ID generation. The default is *85.
CWCID Act Code	Enables call waiting, caller ID generation. The default is *25.
CWCID Deact Code	Disables call waiting, caller ID generation. The default is *45.
Dist Ring Act Code	Enables the distinctive ringing feature. The default is *26
Dist Ring Deact Code	Disables the distinctive ringing feature. The default is *46.
Speed Dial Act Code	Assigns a speed dial number. The default is *74.
Secure All Call Act Code	Makes all outbound calls secure. The default is *16.

Field	Description
Secure No Call Act Code	Makes all outbound calls not secure. The default is *17.
Secure One Call Act Code	Makes the next outbound call secure. (It is redundant if all outbound calls are secure by default.) The default is *18.
Secure One Call Deact Code	Makes the next outbound call not secure. (It is redundant if all outbound calls are not secure by default.) The default is *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.
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.
Modem Line Toggle Code	Toggles the line to a modem. The default is *99. Modem pass-through mode can be triggered only by pre-dialing this code.
FAX Line Toggle Code	Toggles the line to a fax machine. The default is #99.

<table>
<tr><th>Field</th><th>Description</th></tr>
<tr><td>Referral Services Codes</td><td>These codes tell the WRP500 what to do when the user places the current call on hold and is listening to the second dial tone.

One or more *code can be configured into this parameter, such as *98, or *97|*98|*123, etc. Max total length is 79 chars. This parameter applies when the user places the current call on hold (by Hook Flash) and is listening to second dial tone. Each *code (and the following valid target number according to current dial plan) entered on the second dial-tone triggers the WRP500 to perform a blind transfer to a target number that is preceded by the service *code.

For example, after the user dials *98, the WRP500 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 WRP500 sends a blind REFER to the holding party with the Refer-To target equals to *98 target_number. This feature allows the WRP500 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 WRP500. You can empty the corresponding *code that you do not want the WRP500 to process.</td></tr>
</table>

<table>
<tr><th>Field</th><th>Description</th></tr>
<tr><td>Feature Dial Services Codes</td><td>These codes tell the WRP500 what to do when the user is listening to the first or second dial tone.

One or more *code can be configured into this parameter, such as *72, or *72|*74|*67|*82, etc. Max total length is 79 chars. This parameter applies when the user has a dial tone (first or second dial tone). Enter *code (and the following target number according to current dial plan) entered at the dial tone triggers the WRP500 to call the target number preceded by the *code. For example, after user dials *72, the WRP500 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 WRP500 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 WRP500. You can empty the corresponding *code that you do not want to the WRP500 to process.

You can add a parameter to each *code in Features Dial Services Codes to indicate what tone to play after the *code is entered, such as *72‘c‘|*67‘p‘. Below are a list of allowed tone parameters (note the use of back quotes surrounding the parameter w/o 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 WRP500 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 it in this parameter. In that case, simple add that *code in the dial plan and the WRP500 send INVITE *73@..... as usual when user dials *73.</td></tr>
</table>

Outbound Call Codec Selection Codes section

These codes are automatically appended to the dial plan. Thus, they do not need to be included in the dial plan, but there is no harm in doing so.

This table describes the fields in the Outbound Call Codec Section Codes section of the Voice tab > Regional page.

Field	Description
Prefer G711u Code	Makes this codec the preferred codec for the associated call. The default is *017110.
Force G711u Code	Makes this codec the only codec that can be used for the associated call. The default is *027110.
Prefer G711a Code	Makes this codec the preferred codec for the associated call. The default is *017111
Force G711a Code	Makes this codec the only codec that can be used for the associated call. The default is *027111.
Prefer G729a Code	Makes this codec the preferred codec for the associated call. The default is *01729.
Force G729a Code	Makes this codec the only codec that can be used for the associated call. The default is *02729.

Miscellaneous section

This table describes the fields in the Miscellaneous section of the Voice tab > Regional page.

Field	Description
Set Local Date (mm/dd)	Sets the local date (mm stands for months and dd stands for days). The year is optional and uses two or four digits.
Set Local Time (HH/mm)	Sets the local time (hh stands for hours and mm stands for minutes). Seconds are optional.
FXS Port Impedance	Sets the electrical impedance of the FXS port. Choices are 600, 900, 600+2.16uF, 900+2.16uF, 270+750||150nF, 220+850||120nF, 220+820||115nF, or 200+600||100nF. The default is 600.
FXS Port Input Gain	Input gain in dB, up to three decimal places. The range is 6.000 to -12.000. The default is -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. The default is -3.
DTMF Playback Level	Local DTMF playback level in dBm, up to one decimal place. The default is -7.3.

Field	Description
DTMF Playback Length	Local DTMF playback duration in milliseconds. The default is .1.
DTMF Playback Twist	Local DTMF playback duration. The default is 1.3.
Caller ID Method	The following choices are available: • **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 CPE 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 CPE after DTAS for CIDCW. Polarity reversal is applied only if equipment is on hook. The default is Bellcore(N.Amer, China).
Caller ID FSK Standard	The WRP500 supports bell 202 and v.23 standards for caller ID generation. Select the FSK standard you want to use, bell 202 or v.23. The default is bell 202.
Feature Invocation Method	Select the method you want to use, Default or Sweden default. The default is Default.

Line page

You can use the *Voice tab* > *Line* page to configure the lines for voice service. This page includes the following sections:

- Line Enable section, page A-25
- Streaming Audio Server (SAS) section, page A-25
- NAT Settings section, page A-26
- Network Settings section, page A-27
- SIP Settings section, page A-28

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

Line Enable section

This table describes the fields in the Line Enable section of the Voice tab > Line page.

Field	Description
Line Enable	To enable this line for service, select yes. Otherwise, select no. The default is **yes**.

Streaming Audio Server (SAS) section

This table describes the fields in the Streaming Audio Server (SAS) section of the Voice tab > Line page.

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. The default is **no**.

Field	Description
SAS DLG Refresh Intvl	If this value is not zero, it is the interval at which the streaming audio server sends out session refresh (SIP re-INVITE) messages to determine whether the connection to the caller is still active. If the caller does not respond to the refresh message, the WRP500 ends this call with a SIP BYE message. The range is 0 to 255 seconds (0 means that the session refresh is disabled). The default is 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. The default value is empty.

NAT Settings section

This table describes the fields in the NAT Settings section of the Voice tab > Line page.

Field	Description
NAT Mapping Enable	To use externally mapped IP addresses and SIP/RTP ports in SIP messages, select yes. Otherwise, select no. The default is **no**.
NAT Keep Alive Enable	To send the configured NAT keep alive message periodically, select yes. Otherwise, select no. The default is **no**.

Field	Description
NAT Keep Alive Msg	Enter the keep alive message that should be sent periodically to maintain the current NAT mapping. If the value is $NOTIFY, a NOTIFY message is sent. If the value is $REGISTER, a REGISTER message without contact is sent. The default is $**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. The default is $**PROXY**.

Network Settings section

This table describes the fields in the Network Settings section of the Voice tab > Line page.

Field	Description
SIP ToS/DiffServ Value	TOS/DiffServ field value in UDP IP packets carrying a SIP message. The default is **0x68**.
SIP CoS Value [0-7]	CoS value for SIP messages. The default is **3**.
RTP ToS/DiffServ Value	ToS/DiffServ field value in UDP IP packets carrying RTP data. The default is **0xb8**.
RTP CoS Value [0-7]	CoS value for RTP data. The default is **6**.
Network Jitter Min/Max	Determines how jitter buffer range of WRP500 when Network Jitter Mode is adaptive. Jitter buffer size is adjusted dynamically. The default value of Network Jitter Min is **10ms**. The default value of Network Jitter Max is **200ms**.
Network Jitter Mode	Specify whether the jitter buffer should be adjusted or use some constant interval value. Select the appropriate setting: **adaptive**, **static**. The default is **adaptive**.

SIP Settings section

This table describes the fields in the SIP Settings section of the Voice tab > Line page.

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**.
SIP Port	Port number of the SIP message listening and transmission port. The default is **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. The default is **no**.
EXT SIP Port	The external SIP port number.
Auth Resync-Reboot	If this feature is enabled, the WRP500 authenticates the sender when it receives the NOTIFY resync reboot (RFC 2617) message. To use this feature, select yes. Otherwise, select no. The default is **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.
SIP Remote-Party-ID	To use the Remote-Party-ID header instead of the From header, select yes. Otherwise, select no. The default is **yes**.
SIP GUID	The Global Unique ID is generated for each line for each device. When it is enabled, the WRP500 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. The default is **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. The default is none.
RTP Log Intvl	The interval for the RTP log. The default value is 0.
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 WRP500 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). The default is **no**.
Referor Bye Delay	Controls when the WRP500 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. The default is **4**.
Refer Target Bye Delay	For the Refer Target Bye Delay, enter the appropriate period of time in seconds. The default is **0**.

Field	Description
Referee Bye Delay	For the Referee Bye Delay, enter the appropriate period of time in seconds. The default is **0**.
Refer-To Target Contact	To contact the refer-to target, select yes. Otherwise, select no. The default is **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. The default is **no**.
Auth INVITE	When enabled, authorization is required for initial incoming INVITE requests from the SIP proxy.
Use Anonymous With RPID	Set value of Remote Party ID to “anonymous, yes”
Use Local Addr in FROM	Use IP address in From header, no
Reply 182 On Call Waiting	Send 182 response when enter call waiting, no

Call Feature Settings section

This table describes the fields in the Call Feature Settings section of the Voice tab > Line page.

Field	Description
Blind Attn-Xfer Enable	Enables the WRP500 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 WRP500 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. The default is **no**.
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. The default is **yes**.
MoH server	Address of music on hold server
Conference Bridge URL	URL of Conference server

Proxy and Registration section

This table describes the fields in the Proxy and Registration section of the Voice tab > Line page.

Field	Description
Proxy	SIP proxy server for all outbound requests.
Outbound Proxy	SIP Outbound Proxy Server where all outbound requests are sent as the first hop.
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. The default is **no**.
Use OB Proxy In Dialog	Whether to force SIP requests to be sent to the outbound proxy within a dialog. Ignored if the parameter *Use Outbound Proxy* is no, or the *Outbound Proxy* parameter is empty. The default is **yes**.
Register	Enable periodic registration with the *Proxy* parameter. This parameter is ignored if *Proxy* is not specified. The default is **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. The default is **no**.
Register Expires	Allow answering inbound calls without successful (dynamic) registration by the unit. If proxy responded to REGISTER with a smaller Expires value, the WRP500 will renew registration based on this smaller value instead of the configured value. If registration failed with an Expires too brief error response, the WRP500 will retry with the value given in the Min-Expires header in the error response. The default is **3600**.
Ans Call Without Reg	Expires value in sec in a REGISTER request. The WRP500 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
Use DNS SRV	Whether to use DNS SRV lookup for Proxy and Outbound Proxy. The default is **no**.
DNS SRV Auto Prefix	If enabled, the WRP500 will automatically prefix the Proxy or Outbound Proxy name with _sip._udp when performing a DNS SRV lookup on that name. The default is **no**.

Field	Description
Proxy Fallback Intvl	This parameter sets the delay (sec) after which the WRP500 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 WRP500 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 WRP500 will not attempt to fall back after a fail over). The default is **3600**
Proxy Redundancy Method	The WRP500 will make an internal list of proxies returned in DNS SRV records. In normal mode, this list will contain proxies ranked by weight and priority. if Based on SRV port is configured the WRP500 does normal first, and also inspect the port number based on 1st proxy's port on the list. The default is **Normal**.
Voice Mail Server	Enter the URL or IP address of the server.
Mailbox Subscribe Expires	Expiry time to the voice mail server. The time to send another subscribe message to the voice mail server. The default is 2147483647.

Subscriber Information section

This table describes the fields in the Subscriber Information section of the Voice tab > Line page.

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. The default is **no**.
Auth ID	Authentication ID for SIP authentication.
Directory Number	Enter the number for this line.

Supplementary Service Subscription section

The WRP500 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 WRP500.

This table describes the fields in the Supplementary Service Subscription section of the Voice tab > Line page.

Field	Description
Call Waiting Serv	Enable Call Waiting Service. The default is **yes**.
Block CID Serv	Enable Block Caller ID Service. The default is **yes**.
Block ANC Serv	Enable Block Anonymous Calls Service The default is **yes**.
Dist Ring Serv	Enable Distinctive Ringing Service The default is **yes**.
Cfwd All Serv	Enable Call Forward All Service The default is **yes**.
Cfwd Busy Serv	Enable Call Forward Busy Service The default is **yes**.
Cfwd No Ans Serv	Enable Call Forward No Answer Service The default is **yes**.
Cfwd Sel Serv	Enable Call Forward Selective Service The default is **yes**.
Cfwd Last Serv	Enable Forward Last Call Service The default is **yes**.
Block Last Serv	Enable Block Last Call Service The default is **yes**.
Accept Last Serv	Enable Accept Last Call Service The default is **yes**.
DND Serv	Enable Do Not Disturb Service The default is **yes**.
CID_Serv	Enable Caller ID Service The default is **yes**.
CWCID Serv	Enable Call Waiting Caller ID Service The default is **yes**.
Call Return Serv	Enable Call Return Service The default is **yes**.
Call Redial Serv	Enable Call Redial Service.
Call Back Serv	Enable Call Back Service.

Field	Description
Three Way Call Serv	Enable Three Way Calling Service. Three Way Calling is required for Three Way Conference and Attended Transfer. The default is **yes**.
Three Way Conf Serv	Enable Three Way Conference Service. Three Way Conference is required for Attended Transfer. The default is **yes**.
Attn Transfer Serv	Enable Attended Call Transfer Service. Three Way Conference is required for Attended Transfer. The default is **yes**.
Unattn Transfer Serv	Enable Unattended (Blind) Call Transfer Service. The default is **yes**.
MWI Serv	Enable MWI Service. MWI is available only if a Voice Mail Service is set-up in the deployment. The default is **yes**.
VMWI Serv	Enable VMWI Service (FSK). The default is **yes**.
Speed Dial Serv	Enable Speed Dial Service. The default is **yes**.
Secure Call Serv	Enable Secure Call Service. The default is **yes**.
Referral Serv	Enable Referral Service. See the *Referral Services Codes* parameter for more details. The default is **yes**.
Feature Dial Serv	Enable Feature Dial Service. See the *Feature Dial Services Codes* parameter for more details. The default is **yes**.
Service Announcement Serv	Enable Service Announcement Service. The default is **no**.

Audio Configuration section

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 G.729a 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.

This table describes the fields in the Audio Configuration section of the Voice tab > Line page.

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**, **G729a.** The default is **G711u.**
Second Preferred Codec	Second 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: **Unspecified**, **G711u, G711a, G729a.** The default is **Unspecified.**
Third Preferred Codec	Third 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: **Unspecified, G711u, G711a, G729a.** The default is **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. The default is **no**.
Silence Supp Enable	To enable silence suppression so that silent audio frames are not transmitted, select yes. Otherwise, select no. The default is **no**.
G729a Enable	To enable the use of the G.729a codec at 8 kbps, select yes. Otherwise, select no. The default is **yes**.
Echo Canc Enable	To enable the use of the echo canceler, select yes. Otherwise, select no. The default is **yes**.
Echo Supp Enable	To enable the use of the echo suppressor, select yes. Otherwise, select no. The default is **yes**.
FAX CED Detect Enable	To enable detection of the fax Caller-Entered Digits (CED) tone, select yes. Otherwise, select no. The default is **yes**.
FAX V21 Detect Enable	To enable detection of the fax v.21 signal, select yes. Otherwise, select no. The default is **yes**.
FAX Passthru Codec	Select the codec for fax passthrough, G711u or G711a. The default is **G711u**.
DTMF Process INFO	To use the DTMF process info feature, select yes. Otherwise, select no. The default is **yes**.
FAX Codec Symmetric	To force the ATA to use a symmetric codec during fax passthrough, select yes. Otherwise, select no. The default is **yes**.

Field	Description
FAX Passthru Method	Select the fax passthrough method: None, NSE, or ReINVITE. The default is **NSE**.
DTMF Tx Method	Select the method to transmit DTMF signals to the far end: **InBand, AVT, INFO, Auto.** InBand sends DTMF using the audio path. AVT sends DTMF as events. INFO uses the SIP INFO method. Auto uses InBand or AVT based on the outcome of codec negotiation. The default is **Auto**.
FAX Process NSE	To use the fax process NSE feature, select yes. Otherwise, select no. The default is **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. The default is **None**.
Release Unused Codec	This feature allows the release of codecs not used after codec negotiation on the first call, so that other codecs can be used for the second line. To use this feature, select yes. Otherwise, select no. The default is **yes**.
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. The default is **1**.
FAX Tone Detect Mode	If you want the Gateway to detect the fax tone whether the Gateway is a caller or callee, select caller or callee. If you want the Gateway to detect the fax tone only if the Gateway is the caller, select caller only. If you want the Gateway to detect the fax tone only if the Gateway is the callee, select callee only. This parameter has three possible values: caller or callee - The WRP500 will detect FAX tone whether it is callee or caller caller only - The WRP500 will detect FAX tone only if it is the caller callee only - The WRP500 will detect FAX tone only if it is the callee The default is **caller or callee**.
FAX Enable T38	Set to yes to enable fax T.38 mode
FAX T38 ECM Enable	Set to yes to enable T38 error correction mode

Dial Plan section

The default dial plan script for each line is as follows:

(*xx|[3469]11|0|00|[2-9]xxxxxx|1xxx[2-9]xxxxxx|xxxxxxxxxxxx.).

These tables describe the fields in the Dial Plan section of the Voice tab > Line page, which provide the syntax for a dial plan expression.

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

<table>
<tr><th>Field</th><th>Description</th></tr>
<tr><td>Dial Plan</td><td>Dial plan script for this line.

The default is
(*xx|[3469]11|0|00|[2-9]xxxxxx|1xxx[2-9]xxxxxxS0|xxxxxxxxxxxx.)

Each parameter is separated by a semi-colon (;).

Example 1:

<code>*1xxxxxxxxxx<:@fwdnat.pulver.com:5082;uid=jsmith;pwd=xyz</code>

Example 2:

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

Example 3:

<code>[39]11<:@gw0></code></td></tr>
<tr><td>Enable IP Dialing</td><td>Enable or disable IP dialing.

If IP dialing is enabled, one can dial [user-id@]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.

The default is no.</td></tr>
<tr><td>Emergency Number</td><td>Comma separated list of emergency number patterns. If outbound call matches one of the pattern, the WRP500 will disable hook flash event handling. The condition is restored to normal after the phone is on-hook. Blank signifies no emergency number. Maximum number length is 63 characters.

The default is blank.</td></tr>
</table>

FXS Port Polarity Configuration section

This table describes the fields in the FXS Port Polarity Configuration section of the Voice tab > Line page.

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

User page

You can use this page to configure the user settings. This page includes the following sections:

When a call is made from Line 1 or Line 2, the WRP500 uses the user and line settings for that line; there is no user login support. Per user parameter tags must be appended with [1] or [2] (corresponding to line 1 or 2) in the configuration profile. It is omitted below for readability.

Call Forward Settings section

This table describes the fields in the Call Forward Settings section of the Voice tab > User page.

Field	Description
Cfwd All Dest	Forward number for Call Forward All Service The default is blank.
Cfwd Busy Dest	Forward number for Call Forward Busy Service. Same as Cfwd All Dest. The default is blank.

Field	Description
Cfwd No Ans Dest	Forward number for Call Forward No Answer Service. Same as Cfwd All Dest. The default is blank.
Cfwd No Ans Delay	Delay in sec before Call Forward No Answer triggers. Same as Cfwd All Dest. The default is **20**.

Selective Call Forward Settings section

This table describes the fields in the Selective Call Forward Settings section of the Voice tab > User page.

Field	Description
Cfwd Sel1- 8 Caller	Caller number pattern to trigger Call Forward Selective 1, 2, 3, 4, 5, 6, 7, or 8. The default is blank.
Cfwd Sel1 - 8 Dest	Forward number for Call Forward Selective 1, 2, 3, 4, 5, 6, 7, or 8. Same as Cfwd All Dest. The default is blank.
Block Last Caller	ID of caller blocked via the Block Last Caller service. The default is blank.
Accept Last Caller	ID of caller accepted via the Accept Last Caller service. The default is blank.
Cfwd Last Caller	The Caller number that is actively forwarded to *Cfwd Last Dest* by using the Call Forward Last activation code The default is blank.
Cfwd Last Dest	Forward number for the *Cfwd Last Caller* parameter. Same as Cfwd All Dest. The default is blank.

Speed Dial Settings section

This table describes the fields in the Speed Dial Settings section of the Voice tab > User page.

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

Supplementary Service Settings section

The WRP500 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 WRP500.

This table describes the fields in the Supplementary Service Settings section of the Voice tab > User page.

Field	Description
CW Setting	Call Waiting on/off for all calls. The default is **yes**.
Block CID Setting	Block Caller ID on/off for all calls. The default is **no**.
Block ANC Setting	Block Anonymous Calls on or off. The default is **no**.
DND Setting	DND on or off. The default is **no**.
CID Setting	Caller ID Generation on or off. The default is **yes**.
CWCID Setting	Call Waiting Caller ID Generation on or off. The default is **yes**.
Dist Ring Setting	Distinctive Ring on or off. The default is **yes**.
Secure Call Setting	If yes, all outbound calls are secure calls by default. The default is **no**.
Message Waiting	This value is updated when there is voice mail notification received by the WRP500. The user can also manually modify it to clear or set the flag. 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. The default is **no**.
Accept Media Loopback Request	Controls how to handle incoming requests for loopback operation. Choices are: **Never**, **Automatic**, and **Manual**, where: • **never**—never accepts loopback calls; reply 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. The default is **Automatic**.

Field	Description
Media Loopback Mode	The loopback mode to assume locally when making call to request media loopback. Choices are: **Source** and **Mirror.** Default is **Source**. Note that if the WRP500 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 is **Media**. Note that if the WRP500 answers the call, then the loopback type is determined by the caller (the WRP500 always picks the first loopback type in the offer if it contains multiple types.)

Distinctive Ring Settings section

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.

This table describes the fields in the Distinctive Ring Settings section of the Voice tab > User page.

Field	Description
Ring1 - 8 Caller	Caller number pattern to play Distinctive Ring/CWT 1, 2, 3, 4, 5, 6, 7, 8. The default is **blank**.

Ring Settings section

This table describes the fields in the Ring Settings section of the Voice tab > User page.

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