# Configure Voice Services

This chapter describes how to configure your WRP500 to meet customer requirements for voice services.

Analog Telephone Adapter Operations

The WRP500 is equipped with a built-in Analog Telephone Adapter (ATA). An ATA is an intelligent low-density Voice over IP (VoIP) gateway that enables carrier-class residential and business IP Telephony services that are delivered over broadband or high-speed Internet connections. Users can access Internet phone services through standard analog telephone equipment.
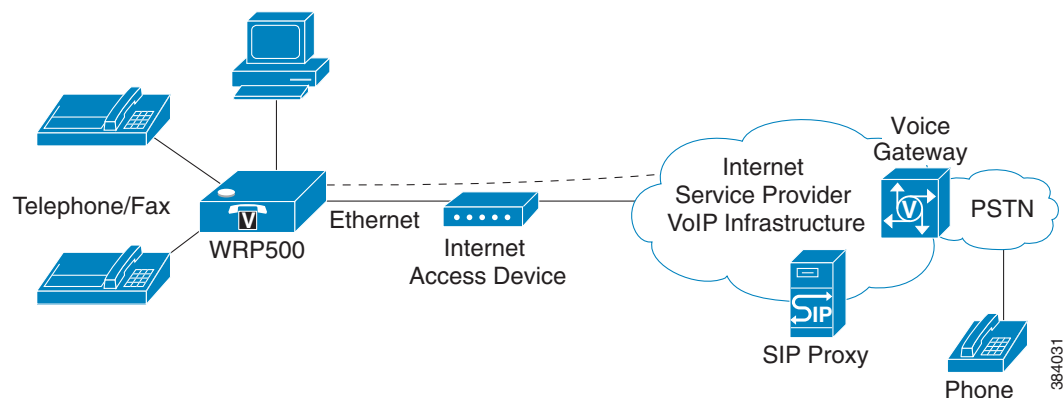
The WRP500 maintains the state of each call it terminates and reacts properly to user input events (such as on/off hook or hook flash). The WRP500 uses the Session Initiation Protocol (SIP) open standard, so little or no involvement by a "middle-man" server or media gateway controller occurs. SIP allows interoperation with all Internet telephony service providers (ITSPs) that support SIP.

ATA Software Features

The WRP500 is equipped with a full featured, fully programmable ATA that can be custom provisioned within a wide range of configuration parameters. These sections describe the factors that contribute to voice quality:

- Supported Codecs, page 3-2
- SIP Proxy Redundancy, page 3-2
- Other ATA Software Features, page 3-3

Supported Codecs

The WRP500 supports the following codecs:

- G.711u (configured by default) and G.711a

  G.711 (A-law and mu-law) are very low complexity codecs that support uncompressed 64 kbps digitized voice transmissions at one through ten 5-millisecond voice frames per packet. This codec provides the highest voice quality and uses the most bandwidth of any of the available codecs.

- G.729a

  The ITU G.729 voice coding algorithm is used to compress digitized speech. G.729a is a reduced complexity version of G.729. It requires about half the processing power as compared to G.729. The G.729 and G.729a bit streams are compatible and interoperable, but not identical.

The administrator can select the preferred codecs to be used for each line. See the "Audio Configuration section" section on page A-34.

In addition, negotiation of the optimal voice codec sometimes depends on the ability of an ATA to match a codec name with the codec that the far-end device uses. You can individually name the various codecs so that the WRP500 can successfully negotiate the codec with the far-end equipment. For more information, see the "Audio Configuration section," on page 34.

SIP Proxy Redundancy

In typical commercial IP Telephony deployments, all calls are established through a SIP proxy server. An average SIP proxy server may handle thousands of subscribers. It is important that a backup server be available so that an active server can be temporarily switched out for maintenance. The WRP500 supports the use of backup SIP proxy servers (via DNS SRV) so that service disruption should be nearly eliminated.

A relatively simple way to support proxy redundancy is to configure your DNS server with a list of SIP proxy addresses. The WRP500 can be instructed to contact a SIP proxy server in a domain named in the SIP message. The WRP500 consults the DNS server to get a list of hosts in the given domain that provides SIP services. If an entry exists, the DNS server returns an SRV record that contains a list of SIP proxy servers for the domain, with their host names, priority, listening ports, and so on. The WRP500 tries to contact the list of hosts in the order of their stated priority.

If the WRP500 is currently using a lower priority proxy server, it periodically probes the higher priority proxy to check whether it is back on line, and switches back to the higher priority proxy when possible. SIP Proxy Redundancy is configured in the Line and PSTN Line pages in the Configuration Utility.

Other ATA Software Features

Table 3-1 summarizes other features that the WRP500 provides.

Table 3-1   ATA Software Features

Feature	Description
Silence Suppression	See "Silence Suppression and Comfort Noise Generation" section on page 3-10.
Modem and Fax Pass-Through	• Modem pass-through mode can be triggered only by predialing the number that is set in the Modem Line Toggle Code. (Set in the Regional tab.) • FAX pass-through mode is triggered by a CED/CNG tone or by an NSE event. • Echo canceler is automatically disabled for Modem pass-through mode.
Adaptive Jitter Buffer	The WRP500 can buffer incoming voice packets to minimize out-of-order packet arrival. This process is known as jitter buffering. The jitter buffer size proactively adjusts or adapts in size, depending on changing network conditions. The WRP500 has a Network Jitter Level control setting for each line of service. The jitter level determines how aggressively the WRP500 tries to shrink the jitter buffer over time to achieve a lower overall delay. If the jitter level is higher, it shrinks more gradually. If jitter level is lower, it shrinks more quickly. Adaptive Jitter Buffer is configured in the Line and PSTN Line tabs. See Appendix A, "Advanced Voice Fields."
International Caller ID Delivery	In addition to support of the Bellcore (FSK) and Swedish/Danish (DTMF) methods of Caller ID (CID) delivery, ATAs provide a large subset of ETSI-compliant methods to support international CID equipment. International CID is configured in the Line and PSTN Line tabs. See Appendix A, "Advanced Voice Fields."
Secure Calls	A user (if enabled by service provider or administrator) has the option to make an outbound call secure in the sense that the audio packets in both directions are encrypted. See the "Secure Call Implementation" section on page 3-17.
Adjustable Audio Frames Per Packet	This feature allows the user to set the number of audio frames that are contained in one RTP packet. Packets can be adjusted to contain audio frames of 10ms to 30ms in length. Increasing the time of packets decreases the bandwidth utilized, but it also increases delay and may affect voice quality. See the RTP Packet Size parameter found in the SIP tab in Appendix A, "Advanced Voice Fields."
DTMF	The WRP500 may relay DTMF digits as out-of-band events to preserve the fidelity of the digits. This can enhance the reliability of DTMF transmission that many IVR applications, such as dial-up banking and airline information, require. DTMF is configured in the *DTMF Tx Mode* parameter that is found in the Line tabs. See Appendix A, "Advanced Voice Fields."

Feature	Description
Call Progress Tone Generation	The WRP500 has configurable call progress tones. Call progress tones are generated locally on the WRP500 so an end user is advised of status (such as ringback). Parameters for each type of tone (for instance, a dial tone that is played back to an end user) may include frequency and amplitude of each component, and cadence information. See the Regional tab in Appendix A, "Advanced Voice Fields."
Call Progress Tone Pass Through	This feature allows the user to hear the call progress tones (such as ringing) that are generated from the far-end network. See the Regional tab in Appendix A, "Advanced Voice Fields."
Echo Cancellation	Impedance mismatch between the telephone and the IP Telephony gateway phone port can lead to near-end echo. The WRP500 has a near-end echo canceler that compensates for impedance match. The WRP500 also implements an echo suppressor with comfort noise generator (CNG) so that any residual echo is not noticeable. Echo Cancellation is configured in the Regional, Line, and PSTN Line tabs. See Appendix A, "Advanced Voice Fields."
Signaling Hook Flash Event	The WRP500 can signal hook flash events to the remote party on a connected call. This feature can be used to provide advanced mid-call services with third-party-call-control. Depending on the features that the service provider offers using third-party-call-control, the following ATA features may be disabled to correctly signal a hook-flash event to the softswitch: • Call Waiting Service (parameter *call waiting serv* set in the Line tab) • Three Way Conference Service (parameter *three-way conf serv* set in the Line tab) • Three Way Call Service (parameter *three-way call serv* set in the Line tab) You can configure the length of time allowed for detection of a hook flash using the Hook Flash Timer parameter on the Regional tab of the Configuration Utility. See Appendix A, "Advanced Voice Fields."
Configurable Dial Plan with Interdigit Timers	The WRP500 has three configurable interdigit timers: • Initial timeout (T)—Signals that the handset is off the hook and that no digit has been pressed yet. • Long timeout (L)—Signals the end of a dial string; that is, no more digits are expected. • Short timeout (S)—Used between digits; that is, after a digit is pressed, a short timeout prevents the digit from being recognized a second time. See "Configure Dial Plans" section on page 3-10 for more information.
Polarity Control	The WRP500 allows the polarity to be set when a call is connected and when a call is disconnected. This feature is required to support some pay phone system and answering machines. Polarity Control is configured in the Line and PSTN Line tabs. See Appendix A, "Advanced Voice Fields."

Feature	Description
Calling Party Control	Calling Party Control (CPC) signals to the called party equipment that the calling party has hung up during a connected call by removing the voltage between the tip and ring momentarily. This feature is useful for auto-answer equipment, which then knows when to disengage. CPC is configured in the Regional, Line, and PSTN Line tabs. See Appendix A, "Advanced Voice Fields."
Syslog and Debug Server Records	Syslog and Debug Sever Records log more details than Report Generation and Event Logging. Using the configuration parameters, the WRP500 allows you to select which type of activity/events should be logged. Syslog and Debug Server allow the information to be sent to a Syslog Server. Syslog and Debug Server Records are configured in the System, Line, and PSTN Line tabs. See Appendix A, "Advanced Voice Fields."
SIP Over TLS	The WRP500 allows the use of SIP over Transport Layer Security (TLS). SIP over TLS is designed to eliminate the possibility of malicious activity by encrypting the SIP messages of the service provider and the end user. SIP over TLS relies on the widely deployed and standardized TLS protocol. SIP Over TLS encrypts only the signaling messages and not the media. A separate secure protocol, such as Secure Real-Time Transport Protocol (SRTP), can be used to encrypt voice packets. SIP over TLS is configured in the SIP Transport parameter configured in the Line tab(s). See Appendix A, "Advanced Voice Fields."

Register to the Service Provider

To use VoIP phone service, you must configure your WRP500 to the Internet Telephony Service Provider (ITSP).

Note   Each line tab must be configured separately. Each line tab can be configured for a different ITSP.

Step 1   Log in as administrator.

Step 2   Under the **Voice** menu, click **Line 1** or **Line 2** to choose the line interface that you want to modify.

Step 3   In the **Proxy and Registration** section, enter the **Proxy.**

Step 4   In the **Subscriber Information** section, enter the **User ID** and **Password.**

Proxy and Registration

Proxy:	10.74.51.158		
Outbound Proxy:			
Use Outbound Proxy:	no	Use OB Proxy In Dialog:	yes
Register:	yes	Make Call Without Reg:	no
Register Expires:	3600	Ans Call Without Reg:	no
Use DNS SRV:	no	DNS SRV Auto Prefix:	no
Proxy Fallback Intvl:	3600	Proxy Redundancy Method:	Normal
Voice Mail Server:		Mailbox Subscribe Expires:	2147483647

Subscriber Information

Display Name:		User ID:	8205
Password:		Use Auth ID:	no
Auth ID:		Directory Number:	

> **Note** These are the minimum settings for most ITSP connections. Enter the account information as required by your ITSP.

Step 5 Click **Submit.** The devices reboot.

Step 6 To verify your progress, perform the following tasks:

- Under the **Voice** menu, click **Info**. Scroll down to the **Line 1 Status** or **Line 2 Status** section of the page, depending on which line you configured. Verify that the line is registered. Refer to the following example.

Line 1 Status

Hook State:	On	Registration State:	Registered
Last Registration At:	1/27/2015 04:33:05	Next Registration In:	3553 s
Message Waiting:	No	Call Back Active:	No

- Use an external phone to place an inbound call to the telephone number that was assigned by your ITSP. Assuming that you have left the default settings in place, the phone should ring and you can pick up the phone to get two-way audio.

- If the line is not registered, you may need to refresh the browser several times because it can take a few seconds for the registration to succeed. Also verify that your DNS is configured properly.
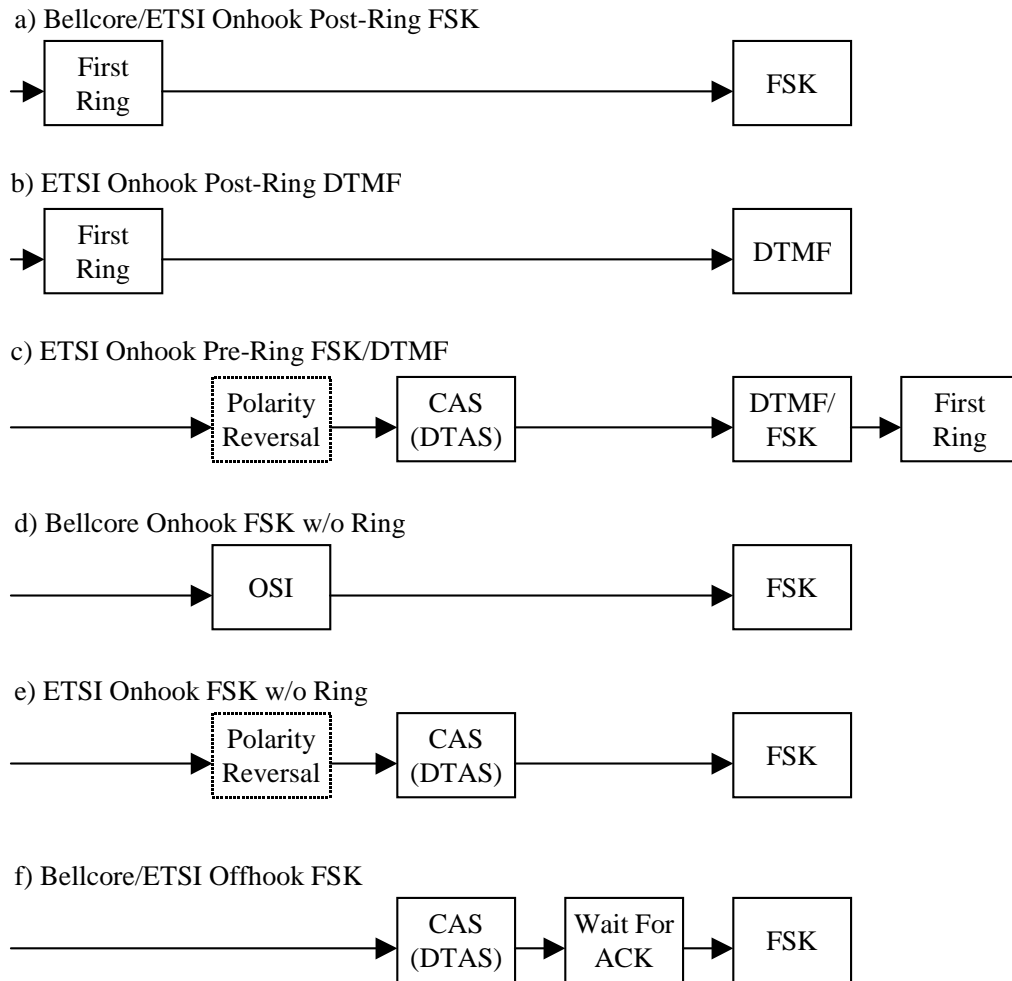
Manage Caller ID Service

The choice of caller ID (CID) method is dependent on your area/region. To configure CID, use the following parameters:

Parameter	Tab	Description and Value
Caller ID Method	Regional	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	Regional	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.**

Three types of Caller ID exist:

- On Hook Caller ID Associated with Ringing — This type of Caller ID is used for incoming calls when the attached phone is on hook. See the following figure (a) – (c). All CID methods can be applied for this type of CID.

- On Hook Caller ID Not Associated with Ringing — This feature is used to send VMWI signal to the phone to turn the message waiting light on and off. See the following figure (d) and (e). This is available only for FSK-based CID methods: Bellcore, ETSI FSK, and ETSI FSK With PR.

- Off Hook Caller ID — This is used to delivery caller-id on incoming calls when the attached phone is off hook. (See the following figure.) This can be call waiting caller ID (CIDCW) or to notify the user that the far-end party identity has changed or updated (such as due to a call transfer). This is available only for FSK-based CID methods: Bellcore, ETSI FSK, and ETSI FSK With PR.

a) Bellcore/ETSI Onhook Post-Ring FSK

```
→  ┌─────────┐                    ┌─────────┐
   │  First  │────────────────────│   FSK   │
   │  Ring   │                    │         │
   └─────────┘                    └─────────┘
```

b) ETSI Onhook Post-Ring DTMF

```
→  ┌─────────┐                    ┌─────────┐
   │  First  │────────────────────│  DTMF   │
   │  Ring   │                    │         │
   └─────────┘                    └─────────┘
```

c) ETSI Onhook Pre-Ring FSK/DTMF

```
   ┌··········┐   ┌─────────┐     ┌─────────┐   ┌─────────┐
───┊ Polarity ┊───│  CAS    │─────│  DTMF/  │───│  First  │
   ┊ Reversal ┊   │ (DTAS)  │     │  FSK    │   │  Ring   │
   └··········┘   └─────────┘     └─────────┘   └─────────┘
```

d) Bellcore Onhook FSK w/o Ring

```
   ┌─────────┐                    ┌─────────┐
───│  OSI    │────────────────────│   FSK   │
   │         │                    │         │
   └─────────┘                    └─────────┘
```

e) ETSI Onhook FSK w/o Ring

```
   ┌··········┐   ┌─────────┐     ┌─────────┐
───┊ Polarity ┊───│  CAS    │─────│   FSK   │
   ┊ Reversal ┊   │ (DTAS)  │     │         │
   └··········┘   └─────────┘     └─────────┘
```

f) Bellcore/ETSI Offhook FSK

```
             ┌─────────┐  ┌─────────┐  ┌─────────┐
─────────────│  CAS    │──│ Wait For│──│   FSK   │
             │ (DTAS)  │  │  ACK    │  │         │
             └─────────┘  └─────────┘  └─────────┘
```

Optimize Fax Completion Rates

Issues can occur with fax transmissions over IP networks, even with the T.38 standard, which is supported by the WRP500. You can adjust several settings on your WRP500 to optimize your fax completion rates.

Note   Only T.38 Fax is supported. The WRP500 supports one connection.

Step 1   Ensure that you have enough bandwidth for the uplink and the downlink:

- For G.711 fallback, approximately 100 kbps are recommended.
- For T.38, allocate at least 50 kbps.

Step 2   To optimize G.711 fallback fax completion rates, set the following on the Line tab of your ATA device:

- **Call Waiting Serv:** no
- **Three Way Call Serv:** no

- **Preferred Codec:** G.711
- **Use pref. codec only:** yes

Step 3   If you are using a Cisco media gateway for PSTN termination, disable T.38 (fax relay) and enable fax using modem passthrough.

For example:

```
modem passthrough nse payload-type 110 codec g711ulaw
fax rate disable
fax protocol pass-through g711ulaw
```

Step 4   Enable T.38 fax on the WRP500 by configuring the following parameter on the Line tab for the FXS port to which the FAX machine is connected:

FAX_Enable T38: **Yes**

Note   If a T.38 call cannot be set up, the call automatically reverts to G.711 fallback.

Step 5   If you are using a Cisco media gateway, use the following settings:

Make sure the Cisco gateway is correctly configured for T.38 with the SPA dial peer. For example:

```
fax protocol T38
fax rate voice
fax-relay ecm disable
fax nsf 000000
no vad
```

Fax Troubleshooting

If you have problems sending or receiving faxes, complete the following steps:

Step 1   Verify that your fax machine is set to a speed between 7200 and 14400.

Step 2   Send a test fax in a controlled environment between two ATAs.

Step 3   Determine the success rate.

Step 4   Monitor the network and record the following statistics:

- Jitter
- Loss
- Delay

Step 5   If faxes fail consistently, capture a copy of the voice settings by selecting **Save As > Web page, complete** from the administration web server page. You can send this configuration file to Technical Support.

Step 6   Enable and capture the debug log. For instructions, refer to Appendix C, "Troubleshooting."

Note   You may also capture data by using a sniffer trace.

Step 7   Identify the type of fax machine that is connected to the ATA device.

Step 8   Contact technical support:

– If you are an end user of VoIP products, contact the reseller or Internet telephony service provider (ITSP) that supplied the equipment.

– If you are an authorized Cisco partner, contact Cisco technical support.

Silence Suppression and Comfort Noise Generation

Voice Activity Detection (VAD) with Silence Suppression is a means of increasing the number of calls that the network supports by reducing the required bandwidth for a single call. VAD uses a sophisticated algorithm to distinguish between speech and non-speech signals. Based on the current and past statistics, the VAD algorithm decides whether speech is present. If the VAD algorithm decides speech is not present, silence suppression and comfort noise generation is activated. This is accomplished by removing and not transmitting the natural silence that occurs in a normal two-way connection. The IP bandwidth is used only when someone is speaking. During the silent periods of a telephone call, additional bandwidth is available for other voice calls or data traffic because the silence packets are not being transmitted across the network.

Comfort Noise Generation provides artificially-generated background white noise (sounds), designed to reassure callers that their calls are still connected during silent periods. If Comfort Noise Generation is not used, the caller may think the call has been disconnected because of the "dead silence" periods that the VAD and Silence Suppression feature creates.

Silence suppression is configured in the Line tab.

Configure Dial Plans

Dial plans determine how the digits are interpreted and transmitted. They also determine whether the dialed number is accepted or rejected. You can use a dial plan to facilitate dialing or to block certain types of calls, such as long distance or international.

This section includes information that you need to understand dial plans, as well as procedures for configuring your own dial plans. This section includes the following topics:

About Dial Plans

This section provides information to help you understand how dial plans are implemented.

Refer to the following topics:

Digit Sequences

A dial plan contains a series of digit sequences, separated by the | character. The entire collection of sequences is enclosed within parentheses. Each digit sequence within the dial plan consists of a series of elements, which are individually matched to the keys that the user presses.

> **Note** White space is ignored, but may be used for readability.

Digit Sequence	Function
0 1 2 3 4 5 6 7 8 9 0 * #	Enter any of these characters to represent a key that the user must press on the phone keypad.
x	Enter x to represent any character on the phone keypad.
[sequence]	Enter characters within square brackets to create a list of accepted key presses. The user can press any one of the keys in the list. • Numeric range For example, enter [2-9] to allow the user to press any one digit from 2 through 9. • Numeric range with other characters For example, enter [35-8*] to allow the user to press 3, 5, 6, 7, 8, or *.
. (period)	Enter a period for element repetition. The dial plan accepts 0 or more entries of the digit. For example, 01. allows users to enter 0, 01, 011, 0111, and so on.
<dialed:substituted>	Use this format to indicate that certain dialed digits are replaced by other characters when the sequence is transmitted. The dialed digits can be zero or more characters. **EXAMPLE 1:** <8:1650>xxxxxxx When the user presses 8 followed by a seven-digit number, the system automatically replaces the dialed 8 with 1650. If the user dials **85550112**, the system transmits **16505550112**. **EXAMPLE 2:** <:1>xxxxxxxxxx In this example, no digits are replaced. When the user enters a 10-digit string of numbers, the number 1 is added at the beginning of the sequence. If the user dials **9725550112**, the system transmits **19725550112**
, (comma)	Enter a comma between digits to play an "outside line" dial tone after a user-entered sequence. **EXAMPLE:** 9, 1xxxxxxxxxx An "outside line" dial tone is sounded after the user presses 9, and the tone continues until the user presses 1.

Digit Sequence	Function
! (exclamation point)	Enter an exclamation point to prohibit a dial sequence pattern. **EXAMPLE:** 1900xxxxxxx! The system rejects any 11-digit sequence that begins with 1900.
*xx	Enter an asterisk to allow the user to enter a 2-digit star code.
S0 or L0	Enter S0 to reduce the short inter-digit timer to 0 seconds, or enter L0 to reduce the long inter-digit timer to 0 seconds.

Digit Sequence Examples

The following examples show digit sequences that you can enter in a dial plan.

In a complete dial plan entry, sequences are separated by a pipe character (|), and the entire set of sequences is enclosed within parentheses.

EXAMPLE: ([1-8]xx | 9, xxxxxxx | 9, <:1>[2-9]xxxxxxxxx | 8, <:1212>xxxxxxx | 9, 1 [2-9] xxxxxxxxx | 9, 1 900 xxxxxxx ! | 9, 011xxxxxx. | 0 | [49]11)

- Extensions on your system

  EXAMPLE: (**[1-8]xx** | 9, xxxxxxx | 9, <:1>[2-9]xxxxxxxxx | 8, <:1212>xxxxxxx | 9, 1 [2-9] xxxxxxxxx | 9, 1 900 xxxxxxx ! | 9, 011xxxxxx. | 0 | [49]11)

  [1-8]xx Allows a user to dial any three-digit number that starts with the digits 1 through 8. If your system uses four-digit extensions, you would instead enter the following string: **[1-8]xxx**

- Local dialing with seven-digit number

  EXAMPLE: ([1-8]xx | **9, xxxxxxx** | 9, <:1>[2-9]xxxxxxxxx | 8, <:1212>xxxxxxx | 9, 1 [2-9] xxxxxxxxx | 9, 1 900 xxxxxxx ! | 9, 011xxxxxx. | 0 | [49]111)

  9, xxxxxxx After a user presses 9, an external dial tone sounds. The user can enter any seven-digit number, as in a local call.

- Local dialing with 3-digit area code and a 7-digit local number

  EXAMPLE: ([1-8]xx | 9, xxxxxxx | **9, <:1>[2-9]xxxxxxxxx** | 8, <:1212>xxxxxxx | 9, 1 [2-9] xxxxxxxxx | 9, 1 900 xxxxxxx ! | 9, 011xxxxxx. | 0 | [49]11)

  9, <:1>[2-9]xxxxxxxxx This example is useful where a local area code is required. After a user presses 9, an external dial tone sounds. The user must enter a 10-digit number that begins with a digit 2 through 9. The system automatically inserts the 1 prefix before transmitting the number to the carrier.

- Local dialing with an automatically inserted 3-digit area code

  EXAMPLE: ([1-8]xx | 9, xxxxxxx | 9, <:1>[2-9]xxxxxxxxx | **8, <:1212>xxxxxxx** | 9, 1 [2-9] xxxxxxxxx | 9, 1 900 xxxxxxx ! | 9, 011xxxxxx. | 0 | [49]11)

  8, <:1212>xxxxxxx This example is useful where a local area code is required by the carrier but the majority of calls go to one area code. After the user presses 8, an external dial tone sounds. The user can enter any seven-digit number. The system automatically inserts the 1 prefix and the 212 area code before transmitting the number to the carrier.

- U.S. long distance dialing

  EXAMPLE: ([1-8]xx | 9, xxxxxxx | 9, <:1>[2-9]xxxxxxxxx | 8, <:1212>xxxxxxx | **9, 1 [2-9] xxxxxxxxx** | 9, 1 900 xxxxxxx ! | 9, 011xxxxxx. | 0 | [49]11)

9, 1 [2-9] xxxxxxxxx After the user presses 9, an external dial tone sounds. The user can enter any 11-digit number that starts with 1 and is followed by a digit 2 through 9.

- Blocked number

  EXAMPLE: ([1-8]xx | 9, xxxxxxx | 9, <:1>[2-9]xxxxxxxxx | 8, <:1212>xxxxxxx | 9, 1 [2-9] xxxxxxxxx | **9, 1 900 xxxxxxx !** | 9, 011xxxxxx. | 0 | [49]11)

  9, 1 900 xxxxxxx ! This digit sequence is useful if you want to prevent users from dialing numbers that are associated with high tolls or inappropriate content, such as 1-900 numbers in the U.S. After the user presses 9, an external dial tone sounds. If the user enters an 11-digit number that starts with the digits 1900, the call is rejected.

- U.S. international dialing

  EXAMPLE: ([1-8]xx | 9, xxxxxxx | 9, <:1>[2-9]xxxxxxxxx | 8, <:1212>xxxxxxx | 9, 1 [2-9] xxxxxxxxx | 9, 1 900 xxxxxxx ! | **9, 011xxxxxx.** | 0 | [49]11)

  9, 011xxxxxx. After the user presses 9, an external dial tone sounds. The user can enter any number that starts with 011, as in an international call from the U.S.

- Informational numbers

  EXAMPLE: ([1-8]xx | 9, xxxxxxx | 9, <:1>[2-9]xxxxxxxxx | 8, <:1212>xxxxxxx | 9, 1 [2-9] xxxxxxxxx | 9, 1 900 xxxxxxx ! | 9, 011xxxxxx. | **0 | [49]11**)

  0 | [49]11 This example includes two digit sequences, separated by the pipe character. The first sequence allows a user to dial 0 for an operator. The second sequence allows the user to enter 411 for local information or 911 for emergency services.

Acceptance and Transmission of Dialed Digits

When a user dials a series of digits, each sequence in the dial plan is tested as a possible match. The matching sequences form a set of candidate digit sequences. As the user enters more digits, the set of candidates diminishes until only one or none are valid. When a terminating event occurs, the WRP500 either accepts the user-dialed sequence and initiates a call, or rejects the sequence as invalid. The user hears the reorder (fast busy) tone if the dialed sequence is invalid.

The following table explains how terminating events are processed.

Terminating Event	Processing
The dialed digits do not match any sequence in the dial plan.	The number is rejected.
The dialed digits exactly match one sequence in the dial plan.	If the sequence is allowed by the dial plan, the number is accepted and is transmitted according to the dial plan.
	If the sequence is blocked by the dial plan, the number is rejected.

Terminating Event	Processing
A timeout occurs.	The number is rejected if the dialed digits are not matched to a digit sequence in the dial plan within the time specified by the applicable interdigit timer. • The Interdigit Long Timer applies when the dialed digits do not match any digit sequence in the dial plan. The default value is 10 seconds. • The Interdigit Short Timer applies when the dialed digits match one or more candidate sequences in the dial plan. The default value is 3 seconds.
The user presses the **#** key or the **dial** softkey on the phone display.	If the sequence is complete and is allowed by the dial plan, the number is accepted and is transmitted according to the dial plan. If the sequence is incomplete or is blocked by the dial plan, the number is rejected.

Dial Plan Timer (Off-Hook Timer)

You can think of the Dial Plan Timer as "the off-hook timer." This timer starts counting when the phone goes off hook. If no digits are dialed within the specified number of seconds, the timer expires and the null entry is evaluated. Unless you have a special dial plan string to allow a null entry, the call is rejected. The default value is 5.

Syntax for the Dial Plan Timer

SYNTAX: (P*s*<:*n*> | *dial plan*)

• **s:** The number of seconds; if no number is entered after P, the default timer of 5 seconds applies.

• **n:** (optional): The number to transmit automatically when the timer expires; you can enter an extension number or a DID number. No wildcard characters are allowed because the number will be transmitted as shown. If you omit the number substitution, <:n>, the user hears a reorder (fast busy) tone after the specified number of seconds.

Examples for the Dial Plan Timer

• Allow more time for users to start dialing after taking a phone off hook.

    EXAMPLE: (**P9** | (9,8<:1408>[2-9]xxxxxx | 9,8,1[2-9]xxxxxxxxx | 9,8,011xx. | 9,8,xx.|[1-8]xx)

    P9 After taking a phone off hook, a user has 9 seconds to begin dialing. If no digits are pressed within 9 seconds, the user hears a reorder (fast busy) tone. By setting a longer timer, you allow more time for users to enter the digits.

• Create a hotline for all sequences on the System Dial Plan

    EXAMPLE: (**P9<:23>** | (9,8<:1408>[2-9]xxxxxx | 9,8,1[2-9]xxxxxxxxx | 9,8,011xx. | 9,8,xx.|[1-8]xx)

    P9<:23> After taking the phone off hook, a user has 9 seconds to begin dialing. If no digits are pressed within 9 seconds, the call is transmitted automatically to extension 23.

- Create a hotline on a line button for an extension

  EXAMPLE: (P0 <:1000>)

  With the timer set to 0 seconds, the call is transmitted automatically to the specified extension when the phone goes off hook. Enter this sequence in the Phone Dial Plan for Ext 2 or higher on a client station.

Interdigit Long Timer (Incomplete Entry Timer)

You can think of this timer as the "incomplete entry" timer. This timer measures the interval between dialed digits. It applies as long as the dialed digits do not match any digit sequences in the dial plan. Unless the user enters another digit within the specified number of seconds, the entry is evaluated as incomplete, and the call is rejected. The default value is 10 seconds.

✎

Note   This section explains how to edit a timer as part of a dial plan. Alternatively, you can modify the Control Timer that controls the default interdigit timers for all calls. See the "Reset the Control Timers" section on page 3-16.

Syntax for the Interdigit Long Timer

SYNTAX: L:*s,* (*dial plan*)

- **s:** The number of seconds; if no number is entered after L:, the default timer of 5 seconds applies.

- Note that the timer sequence appears to the left of the initial parenthesis for the dial plan.

Example for the Interdigit Long Timer

EXAMPLE: L:15, (9,8<:1408>[2-9]xxxxxx | 9,8,1[2-9]xxxxxxxxx | 9,8,011xx. | 9,8,xx.|[1-8]xx)

L:15, This dial plan allows the user to pause for up to 15 seconds between digits before the Interdigit Long Timer expires. This setting is especially helpful to users such as sales people, who are reading the numbers from business cards and other printed materials while dialing.

Interdigit Short Timer (Complete Entry Timer)

You can think of this timer as the "complete entry" timer. This timer measures the interval between dialed digits. It applies when the dialed digits match at least one digit sequence in the dial plan. Unless the user enters another digit within the specified number of seconds, the entry is evaluated. If it is valid, the call proceeds. If it is invalid, the call is rejected. The default value is 3 seconds.

Syntax for the Interdigit Short Timer

- **SYNTAX 1:** S:*s,* (*dial plan*)

  Use this syntax to apply the new setting to the entire dial plan within the parentheses.

- **SYNTAX 2:** *sequence* Ss

  Use this syntax to apply the new setting to a particular dialing sequence.

  s: The number of seconds; if no number is entered after S, the default timer of 5 seconds applies.

Examples for the Interdigit Short Timer

- Set the timer for the entire dial plan.

    EXAMPLE: S:6, (9,8<:1408>[2-9]xxxxxx | 9,8,1[2-9]xxxxxxxxx | 9,8,011xx. | 9,8,xx.|[1-8]xx)

    S:6, While entering a number with the phone off hook, a user can pause for up to 15 seconds between digits before the Interdigit Short Timer expires. This setting is especially helpful to users such as sales people, who are reading the numbers from business cards and other printed materials while dialing.

- Set an instant timer for a particular sequence within the dial plan.

    EXAMPLE: (9,8<:1408>[2-9]xxxxxx | **9,8,1[2-9]xxxxxxxxxS0** | 9,8,011xx. | 9,8,xx.|[1-8]xx)

    9,8,1[2-9]xxxxxxxxxS0 With the timer set to 0, the call is transmitted automatically when the user dials the final digit in the sequence.

Edit Dial Plans

You can edit dial plans and can modify the control timers.

Enter the Line Interface Dial Plan

This dial plan is used to strip steering digits from a dialed number before it is transmitted out to the carrier.

Step 1 Start Internet Explorer, connect to the Configuration Utility, choose **Voice**. If prompted, enter the administrative login provided by the Service Provider. (The default username and password are both **admin**.)

Step 2 Under the **Voice** menu, click **Line 1** or **Line 2**, depending on the line interface that you want to configure.

Step 3 Scroll down to the *Dial Plan* section.

Step 4 Enter the digit sequences in the *Dial Plan* field. For more information, see the "About Dial Plans" section on page 3-10.

Step 5 Click **Submit.**

Reset the Control Timers

You can use the following procedure to reset the default timer settings for all calls.

Note If you need to edit a timer setting only for a particular digit sequence or type of call, you can edit the dial plan. See the "About Dial Plans" section on page 3-10.

Step 1 Start Internet Explorer, connect to the Configuration Utility, choose **Voice**. If prompted, enter the administrative login provided by the Service Provider. (The default username and password are both **admin**.)

Step 2 Under the **Voice** menu, click **Regional.**

Step 3 Scroll down to the *Control Timer Values* section.

Step 4   Enter the desired values in the *Interdigit Long Timer* field and the *Interdigit Short Timer* field. Refer to the definitions at the beginning of this section.

Secure Call Implementation

This section describes secure call implementation with the WRP500. It includes the following topics:

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Note   This is an advanced topic meant for experience installers. Also see the *Provisioning Guide* at the following URL:

http://www.cisco.com/c/dam/en/us/td/docs/voice_ip_comm/csbpvga/ata/provisioning/guide/Provisioning.pdf

Enable Secure Calls

WRP500 does not support establishing secure call by "mini certificate" as WRP400 did. The only method to enable a secure call requires use of SRTP, while the SRTP key parameters are transferred in SIP messages that are encrypted by TLS.

To enable SRTP on Line 1:

- Voice > Line 1 > Secure Call Serv, set to Yes
- Voice > User 1 > Secure Call Setting, set to Yes

To enable SIP over TLS on Line:

- Voice > Line 1 > SIP Transport, set to TLS