194559

ADMINISTRATION GUIDE

Cisco Small Business Pro
WRP400

Wireless-G Broadband Router with 2 Phone Ports and Built-In Analog Telephone Adapter

Contents

1

Product Overview and Deployment Guidelines

This chapter describes the features and benefits of the WRP400, describes deployment scenarios, and offers guidelines to help you plan your network.

WRP400 Features and Benefits

With a variety of features, the WRP400 offers the benefits of five devices in one:

1. **Router:** The WRP400 is a broadband router with a robust security firewall to protect your network.
2. **Switch:** The WRP400 includes a built-in, 4-port, full-duplex, 10/100 Ethernet switch to connect computers, printers, and other equipment directly or to attach additional hubs and switches. Advanced Quality of Service functionality ensures that you can prioritize traffic for data, voice, and video applications.
3. **Analog Telephone Adapter:** The WRP400 includes a two-port Analog Telephone Adapter (ATA) that allows you to connect your analog phones or fax machines to your configured Internet telephone service. Two traditional phone lines also can be connected for support of legacy phone numbers and fax numbers.

4. **Wireless Access Point:** The WRP400 has an integrated 802.11b/g wireless access point that secures your communications with WEP and WPA security protocols. It is preconfigured to support two wireless networks: one for private use by your business and one for guest use by customers, temporary employees, and other visitors.

5. **Mobile Broadband Router:** When you attach a compatible Mobile Broadband Modem to the USB port, the WRP400 allows multiple Wi-Fi devices to share a mobile broadband connection. This feature also can be used to provide continuous Internet service by providing automatic failover to the mobile network when the primary Internet connection is unavailable. For the latest copy of the USB Modem Compatibility List, visit the following URL: www.cisco.com/en/US/products/ps10028/index.html

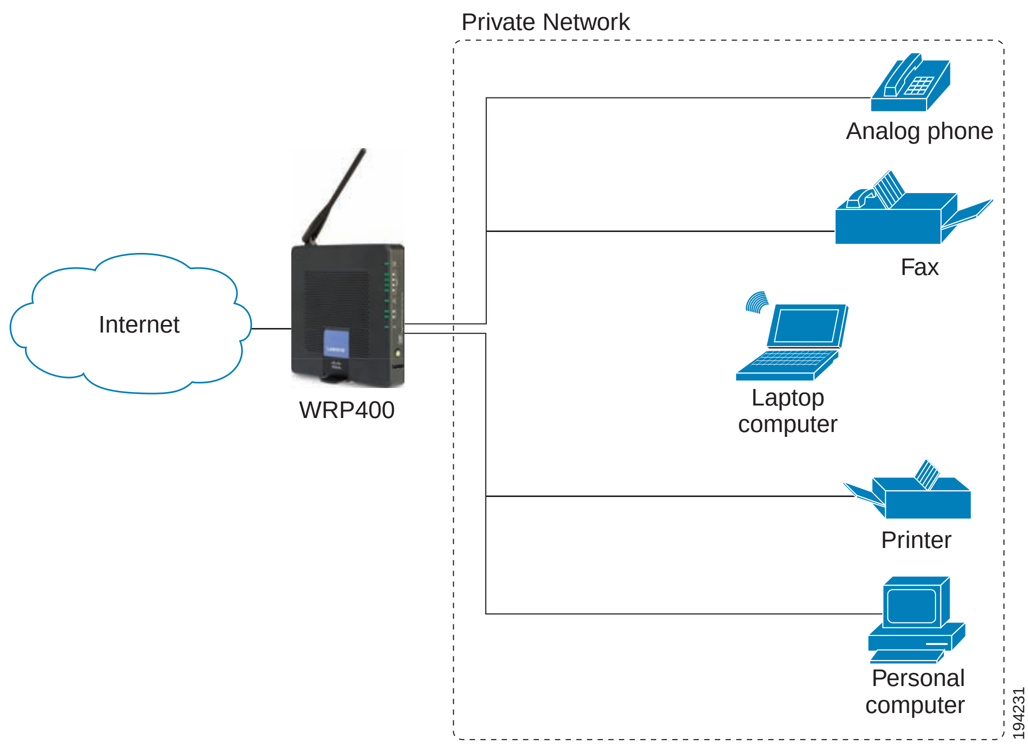
NOTE Because this device has many unique functions, the administrative tasks for the WRP400 may be different from corresponding tasks on other Cisco Small Business routers, switches, and ATAs. Administrators should refer to this guide for the proper procedures for installation, configuration, and management of the WRP400.

Deployment Models

The versatility of the WRP400 makes it useful for a variety of deployments. Three are described in this section.

- **Deploying the WRP400 in a Basic Network, page 7**
- **Deploying the WRP400 with a Wireless Guest Network, page 8**
- **Deploying the WRP400 with Mobile Broadband, page 9**

Deploying the WRP400 in a Basic Network



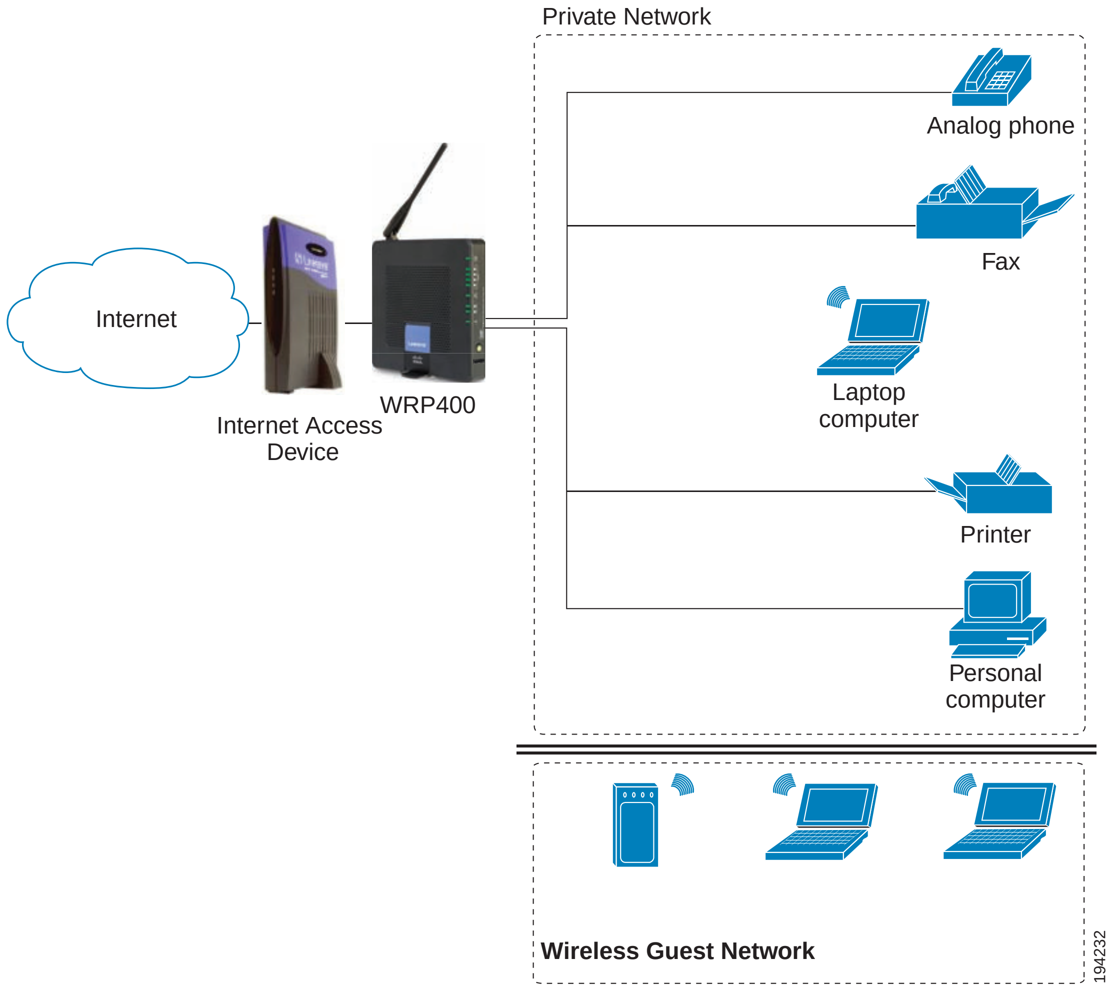
In this scenario, the WRP400 is deployed in a small business that has a basic network configuration.

- The WRP400 is preconfigured by the Service Provider to act as the edge device that routes traffic between the small business network and the Service Provider network.

NOTE The WRP400 may be configured as an edge device or can be connected to another device that provides access to the Service Provider network.

- The WRP400 connects the computers to the Internet. Computers may be connected by network cables or may operate wirelessly. All computers have access to the printer on the local network.
- An analog phone and a fax machine are connected to the WRP400 phone ports and have access to the configured Voice over IP services.

Deploying the WRP400 with a Wireless Guest Network



In this example, the WRP400 is deployed in an Internet cafe.

- The WRP400 is connected to a cable modem that provides Internet access.

NOTE The WRP400 may be configured as an edge device or can be connected to another device that provides access to the Service Provider network.

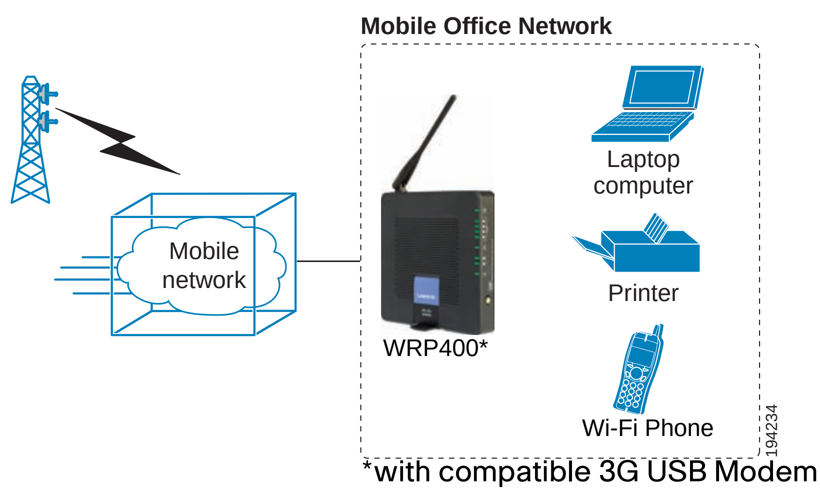
- In the private network, a computer is connected to the WRP400 by an Ethernet cable. The manager also has a laptop computer that can be used wirelessly from anywhere on the premises, using the main wireless network, SSID1. The manager and employees using SSID1 have access to the printer. If desired, a wireless phone also could be connected to this network for business use.

- An analog phone and a fax machine are in the private network. The WRP400 is configured for Internet telephone service and for traditional telephone service through a connected phone line.
- The WRP400 is configured with a guest network, SSID2, that enables the business to provide its customers with a free wireless hotspot for their laptop computers and other mobile devices. Because this network is separate from the main wireless network, the customers have no access to the manager's computer, the printer, or the telephone service.

Deploying the WRP400 with Mobile Broadband

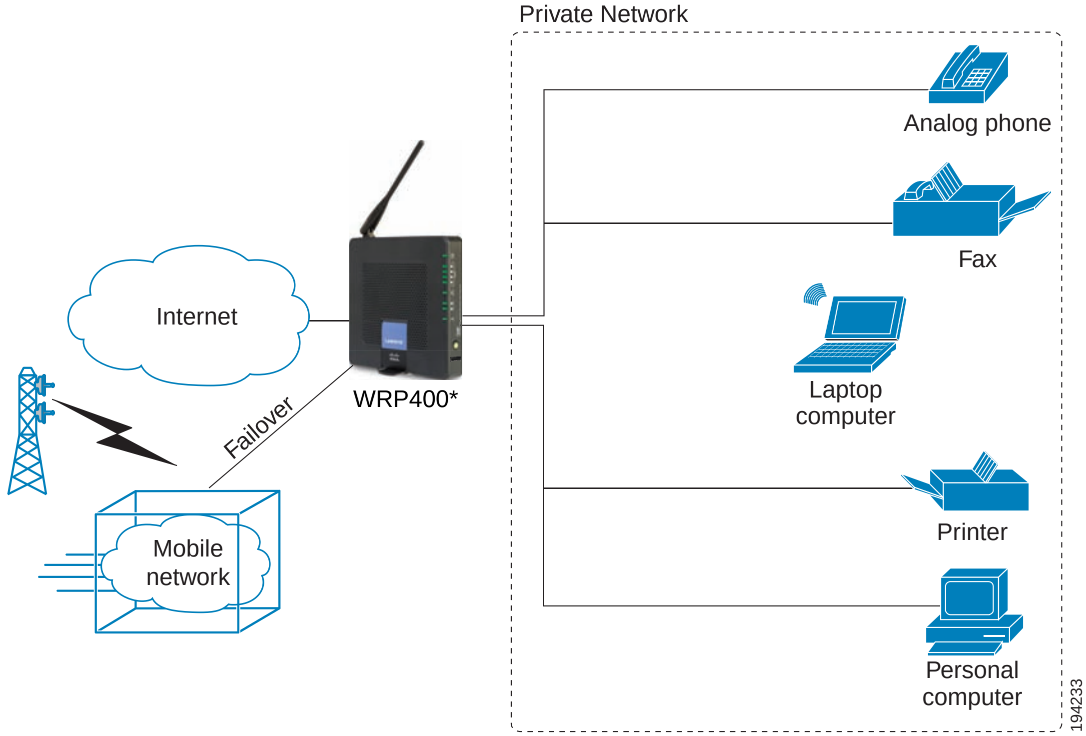
When a compatible mobile broadband modem is connected to the USB port, the WRP400 can connect to a mobile broadband network. The mobile network can be the primary network or can serve as a backup network to ensure continuous Internet connectivity. Consider the two scenarios illustrated below.

Mobile Office Using the Mobile Network for Internet Access



In this example, a team has set up a temporary network at a construction site. The team members have laptop computers and Wi-Fi phones that share a mobile broadband connection for Internet access. All computers can connect to the printer on the local network. If a Virtual Private Network (VPN) tunnel is configured on the laptop computer, team members also can securely connect to resources at the main office (not illustrated).

Basic Office Deployment Using the Mobile Network as a Backup Connection



*with compatible 3G USB Modem

In this example, the business has the same network as illustrated in **Deploying the WRP400 in a Basic Network, page 7**. However, this business has the added benefit of using the mobile broadband network as a backup network to ensure continuous Internet connectivity. In the event that the Internet connection fails, the WRP400 fails over to the configured mobile network. When the Internet connection becomes available, the WRP400 recovers the connection.

Local Area Network Guidelines

This section offers guidelines for setting up your Local Area Network (LAN).

NOTE As you design your network, be aware that the WRP400 is intended for deployment in a very small business. The router is designed to handle the data, voice, and video traffic that would be expected by office personnel who use the Internet to find data, conduct phone conversations, transmit email, and participate in videoconferences. For large-scale operations with heavy data, voice, and video requirements, consider other models of Cisco Small Business routers.

Power, Cabling and Telephone Lines

- **AC outlets:** Ensure there is an AC outlet available for every network device that requires AC power.
  - The WRP400 requires power, and Ethernet switches (optional) require power.
  - Some analog telephones require AC power.
- **Ethernet cabling:** If an Internet access device is present, you will need to connect it to the WRP400 with an Ethernet cable. You also will need Ethernet cable for any devices that do not have wireless connectivity. It is recommended that Ethernet cables are UTP Cat5e or better.
- **PSTN lines:** Ensure that the lines are operative and that any features, such as caller identification, operate properly before starting the installation.
- **UPS:** It is strongly recommended that you included an Uninterrupted Power Supply (UPS) mechanism in your network to ensure continuous operation during a power failure. Connect all essential devices, including the Internet access device, WRP400, and the Ethernet switch (if present).

Basic Services and Equipment

The following basic services and equipment are required:

- An Integrated access device or modem for broadband access to the Internet
- Business grade Internet service

- Internet Telephony Service Provider (ITSP) for Voice Over IP telephone service, supporting a "bring your own device" model
- A computer with Microsoft Windows XP or Windows Vista for system configuration

Special Requirements for Voice Deployments

Voice deployments have special requirements that you must meet to ensure voice quality.

- **"Bandwidth for Voice Deployments," on page 12**
- **"NAT Mapping for Voice over IP Deployments," on page 14**
- **"Local Area Network Design for Voice Deployments," on page 14**

Bandwidth for Voice Deployments

You can choose from several types of broadband access technologies to provide symmetric or asymmetric connectivity to a small business. These technologies vary on the available bandwidth and on the quality of service. For voice deployments, it is generally recommended that you use broadband access with a Service Level Agreement that provides quality of service. If there is not a Service Level Agreement with regard to the broadband connection quality of service, the downstream audio quality may be affected negatively under heavy load conditions (bandwidth utilization beyond 80%).

To eliminate or minimize this effect, Cisco recommends one of the following actions:

- For broadband connections with a bandwidth lower than 2 Mbps, perform the call capacity calculations by assuming a bandwidth value of 50% of the existing broadband bandwidth. For example, in the case of a 2 Mbps uplink broadband connection, assume 1 Mbps. Limit the uplink bandwidth in the Integrated Access Device to this value. This setting helps to maintain the utilization levels below 60%, thus reducing jitter and packet loss.
- Use an additional broadband connection for voice services only. A separate connection is required when the broadband connection services do not offer quality of service and when it is not possible to apply the above mentioned utilization mechanism.

The available connection bandwidth determines the maximum number of simultaneous calls that the system can support with the appropriate audio quality. Use this information to determine the maximum number of simultaneous VoIP connections that the system can support.

For asymmetric connections, such as ADSL, the maximum number of calls is determined by the upstream bandwidth. In general it is a good practice to use no more than 75% of the total available bandwidth for calls. This provides space for data traffic and helps ensure good voice quality.

NOTE Some ITSP SIP trunk services limit the maximum number of simultaneous calls. Please check with your Service Provider to understand the maximum number of simultaneous calls each SIP trunk supports.

The following table provides the approximate bandwidth budget for different codecs.

Codec	Approximate bandwidth budget for each side of conversation	2 calls	4 calls	6 calls	8 calls
G.711	110 kbps	220 kbps	440 kbps	660 kbps	880 kbps
G.726-40	87 kbps	174 kbps	348 kbps	522 kbps	696 kbps
G.726-32	79 kbps	158 kbps	316 kbps	474 kbps	632 kbps
G.726-24	71 kbps	142 kbps	284 kbps	426 kbps	568 kbps
G.726-16	63 kbps	126 kbps	252 kbps	378 kbps	504 kbps
G.729	55 kbps	110 kbps	220 kbps	330 kbps	440 kbps

For more information about bandwidth calculation, refer to the following web sites:
www.erlang.com/calculator/lipb/
www.bandcalc.com/

NAT Mapping for Voice over IP Deployments

Network Address Translation (NAT) is the function that allows multiple devices in your small business network to share one external (public) IP address that you receive from your Internet Service Provider. Voice over IP can co-exist with NAT only when some form of NAT traversal is provided.

Some Internet Telephone Service Providers (ITSPs) provide NAT traversal, but some do not. **For voice deployments, it is strongly recommended that you choose an ITSP that supports NAT mapping through a Session Border Controller.**

If your ITSP does not provide NAT mapping through a Session Border Controller (the preferred method), you have three options for providing NAT traversal on your WRP400:

- Deploy an edge device that has a SIP ALG (Application Layer Gateway). The Cisco Small Business WRV200 is suited for this purpose, but other SIP-ALG routers can be used. If your Internet Service Provider is providing the edge device, check with your provider to determine if the router has a SIP ALG.
- Configure NAT mapping with the EXT IP setting. This option requires that you have (1) a static external (public) IP address from your Internet Service Provider and (2) an edge device with a symmetric NAT mechanism. If the WRP400 is the edge device, the second requirement is met. For more information about the EXT IP setting, see **NAT Support Parameters section, page 70**.
- Configure Simple Traversal of UDP through NAT (STUN). This option requires that you have (1) a dynamic external (public) IP address from your service provider, (2) a computer running STUN server software, and (3) an edge device with an asymmetric NAT mechanism. If the WRP400 is the edge device, the third requirement *is not* met. For more information about the STUN Enable setting and the STUN Test Enable setting, see **NAT Support Parameters section, page 70**.

Local Area Network Design for Voice Deployments

Use the following guidelines to manage the LAN setup for voice deployments.

- Ensure that all telephones are located in the same local area network subnet.
- Configure your WRP400 as a DHCP server for the purpose of easily adding network devices to the system. Ensure that the DHCP server can assign

enough IP addresses to serve the devices that you need to connect to your network.

- Use stable DNS server addresses for URL name resolution. Your Internet Service Provider can provide the primary and secondary DNS server IP addresses.
- If you need to directly connect more than four network devices (other than wireless devices), you will need to connect an Ethernet switch to the WRP400. For voice deployments, Cisco recommends use of the SLMxxxP, SRWxxxP and SRWxxxMP switch product families. The SLM224P is a popular choice. For more information about these switches, visit the following URL: www.cisco.com/cisco/web/solutions/small_business/products/routers_switches/index.html
- If you use an Ethernet switch, configure it to ensure voice quality. The following settings are recommended:
  - Enable Port Fast and Spanning Tree Protocol on the ports to which your voice devices are connected. The Cisco phones are capable of rebooting in a few seconds and will attempt to locate network services while a switch port is being blocked by STP after it senses a device reboot. Enabling Port Fast means that the network will be available to the phones when needed. If the switch does not provide a way to enable Port Fast, then you must disable Spanning Tree Protocol.
  - In the administrative web pages for the switch, you should enable QoS and choose DSCP as the Trust Mode.

WRP400 Maintenance Operations

Due to its unique functions, the WRP400 has unique maintenance operations as compared to other Cisco Small Business IP telephony devices.

NOTE For complete instructions about the settings mentioned below, see the *WRP400 User Guide.*

- **Remote Management:** For security purposes, remote management is disabled by default.
  - When you first configure the WRP400, connect your administrative computer directly to one of the LAN ports and enter the default static IP

address into your web browser to log on to the configuration utility.

NOTE The default LAN IP address of the WRP400 is 192.168.15.1. If another device on the network has the same IP address, the WRP400 will take the address 192.168.16.1. You can modify the Local IP Address on the Setup tab > Basic Setup page, Network Setup section.

If you are using the IVR, be aware that this address is NOT the address reported by the 110 option of the IVR. The device does not respond to the 110 option address.

- If you wish to enable web access and wireless access to the configuration utility, you can use the Administration tab > Management page, Web Access section.

- **DHCP Server:** The DCHP server is disabled by default. If there are no other DHCP servers on your network, you can enable the DHCP server option to allow your WRP400 to assign IP addresses to connected devices automatically. This setting is on the Setup tab > Basic Setup page, DHCP Server Setting section.
- **System Logging:** If you wish to enable system logging, be aware that there are two sets of system logs: one for the data (router) functions and another for the voice functions.
  - **Data (router) logging:** See the Administration tab > Logging page.
  - **Voice logging:** See the Voice tab > System page, Miscellaneous Settings section.
- **Factory Reset:** If you wish to reset your WRP400 to the factory default settings, you can reset the data (router) settings and the voice settings separately.

  Factory reset of data (router) settings: Use one of the following methods:
  - **Option 1:** Log on to the configuration utility, and then click **Administration > Factory Defaults**. Next to **Restore Router Factory Defaults**, click **Yes**. Then click **Save Settings** to begin the operation.
  - **Option 2:** Press and hold the reset button located on the side panel for approximately ten seconds.

Factory reset of voice settings: Use one of the following methods:

- **Option 1:** Log on to the configuration utility, and then click **Administration tab > Factory Defaults**. Next to **Restore Voice Factory Defaults**, click **Yes**. Then click **Save Settings** to begin the operation.
- **Option 2:** Connect an analog phone to the Phone 1 or Phone 2 port. Press **** to access the Interactive Voice Response menu. After you hear the greeting, press 73738 for factory reset. Listen to the prompts and then press 1 to confirm or * to cancel. After you hear "Option successful," you can hang up the phone.

Remote Provisioning

Like other Cisco Small Business IP Telephony Devices, the WRP400 provides for secure provisioning and remote upgrade. Provisioning is achieved through configuration profiles transferred to the device via TFTP, HTTP, or HTTPS. To configure Provisioning, go to the Provisioning tab in the Configuration Utility.

NOTE For complete details, see the *Provisioning Guide* at the following URL: www.cisco.com/en/US/docs/voice_ip_comm/csbpvga/ata/provisioning/guide/Cisco_Small_Business_IP_Telephony_Provisioning_Guide.pdf

Upgrade URL

Remote firmware upgrade is achieved via TFTP or HTTP (firmware upgrades using HTTPS are not supported). Remote upgrades are initiated by causing the WRP400 to request the upgrade firmware image by providing a URL for the WRP400 to retrieve the firmware.

NOTE If the value of the *Upgrade Enable* parameter in the Provisioning page is **No**, you cannot upgrade the WRP400 even if the web page indicates otherwise.

The syntax of the Upgrade URL is as follows:

```
http://WRP400_ip_address/admin/upgrade?[protocol://][server-name[:port]][/firmware-pathname]
```

Both HTTP and TFTP are supported for the upgrade operation.

If no *protocol* is specified, TFTP is assumed. If no *server-name* is specified, the host that requests the URL is used as *server-name*.

If no port specified, the default port of the protocol is used. (69 for TFTP or 80 for HTTP)

The *firmware-pathname* is typically the file name of the binary located in a directory on the TFTP or HTTP server. If no *firmware-pathname* is specified, */spa.bin* is assumed, as in the following example:

```
http://192.168.2.217/admin/upgrade?tftp://192.168.2.251/
spa.bin
```

Resync URL

The WRP400 can be configured to automatically resync its internal configuration state to a remote profile periodically and on power up. The automatic resyncs are controlled by configuring the desired profile URL into the device.

The Resync URL lets you force the WRP400 to do a resync to a profile specified in the URL, which can identify either a TFTP, HTTP, or HTTPS server. The syntax of the Resync URL is as follows:

```
http://WRP400_ip_address/admin/resync?[[protocol://][server-
name[:port]]/profile-pathname]
```

NOTE The WRP400 resyncs only when it is idle.

If no parameter follows */resync?,* the Profile Rule setting from the Provisioning page is used.

If no *protocol* is specified, TFTP is assumed. If no *server-name* is specified, the host that requests the URL is used as *server-name*.

If no port is specified, the default port is used (69 for TFTP, 80 for HTTP, and 443 for HTTPS).

The profile-path is the path to the new profile with which to resync, for example:

```
http://192.168.2.217/admin/resync?tftp://192.168.2.251/
spaconf.cfg
```

Reboot URL

The Reboot URL lets you reboot the WRP400. The Reboot URL is as follows:

http://*WRP400_ip_address***/admin/reboot**

NOTE The WRP400 reboots only when it is idle.

Configuration Profile

Because the WRP400 has two sets of parameters, one set for data and one set for voice, the requirements vary from the provisioning of other Cisco Small Business IP Telephony Devices. You will have two profiles: one for the data (router) parameters and one for the voice parameters. One benefit of having separate profiles for voice parameters and data parameters is that you can deploy the common data parameters to all of your customer sites and deploy the custom voice parameters to each site individually.

- **Data (router) parameters:** Use the XML format only, as described in the *Provisioning Guide*. Binary files are not supported for the configuration of data (router) parameters. For more information about the data parameters, see **Appendix B, "Data Fields."**
- **Voice parameters:** Use the binary or XML format. The binary format is generated by a profile compiler tool available from Cisco. Find the correct SPA Profiler Compiler (SPC) for the firmware that you have installed on your WRP400. For more information about the data parameters, see **Appendix A, "Advanced Voice Fields."**

  NOTE You can download the SPC at the following URL: tools.cisco.com/support/ downloads/go/Redirect.x?mdfid=282414113

XML Format

Use the XML format for data (router) parameters. The XML file consists of a series of elements (one per configuration parameter), encapsulated within the element tags <flat-profile> ... </flat-profile>. The encapsulated elements specify values for individual parameters. Here is an example of a valid XML profile:

```
<flat-profile>

<Admin_Passwd>some secret</Admin_Passwd>

<Upgrade_Enable>Yes</Upgrade_Enable>

</flat-profile>
```

The names of parameters in XML profiles can generally be inferred from the WRP400 Configuration Utility, by substituting underscores (_) for spaces and other control characters. To distinguish between Lines 1, 2, 3, and 4, corresponding parameter names are augmented by the strings _1_, _2_, _3_, and _4_. For example, Line 1 Proxy is named Proxy_1_ in XML profiles. For more information, see Appendix C, "WRP400 Provisioning Reference."

Binary Format

Binary format profiles contain voice parameter values and user access permissions for the parameters. By convention, the profile uses the extension .cfg (for example, spa2102.cfg). The Profile Compiler (SPC) tool compiles a plain-text file containing parameter-value pairs into a properly formatted and encrypted .cfg file.

The syntax of the plain-text file accepted by the profile compiler is a series of parameter-value pairs, with the value in double quotes. Each parameter-value pair is followed by a semicolon. Here is an example of a valid text source profile for input to the SPC tool:

```
Admin_Passwd  “some secret”;

Upgrade_Enable “Yes”;
```

The names of parameters in the source text files for the SPC tool can generally be inferred from the WRP400 Configuration Utility, by substituting underscores (_) for spaces and other control characters. To distinguish between Line 1, 2, 3, and 4, corresponding parameter names are augmented by adding [1], [2], [3], or [4]. For example, the Line 1 Proxy is named Proxy[1] in source text profiles for input to the SPC.

2

Configuring Your System for ITSP Interoperability

This chapter provides configuration details to help you to ensure that your infrastructure properly supports voice services.

Configuring NAT Mapping

As discussed in **Chapter 1, "Product Overview and Deployment Guidelines,"** some form of NAT mapping is needed to support VoIP. If your ITSP does not support NAT mapping through a Session Border Controller, and your edge device is not a SIP-ALG router, you can address this issue through one of the following methods:

Configuring NAT Mapping with a Static IP Address

This option can be used if the following requirements are met:

- You must have a static external (public) IP address from your ISP.
- The edge device—that is, the router between your local area network and your ISP network—must have a symmetric NAT mechanism. If the WRP400 is the edge device, this requirement is met. If another device is used as the edge device, see **"Determining Whether the Router Uses Symmetric or Asymmetric NAT," on page 25**.

- If the WRP400 is connected to an Ethernet switch, the switch must be configured to enable Spanning Tree Protocol and Port Fast on the port to which the WRP400 is connected.

NOTE Use NAT mapping only if the ITSP network does not provide a Session Border Controller functionality.

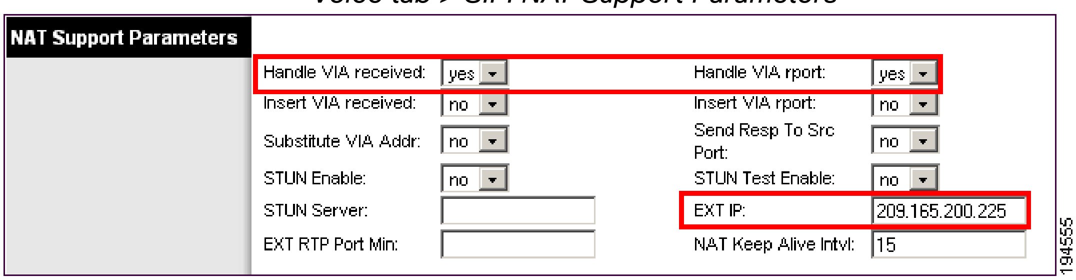
STEP 1 Start Internet Explorer, connect to the Configuration Utility, and choose **Voice > Admin Login**. If prompted, enter the administrative login provided by the Service Provider. (The default username and password are both **admin**.)

STEP 2 Under the **Voice** menu, click **SIP**.

STEP 3 In the *NAT Support Parameters* section, enter the following settings:

- **Handle VIA received, Insert VIA received, Substitute VIA Addr:** Choose **yes**.
- **Handle VIA rport, Insert VIA rport, Send Resp To Src Port:** Choose **yes**.
- **EXT IP:** Enter the public IP address that was assigned by your ISP.

Voice tab > SIP: NAT Support Parameters

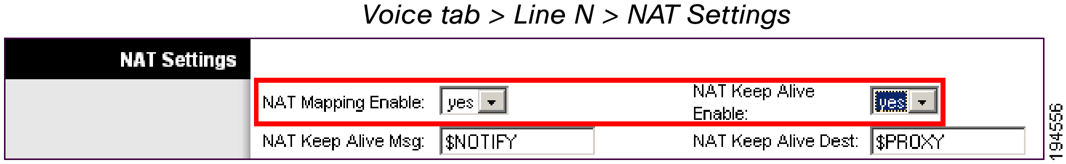


STEP 4 Under the **Voice** menu, click **Line 1** or **Line 2** to choose the line interface that you want to modify.

STEP 5 In the *NAT Settings* section, enter the following settings:

- **NAT Mapping Enable:** Choose **yes**.
- **NAT Keep Alive Enable:** Choose **yes**.

Voice tab > Line N > NAT Settings



STEP 6 Click **Save Settings**.

NOTE You also need to configure the firewall settings on your router to allow SIP traffic. See **"Firewalls and SIP," on page 26**.

Configuring NAT Mapping with STUN

This option is considered a practice of last resort and should be used only if the other methods are unavailable. This option can be used if the following requirements are met:

- You have a dynamically assigned external (public) IP address from your ISP.
- You must have a computer running STUN server software.
- The edge device uses an asymmetric NAT mechanism. If the WRP400 is the edge device, this requirement *is not met*. For more information, see **"Determining Whether the Router Uses Symmetric or Asymmetric NAT," on page 25**.
- If the WRP400 is connected to an Ethernet switch, the switch must be configured to enable Spanning Tree Protocol and Port Fast on the port to which the WRP400 is connected.

NOTE Use NAT mapping only if the ITSP network does not provide a Session Border Controller functionality.

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

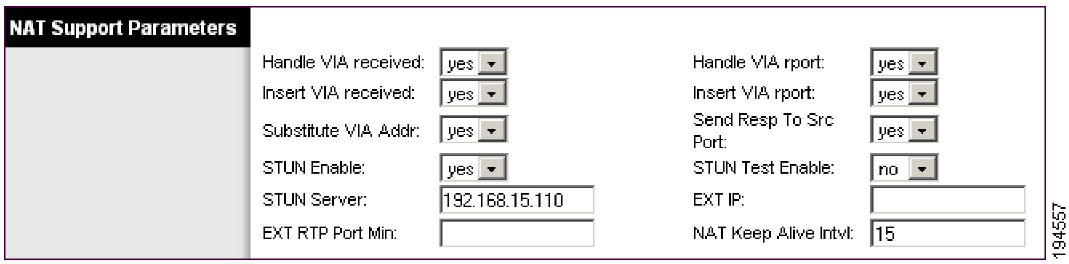
STEP 2 Under the **Voice** menu, click **SIP**.

STEP 3 In the *NAT Support Parameters* section, enter the following settings:

- **Handle VIA received:** yes
- **Handle VIA rport:** yes

- **Insert VIA received:** yes
- **Insert VIA rport:** yes
- **Substitute VIA Addr:** yes
- **Send Resp To Src Port:** yes
- **STUN Enable**: Choose **yes.**
- **STUN Server:** Enter the IP address for your STUN server.

Voice tab > SIP > NAT Support Parameters

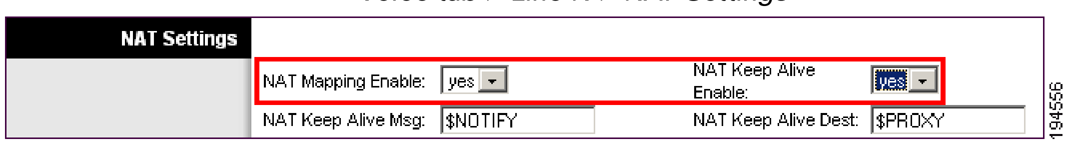


STEP 4 Under the **Voice** menu, click **Line 1** or **Line 2** to choose the line interface that you want to modify.

STEP 5 In the *NAT Settings* section, enter the following settings:

- **NAT Mapping Enable:** Choose **yes.**
- **NAT Keep Alive Enable:** Choose **yes** (optional).

Voice tab > Line N > NAT Settings



NOTE Your ITSP may require the WRP400 to send NAT keep alive messages to keep the NAT ports open permanently. Check with your ITSP to determine the requirements.

STEP 6 Click **Save Settings**.

NOTE You also need to configure the firewall settings on your router to allow SIP traffic. See **"Firewalls and SIP," on page 26**.

Determining Whether the Router Uses Symmetric or Asymmetric NAT

To use a STUN server, the edge device—that is, the device that routes traffic between your private network and your ISP network—must have an asymmetric NAT mechanism. You need to determine which type of NAT mechanism is available on that device.

STUN does not work on routers with symmetric NAT. With symmetric NAT, IP addresses are mapped from one internal IP address and port to one external, routable destination IP address and port. If another packet is sent from the same source IP address and port to a different destination, then a different IP address and port number combination is used. This method is restrictive because an external host can send a packet to a particular port on the internal host *only if* the internal host first sent a packet from that port to the external host.

NOTE This procedure assumes that a syslog server is configured and is ready to receive syslog messages.

STEP 1 Make sure you do not have firewall running on your computer that could block the syslog port (port 514 by default).

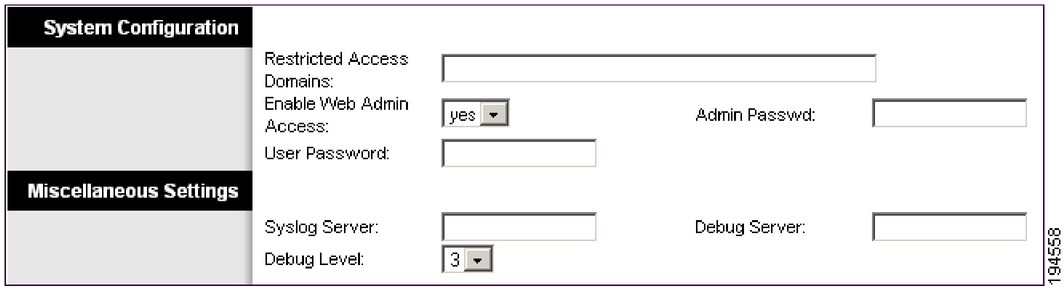
STEP 2 Start Internet Explorer, connect to the Configuration Utility, choose **Voice > Admin Login**. If prompted, enter the administrative login provided by the Service Provider. (The default username and password are both **admin**.)

STEP 3 To enable debugging, complete the following tasks:

a. Under the **Voice** menu, click **System**.

b. In the *Debug Server* field, enter the IP address of your syslog server. This address and port number must be reachable from the WRP400.

c. From the *Debug level* drop-down list, choose **3**.



STEP 4 To collect information about the type of NAT your router is using, complete the following tasks:

a. Under the **Voice** menu, click **SIP**.

b. Scroll down to the *NAT Support Parameters* section.

c. From the *STUN Test Enable* field, choose **yes**.

STEP 5 To enable SIP signalling, complete the following task:

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

b. In the *SIP Settings* section, choose **full** from the *SIP Debug Option* field.

STEP 6 Click **Save Settings**.

STEP 7 View the syslog messages to determine whether your network uses symmetric NAT. Look for a warning header in the REGISTER messages, such as Warning: 399 spa "Full Cone NAT Detected."

Firewalls and SIP

To enable SIP requests and responses to be exchanged with the SIP proxy at the ITSP, you must ensure that your firewall allows both SIP and RTP unimpeded access to the Internet.

- Make sure that the following ports are not blocked:
  - SIP ports—UDP port 5060 through 5063, which are used for the ITSP line interfaces

  - RTP ports—16384 to 16482
- Also disable SPI (Stateful Packet Inspection) if this function exists on your firewall.

Configuring SIP Timer Values

The default timer values should be adequate in most circumstances. However, you can adjust the SIP timer values as needed to ensure interoperability with your ISTP. For example, if SIP requests are returned with an “invalid certificate” message, you may need to enter a longer SIP T1 retry value.

For more information, see **”SIP Timer Values (sec) section,” on page 65 of Appendix A**.

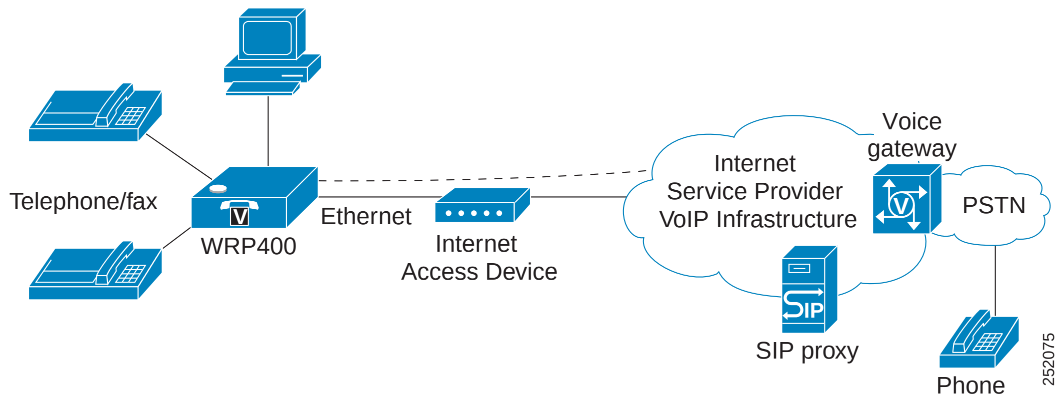
3

Configuring Voice Services

This chapter describes how to configure your WRP400 to meet the customer's requirements for voice services.

Understanding Analog Telephone Adapter Operations

The WRP400 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 delivered over broadband or high-speed Internet connections. Users can access Internet phone services using standard analog telephone equipment. In addition, the WRP400 has two line ports that can be connected to the Public Switched Telephone Network (PSTN) so that your business can support legacy phone numbers and fax numbers.



The WRP400 maintains the state of each call it terminates and makes the proper reaction to user input events (such as on/off hook or hook flash). The WRP400 uses the Session Initiation Protocol (SIP) open standard, so there is little or no involvement by a “middle-man” server or media gateway controller. SIP allows interoperation with all ITSPs that support SIP.

ATA Software Features

The WRP400 is equipped with a full featured, fully programmable ATA that can be custom provisioned within a wide range of configuration parameters. The following sections describe the factors that contribute to voice quality:

- **“Supported Codecs,” on page 29**
- **“SIP Proxy Redundancy,” on page 30**
- **“Other ATA Software Features,” on page 31**

Supported Codecs

The WRP400 supports the following codecs:

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

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

- G.726-32

  This low complexity codec supports compressed 16, 24, 32, and 40 kbps digitized voice transmission at one through ten 10 ms voice frames per packet. This codec provides high voice quality.

- 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 **"Audio Configuration section," on page 104**.

In addition, negotiation of the optimal voice codec sometimes depends on the ability of an ATA to match a codec name with the codec used by the far-end device. You can individually name the various codecs so that the WRP400 can successfully negotiate the codec with the far-end equipment. For more information, see **Audio Configuration section, page 104**.

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 WRP400 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 WRP400 can be instructed to contact a SIP proxy server in a domain named in the SIP message. The WRP400 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 WRP400 tries to contact the list of hosts in the order of their stated priority.

If the WRP400 is currently using a lower priority proxy server, it periodically probes the higher priority proxy to see 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. See **Appendix B, "Data Fields."**.

Other ATA Software Features

The following table summarizes other features provided by the WRP400.

Feature	Description
Silence Suppression	See **"Silence Suppression and Comfort Noise Generation," on page 41**.
Modem and Fax Pass-Through	▪ Modem pass-through mode can be triggered only by predialing the number set in the *Modem Line Toggle Code.* (Set in the Regional tab.) ▪ FAX pass-through mode is triggered by a CED/CNG tone or an NSE event. ▪ Echo canceller is automatically disabled for Modem pass-through mode. ▪ Echo canceller is disabled for FAX pass-through if the parameter *FAX Disable ECAN* (Line 1 or 2 tab) is set to "yes" for that line (in that case FAX pass-through is the same as Modem pass-through). ▪ Call waiting and silence suppression is automatically disabled for both FAX and Modem pass-through. In addition, out-of-band DTMF Tx is disabled during modem or fax pass-through.
Adaptive Jitter Buffer	The WRP400 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 WRP400 has a Network Jitter Level control setting for each line of service. The jitter level determines how aggressively the WRP400 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 **"Advanced Voice Fields," on page 57**.
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 **"Advanced Voice Fields," on page 57**.

Feature	Description
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 **"Secure Call Implementation" section on page 52**.
Adjustable Audio Frames Per Packet	This feature allows the user to set the number of audio frames contained in one RTP packet. Packets can be adjusted to contain from 1–10 audio frames. Increasing the number 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 the **"Advanced Voice Fields," on page 57**.
DTMF	The WRP400 may relay DTMF digits as out-of-band events to preserve the fidelity of the digits. This can enhance the reliability of DTMF transmission required by many IVR applications such as dial-up banking and airline information. DTMF is configured in the *DTMF Tx Mode* parameter found in the Line tabs. See the **"Advanced Voice Fields," on page 57**.
Call Progress Tone Generation	The WRP400 has configurable call progress tones. Call progress tones are generated locally on the WRP400 so an end user is advised of status (such as ringback). Parameters for each type of tone (for instance a dial tone played back to an end user) may include frequency and amplitude of each component, and cadence information. See the Regional tab in the **"Advanced Voice Fields," on page 57**.
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 the **"Advanced Voice Fields," on page 57**.
Echo Cancellation	Impedance mismatch between the telephone and the IP Telephony gateway phone port can lead to near-end echo. The WRP400 has a near-end echo canceller that compensates for impedance match. The WRP400 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 **"Advanced Voice Fields," on page 57**.

Feature	Description
Signaling Hook Flash Event	The WRP400 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 **"Advanced Voice Fields," on page 57**.
Configurable Dial Plan with Interdigit Timers	The WRP400 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 **"Configuring Dial Plans," on page 42** for more information.
Polarity Control	The WRP400 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 **"Advanced Voice Fields," on page 57**.

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 **"Advanced Voice Fields," on page 57**.
Report Generation and Event Logging	The WRP400 reports a variety of status and error reports to assist service providers to diagnose problems and evaluate the performance of their services. The information can be queried by an authorized agent, using HTTP with digested authentication, for instance. The information may be organized as an XML page or HTML page. Report Generation and Event Logging are configured in the System, Line, and PSTN Line tabs. See **"Advanced Voice Fields," on page 57**.
Syslog and Debug Server Records	Syslog and Debug Sever Records log more details than Report Generation and Event Logging. Using the configuration parameters, the WRP400 allows you to select which type of activity/events should be logged. Syslog and Debug Server allow the information captured to be sent to a Syslog Server. Syslog and Debug Server Records are configured in the System, Line, and PSTN Line tabs. See **"Advanced Voice Fields," on page 57**.
SIP Over TLS	The WRP400 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 **"Advanced Voice Fields," on page 57**.

Registering to the Service Provider

To use VoIP phone service, you must configure your WRP400 to the Internet Telephony Service Provider (ITSP).

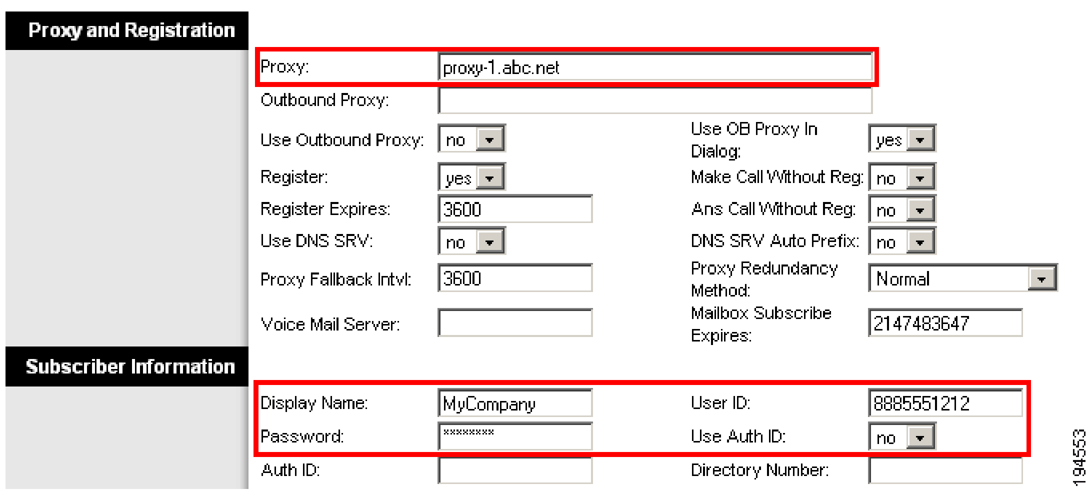
NOTE Each line tab must be configured separately. Each line tab can be configured for a different ITSP.

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

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**.

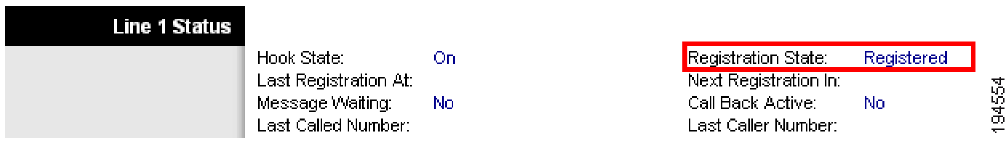


NOTE These are the minimum settings for most ITSP connections. Enter the account information as required by your ITSP.

STEP 5 Click **Save Settings**. 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.



- 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.

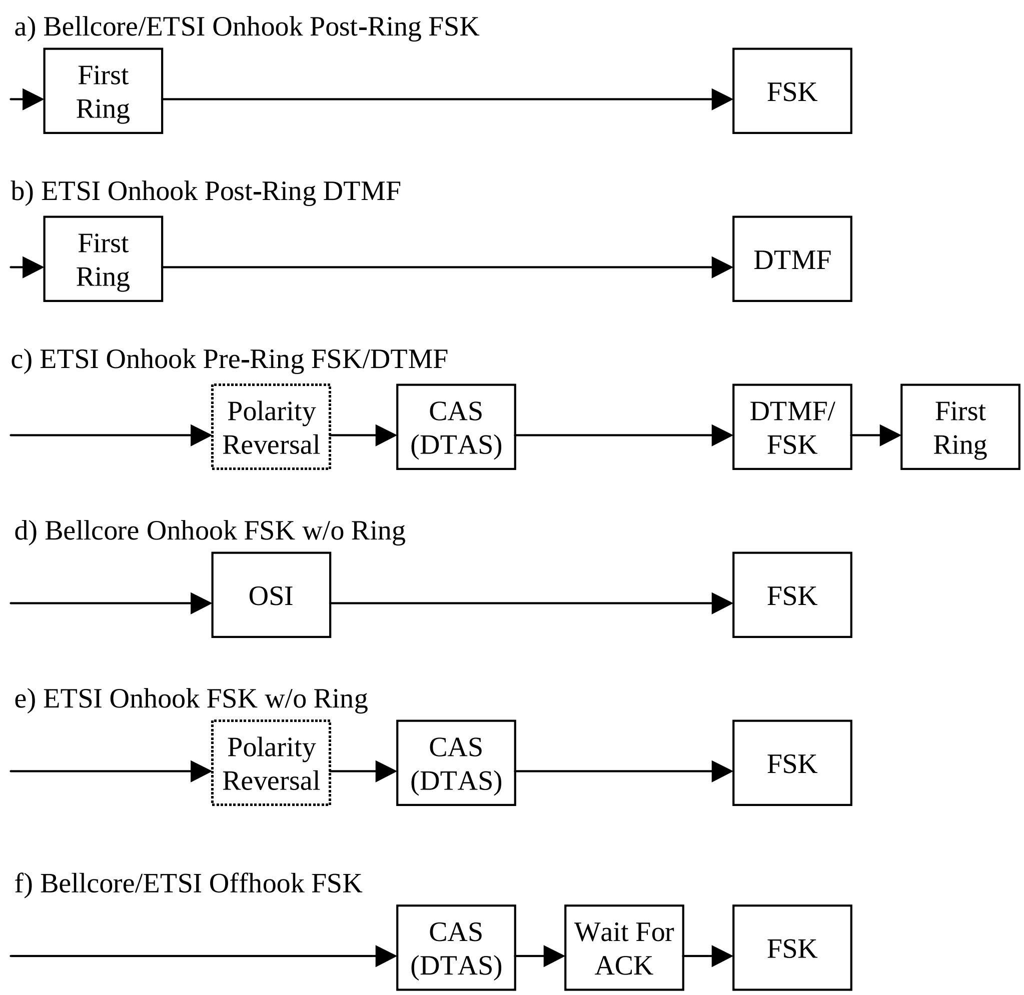
Managing Caller ID Service

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. ▪ **DTMF (Denmark) With PR**—CID only. DTMF sent after polarity reversal (and no DTAS) and before first ring. The default is **Bellcore(N.Amer, China)**.
Caller ID FSK Standard	Regional	The WRP400 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**.

There are three types of Caller ID:

- 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 Figure 1 (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).



Optimizing Fax Completion Rates

Issues can occur with fax transmissions over IP networks, even with the T.38 standard, which is supported by the WRP400. You can adjust several settings on your WRP400 to optimize your fax completion rates.

NOTE Only T.38 Fax is supported. The WRP400 supports one connection.

STEP 1 Ensure that you have enough bandwidth for the uplink and the downlink.

- For G.711 fallback, it is recommend to have approximately 100Kbps.
- 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:

- **Network Jitter Buffer**: very high
- **Jitter buffer adjustment**: disable
- **Call Waiting**: no
- **3 Way Calling**: no
- **Echo Canceller**: no
- **Silence suppression**: 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 WRP400 by configuring the following parameter on the Line tab for the FXS port to which the FAX machine is connected:

```
FAX_Passthru_Method: ReINVITE
```

NOTE If a T.38 call cannot be set-up, then 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 D, "Troubleshooting."**

NOTE You may also capture data using a sniffer trace.

STEP 7 Identify the type of fax machine 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 supported by the network 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 or not speech is present. If the VAD algorithm decides speech is not present, the silence suppression and comfort noise generation is activated. This is accomplished by removing and not transmitting the natural silence that occurs in 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 created by the VAD and Silence Suppression feature.

Silence suppression is configured in the Line and PSTN Line tabs. See **Appendix B, “Data Fields.”**

Configuring 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," on page 42**
- **"Editing Dial Plans," on page 50**

About Dial Plans

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

Refer to the following topics:

- **"Digit Sequences," on page 42**
- **"Digit Sequence Examples," on page 44**
- **"Acceptance and Transmission the Dialed Digits," on page 47**
- **"Dial Plan Timer (Off-Hook Timer)," on page 48**
- **"Interdigit Long Timer (Incomplete Entry Timer)," on page 49**
- **"Interdigit Short Timer (Complete Entry Timer)," on page 49**

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, you would enter [2-9] to allow the user to press any one digit from 2 through 9. ▪ Numeric range with other characters For example, you would 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**

Digit Sequence	Function
, (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.
! (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 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 is 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 press 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 the 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 more digits are entered by the user, the set of candidates diminishes until only one or none are valid. When a terminating event occurs, the WRP400 either accepts the user-dialed sequence and initiates a call, or else 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.
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: `(Ps<: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>, then 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 **“Resetting the Control Timers,” on page 51**.

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.

Editing Dial Plans

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

Entering 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 > Admin Login**. If prompted, enter the administrative login provided by the Service

Provider. (The default username and password are both **admin**.)rovided by your Service Provider.

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 **"About Dial Plans," on page 42**.

STEP 5 Click **Save Settings**.

Resetting 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 **"About Dial Plans," on page 42**.

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

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 WRP400 . It includes the following topics:

NOTE This is an advanced topic meant for experience installers. Also see the *Provisioning Guide* at the following URL:
www.cisco.com/en/US/docs/voice_ip_comm/csbpvga/ata/provisioning/guide/Cisco_Small_Business_IP_Telephony_Provisioning_Guide.pdf

Enabling Secure Calls

A secure call is established in two stages. The first stage is no different from normal call setup. The second stage starts after the call is established in the normal way with both sides ready to stream RTP packets.

In the second stage, the two parties exchange information to determine if the current call can switch over to the secure mode. The information is transported by base64 encoding embedded in the message body of SIP INFO requests, and responses using a proprietary format. If the second stage is successful, the WRP400 plays a special Secure Call Indication Tone for a short time to indicate to both parties that the call is secured and that RTP traffic in both directions is being encrypted.

If the user has a phone that supports call waiting caller ID (CIDCW) and that service is enabled, the CID will be updated with the information extracted from the Mini-Certificate received from the remote party. The Name field of the CID will be prepended with a '$' symbol. Both parties can verify the name and number to ensure the identity of the remote party.

The signing agent is implicit and must be the same for all ATAs that communicate securely with each other. The public key of the signing agent is pre-configured into the WRP400 by the administrator and is used by the WRP400 to verify the Mini-Certificate of its peer. The Mini-Certificate is valid if it has not expired, and it has a valid signature.

The WRP400 can be configured so that, by default, all outbound calls are either secure or not secure. If secure by default, the user has the option to disable security when making a call by dialing *19 before dialing the target number. If not secure by default, the user can make a secure outbound call by dialing *18 before dialing the target number. However, the user cannot force inbound calls to be secure or not secure; that depends on whether the caller has security enabled or not.

The WRP400 will not switch to secure mode if the CID of the called party from its Mini-Certificate does not agree with the user-id used in making the outbound call. The WRP400 performs this check after receiving the Mini-Certificate of the called party

Secure Call Details

Looking at the second stage of setting up a secure call in greater detail, this stage can be further divided into two steps.

STEP 1 The caller sends a "Caller Hello" message (base64 encoded and embedded in the message body of a SIP INFO request) to the called party with the following information:

- Message ID (4B)
- Version and flags (4B)
- SSRC of the encrypted stream (4B)
- Mini-Certificate (252B)

Upon receiving the Caller Hello, the called party responds with a Callee Hello message (base64 encoded and embedded in the message body of a SIP response to the caller's INFO request) with similar information, if the Caller Hello message is valid. The caller then examines the Callee Hello and proceeds to the next step if the message is valid.

STEP 2 The caller sends the “Caller Final” message to the called party with the following information:

- Message ID (4B)
- Encrypted Master Key (16B or 128b)
- Encrypted Master Salt (16B or 128b)

Using a Mini-Certificate

The Master Key and Master Salt are encrypted with the public key from the called party mini-certificate. The Master Key and Master Salt are used by both ends for deriving session keys to encrypt subsequent RTP packets. The called party then responds with a Callee Final message (which is an empty message).

The Mini-Certificate (MC) contains the following information:

- User Name (32B)
- User ID or Phone Number (16B)
- Expiration Date (12B)
- Public Key (512b or 64B)
- Signature (1024b or 512B)

The MC has a 512-bit public key used for establishing secure calls. The administrator must provision each subscriber of the secure call service with an MC and the corresponding 512-bit private key. The MC is signed with a 1024-bit private key of the service provider, which acts as the CA of the MC. The 1024-bit public key of the CA signing the MC must also be provisioned for each subscriber.

The CA public key is used to verify the MC received from the other end. If the MC is invalid, the call will not switch to secure mode. The MC and the 1024-bit CA public key are concatenated and base64 encoded into the single parameter *Mini Certificate*. The 512-bit private key is base64 encoded into the *SRTP Private Key* parameter, which should be kept secret, like a password. (*Mini Certificate* and *SRTP Private Key* are configured in the Line tabs.)

Because the secure call establishment relies on exchange of information embedded in message bodies of SIP INFO requests/responses, the service provider must ensure that the network infrastructure allows the SIP INFO messages to pass through with the message body unmodified.

Generating a Mini Certificate

Cisco provides a Mini Certificate Generator for the generation of mini certificates and private keys. Partners can download the Mini Certificate Generator by going to Cisco Partner Central, Voice & Conferencing page, Technical Resources section. Use the following URL:
www.cisco.com/web/partners/sell/smb/products/voice_and_conferencing.html#~vc_technical_resources

NOTE The partner sites require a logon.

The Mini Certificate Generator uses the following syntax:

`gen_mc` *`ca-key user-name user-id expire-date`*

Where:

- *ca-key* is a text file with the base64 encoded 1024-bit CA private/public key pairs for signing/verifying the MC, such as the following:


```
9CC9aYU1X5lJuU+EBZmi3AmcqE9U1LxEOGwopaGyGOh3VyhKgi6JaVtQZt87PiJINKW8XQj3B9Qq
e3VgYxWCQNa335YCnDsenASeBxuMIEaBCYd1l1fVEodJZOGwXwfAde0MhcbD0kj7LVlzcsTyk2TZ
YTccnZ75TuTjj13qvYs=5nEtOrkCa84/mEwl3D9tSvVLyliwQ+u/
Hd+C8u5SNk7hsAUZaA9TqH8Iw0J/
IqSrsf6scsmundY5j7Z5mK5J9uBxSB8t8vamFGD0pF4zhNtbrVvIXKI9kmp4vph1C5jzO9gDfs3M
F+zjyYrVUFdM+pXtDBxmM+fGUfrpAuXb7/k=
```


- *user-name* is the name of the subscriber, such as "Joe Smith". Maximum length is 32 characters
- *user-id* is the User ID of the subscriber, which must match exactly the user-id used in the INVITE when making the call, such as "14083331234". The maximum length is 16 characters.
- *expire-date* is the expiration date of the MC, such as "00:00:00 1/1/34" (34=2034). Internally the date is encoded as a fixed 12B string: 000000010134

The tool generates the *Mini Certificate* and *SRTP Private Key* parameters that can be provisioned.

EXAMPLE:

```
gen_mc ca_key “Joe Smith” 14085551234 “00:00:00 1/1/34”
```

This example produces the following Mini Certificate and SRTP Private Key:


```
<Mini Certificate>
Sm9lIFNtaXRoAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAAxNDA4NTU1MTIzNAAAAAAAMDAwMDAwMDEw
MTM0OOvJakde2vVMF3Rw4pPXL7lAgIagMpbLSAG2+++YlSqt198Cp9rP/
xMGFfoPmDKGx6JFtkQ5sxLcuwgxpxpxkeXvpZKlYlpsb28L4Rhg5qZA+Gqj1hDFCmG6dffZ9SJhx
ES767G0JIS+N8lQBLr0AuemotknSjjjOy8c+1lTCd2t44Mh0vmwNg4fDck2YdmTMBR516xJt4/
uQ/
LJQlni2kwqlm7scDvll5k232EvvvVtCK0AYa4eWd6fQOpiESCO9CC9aYU1X5lJuU+EBZmi3AmcqE
9U1LxEOGwopaGyGOh3VyhKgi6JaVtQZt87PiJINKW8XQj3B9Qqe3VgYxWCQNa335YCnDsenASeBx
uMIEaBCYd1l1fVEodJZOGwXwfAde0MhcbD0kj7LVlzcsTyk2TZYTccnZ75TuTjj13qvYs=
<SRTP Private Key>
b/DWc96X4YQraCnYzl5en1CIUhVQQqrvcr6Qd/8R52IEvJjOw/
e+Klm4XiiFEPaKmU8UbooxKG36SEdKusp0AQ==
```

A

Advanced Voice Fields

This appendix describes the Advanced settings that are available after you login from the Voice > Admin Login page.

NOTE For information about the other pages in the Configuration Utility, see the *WRP400 User Guide*.

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

- **"Info page," on page 57**
- **"System page," on page 61**
- **"SIP page," on page 62**
- **"Regional page," on page 72**
- **"Line page," on page 92**
- **"User page," on page 111**

Info page

You can use the *Voice tab* > *Info* page to view information about the WRP400. This page includes the following sections:

- **"Product Information section," on page 58**
- **"System Status section," on page 58**
- **"Line Status section," on page 59**

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

Voice tab > Info page >

Product Information section

Product Name	Model number/name.
Serial Number	Serial number.
Software Version	Software version number.
Hardware Version	Hardware version number.
MAC Address	MAC address.
Client Certificate	Status of the client certificate, which can indicate if the WRP400 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 tab > Info page >

System Status section

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).

Current Time	Current date and time of the system; for example, 10/3/2003 16:43:00.
SIP Bytes Recv	Total number of bytes of SIP messages received (including retransmissions).
External IP	External IP address used for NAT mapping.

Voice tab > Info page >

Line Status section

(PSTN) Hook State	Hook state of the FXO 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.
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 FXO 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 ▪ Collecting PSTN Pin ▪ Invalid PSTN PIN ▪ PSTN Caller Accepted ▪ Connected to PSTN
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 pass-through mode.
Call 1 and 2 Type	Direction of the call. May take one of the following values: ▪ PSTN Gateway Call = VoIP-To-PSTN Call ▪ VoIP Gateway Call = PSTN-To-VoIP Call ▪ PSTN To Line 1 = PSTN call ring through and answered by Line 1 ▪ Line 1 Forward to PSTN Gateway = VoIP calls Line 1 then forwarded to PSTN GW ▪ Line 1 Forward to PSTN Number =VoIP calls Line 1 then forwarded to PSTN number ▪ Line 1 To PSTN Gateway ▪ Line 1 Fallback To PSTN Gateway
Call 1 and 2 Remote Hold	Indicates whether the far end has placed the call on hold.
Call 1 and 2 Callback	Indicates whether the call was triggered by a call back request.
Call 1 and 2 Peer Name	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.
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 Round Trip Delay	Number of milliseconds for delay.
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 experienced by the end user.

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" section on page 61**
- **"Miscellaneous Settings section" section on page 62**

Voice tab > System page >

System Configuration section

Restricted Access Domains	This feature is used when implementing software customization.
Enable Web Admin Access	Lets you enable or disable local access to the Configuration Utility. Select yes or no from the drop-down menu. The default is **yes**.
Admin Password	Password for the administrator. The default is no password.
User Password	Password for the user. The default is no password.

Voice tab > System page >

Miscellaneous Settings section

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.

SIP page

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

- **"SIP Parameters section" section on page 63**
- **"SIP Timer Values (sec) section" section on page 65**
- **"Response Status Code Handling section" section on page 67**
- **"RTP Parameters section" section on page 67**
- **"SDP Payload Types section" section on page 69**
- **"NAT Support Parameters section" section on page 70**

Voice tab > SIP page >

SIP Parameters section

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 $D corresponding to GPP_A to GPP_D allowed.
SIP Server Name	Server header used in responses to inbound responses. The default is **$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 WRP400 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**.
Hook Flash MIME Type	MIME Type used in a SIP INFO message to signal a hook flash event. The default is **application/hook-flash**.
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 WRP400 uses compact SIP headers in outbound SIP messages. If set to no, the WRP400 uses normal SIP headers. If inbound SIP requests contain compact headers, the WRP400 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 WRP400 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 WRP400 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.
SIP TCP Port Max	Specifies the highest TCP port number that can be used for SIP sessions.

Voice tab > SIP page >

SIP Timer Values (sec) section

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.**
Reg Retry Intvl	Interval to wait before the WRP400 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 WRP400 waits for the specified length of time before retrying. If this interval is 0, the WRP400 stops trying. This value should be much larger than the Reg Retry Intvl value, which should not be 0. The default is **1200**.
Reg Retry Random Delay	Random delay range (in seconds) to add to *Register Retry Intvl* when retrying REGISTER after a failure. The default is **0**, which disables this feature.
Reg Retry Long Random Delay	Random delay range (in seconds) to add to *Register Retry Long Intvl* when retrying REGISTER after a failure. The default is **0**, which disables this feature.
Reg Retry Intvl Cap	The maximum value to cap the exponential back-off retry delay (which starts at *Register Retry Intvl* and doubles on every REGISTER retry after a failure). In other words, the retry interval is always at *Register Retry Intvl* seconds after a failure. If this feature is enabled, *Reg Retry Random Delay* is added on top of the exponential back-off adjusted delay value. The default value is **0**, which disables the exponential back-off feature.

Voice tab > SIP page >

Response Status Code Handling section

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 WRP400 retries registration after failing during the last registration. The default is **30**.

Voice tab > SIP page >

RTP Parameters section

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**.
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**.

Max RTP ICMP Err	Number of successive ICMP errors allowed when transmitting RTP packets to the peer before the WRP400 terminates the call. If value is set to 0, the WRP400 ignores the limit on ICMP errors. The default is **0**.
RTCP Tx Interval	Interval for sending out RTCP sender reports on an active connection. It can range from 0 to 255 seconds. During an active connection, the WRP400 can be programmed to send out compound RTCP packet on the connection. Each compound RTP packet except the last one contains a SR (Sender Report) and a SDES.(Source Description). The last RTCP packet contains an additional BYE packet. Each SR except the last one contains exactly 1 RR (Receiver Report); the last SR carries no RR. The SDES contains CNAME, NAME, and TOOL identifiers. The CNAME is set to <User ID>@<Proxy>, NAME is set to <Display Name> (or Anonymous if user blocks caller ID), and TOOL is set to the Vendor/Hardware-platform-software-version (such as Cisco/wrp400-1.0.31(b)). The NTP timestamp used in the SR is a snapshot of the WRP400's local time, not the time reported by an NTP server. If the WRP400 receives a RR from the peer, it attempts to compute the round trip delay and show it as the <Call Round Trip Delay> value (ms) in the Info section of the WRP400 Configuration Utility. The default is **0**.
No UDP Checksum	Select yes if you want the WRP400 to calculate the UDP header checksum for SIP messages. Otherwise, select no. The default is **no**.
Stats In BYE	Determines whether the WRP400 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**.

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SDP Payload Types section

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.
G726r16 Dynamic Payload	G.726-16 dynamic payload type. The valid range is 96-127. The default is **98**.
G726r24 Dynamic Payload	G.726-24 dynamic payload type. The valid range is 96-127. The default is **97**.
G726r32 Dynamic Payload	G726r32 dynamic payload type. The default is 2.
G726r40 Dynamic Payload	G.726-40 dynamic payload type. The valid range is 96-127. The default is **96**.
G729b Dynamic Payload	G.729b dynamic payload type. The valid range is 96-127. The default is **99**.
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 **computerMU**.
G711a Codec Name	G.711a codec name used in SDP. The default is **computerMA**.
G726r16 Codec Name	G.726-16 codec name used in SDP. The default is **G726-16.**

G726r24 Codec Name	G.726-24 codec name used in SDP. The default is **G726-24**.
G726r32 Codec Name	G.726-32 codec name used in SDP. The default is **G726-32**.
G726r40 Codec Name	G.726-40 codec name used in SDP. The default is **G726-40**.
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**.
G723 Codec Name	G.723 codec name used in SDP. The default is **G723**.
EncapRTP Codec Name	EncapRTP codec name used in SDP. The default is **EncapRTP**.

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NAT Support Parameters section

Handle VIA received	If you select yes, the WRP400 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 WRP400 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**.
STUN Test Enable	If the STUN Enable feature is enabled and a valid STUN server is available, the WRP400 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 WRP400 detects symmetric NAT or a symmetric firewall, NAT mapping is disabled. The default is **no**.
STUN Server	IP address or fully-qualified domain name of the STUN server to contact for NAT mapping discovery.

EXT IP	External IP address to substitute for the actual IP address of the WRP400 in all outgoing SIP messages. If 0.0.0.0 is specified, no IP address substitution is performed. If this parameter is specified, the WRP400 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 WRP400 is the edge device, the second requirement is met. The default is **0.0.0.0**.
EXT RTP Port Min	External port mapping number of the RTP Port Min. number. If this value is not zero, the RTP port number in all outgoing SIP messages is substituted for the corresponding port value in the external RTP port range. The default is **0**.
NAT Keep Alive Intvl	Interval between NAT-mapping keep alive messages. The default is **15**.

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" section on page 73**
- **"Distinctive Ring Patterns section" section on page 75**
- **"Distinctive Call Waiting Tone Patterns section" section on page 76**
- **"Distinctive Ring/CWT Pattern Names section" section on page 77**
- **"Ring and Call Waiting Tone Spec section" section on page 78**
- **"Control Timer Values (sec) section" section on page 78**
- **"Vertical Service Activation Codes section" section on page 80**

- **"Outbound Call Codec Selection Codes section" section on page 86**
- **"Miscellaneous section" section on page 88**

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Call Progress Tones section

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 WRP400 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)**.
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)**.

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Distinctive Ring Patterns section

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(.3/.2, 1/.2,.3/4)**.
Ring3 Cadence	Cadence script for distinctive ring 3. The default is **60(.8/.4,.8/4)**.
Ring4 Cadence	Cadence script for distinctive ring 4. The default is **60(.4/.2,.3/.2,.8/4)**.
Ring5 Cadence	Cadence script for distinctive ring 5. The default is **60(.2/.2,.2/.2,.2/.2,1/4)**.
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)**.

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Distinctive Call Waiting Tone Patterns section

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, .1/9.3)**.
CWT4 Cadence	Cadence script for distinctive CWT 4. The default is **30(.1/.1, .3/.1, .1/9.5)**.
CWT5 Cadence	Cadence script for distinctive CWT 5. The default is **30(.3/.1,.1/.1,.3/9.1)**.
CWT6 Cadence	Cadence script for distinctive CWT 6. The default is **30(.3/.1,.3/.1,.1/9.1).**
CWT7 Cadence	Cadence script for distinctive CWT 7. The default is **30(.1/.1, .3/.1, .1/9.3)**.
CWT8 Cadence	Cadence script for distinctive CWT 8. The default is **2.3(.3/2).**

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Distinctive Ring/CWT Pattern Names section

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

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Ring and Call Waiting Tone Spec section

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

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

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**.

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Control Timer Values (sec) section

Hook Flash Timer Min	Minimum on-hook time before off-hook qualifies as hook-flash. Less than this the on-hook event is ignored. Range: 0.1–0.4 seconds. The default is **0.1**.
Hook Flash Timer Max	Maximum on-hook time before off-hook qualifies as hook-flash. More than this the on-hook event is treated as on-hook (no hook-flash event). Range: 0.4–1.6 seconds. The default is **0.9**.

Callee On Hook Delay	Phone must be on-hook for at this time in sec before the WRP400 will tear down the current inbound call. It 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 WRP400 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 **0.5**.
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**.

Ccomputer Delay	Delay in seconds after caller hangs up when the WRP400 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 Ccomputer feature should be used instead. Without Ccomputer enabled, reorder tone will is played after a configurable delay. If Ccomputer is enabled, dial tone will be played when tip-to-ring voltage is restored Resolution is 1 second. The default is **2**.
Ccomputer 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. Ccomputer is disabled if this value is set to 0. Range: 0 to 1.000 second. Resolution is 0.001 second. The default is **0** (Ccomputer disabled).

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Vertical Service Activation Codes section

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

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**.

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**.
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**.
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**.
Referral Services Codes	These codes tell the WRP400 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 WRP400 to perform a blind transfer to a target number that is prepended by the service *code. For example, after the user dials *98, the WRP400 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 WRP400 sends a blind REFER to the holding party with the Refer-To target equals to *98 *target_number*. This feature allows the WRP400 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 WRP400. You can empty the corresponding *code that you do not want the WRP400 to process.

Feature Dial Services Codes	These codes tell the WRP400 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 WRP400 to call the target number prepended by the *code. For example, after user dials *72, the WRP400 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 WRP400 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 WRP400. You can empty the corresponding *code that you do not want to the WRP400 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 WRP400 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 WRP400 send INVITE *73@..... as usual when user dials *73.

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Outbound Call Codec Selection Codes section

These codes automatically appended to the dial-plan. So no need to include them in dial-plan (although no harm to do so either).

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 G723 Code	Makes this codec the preferred codec for the associated call. The default is ***01723**.
Force G723 Code	Makes this codec the only codec that can be used for the associated call. The default is ***02723**.
Prefer G726r16 Code	Makes this codec the preferred codec for the associated call. The default is ***0172616**.
Force G726r16 Code	Makes this codec the only codec that can be used for the associated call. The default is ***0272616**.
Prefer G726r24 Code	Makes this codec the preferred codec for the associated call. The default is ***0172624**.

Force G726r24 Code	Makes this codec the only codec that can be used for the associated call. The default is ***0272624**.
Prefer G726r32 Code	Makes this codec the preferred codec for the associated call. The default is ***0172632**.
Force G726r32 Code	Makes this codec the only codec that can be used for the associated call. The default is ***0272632**.
Prefer G726r40 Code	Makes this codec the preferred codec for the associated call. The default is ***0172640**.
Force G726r40 Code	Makes this codec the only codec that can be used for the associated call. The default is ***0272640**.
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**.

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Miscellaneous section

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.
Time Zone	Selects the number of hours to add to GMT to generate the local time for caller ID generation. Choices are GMT-12:00, GMT-11:00,..., GMT, GMT+01:00, GMT+02:00, ..., GMT+13:00. The default is **GMT-08:00**.
FXS Port Impedance	Sets the electrical impedance of the FXS port. Choices are **600**, **900**, **600+2.16uF**, **900+2.16uF**, **270+750ll150nF**, **220+850ll120nF**, **220+820ll115nF**, or **200+600ll100nF**. The default is **600**.

Daylight Saving Time Rule	Enter the rule for calculating daylight saving time; it should include the start, end, and save values. This rule is comprised of three fields. Each field is separated by ; (a semicolon) as shown below. Optional values inside [ ] (the brackets) are assumed to be 0 if they are not specified. Midnight is represented by 0:0:0 of the given date. SYNTAX: Start = <start-time>; end=<end-time>; save = <save-time>. The <start-time> and <end-time> values specify the start and end dates and times of daylight saving time. Each value is in this format: <month> /<day> / <weekday>[/ HH:[mm[:ss]]] The <save-time> value is the number of hours, minutes, and/or seconds to add to the current time during daylight saving time. The <save-time> value can be preceded by a negative (-) sign if subtraction is desired instead of addition. The <save-time> value is in this format: [/[+|-]HH:[mm[:ss]]] The <month> value equals any value in the range 1-12 (January-December). The <day> value equals [+|-] any value in the range 1-31. If <day> is 1, it means the <weekday> on or before the end of the month (in other words the last occurrence of < weekday> in that month). The <weekday> value equals any value in the range 1-7 (Monday-Sunday). It can also equal 0. If the <weekday> value is 0, this means that the date to start or end daylight saving is exactly the date given. In that case, the <day> value must not be negative. If the <weekday> value is not 0 and the <day> value is positive, then daylight saving starts or ends on the <weekday> value on or after the date given. If the <weekday> value is not 0 and the <day> value is negative, then daylight saving starts or ends on the <weekday> value on or before the date given. The abbreviation HH stands for hours (0-23). The abbreviation mm stands for minutes (0-59). The abbreviation ss stands for seconds (0-59). The default Daylight Saving Time Rule is **start=4/1/7;end=10/-1/7;save=1**.

Daylight Saving Time Enable	Daylight Saving Time can be turned on or off. This option affects the time stamp on CallerID and affects all the lines and extensions of the phone. Default is **Yes** (on).
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 **-16.0**.
DTMF Playback Length	Local DTMF playback duration in milliseconds. The default is **.1**.
Detect ABCD	To enable local detection of DTMF ABCD, select yes. Otherwise, select no. The default is **yes**. Setting has no effect if DTMF Tx Method is INFO; ABCD is always sent OOB regardless in this setting.
Playback ABCD	To enable local playback of OOB DTMF ABCD, select yes. Otherwise, select no. The default is **yes**.

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 sentbefore 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. ▪ **DTMF (Denmark) With PR**—CID only. DTMF sent after polarity reversal (and no DTAS) and before first ring. The default is **Bellcore(N.Amer, China)**.
Caller ID FSK Standard	The WRP400 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**.
More Echo Suppression	Enable or disable more echo suppresion. The default is **no**.

Line page

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

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.

Voice tab > Line page >

Line Enable section

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

Voice tab > Line page >

Streaming Audio Server (SAS) section

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**.
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 WRP400 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.

Voice tab > Line page >

NAT Settings section

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**.
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**.

Voice tab > Line page >

Network Settings section

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 Level	Determines how jitter buffer size is adjusted by the WRP400. Jitter buffer size is adjusted dynamically. The minimum jitter buffer size is 30 milliseconds or (10 milliseconds + current RTP frame size), whichever is larger, for all jitter level settings. However, the starting jitter buffer size value is larger for higher jitter levels. This setting controls the rate at which the jitter buffer size is adjusted to reach the minimum. Select the appropriate setting: **low**, **medium**, **high**, **very high**, or **extremely high**. The default is **high**.
Jitter Buffer Adjustment	Controls how the jitter buffer should be adjusted. Select the appropriate setting: **up and down**, **up only**, **down only**, or **disable**. The default is **up and down**.

Voice tab > Line page >

SIP Settings section

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 WRP400 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 WRP400 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 **yes**.

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.
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 WRP400 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 WRP400 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**.
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.

Voice tab > Line page >

Call Feature Settings section

Blind Attn-Xfer Enable	Enables the WRP400 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 WRP400 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**.

Voice tab > Line page >

Proxy and Registration section

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	Enablse 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 WRP400 will renew registration based on this smaller value instead of the configured value. If registration failed with an Expires too brief error response, the WRP400 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 WRP400 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 WRP400 will automatically prepend the Proxy or Outbound Proxy name with _sip._udp when performing a DNS SRV lookup on that name. The default is **no**.
Proxy Fallback Intvl	This parameter sets the delay (sec) after which the WRP400 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 WRP400 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 WRP400 will not attempt to fall back after a fail over). The default is **3600**
Proxy Redundancy Method	The WRP400 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 WRP400 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.

Voice tab > Line page >

Subscriber Information section

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.
Call Capacity	Maximum number of calls allowed on this line interface. Choices: {unlimited,1,2,3,...25 }. Default is **16**. Note that the the WRP400 does not distinguish between incoming and outgoing calls when talking about call capacity. **NOTE:** unlimited = 16
Cfwd No Ans Delay	Delay, in seconds, before the call forwarding of no-answer calls feature is triggered. The default is **20**.
Mini Certificate	Base64 encoded of Mini-Certificate concatenated with the 1024-bit public key of the CA signing the MC of all subscribers in the group. The default is empty.
SRTP Private Key	Base64 encoded of the 512-bit private key per subscriber for establishment of a secure call. The default is empty.

Voice tab > Line page >

Supplementary Service Subscription section

The WRP400 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 WRP400.

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.
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 **yes**.

Voice tab > Line page >

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. On the other hand, two G.723.1/G.726 resources are available per device.

Therefore it is important to disable the use of G.729a in order to guarantee the support of two simultaneous G.723/G.726 codec.

Preferred Codec	Preferred codec for all calls. (The actual codec used in a call still depends on the outcome of the codec negotiation protocol.) Select one of the following: **G711u, G711a, G726-16, G726-24, G726-32, G726-40, G729a,** or **G723.** 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, G726-16, G726-24, G726-32, G726-40, G729a,** or **G723.** 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, G726-16, G726-24, G726-32, G726-40, G729a,** or **G723.** 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**.
Silence Threshold	Select the appropriate setting for the threshold: high, medium, or low. The default is **medium**.
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 canceller, select yes. Otherwise, select no. The default is **yes**.
G723 Enable	To enable the use of the G.723a codec at 6.3 kbps, select yes. Otherwise, select no. The default is **yes**.
Echo Canc Adapt Enable	To enable the echo canceller to adapt, select yes. Otherwise, select no. The default is **yes**.
G726-16 Enable	To enable the use of the G.726 codec at 16 kbps, 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**.

G726-24 Enable	To enable the use of the G.726 codec at 24 kbps, 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**.
G726-32 Enable	To enable the use of the G.726 codec at 32 kbps, select yes. Otherwise, select no. The default is **yes**.
FAX CNG Detect Enable	To enable detection of the fax Calling Tone (CNG), select yes. Otherwise, select no. The default is **yes**.
G726-40 Enable	To enable the use of the G.726 codec at 40 kbps, 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**.
DTMF Process AVT	To use the DTMF process AVT feature, select yes. Otherwise, select no. The default is **yes**.
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+INFO**, or **AVT+INFO.** InBand sends DTMF using the audio path. AVT sends DTMF as eypents. INFO uses the SIP INFO method. Auto uses InBand or AVT based on the outcome of codec negotiation. The default is **Auto**.
DTMF Tx Mode	DTMF Detection Tx Mode is available for SIP information and AVT . Options are: **Strict** or **Normal**. The default is **Strict** for which the following are true: ▪ A DTMF digit requires an extra hold time after detection. ▪ The DTMF level threshold is raised to -20 dBm. ▪ The minimum and maximum duration thresholds are: • strict mode for AVT: 70 ms • normal mode for AVT: 40 ms • strict mode for SIP info: 90 ms • normal mode for SIP info: 50 ms
DTMF Tx Strict Hold Off Time:	This parameter is in effect only when "DTMF Tx Mode" is set to "strict," and when"DTMF Tx Method" is set to out-of-band; i.e. either AVT or SIP-INFO. If a user inadvertently sets the value to less than the default value, the system checks and reverts to the default value. There is no max limit on what the user can set of this parameter. A larger value will reduce the chance of talk-off (beeping) during conversation, at the expense of reduced performance of dtmf detection, which is needed for interactive voice response systems (IVR). Default is **90 ms**.
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**.

FAX Disable ECAN	If enabled, this feature automatically disables the echo canceller when a fax tone is detected. To use this feature, select yes. Otherwise, select no. The default is **no**.
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, then select caller or callee. If you want the Gateway to detect the fax tone only if the Gateway is the caller, then select caller only. If you want the Gateway to detect the fax tone only if the Gateway is the callee, then select callee only.
FAX Tone Detect Mode	This parameter has three possible values: ▪ caller or callee - The WRP400 will detect FAX tone whether it is callee or caller ▪ caller only - The WRP400 will detect FAX tone only if it is the caller ▪ callee only - The WRP400 will detect FAX tone only if it is the callee The default is **caller or callee**.

Voice tab > Line page >

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.). The syntax for a dial plan expression is as follows:

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

Dial Plan	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:
	`*1xxxxxxxxxx<:@fwdnat.pulver.com:5082;uid=jsmith;pwd=xyz`
	Example 2:
	`*1xxxxxxxxxx<:@fwd.pulver.com;nat;uid=jsmith;pwd=xyz`
	Example 3:
	`[39]11<:@gw0>`

Enable IP Dialing	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**.
Emergency Number	Comma separated list of emergency number patterns. If outbound call matches one of the pattern, the WRP400 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.

Voice tab > Line page >

FXS Port Polarity Configuration section

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:

- **"Call Forward Settings section" section on page 111**
- **"Selective Call Forward Settings section" section on page 112**
- **"Speed Dial Settings section" section on page 112**
- **"Supplementary Service Settings section" section on page 113**
- **"Distinctive Ring Settings section" section on page 114**
- **"Ring Settings section" section on page 115**

When a call is made from Line 1 or Line 2, the WRP400 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.

Voice tab > User page >

Call Forward Settings section

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.
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**.

Voice tab > User page >

Selective Call Forward Settings section

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.

Voice tab > User page >

Speed Dial Settings section

This section does not apply to the WIP310 wireless phone.

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.

Voice tab > User page >

Supplementary Service Settings section

The WRP400 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 WRP400.

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 WRP400. 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**.
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 WRP400 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 WRP400 answers the call, then the loopback type is determined by the caller (the WRP400 always picks the first loopback type in the offer if it contains multiple types.)

Voice tab > User page >

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.

Ring1 - 9 Caller	Caller number pattern to play Distinctive Ring/CWT 1, 2, 3, 4, 5, 6, 7, 8, or 9. The default is **blank**.

Voice tab > User page >

Ring Settings section

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 **2**.
Hold Reminder Ring	Ring pattern for reminder of a holding call when the phone is on-hook. The default is **None**.
Call Back Ring	Ring pattern for call back notification. The default is **None**.
Cfwd Ring Splash Len	Duration of ring splash when a call is forwarded (0 – 10.0s). The default is **0**.
Cblk Ring Splash Len	Duration of ring splash when a call is blocked (0 – 10.0s). The default is **0**.
VMWI Ring Splash Len	Duration of ring splash when new messages arrive before the VMWI signal is applied (0 – 10.0s). The default is **.5**.
VMWI Ring Policy	The parameter controls when a ring splash is played when a the VM server sends a SIP NOTIFY message to the WRP400 indicating the status of the subscriber's mail box. 3 settings are available: ▪ **New VM Available**—ring as long as there is 1 or more unread voice mail ▪ **New VM Becomes Available**—ring when the number of unread voice mail changes from 0 to non-zero ▪ **New VM Arrives**—ring when the number of unread voice mail increases. The default is **New VM Available**.

Ring On No New VM	If enabled, the WRP400 will play a ring splash when the VM server sends SIP NOTIFY message to the WRP400 indicating that there are no more unread voice mails. Some equipment requires a short ring to precede the FSK signal to turn off VMWI lamp. The default is **no**.

B

Data Fields

This appendix describes the fields for the data parameters.

Setup

The Setup module includes the following pages:

Setup > Basic Setup

Internet Setup	
Internet Connection Type	The type of Internet connection: Automatic Configuration - DHCP, Static IP, PPPoE, PPTP, L2TP, Telstra Cable
	Static IP: ▪ Internet IP Address: The IP address of your WRP400, as seen from the Internet. ▪ Subnet Mask: The subnet mask, as seen by users on the Internet (including your service provider). ▪ Default Gateway: The IP address of your service provider server.

	PPPoE: ▪ User Name: The user name for your account with your service provider. ▪ Password: The password for your account with your service provider. ▪ Service Name (optional): For the PPPoE connection type, the service name (if provided). ▪ Connect on Demand: For a PPPoE, PPTP, L2TP, or Telstra Cable connection type, a feature that allows your WRP400 to re-establish a terminated connection when a user attempts to access the Internet. ▪ Max Idle Time: When Connection on Demand is enabled, use the Max Idle Time field to specify the period of inactivity that causes a connection to terminate. Default: 5 minutes ▪ Keep Alive: a feature that allows your WRP400 to check your Internet connection at a specified interval (Redial Period). If you are disconnected, then the WRP400 automatically re-establishes your connection. ▪ Redial Period: When Keep Alive is enabled, this period is the interval in seconds at which the Internet connection is checked. Default: 30 seconds

	PPTP: ▪ Gateway: The IP address of your service provider server. ▪ User Name: The user name for your account with your service provider. ▪ Password: The password for your account with your service provider. ▪ Connect on Demand: For a PPPoE, PPTP, L2TP, or Telstra Cable connection type, a feature that allows your WRP400 to re-establish a terminated connection when a user attempts to access the Internet. ▪ Max Idle Time: When Connection on Demand is enabled, use the Max Idle Time field to specify the period of inactivity that causes a connection to terminate. Default: 5 minutes ▪ Keep Alive: a feature that allows your WRP400 to check your Internet connection at a specified interval (Redial Period). If you are disconnected, then the WRP400 automatically re-establishes your connection. ▪ Redial Period: When Keep Alive is enabled, this period is the interval in seconds at which the Internet connection is checked. Default: 30 seconds

	L2TP: ▪ Server IP Address: The IP address of your service provider server. ▪ User Name: The user name for your account with your service provider. ▪ Password: The password for your account with your service provider. ▪ Connect on Demand: For a PPPoE, PPTP, L2TP, or Telstra Cable connection type, a feature that allows your WRP400 to re-establish a terminated connection when a user attempts to access the Internet. ▪ Max Idle Time: When Connection on Demand is enabled, use the Max Idle Time field to specify the period of inactivity that causes a connection to terminate. Default: 5 minutes ▪ Keep Alive: a feature that allows your WRP400 to check your Internet connection at a specified interval (Redial Period). If you are disconnected, then the WRP400 automatically re-establishes your connection. ▪ Redial Period: When Keep Alive is enabled, this period is the interval in seconds at which the Internet connection is checked. Default: 30 seconds

	Telstra Cable: ▪ User Name: The user name for your account with your service provider. ▪ Password: The password for your account with your service provider. ▪ Connect on Demand: For a PPPoE, PPTP, L2TP, or Telstra Cable connection type, a feature that allows your WRP400 to re-establish a terminated connection when a user attempts to access the Internet. ▪ Max Idle Time: When Connection on Demand is enabled, use the Max Idle Time field to specify the period of inactivity that causes a connection to terminate. Default: 5 minutes ▪ Keep Alive: a feature that allows your WRP400 to check your Internet connection at a specified interval (Redial Period). If you are disconnected, then the WRP400 automatically re-establishes your connection. ▪ Redial Period: When Keep Alive is enabled, this period is the interval in seconds at which the Internet connection is checked. Default: 30 seconds ▪ Heart Beat Server: The IP address of the Heart Beat Server.
Host Name	A host name for the WRP400. Some service providers, usually cable service providers, require a host name and a domain name as identification. In most cases, these fields can be left blank.
Domain Name	A domain name for the WRP400. Some service providers, usually cable service providers, require a host name and a domain name as identification. In most cases, these fields can be left blank.

MTU	Maximum Transmission Unit. The largest packet size that is permitted for Internet transmission. Select Manual if you want to manually enter the largest packet size that is transmitted. To have the WRP400 select the best MTU for your Internet connection, keep the default setting, Auto.
MTU Size	When Manual is selected in the MTU field, this option is enabled. Set this value in the 576 to 1500 range. The default size depends on the Internet Connection Type: ▪ DHCP or Static IP: 1500 ▪ PPPoE: 1492 ▪ PPTP or L2TP: 460 ▪ Telstra Cable: 1500
Static DNS 1, 2, 3	The Domain Name System (DNS) is how the Internet translates domain or website names into Internet addresses or URLs. Enter the IP address of the DNS server, which is provided by your service provider. If you wish to use a different DNS server, enter its IP address in one of these fields. You can enter up to three DNS server IP addresses here. The WRP400 will use these for quicker access to functioning DNS servers. By default, the WRP400 uses 192.168.15.1 for DNS.
Network Setup	
Local IP Address	The address of the WRP400 on the local area network.
Subnet Mask	The subnet mask for the local area network.
DHCP Server	When this feature is enabled, the WRP400 assigns IP addresses dynamically to the connected devices. Default:: Enabled
DHCP Reservation	You can use this feature to reserve IP addresses for use by specified devices on your network.

DNS Proxy	The DNS proxy relays DNS requests to the current public network DNS server for the proxy, and it replies as a DNS resolver to the client device on the network. Default: Disabled
Starting IP Address	The first IP address in the range of addresses that the DHCP server issues to connected devices. The Starting IP Address must be greater than the default IP address of the WRP400, 192.168.15.1, and less than 192.168.15.253. Default: 192.168.15.100
Maximum DHCP Users	The maximum number of computers that will receive IP addresses from the DHCP server. This number cannot be greater than 253. Default: 50
IP Address Range	The range of available IP addresses
Client Lease Time	The maximum connection time in minutes that a a dynamic IP address is “leased” to a network user. When the time elapses, the user is automatically assigned a new dynamic IP address. Default: 0 minutes (1 day)
Static DNS	The local IP address of the DNS server, which is provided by your service provider. If you wish to use a different DNS server, enter that IP address in this field. The Domain Name System (DNS) is how the Internet translates domain or website names into Internet addresses or URLs.
WINS	The Windows Internet Naming Service (WINS) manages each PC’s interaction with the Internet. Enter the IP address of the WINS server, if applicable. Default: 0.0.0.0.

Time Setting	
Time Zone	The time zone for the location.
Automatically adjust clock for daylight saving changes	When this feature is enabled, the WRP400 automatically adjusts the clock for daylight saving time. Default: Enabled
Time Server Address	The time server that is use to obtain time settings. When the Time Server Address is set to Manual, the IP address can be entered in the NTP Server Address field. Default: Auto
Resync Timer	The number of seconds that elapse before the WRP400 resyncs with the NTP server. Default: 3600 seconds

Setup > DDNS

User Name	For DynDNS.org service, the user name for your DDNS account.
Password	For DynDNS.org service, the password for your DDNS account.
Host Name	For DynDNS.org service, the DDNS URL assigned by the DDNS service.
System	For DynDNS.org service, the type of DynDNS service you use: Dynamic, Static, or Custom. Default: Dynamic
Mail Exchange (Optional)	For DynDNS.org service, the address of your mail exchange server. Emails addressed to your DynDNS address will go to this mail server.
Backup MX	For DynDNS.org service, this feature allows the mail exchange server to be a backup. To enable the feature, select Enabled. Default: Disabled

Wildcard	For DynDNS.org service, this setting enables or disables wildcards for your host. For example, if your DDNS address is myplace.dyndns.org and you enable wildcards, then x.myplace.dyndns.org will work as well (x is the wildcard). To enable wildcards, select Enabled. Default: Disabled
E-mail Address	For TZO.com service, the email address for your account.
TZO Key	For TZO.com service, the key for your account.
Domain Name	For TZO.com service, the domain name for your WRP400.

Setup > MAC Address Clone

Enabled, Disabled	When this feature is enabled, you can assign a previously registered MAC address to the WRP400 if needed to meet the requirements of your service provider.
MAC Address	The MAC address that you previously registered with your service provider for this account.

Setup > Advanced Routing

NAT	If the WRP400 is hosting your network's connection to the Internet, keep the default, **Enabled**. If another router exists on your network, select **Disabled**. When the NAT setting is disabled, dynamic routing will be enabled.

Dynamic Routing (RIP)	This feature enables the WRP400 to automatically adjust to physical changes in the network's layout and to exchange routing tables with the other router(s). The WRP400 determines the route of the network packets based on the fewest number of hops between the source and the destination. When the NAT setting is enabled, the Dynamic Routing feature is automatically disabled. When the NAT setting is disabled, this feature is available.
Static Routing	Pre-determined pathways that network information travels to reach a specific host or network. ▪ Route Name: A name for the route, including up to 25 alphanumeric characters. ▪ Destination LAN IP: The address of the remote network or host to which you want to assign a static route. ▪ Subnet Mask: The subnet mask for the destination network. ▪ Gateway: The IP address of the gateway device that allows for contact between the WRP400 and the remote network or host. ▪ Interface: The location of the destination network, LAN and Wireless (Ethernet and wireless networks) or the Internet (WAN).

Setup > Mobile Network

Connect Mode	▪ Auto: When this mode is selected, the modem can establish a connection automatically. ▪ Manual: When this mode is selected, a connection is established manually.
Connect on Demand	When this feature is enabled, the modem can automatically re-establish a terminated connection when a user attempts to access the Internet again.

Max. Idle Time	The number of minutes of inactivity that can elapse before an Internet connection is terminated. Default: 5 minutes
Keep Alive	When this feature is enabled, the WRP400 periodically checks the Internet connection. If the connection is terminated, then the WRP400 automatically re-establishes the connection.
Card Status	The current modem connection status as Detecting, Connecting, or Connected.
Configure Mode	▪ Auto: When this mode is enabled, the WRP400 automatically detects which card model was inserted and which carrier is available. ▪ Manual: When this mode is enabled, the connection is established manually.
Card Model	The data card model that is inserted in the USB port.
Carrier	The mobile network service provider for Internet connection. This setting is required when you are using HSDPA/UMTS/GPRS Internet service.
Country	The card issue country
Carrier	The card issue provider
Access Point Name (APN)	The name that the mobile network service provider has assigned to the particular Internet network for this connection.
Dial Number	The dial number that is used to access the mobile network service.
User Name and Password (Optional)	The user name and password, if any, provided by your mobile network service provider.
SIM PIN (Optional)	The PIN code associated with your SIM card, if required.
Server Name (Optional)	The name of the server for the Internet connection, if required.
Authentication	The type of authentication used by your service provider. Default: Auto

Service Type	The most commonly available type of mobile data service connection based on your area service signal. If your location supports only one mobile data service, you may set up for enhance build up connection. The first selection will always search for HSPDA/3G/UMTS service or switch to GPRS automatically only when it is available.

Setup > Connection Recovery

Recovery & Failover	
Ethernet Connection Recovery	When this feature is enabled, the Internet connection is made through the Ethernet interface if it is available. This feature also enables the Interface Connection Failover feature, so that a connection failure on the Ethernet interface causes the WRP400 to attempt to bring up the connection through the mobile network if available. Whenever the Ethernet Internet connection recovers, the WRP400 automatically attempts to bring back and recover the Ethernet Internet connection. Ethernet Connection Recovery requires that the Mobile Connection Mode is set to Auto and the Ethernet interface is set to the high priority.
Interface Connection Failover	When this feature is enabled, the WRP400 detects the physical connection and/or presence of traffic on the Internet link. If the link is idle for some time, the WRP400 attempts to ping a destination. If the ping does not reply, the WRP400 assumes the link is down and attempts to fail over to another interface. This feature is automatically enabled if Ethernet Connection Recovery is enabled.
Timeout	Specify the time interval at which the WRP400 detects the status of the Internet connection. Default: 60 seconds

Failover Validation Site	Optional. A ping target for the WRP400 to use to detect the status of the Internet connection. If you do not specify an IP address here, the WRP400 uses the Network Time Protocol (NTP) server as the ping target.
WAN Interfaces	
Interface	The interface: Ethernet or USB
Status	The status of the interface: Connected or Disconnected
Priority	Determines which interface is used when both interfaces are available. The priority is indicated by the order in which the interfaces appear in the Summary Table. This setting is configurable only when Ethernet Connection Recovery is disabled. Default: Ethernet interface has top priority.

Wireless Configuration

The Wireless module includes the following pages:

- **"Wireless > Basic Wireless Settings" on page 131**
- **"Wireless > Wireless Security" on page 132**
- **"Wireless > Wireless MAC Filter" on page 133**
- **"Wireless > Advanced Wireless Settings" on page 134**

Wireless > Basic Wireless Settings

Network Mode	The wireless standards that are running on your network: ▪ Mixed: Choose this setting if the network has Wireless-G and Wireless-B devices. ▪ Wireless-G only: Choose this setting if the network has only Wireless-G devices ▪ Wireless-B only: Choose this setting if the network has only Wireless-B devices. Default: Mixed
Wireless Channel	The channel that is used by the wireless nework. To enable the WRP400 to select the best available wireless channel, choose Auto. Default: Auto
SSID1 Network Enabled, SSID2 Network Enabled	When this feature is enabled, the network is active.
Wireless Network Name (SSID)	A name for your wireless network, including up to 32 characters. Any character on the keyboard can be used. The name is case sensitive. The default wireless network (SSID1) uses a name with the following pattern: cisco<*MAC*> where <*MAC*> represents the last four digits of the wireless MAC address of the WRP400.
SSID Broadcast Enabled	When this feature is enabled, the WRP400 allows its SSID to be detected by wireless clientsdevices within range. When this feature is disabled, a wireless device can connect to the network only if the user enters the network name to establish a connection.
For Internet Access Only	Applies to SSID2 only. When this feature is enabled, connected devices have access to the Internet but are blocked from accessing to your local network. This feature is useful for establishing a guest wireless network for use by customers and visitors.

Wireless > Wireless Security

SSID	The name of the wireless network
Security Mode	The type of security that is used on the network: WEP, WPA Personal, WPA2 Personal, WPA Enterprise, WPA2 Enterprise.
	WEP: ▪ Encryption: 64 bits 10 hex digits or 28 bits 26 hex digits. Default: 64 bits 10 hex digits ▪ Passphrase: A passphrase used to automatically generate WEP keys. ▪ Key 1-4: Instead of using a passphrase to generate WEP keys, enter the WEP key(s) manually. ▪ TX Key: The TX (Transmit) Key to use. Default: 1
	WPA Personal and WPA2 Personal: ▪ WPA Algorithms: The encryption method, TKIP or AES for dynamic encryption keys. Default: TKIP ▪ WPA Shared Key: A WPA Shared Key of 8-63 characters. ▪ Group Key Renewal: The period of time in seconds that can elapse before the WRP400 changes the encryption keys. Default: 3600 seconds (1 hour)

	WPA Enterprise and WPA2 Enterprise: ▪ WPA Algorithms: The encryption method for dynamic encryption keys. For WPA Enterprise, the options are AES or TKIP (default). For WPA2 Enterprise, the options are AES or TKIP+AES (default). ▪ RADIUS Server Address: The IP address of the RADIUS server to be used for authentication. ▪ RADIUS Port: The port number of the RADIUS server. The default value is 1812. ▪ Shared Key: The key shared to be used to establish a connection between the WRP400 and the server. ▪ Key Renewal Timeout: The period of time in minutes that can elapse before the WRP400 changes the encryption keys. Default: 600 seconds (10 minutes)

Wireless > Wireless MAC Filter

SSID	The network name of the network for this MAC filter.
Wireless MAC Filter	When this feature is enabled, the specified Access Restriction is applied to the specified clients. When this feature is disabled, access is not filtered by MAC address. Default: Disabled
Access Restriction	Prevent: Devices in the MAC Address Filter List are prevented from connecting to the specified wireless network. Permit: Only devices in the MAC Address Filter List are allowed to connect to the specified wireless network.
MAC Address Filter List MAC 01 through MAC 40	The MAC address of each machine that is subject to the specified access restrictions.

Wireless > Advanced Wireless Settings

Authentication Type	When the type is set to Auto, either Open System or Shared Key authentication can be used. ▪ With Open System authentication, the sender and the recipient do not use a WEP key for authentication. ▪ With Shared Key authentication, the sender and recipient use a WEP key for authentication. Default: Auto
Transmission Rate	The rate of data transmission. Choose an appropriate setting based on the speed of your wireless network(s). Select Auto to have the WRP400 automatically use the fastest possible data rate and to enable the Auto-Fallback feature. Auto-Fallback will negotiate the best possible connection speed between the WRP400 and a wireless client. Default: Auto
CTS Protection Mode	Clear-To-Send Protection Mode. This function boosts the ability of the WRP400 to catch all Wireless-G transmissions but will severely decrease performance. When this setting is on Auto, the WRP400 switches to CTS Protection Mode whenever your Wireless-G devices are experiencing severe problems and are not able to transmit to the WRP400 in an environment with heavy 802.11b traffic. Default: Auto
Beacon Interval	The interval in milliseconds when the WRP400 transmits a beacon, which is a packet broadcast to synchronize the wireless network(s). Enter a value between 20 and 65,535 milliseconds. Default: 100 Milliseconds

DTIM Interval	The interval for sending the Delivery Traffic Indication Message (DTIM). A DTIM field is a countdown field informing clients of the next window for listening to broadcast and multicast messages. When the WRP400 has buffered broadcast or multicast messages for associated clients, it sends the next DTIM with a DTIM Interval value. Its clients hear the beacons and awaken to receive the broadcast and multicast messages. Enter a value between 1 and 255. Default: 1
RTS Threshold	The WRP400 sends Request to Send (RTS) frames to a particular receiving station and negotiates the sending of a data frame. After receiving an RTS, the wireless station responds with a Clear to Send (CTS) frame to acknowledge the right to begin transmission. If you encounter inconsistent data flow, only minor reduction of the default value, 2347, is recommended. If a network packet is smaller than the preset RTS threshold size, the RTS/ CTS mechanism will not be enabled. Default: 2347

Security

The Security module includes the following pages:

- **"Security > Firewall" on page 136**
- **"Security > VPN Passthrough" on page 137**

Security > Firewall

Firewall	
SPI Firewall Protection	To use firewall protection, keep the default, Enabled. To turn off firewall protection, select Disabled. Default: Enabled
Internet Filter	
Filter Anonymous Internet Requests	When thisfeature is enabled, it is more difficult for outside users to access your network. Disable this feature if you want to allow anonymous Internet requests. Default: Enabled
Filter Internet NAT Redirection	This feature uses port forwarding to block access to local servers from local networked computers. Select this feature to filter Internet NAT redirection. Default: Disabled
Filter IDENT (Port 113)	This feature keeps port 113 from being scanned by devices outside of your local network. Default: Enabled
Web Filter	
Proxy	The use of WAN proxy servers may compromise the security of your network. Denying Proxy will disable access to any WAN proxy servers. Select this feature to enable proxy filtering. Deselect the feature to allow proxy access. Default: Disabled
Java	Java is a programming language for websites. If you deny Java, you run the risk of not having access to Internet sites created using this programming language. Select this feature to enable Java filtering. Deselect the feature to allow Java usage. Default: Disabled

ActiveX	ActiveX is a programming language for websites. If you deny ActiveX, you run the risk of not having access to Internet sites created using this programming language. Select this feature to enable ActiveX filtering. Deselect the feature to allow ActiveX usage. Default: Disabled
Cookies	A cookie is data stored on your computer and used by Internet sites when you interact with them. Select this feature to filter cookies. Deselect the feature to allow cookie usage. Default: Disabled

Security > VPN Passthrough

IPSec Passthrough	Internet Protocol Security (IPSec) is a suite of protocols used to implement secure exchange of packets at the IP layer. When this feature is enabled, IPSec tunnels are allowed to pass through the WRP400. Default: Enabled
PPTP	Passthrough Point-to-Point Tunneling Protocol (PPTP) allows the Point-to-Point Protocol (PPP) to be tunneled through an IP network. When this feature is enabled, PPTP tunnels are allowed to pass through the WRP400. Default: Enabled
L2TP Passthrough	Layer 2 Tunneling Protocol is the method used to enable Point-to-Point sessions via the Internet on the Layer 2 level. When this feature is enabled, L2TP tunnels are allowed to pass through the WRP400. Default: Enabled

Access Restrictions

The Access Restrictions module includes the following pages:

- "Access Restrictions > Internet Access" on page 138

Access Restrictions > Internet Access

Enter Policy Name	A name for the policy
Status	Policies are disabled by default. To enable the selected policy, select **Enabled**.
Applied PCs	The computers that will be affected by the policy that you selected in the Access Policy list. - MAC Address: The MAC address of the device. - IP Address: The final octet of the IP address. - IP Address Range: A range of devices, identfied by the final octet of the starting IP address and the final octect of the ending IP address.
Access Restriction	Deny: Prevent the listed computers from accessing the Internet. Allow: Permit the listed computers to access the Internet.
Schedule	The days and times when the policy is enforced. The Everyday option applies the policy to all days of the week, or you can specify the days. The 24 Hours option applies the policy to all hours of the specified days, or you can speciy the time range.
Website Blocking by URL Address	A list of website addresses that users are prevented from accessing.
Website Blocking by Keyword	A list of keywords that are used to prevent access to inappropriate websites.
Blocked Applications	A list of applications, such as FTP, that users are prevented from using. Up to three applications can be blocked for each policy.

Applications and Gaming

The Applications and Gaming module includes the following pages:

Applications and Gaming > Single Port Forwarding

Application Name	A name for the application. Each name can be up to 12 characters.
External Port	The external port number used by the server or Internet application. Check with the Internet application documentation for more information.
Internal Port	The internal port number used by the server or Internet application. Check with the Internet application documentation for more information.
Protocol	The protocol used for this application, either TCP, UDP, or both.
To IP Address	The IP address of the PC that should receive the requests.
Enabled	When selected, port forwarding is active.

Applications and Gaming > Port Range Forwarding

Application Name	A name for the application. Each name can be up to 12 characters.
Start~End Port	The number or range of port(s) used by the server or Internet applications. Check with the Internet application documentation for more information.

Protocol	The protocol used for this application, either TCP, UDP, or both.
To IP Address	The IP address of the PC running the specific application.
Enabled	When selected, port forwarding is active.

Applications & Gaming > Port Range Triggering

Application Name	The application name of the trigger.
Triggered Range	The starting and ending port numbers of the triggered port number range.
Forwarded Range	The starting and ending port numbers of the forwarded port number range.

Applications & Gaming > DMZ

Enabled/Disabled	To disable DMZ hosting, select **Disabled**. To expose one PC, select **Enabled**. Then configure the Source IP Address and Destination.
Source IP Address	Select **Any IP Address**, or specify an IP address or range of IP addresses as the designated source.
Destination	If you want to specify the DMZ host by IP address, select **IP Address** and enter the IP address in the field provided. If you want to specify the DMZ host by MAC address, select **MAC Address** and enter the MAC address in the field provided.

Applications and Gaming > QoS (Quality of Service)

Wireless	
WMM Support	To support Wi-Fi Multimedia (WMM) on your network, select Enabled. Default: Disabled
No Acknowledgement	To prevent the WRP400 from resending data if an error occurs, select Enabled. Default: Disabled
Internet Access Priority	
Enabled/Disabled	To use the QoS policies you have set, keep the default, Enabled. Otherwise, select Disabled.

Upstream Bandwidth	To allow the WRP400 to control the maximum bandwidth for upstream data transmissions, keep the default, **Auto**. To manually set the maximum, select **Manual**, and enter the appropriate number in the field provided.
Category	Identify the category by choosing Application, Online Games, MAC Address, or Ethernet Port.
	Application: Application: Select an application from the list or click **Add a New Application**. ▪ Enter a Name: A name to identify the application. ▪ Port Range: The range of ports for this application. You can have up to three ranges to define for this bandwidth allocation. Port numbers can range from 1 to 65535. Check your application's documentation for details on the service ports used. ▪ Protocol: Choose TCP, UDP, or Both. ▪ Priority: Select the appropriate priority: High, Medium (Recommend), Normal, or Low.
	Online Games: Select a game from the list, or click **Add a New Game**. ▪ Enter a Name: Enter any name to indicate the name of the entry. ▪ Port Range: The range of ports for this game. For example, ito allocate bandwidth for FTP, enter 21-21. If you need services for an application that uses from 1000 to 1250, you enter 1000-1250 as your settings. You can have up to three ranges to define for this bandwidth allocation. Port numbers can range from 1 to 65535. Check your application's documentation for details on the service ports used. ▪ Protocol: Choose TCP, UDP, or Both. ▪ Priority: Select the appropriate priority: High, Medium (Recommend), Normal, or Low.

	MAC Address: ▪ Enter a Name: A name for the device. ▪ MAC Address: The MAC address of the device. ▪ Priority: The appropriate priority: High, Medium (Recommend), Normal, or Low.
	Ethernet Port ▪ Ethernet: Select the appropriate Ethernet port. ▪ Priority: Select the appropriate priority: High, Medium (Recommend), Normal, or Low.

Administration

The Administration module includes the following pages:

- "Administration > Management" on page 143
- "Administration > Log" on page 146
- "Administration > Diagnostics" on page 147
- "Administration > Factory Defaults" on page 147

Administration > Management

Router Access	
Router Password	The administrative password for the WRP400. When changing the password, re-enter the password in the **Re-enter to Confirm** field.

Web Access	
Web Utility Access	The protocol that is used for access to the web-based configuration utility. The options are HTTP or HTTPS. HTTP (HyperText Transport Protocol) is the communications protocol used to connect to servers on the World Wide Web. HTTPS uses SSL (Secured Socket Layer) to encrypt data transmitted for higher security. Default: HTTP
Web Utility Access via Wireless	You can enable or disable wireless access to the web-based configuration utility. If you are using the WRP400 in a public domain where you are giving wireless access to your guests, you can disable wireless access to the web-based configuration utility of the WRP400. In this case, you will only be able to access the utility via a wired connection. Default: Enabled
Remote Access	
Remote Management	You can enable or disable remote access to the WRP400 from outside the local network. If you need to manage your WRP400 from a PC on the Internet, you can enable this feature. Default: Disabled
Web Utility Access	The protocol that is used for access to the web-based configuration utility. The options are HTTP or HTTPS. HTTP (HyperText Transport Protocol) is the communications protocol used to connect to servers on the World Wide Web. HTTPS uses SSL (Secured Socket Layer) to encrypt data transmitted for higher security. Default: HTTP

Remote Upgrade	You can enable or disable remote upgrades for your WRP400. If you need to upgrade your WRP400 from a PC on the Internet, you can enable this feature. The the Remote Management feature must be enabled as well. Default: Disabled
Allowed Remote IP Address	You can allow remote access from Any IP Address or restrict remote access to a specified IP address or range of IP addresses.
Remote Management Port	The port number the is open for remote access. To access your WRP400 from a remote location, enter the WRP400 IP address and the remote management port number as shown below: http://<Internet_IP_address>:port OR https://<Internet_IP_address>:port
UPnP	
UPnP	When Universal Plug and Play (UPnP) is enabled, Windows XP and Vista can automatically configure the WRP400 for various Internet applications, such as gaming and videoconferencing. Default: Enabled
Allow Users to Configure	When this feature is enabled, you can make manual changes to the automatic UPnP settings. Default: Enabled
Keep UPnP Configurations After System Reboot	When this feature is enabled, any manual changes in the UPnP configurations are saved when the WRP400 reboots. This feature requires enabling the Allow Users to Configure feature. Default: Disabled
Allow Users to Disable Internet Access	When this feature is enabled, you can prohibit any and all Internet connections. Default: Disabled

Multimedia Streaming	
RTSP Support	If you experience issues with video-on-demand applications, select Enabled to improve multimedia transmissions. Using this option, the WRP400 will establish channels with the Real Time Streaming Protocol) RTSP server, which is located at the service provider. Default: Disabled
IGMP	
Support IGMP Version	Select the version that you want to support, IGMP 1, IGMP v2, or IGMP 3. Default: IGMP v2
IGMP Proxy	When this feature is enabled, the WRP400 allows multicast traffic through the WRP400 for your multimedia application devices. Default: Enabled
Immediate Leave	When this feature is enabled, IPTV applications are allowed to do immediate channel swapping or flipping without lag or delays. Default: Disabled

Administration > Log

Log	To disable the Log function, keep the default, **Disabled**. To monitor traffic between the network and the Internet, select **Enabled**. With logging enabled, you can choose to view temporary logs. The logs can be viewed on the Administration > Log > View Log page.

Administration > Diagnostics

Ping Test	
IP or URL Address	The address of the PC whose connection you wish to test.
Packet Size	The packet size you want to use. Default: 32 bytes
Times to Ping	The number of times to run the ping test.
Traceroute Test	
IP or URL Address	The address of the PC whose connection you wish to test.
Detect Active LAN Client(s)	
Search Time	The duration of the search in seconds: 5, 10, or 15.

Administration > Factory Defaults

Restore Router Factory Defaults	To reset the router settings to the default values, select Yes. Then click Save Settings. Any custom router settings you have saved will be lost when the default settings are restored.
Restore Voice Factory Defaults	To reset the voice settings to the default values, select Yes. Then click Save Settings. Any custom Voice settings you have saved will be lost when the default settings are restored.

Status

The Status module includes the following pages:

Status > Router

Router Information	
Firmware Version	The version number of the current firmware is displayed.
Current Time	The time set on the WRP400 is displayed.
Internet MAC Address	The MAC address, as seen by your service provider, is displayed.
Router Name	The name of the WRP400 is displayed.
Host Name	If required by your service provider, this was entered on the Basic Setup screen.
Domain Name	If required by your service provider, this was entered on the Basic Setup screen.
Internet Connection	
Connection Type	The type of Internet connection: Automatic Configuration - DHCP, Static IP, PPPoE, PPTP, L2TP, Telstra Cable
Internet IP Address	The IP address of your WRP400, as seen from the Internet.
Subnet Mask	The subnet mask, as seen by users on the Internet
Default Gateway	The IP address of your service provider server.
DNS1, DNS2, DNS3	The addresses of the Domain Name Servers (DNS) servers for your Service Provider.
MTU	Maximum Transmission Unit. The largest packet size that is permitted for Internet transmission. Can be set manually or automatically.

Status > Mobile Network

Mobile Network Status	
Connection	The status of the mobile network connection, either Disconnected or Connected.
Connection Up Time	The period of time that the Mobile USB modem has been connected to the Internet during this session.
Current Session Usage	The number of packets have been downloaded and uploaded during this session.
Data Card Status	
Manufacturer	The manufacturer of the Mobile USB modem data card.
Card Model	The model number of your Mobile USB modem data card.
Card Firmware	The firmware that is installed on your Mobile USB modem data card.
SIM Status	The status of your SIM card.
IMSI	International Mobile Subscriber Identity is a unique number that is stored in the Subscriber Identity Module (SIM) associated with all GSM and Universal Mobile Telecommunications System (UMTS) network mobile phone users.
Carrier	The network service provider that is used for Internet connection.
Service Type	The current UMTS/GPRS/EVDO service for Internet connection.
Signal Strength	The signal strength of your current UMTS/GPRS/EVDO service to your location.
Card Status	The current Mobile WAN connection status.

Status > Local Network

Local Network	
Local MAC Address	The MAC address of the local, wired interface of the WRP400 is displayed.
Router IP Address	The IP address of the WRP400, as it appears on your local network, is displayed.
Subnet Mask	The Subnet Mask of the WRP400 is displayed.
DHCP Server	
DHCP Server	The status of the DHCP server function.
Start IP Address	The starting IP address for the range of IP addresses used by devices on your local network.
End IP Address	The ending IP address for the range of IP addresses used by devices on your local network.

Status > Wireless Network

Channel	The channel used by the wireless network(s) .
Mode	The wireless mode, which may be Mixed, Wireless-G only, or Wireless-B only.
Wireless Network 1, Wireless Network 2	
Wireless MAC Address	The wireless MAC address of the local, wireless interface.
Network Name (SSID)	The network name, which is also called the SSID.
Security	The wireless security method, which may be WEP, WPA Personal, WPA2 Personal, WPA Enterprise, WPA2 Enterprise.
SSID Broadcast	The status of the SSID Broadcast feature, which may be Enabled or Disabled.

C

WRP400 Provisioning Reference

This chapter provides information about the parameters that can be provisioned from an XML profile by using the profile compiler tool (Scomputer).

NOTE For instructions about provisioning, see the *SPA Provisioning Guide* at the following URL: www.cisco.com/en/US/docs/voice_ip_comm/csbpvga/ata/provisioning/guide/Cisco_Small_Business_IP_Telephony_Provisioning_Guide.pdf

Feature/XML Tag	Parameters	Examples
Wireless QoS <WL_QOS>	<WL_QOS>wl_wme,wl_wme_no_ack </WL_QOS> **wl_wme:** WMM support (Wi-Fi Multimedia); on (enabled) or off (disabled) **wl_wme_no_ack:** No-acknowledgement option; on (enabled) or off (disabled)	To enable WMM with the No-acknowledgement option turned off: <WL_QOS>wl_wme=on,wl_wme_no_ack=off</WL_QOS>
Internet Access Priority <RT_QOS>	<RT_QOS>QoS,rate_mode,manual_rate</RT_QOS> QoS: Internet access priority; 1 (enabled) or 0 (disabled) rate_mode: Upstream bandwidth type; 0 (manual) or 1 (automatic) manual_rate: Upstream bandwidth rate; numerals from 64 to 50000	To enable Manual QoS and specify the upstream bandwidth rate: <RT_QOS>QoS=1,rate_mode=0, manual_rate=5000</RT_QOS> To enable Auto QoS: <RT_QOS> QoS=1, rate_mode=1</RT_QOS> To disable QoS: <RT_QOS>QoS=0 </RT_QOS>

Feature/XML Tag	Parameters	Examples
RTSP <RTSP>	<RTSP>rtsp_enable</RTSP> rtsp_enable: Real Time Streaming Protocol (RTSP); 1 (enabled) or 0 (disabled)	To enable RTSP: <RTSP>rtsp_enable=1 </RTSP> To disable RTSP: <RTSP>rtsp_enable=0 </RTSP>
IGMP <IGMP>	<IGMP>force_igmp_version,multicast_pass,multicast_immediate_leave </IGMP> force_igmp_version: Specifies the version of IGMP that is supported; 1 (IGMP v1, RFC 1112), 2 (IGMP v2, RFC 2236) or 3 (IGMP v3, RFC 3376) multicast_pass: IGMP proxy, allows multicast traffic through the router for your multimedia application devices; 1 (enabled) or 0 (disabled) multicast_immediate_leave: Allows immediate channel swapping or flipping without lag or delays; 1 (eanbled) or 0 (disabled)	To specify IGMP version 1 with multicast pass through and immediate leave: <IGMP>force_igmp_version=1, multicast_pass=1,multicast_immediate_leave=1</IGMP>
UPnP <UPNP>	<UPNP>upnp_enable,upnp_config, upnp_keep_portmap</UPNP> upnp_enable: UPnP status; 1 (enabled) or 0 (disabled) upnp_config: Allows configuration of UPnP; 1 (enabled) or 0 (disabled) upnp_keep_portmap: Keeps UPnP configurations after system reboot; 1 (enabled) or 0 (disabled) **NOTE:** This paramater applies only if upnp_config is enabled. upnp_internet_dis: Prevents Internet access; 1 (Internet access is disabled) or 0 (Internet access is allowed)	To allow users to config UPnP: <UPNP> upnp_enable=1,upnp_config=1 </UPNP> To allow user to config UPnP ,and save this config even after system reboot: <UPNP>upnp_enable=1,upnp_config=1,upnp_keep_portmap=1 </UPNP> To allow user to enable or disable Internet access::<UPNP>upnp_enable= 1,upnp_internet_dis=1</UPNP> To allow user to do any UPnP function: <UPNP>upnp_enable=1,upnp_config=1,upnp_keep_portmap=1,upnp_ internet_dis=1</UPNP>

Feature/XML Tag	Parameters	Examples
QoS Category Priority Rule <QOS_PRIORITY _ RULE>	<QOS_PRIORITY_RULE>category_ number,name, priority,port_range</ QOS_PRIORITY_RULE> category_num: QoS Category number; 1 (application), 2 (online game), 3 (MAC address), 4 (Ethernet port) name: Name string, corresponding to the selected category Application: The name of the application Online Games: The name of the game MAC Address: The MAC address in the format xx:xx:xx:xx:xx:xx Ethernet Port: The port; Ethernet Port 1, Ethernet Port 2, Ethernet Port 3, or Ethernet Port 4 priority: Priority; 0 (Low), 1 (Normal), 2 (Medium), 3 (High) port_range: The port range; *start;end;protocol* start : The first port number in the range end: The final port number in the range protocol : 0 (Both, 1 (TCP, 2 (UDP	To configure a rule for an application: <QOS_PRIORITY_RULE>category_num= 1,name= ap1, priority=3,port_range= 111;222; 0;333;444;1</QOS_PRIORITY_ RULE> To configure a rule for an online game: Format 1 (default game): <QOS_ PRIORITY_RULE> category_ number=2, name,priority</QOS_ PRIORITY_RULE> Example: <QOS_PRIORITY_RULE>category_num= 2, name=Age of Empires,priority=2 </QOS_PRIORITY_RULE> Format 2 (with port range): <QOS_ PRIORITY_RULE>category_ number=2, name,priority,port_range</QOS_ PRIORITY_RULE> <QOS_ PRIORITY_ RULE>category_num=2, name= game1,priority=1, port_range= 555; 666;1</ QOS_ PRIORITY_RULE> To configure a rule for a MAC Address: <QOS_PRIORITY_RULE>category_num=3, name=mac1,priority=1,mac= 00:02: 03:04:05:06</QOS_ PRIORITY_RULE> To configure a rule for an Ethernet port: <QOS_PRIORITY_RULE>category_num= 4,name= Ethernet Port 1,priority=0 </QOS_PRIORITY_RULE> To delete all rules: <QOS_PRIORITY_ RULE></QOS_PRIORITY_RULE>

Feature/XML Tag	Parameters	Examples
Basic Wireless Settings for Primary Network <WL_BASIC_SET_1>	<WL_BASIC_SET_1>wl_net_mode,wl_closed,wl_ssid</WL_BASIC_SET_1> wl_net_mode: Network mode; mixed, b-only, g-only, or disabled wl_closed: SSID broadcast status; 1 (disabled) or 0 (enabled) wl_ssid: Wireless network name; enter 1 to 32 ASCII characters (backslash character not allowed)	To enable SSID-1 and specify the SSID name: <WL_BASIC_SET_1> wl_net_mode =g-only,wl_closed=0, wl_ssid=aaabbb</WL_BASIC_SET_1> To configure SSID-1 as a Wireless B network: <WL_BASIC_SET_1>wl_net_mode=b-only,wl_ssid= aaabbb</WL_BASIC_SET_1> To disable SSID-1: <WL_BASIC_SET_1> wl_net_mode=disabled</WL_BASIC_SET_1>
Basic Wireless Settings for Secondary or Guest Network <WL_BASIC_SET_2>	<WL_BASIC_SET_2>wl1_net_mode_tmp,wl1_closed,wl1_ssid,ap_isolation</WL_BASIC_SET_2> IMPORTANT: The secondary network can be enabled only when when wl_net_mode is enabled for the primary network. wl1_net_mode_tmp: Network mode; 1 (enabled), 0 (disabled) wl1_closed: SSID broadcast status; 1 (disabled) or 0 (enabled) wl1_ssid: Wireless network name; enter 1-32 ASCII characters (backslash character not allowed) ap_isolation: For Internet Only Access (Guest Network); 1 (disabled) or 0 (enabled) ctrl_ssid2: Allows Service Provider to lock SSID2; when enabled, user will not be able to configure SSID2 from the device GUI; 1 (enabled) or 0 (disabled)	To enable SSID-2 and specify the SSID name, with guest network: <WL_ BASIC_SET_2>wl1_net_mode_tmp= 1,wl1_closed=0,wl1_ssid= cccddd, ap_isolation=1</WL_BASIC_ SET_2> To disable SSID-2: <WL_BASIC_SET_2> wl1_net_mode_tmp=0</WL_BASIC_SET_2> To enable SSID-2 guest network: <WL_BASIC_SET_2>ap_isolation=1</WL_BASIC_SET_2> To prevent SSID-2 configuration from the device GUI: <WL_BASIC_SET_2> ctrl_ssid2=0</WL_BASIC_SET_2>

Feature/XML Tag	Parameters	Examples
Wireless Security for SSID1 <WL_SECURITY_ SET_1> Wireless Security for SSID2 <WL_SECURITY SET_2>	<WL_SECURITY_SET_1>wl_security _mode2= [mode],[parameters]</ WL_SECURITY_SET_1> <WL_SECURITY_SET_2>wl1_securit y_mode2= [mode],[parameters]</ WL_SECURITY_SET_1> wl_security_mode2: Security mode for SSID1 wl1_security_mode2: Security mode for SSID2 Acceptable values are WEP, WPA Personal, WPA2 Personal, WPA Enterprise, WPA2 Enterprise, or Disabled	To disable Wireless Security 1: <WL_ SECURITY_SET_1>wl_security_mode2= disabled </WL_SECURITY_ SET_1> To disable Wireless Security 2: <WL_ SECURITY_SET_1>wl1_security_mode2=di sabled</WL_SECURITY_SET_1>
	WEP Parameters wl_wep_bit: WEP encryption; 64 (64 bits 10 hex digits) or 128 (128 bits 26 hex digits) wl_passphrase: WEP passphrase; enter 1 to 16 ASCII characters wl_key1: Key 1; 10 or 26 hex wl_key2: Key 2; 10 or 26 hex wl_key3: Key 3; 10 or 26 hex wl_key4: Key 4; 10 or 26 hex wl_key: WEP transmission key; numerals from 1 to 4	To enable Wireless WEP 1 and specify the passphrase and keys: <WL_ SECURITY_ SET_1>wl_security_ mode2= wep,wl_wep_bit=64,wl_ passphrase=test1,wl_key1= 81461A6 88C,wl_key2=A8B0AFDB8F,wl_key3= B99D3E230B,wl_key4=B9EF3E6ACD, wl_key=4</WL_SECURITY_ SET_1> To enable Wireless WEP 2 and specify the passphrase and keys: <WL_ SECURITY_SET_2>wl1_security_mode2=w ep,wl1_wep_bit=64,wl1_ passphrase=test2,wl1_ key1=8542E268 D6,wl1_key2=FFD9405B 8B,wl1_key3= 25C9B8C5BB,wl1_key4=73B13791B2, wl1_key=4</WL_SECURITY_ SET_2>

Feature/XML Tag	Parameters	Examples
	WPA Personal and WPA2 Personal Parameters wl_crypto: WPA algorithms; tkip (TKIP) or aes (AES) wl_wpa_psk: WPA shared key; enter from 8 to 63 ASCII characters wl_wpa_gtk_rekey: WPA group key renewal; numerals from 600 to 7200	To enable Wireless WPA Personal, specify the keys and set the renewal rate: <WL_ SECURITY_SET_1>wl_ security_ mode2= wpa_personal,wl_ crypto=aes, wl_wpa_ psk=personal, wl_wpa_gtk_ rekey=700</ WL_ SECURITY_SET_1> To enable Wireless WPA2 Personal, specify the keys and set the group key renewal: <WL_SECURITY_SET_1>wl_ security_mode2=wpa2_personal,wl_ crypto=aes,wl_wpa_psk=personal,wl_wpa _gtk_ rekey=700</WL_SECURITY_ SET_1>
	WPA Enterprise and WPA2 Enterprise Parameters wl_crypto: WPA algorithms; tkip (TKIP) or aes (AES) wl_radius_ipaddr: RADIUS server address wl_radius_port: RADIUS port number; numerals from 1 to 65535 wl_radius_key: RADIUS shared key; enter from 1 to 79 ASCII characters wl_wpa_gtk_rekey: Key renewal timeout; numerals from 600 to 7200	To enable WPA Enterprise and specify the RADIUS information: <WL_ SECURITY_SET_1> wl_security_ mode2=wpa_enterprise,wl_crypto= aes,wl_radius_ipaddr=192.168.15.111, wl_radius_port=6666,wl_radius_key= enterprise,wl_wpa_gtk_rekey=666 </WL_ SECURITY_SET_1> To enable WPA2 Enterprise and specify the RADIUS information: <WL_ SECURITY_ SET_1>wl_security_ mode2= wpa2_ enterprise,wl_crypto= aes,wl_radius_ipaddr=192.168.15.111,wl_ra dius_port=6666,wl_radius_key= enterprise,wl_wpa_gtk_rekey=666 </WL_SECURITY_SET_1>

Feature/XML Tag	Parameters	Examples
LAN DHCP <LAN_DHCP>	<LAN_DHCP>dhcp_lease,dhcp_default_lease</LAN_DHCP> dhcp_lease: Client lease time in minutes; numerals from 1 to 9999 dhcp_default_lease: Default lease time in minutes; numerals from 1 to 9999 **NOTE:** Dhcp_default_lease allows the Service Provider to configure the length of the "default lease time." By default, the client lease time is set to "0," meaning 1 day.	To set the client lease time: <LAN_DHCP> dhcp_default_lease=888 </LAN_DHCP> To set lease time and default lease time: <LAN_DHCP>dhcp_lease=777,dhcp_default_lease=888</LAN_DHCP>
Switch Rate <SWITCH_RATE>	<SWITCH_RATE>mv_switch_total_rate_limit,mv_switch_ingress_mcast_rate</SWITCH_RATE> mv_switch_total_rate_limit: Limits the switch throughput; numerals from 1 to 200 (default is 4) mv_switch_ingress_mcast_rate: Ingress multicast rate in Mbps; numerals from 1 to 100 (default is 80) **NOTE:** The switch rate is set by dividing 200 by the mv_swtich_total_rate_limit. With the default value of 4, the throughput is limited to 50Mbps. MPORTANT: It is highly recommended to keep the default switch rate settings. Default settings hae been tested to support the appropriate Quality of Service for the IPTV video transmission towards the et-top box, in addition to maintaining the appropriate Quality of Service of the Voice Telephony transmission.	To set the switch rate limit to 40 Mbps and the multicast rate to 40 Mbps: <SWITCH_RATE>mv_switch_total_rate_limit=5,mv_switch_ingress_mcast_rate=40</SWITCH_RATE>

Feature/XML Tag	Parameters	Examples
WAN Type <WAN_TYPE>	<WAN_TYPE>wan_proto=[mode],[parameters]</WAN_TYPE> wan_proto: Internet connection type; dhcp, static, pppoe, pptp, l2tp, heartbeat	
	DHCP Parameters No other settings are required.	To configure a DHCP connection: <WAN_TYPE>wan_proto=dhcp </WAN_TYPE>
	Static IP Parameters wan_ipaddr: WAN IP address wan_netmask: WAN subnet mask wan_gateway: Gateway IP address	To configure a Static IP connection: <WAN_TYPE>wan_proto=static,wan_ipaddr=192.168.0.11,wan_netmask=255.255.255.128,wan_gateway=192.168.0. 252</WAN_TYPE>
	PPPoE (Point-to-Point Protocol over Ethernet) Parameters ppp_username: User name; enter from 1 to 63 ASCII characters ppp_passwd: Password; enter from 1 to 63 ASCII characters ppp_service: Service name; enter from 0 to 63 ASCII characters	To configure a PPPPoE connection: <WAN_TYPE>wan_proto=pppoe,ppp_username=adc,ppp_passwd=def </WAN_TYPE> To configure a PPPPoE connection type and specify a service name: <WAN_TYPE> wan_proto=pppoe, ppp_username=adc,ppp_passwd=def,ppp_service=aaa</WAN_TYPE>
	PPTP (Point-to-Point Tunneling Protocol) Parameters wan_ipaddr: WAN IP address wan_netmask: WAN subnet mask wan_gateway: Gateway IP address	To configure a PPTP connection: <WAN_TYPE> wan_ proto=pptp,ppp_username=adc,ppp_passwd=def,wan_ipaddr=192.168.0.18,wan_netmask=255.255.255.0,pptp_server_ip=192.168.0.251 </WAN_TYPE>
	L2TP (Layer 2 Tunneling Protocol) Parameters l2tp_server_ip: Server IP address ppp_username: User name; enter from 1 to 63 ASCII characters ppp_passwd: Password; enter from 1 to 63 ASCII characters	To configure an L2TP connection: <WAN_TYPE>wan_proto=l2tp, ppp_username=adc,ppp_passwd= def,l2tp_server_ip=192.168.0.15 </WAN_TYPE>

Feature/XML Tag	Parameters	Examples
	Heartbeat for Telstra Cable Network Parameters hb_server_ip: Heartbeat server IP address ppp_username: User name; enter from 1 to 63 ASCII characters ppp_passwd: Password; enter from 1 to 63 ASCII characters	To configure a Telstra Cable connection: <WAN_TYPE>wan_proto= heartbeat,ppp_username=adc,ppp_ passwd=def,hb_ server_ip= 192.168. 0.16</ WAN_TYPE>
		Fail Pattern: <WAN_TYPE>wan_proto=dhcpd </WAN_TYPE> <WAN_TYPE>wan_proto=static,wan_ ipaddr= 192.168.0.11,wan_netmask= 255. 255.255.128</ WAN_TYPE> <WAN_TYPE>wan_proto=l2tp,ppp_ passwd=def,l2tp_server_ip=192.168. 0.15 </WAN_TYPE> <WAN_TYPE>wan_proto=heartbeat, ppp_username=adc,ppp_passwd=def </WAN_TYPE> <WAN_TYPE>wan_proto=static,wan_ ipaddr=aaabbb,wan_netmask=255. 255.255.128,wan_gateway=192.168.0. 252</ WAN_TYPE>
PPP Demand <PPP_DEMAND>	<PPP_DEMAND>ppp_demand,ppp_redia lperiod</PPP_DEMAND> ppp_demand: PPP Demand Type; 1 (Connect on Demand) or 0 (Keep Alive) ppp_idletime: Maximum idle time in minutes; numerals from 1 to 9999 ppp_redialperiod: Redial period in seconds; numerals from 2 to 180	To configure PPP to connect on demand: <PPP_DEMAND>ppp_ demand=1, ppp_ idletime=666</PPP_ DEMAND> To configure PPP to keep alive: <PPP_ DEMAND>ppp_demand=0,ppp_redial period=77</PPP_DEMAND>

Feature/XML Tag	Parameters	Examples
		Fail Pattern: <PPP_DEMAND>ppp_demand=1,ppp_ idletime= 66666</PPP_DEMAND> <PPP_DEMAND>ppp_demand=0,ppp_ redialperiod=777</PPP_DEMAND> <PPP_DEMAND>ppp_demand=1 </PPP_ DEMAND> <PPP_DEMAND>ppp_demand=0 </PPP_ DEMAND> <PPP_DEMAND>ppp_demand=1,ppp_ redialperiod=77</PPP_DEMAND> <PPP_DEMAND>ppp_demand=0,ppp_ idletime= 666</PPP_DEMAND>
WAN Host <WAN_HOST>	<WAN_HOST>wan_hostname=host_test, wan_domain=domain</WAN_HOST> wan_hostname: WAN hostname; enter from 0 to 39 ASCII characters wan_domain: WAN domain name; enter from 0 to 63 ASCII characters	To specify a WAN hostname and WAN domain name: <WAN_HOST> wan_ hostname=host_test,wan_domain= domain_test</WAN_ HOST> To specify a WAN hostname only: <WAN_ HOST>wan_hostname= host_test</ WAN_HOST> To specify a WAN domain name only: <WAN_HOST>wan_domain=domain_ test</ WAN_HOST>
WAN MTU <WAN_MTU>	<WAN_MTU>mtu_enable</WAN_MTU> mtu_enable: MTU mode; 0 (automatic) or 1 (manual) wan_mtu: MTU size; if MTU mode is manual, enter a numeral from 576 to 1500 **NOTE:** The default size depends on the Internet Connection Type: DHCP or Static IP: 1500 PPPoE: 1492 PPTP or L2TP: 1460 Telstra Cable: 1500	To enable MTU in Auto mode: <WAN_ MTU>mtu_enable=0</WAN_MTU> To enable MTU in Manual mode and specify the MTU size: <WAN_MTU> mtu_ enable=1,wan_mtu=888 </WAN_MTU> To enable MTU in Manual mode without specifying the MTU size: <WAN_MTU> mtu_enable=1</WAN_ MTU>

Feature/XML Tag	Parameters	Examples
		Fail Pattern <WAN_MTU>mtu_enable=0,wan_mtu= 999</ WAN_MTU> <WAN_MTU>wan_mtu=777</WAN_ MTU>
WAN DNS <WAN_DNS>	<WAN_DNS>wan_dns</WAN_DNS> wan_dns: DNS IP address; separate multiple addresses with a space	To specify one DNS address: <WAN_ DNS>wan_dns=192.168.0.21</WAN_ DNS> To specify multiple DNS addresses: <WAN_DNS>wan_dns=192.168.0.21 192.168.0.22</WAN_DNS> <WAN_ DNS>wan_dns=192.168.0.21 192.168.0.22 192.168.0.23</WAN_ DNS>
WAN DNS, continued		Fail Pattern <WAN_DNS>wan_dns=aaabbb</WAN_DNS> <WAN_DNS>wan_dns=192.168.0.21 192.168.0.aa</WAN_DNS> <WAN_DNS>wan_dns=192.168.0.21 192.168.0.22 192.168.0.23 192.168.0.23 </WAN_DNS>
DHCP Reservation <DHCP_RESERVATION>	<DHCP_RESERVATION>dhcp_statics=name;mac;ip</DHCP_RESERVATION> dhcp_statics: Identifies the client name: A name for this reservation mac: The MAC address of the client; enter the MAC address without hyphens ip: The IP address of the client	To create two reservations (R51 and R52) for two clients: <DHCP_RESERVATION>dhcp_statics= R51; 00:0E:35:6B:56:78;100</DHCP_ RESERVATION><DHCP_RESERVATION> dhcp_statics=R52;00:0E:35:6B:34:56; 101</ DHCP_ RESERVATION> To delete all reservations: <DHCP_ RESERVATION></DHCP_ RESERVATION>

Feature/XML Tag	Parameters	Examples
Single Port Forwarding <SINGLE_PORT_ FORWARDING>	<SINGLE_PORT_FORWARDING>forward _single=name:on|off:both|tcp|udp:external -port:internal-port:ip</ SINGLE_PORT_FORWARDING> **NOTE:** To configure port forwarding, you also should configure a DHCP reservation for the designated server. forward_single: Supports port forwarding on the specified port name: Application name; enter a name or use the following names for standard applications: FTP, Telnet, SMTP,DNS,TFTP,Finger, HTTP, POP3, NNTP on|off: on (enabled) or off (disabled) both|tcp|udp: Selected protocol; tcp, udp, or both external-port: The external port number internal-port: The internal port number ip: The IP address of the computer that should receive the requests.	To forward FTP to 192.168.15.18: <SINGLE_PORT_FORWARDING>forward_singl e=FTP:on:tcp:21:21:18</SINGLE_ PORT_FORWARDING> To configure port forwarding for a non-standard application: <SINGLE_PORT_ FORWARDING>forward_single=fw1:on: both:1111: 2222:28</SINGLE_PORT_ FORWARDING> To delete all: <SINGLE_PORT_FORWARDING> </SINGLE_PORT_FORWARDING> To configure port forwarding for default standard applications such as FTP, Telnet, SMTP, and others: <SINGLE_PORT_ FORWARDING> forward_single=FTP:on: tcp:21:21:18 </SINGLE_PORT_ FORWARDING> <SINGLE_PORT_FORWARDING> forward_single=Telnet:on:tcp:23:23:19</ SINGLE_ PORT_FORWARDING>

Feature/XML Tag	Parameters	Examples
Port Range Forwarding <PORT_RANGE_ FOWARDING>	<PORT_RANGE_FORWARDING>forward _single=name:on|off:both|tcp|udp:port range start:port range end:ip</ PORT_RANGE_FORWARDING> **NOTE:** To configure port forwarding, you also should configure a DHCP reservation for the designated server. forward_port: Supports port forwarding on a range of ports name: Application name on|off: 0n (Enabled or off (Disabled both|tcp|udp: Selected protocol; tcp, udp, or both external-port: The external port number internal-port: The internal port number ip: The IP address of the computer running the specific application.	To allow forwarding on two specified port ranges: <PORT_RANGE_FORWARDING> forward_port=prf1:on:tcp:555:666:18 </PORT_RANGE_FORWARDING> <PORT_RANGE_FORWARDING> forward_port=prf2:on:both:777:888:19</PORT_ RANGE_FORWARDING> To delete all: <PORT_RANGE_FORWARDING> </PORT_ RANGE_FORWARDING>
Port Range Triggering <PORT_RANGE_ TRIGGERING>	<PORT_RANGE_TRIGGERING>port_trigg er=name:on|off:trigger start:trigger end:forward start:forward end</ PORT_RANGE_TRIGGERING> port_trigger: Supports port range triggering name: Application name on|off: On (enabled) or Off (disabled) trigger start:**trigger end:** Triggered range forward star:**forward end:** Forwarded range	To configure two port range triggers: <PORT_RANGE_TRIGGERING>port_ trigger=prt1:on:111:222:333:444 </PORT_ RANGE_TRIGGERING> <PORT_RANGE_TRIGGERING>port_ trigger=prt2:on:555:666:777:888 </PORT_ RANGE_TRIGGERING> To delete all: <PORT_RANGE_TRIGGERING></ PORT_ RANGE_TRIGGERING>
VLAN <WAN_VLAN>	<WAN_VLAN>wan_vlan_enable,wan_vla n_id</WAN_VLAN> wan_vlan_enable: VLAN status; 1 (enabled) 0 (disabled) wan_vlan_id: VLAN ID number	To enable VLAN and specify the VLAN ID: <WAN_VLAN>wan_vlan_enable=1, wan_vlan_id=123</WAN_VLAN> To disable VLAN: <WAN_VLAN>wan_ vlan_enable=0</WAN_VLAN>

Feature/XML Tag	Parameters	Examples
Router Syslog <ROUTER_SYSLOG>	<ROUTER_SYSLOG>log_provision</ROUTER_SYSLOG> **log_provision:** Type of log; **0** (console display), **1** (system log), or **2** (console display and system log)	To configure console display and system log: <ROUTER_SYSLOG> log_provision=2</ROUTER_SYSLOG>

D

Troubleshooting

This appendix provides solutions to problems that may occur during the installation and operation of the WRP400s.

NOTE If you can't find an answer here, visit Cisco Community Central > Small Business Support Communityat the following URL: www.myciscocommunity.com/community/smallbizsupport/voiceandconferencing/ata

Q. I want to access the Configuration Utility, but the address I entered did not work.

Use the Interactive Voice Response Menu to find out the Internet IP address. Follow these steps:

1. Use a telephone connected to the Phone 1 port of the WRP400.
2. Press **** (in other words, press the star key four times).
3. After the greeting plays, press **110#**.
4. Write down the IP address as it is announced.
5. Press **7932#**.
6. Press **1** to enable WAN access to the Configuration Utility.
7. Open a web browser on a networked computer.
8. Start Internet Explorer and enter the IP address of the WRP400.

Q. I'm trying to access the Configuration Utility, but I do not see the login screen. Instead, I see a screen saying, "404 Forbidden."

If you are using Windows Explorer, perform the following steps until you see the Configuration Utility login screen (Mozilla requires similar steps).

1. Click **File**. Make sure *Work Offline* is NOT checked.
2. Press **CTRL + F5**. This is a hard refresh, which forces Windows Explorer to load new web pages, not cached ones.
3. Click **Tools**. Click **Internet Options**. Click the **Security** tab. Click the **Default level** button. Make sure the security level is Medium or lower. Then click the **OK** button.

Q. How do I save the voice configuration for my WRP400?

1. Start Internet Explorer, connect to the Configuration Utility, and choose **Voice > Admin Login**. If prompted, enter the administrative login provided by the Service Provider. (The default username and password are both **admin**.)
2. Click the **File** menu, and then choose **Save as** > **HTML** to save all the Voice pages into one HTML file. This HTML file is helpful to provide to the support team when you have a problem or technical question.

Q. How do I debug the WRP400? Is there a syslog?

The WRP400 provides the option to send messages to both a syslog and debug server. The ports can be configured (by default the port is 514).

1. Make sure you do not have firewall running on your computer that could block port 514.
2. Start Internet Explorer, connect to the Configuration Utility, and choose **Voice > Admin Login**. If prompted, enter the administrative login provided by the Service Provider. (The default username and password are both **admin**.)
3. Under the **Voice** menu, set *Debug Server* as the IP address and port number of your syslog server. Note that this address has to be reachable from the WRP400. For example, if the WRP400 is at 192.168.15.1, reachable addresses are in the range of 192.168.15.x, for example 192.168.15.100:514.
4. Set *Debug level* to **3**. You do not need to change the value of the *syslog server* parameter.
5. To capture SIP signaling messages, under the Line tab, set *SIP Debug Option* to **Full**. The file output is syslog.<portnum>.log (for the default port setting, syslog.514.log).

Q. How do I access the WRP400 if I forget my password?

By default, the User and Admin accounts have no password. If the ITSP set the password for either account and you do not know what it is, you need to contact the ITSP. If the password for the user account was configured after you received the WRP400, you can reset the device to the user factory default, which preserves any provisioning completed by the ITSP. If the Admin account needs to be reset, you have to perform a full factory reset, which also erases any provisioning.

To reset the WRP400 to the factory defaults, perform the following steps:

1. Connect an analog phone to the WRP400 and access the IVR by pressing ****.
2. Press the appropriate code to reset the unit:
   - Press 877778# to reset the unit to the defaults as it shipped from the ITSP. This will reset the User account password to the default of blank.
   - Press 73738# to perform a full reset of unit to the factory default settings. The Admin account password will be reset to the default of blank.
3. Press 1 to confirm the operation, or press * to cancel the operation.
4. Login to the unit using the User or Admin account without a password and reconfigure the unit as necessary.

Q. The WRP400 is behind a NAT device or firewall and I'm unable to make a call or I'm only receiving a one-way connection. What should I do?

Complete the following steps.

1. Configure your router to port forward "TCP port 80" to the IP address of the WRP400. You should use a static IP address. (For help with port forwarding, consult the documentation for the NAT device or firewall.)
2. On the Line tab of the Configuration Utility, change the value of *Nat Mapping Enable* to **yes**. On the SIP tab; change *Substitute VIA Addr* to **yes**, and the *EXT IP* parameter to the IP address of your router.
3. Make sure you are not blocking the UDP PORT 5060,5061 and port for UDP packets in the range of 16384-16482. Also, disable "SPI" if this feature is provided by your firewall. Identify the SIP server to which the WRP400 is registering, if it supports NAT, using the *Outbound Proxy* parameter.
4. Add a STUN server to allow traversal of UDP packets through the NAT device. On the SIP tab of the Configuration Utility, set *STUN Enable* to **yes**, and enter the IP address of the STUN server in *STUN Server*.

STUN (Simple Traversal of UDP through NATs) is a protocol defined by RFC 3489, that allows a client behind a NAT device to find out its public address, the type of NAT it is behind, and the port associated on the Internet connection with a particular local port. This information is used to set up UDP

communication between two hosts that are both behind NAT routers. Open source STUN software can be obtained at the following address: http://www.voip-info.org/wiki-Open+Source+VOIP+Software

NOTE STUN does not work with a symmetric NAT router. Enable debug through syslog (see FAQ#10), and set *STUN Test Enable* to **yes**. The messages indicate whether you have symmetric NAT or not.

E

Environmental Specifications for the WRP400

Device Dimensions	5.51” x 5.51” x 1.06” (140 x 140 x 27 mm)
Unit Weight	10.05 oz (285 g)
Power	External, Switching 5VDC 2A
Certification	FCC (Part 15 Class B), CE, ICES-003, RoHS, UL, A-Tick, NZ Telepermit, CB, Wi-Fi (802.11b + WPA2, 802.11g + WPA2, WMM, WPS)
Operating Temp	32º to 104º F(0 to 40ºC)
Storage Temp	-20° C to 60° C (-4° F to 140° F)
Operating Humidity	10% to 85% relative humidity, Non-Condensing
Storage Humidity	5% to 90% relative humidity, Non-Condensing

F

Where to Go From Here

This appendix describes additional resources that are available to help you and your customer obtain the full benefits of the WRP400.

Support	
Cisco Small Business Support Community	www.cisco.com/go/smallbizsupport
Online Technical Support and Documentation (Login Required)	www.cisco.com/support
Phone Support Contacts	www.cisco.com/en/US/support/tsd_cisco_small_business_support_ center_contacts.html
Software Downloads (Login Required)	Go to tools.cisco.com/support/downloads, and enter the model number in the Software Search box.
Product Documentation	
Technical Documentation	www.cisco.com/en/US/products/ps10024/tsd_products_support_series_home.html
3G USB Modem Compatibility List	www.cisco.com/en/US/prod/collateral/voicesw/ps6790/gatecont/ps10024/sales_tool_c96-522031.html
Cisco Small Business	
Cisco Partner Central for Small Business (Partner Login Required)	www.cisco.com/web/partners/sell/smb
Cisco Small Business Home	www.cisco.com/smb

Support	
Marketplace	www.cisco.com/go/marketplace