ZIP2

User's Manual

Manual Part Number 90-18200

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Notice

Revision History

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Contents

Chapter 1

Introduction

1.1 Scope

1.1.1 Audience

This manual is intended for networking engineers and network administrators who need to install, maintain, support, and use the ZIP2. The manual can also be used by engineers that want to make a phone system compatible with the ZIP2. The manual assumes you are familiar with networking and telephony principals and practices.

If you use the ZIP2 with the MX250 or the MX1200, you should read this manual in conjunction with the *MX Administrator User's Manual*.[1] That manual describes how certain features of the phone interact with the enterprise media exchange. You can obtain that manual on line at:

http://www.zultys.com

This manual on the ZIP2 can be used by a user who wants to understand in detail how features and functions of the phone operate. End users who do not need the depth of information contained in this manual (which is about 70 pages) should refer to the ZIP2 User's Guide (which is 16 pages). One guide is shipped with each phone, but you can download the guide from the ZIP2 web site at the Zultys website.

1.1.2 Installation and Use

Unpack the ZIP2 and verify the contents as described in section 2.2 on page 7. Install the product as described in chapter 3, starting on page 11.

1.1.3 What this Manual Includes

This manual provides detailed information and instructions on the complete installation and operation of the ZIP2 IP phone.

1.1.4 What this Manual Does Not Include

This manual does not provide technology details, pricing, names of sales representatives, or names of distribution channels. Access the Zultys web site for this information:

1. The MX250 and MX1200 are Enterprise Media Exchanges. They are manufactured by Zultys and provide the communications needs of an enterprise by integrating voice, data, video, and fax.

http://www.zultys.com

1.2 Features

The ZIP2 is a SIP based IP phone that supports all of the common functions of a business telephone set. By using Session Initiation Protocol (SIP) for call control, the ZIP2 is compatible with any standard SIP server. Figure 1-1 displays a frontal view of the ZIP2.

Figure 1-1 Front View of ZIP2

Key features of the phone include:

- two virtual lines to support two simultaneous calls
- handset and speaker modes
- speech quality ensured by QoS at the Ethernet and IP layers
- one 10/100 Base-T Ethernet circuit to connect to the LAN
- supports IEEE 802.1q VLAN tagging
- supports transfer, conference, hold, redial, unconditional call forward, and do not disturb (DND)
- 16 buttons support all commonly used features

- two LEDs provide easy indication of status
- uses standard SIP messages to interface to a variety of call managers from various manufacturers

1.3 Documentation Overview

1.3.1 Organization

This user's manual describes:

- how to install the ZIP2
- how to provision the phone for use within the network
- how to make and receive calls
- how to access the features of the phone
- how to customize the phone to suit your preferences
- what to do when you are convinced there is a problem

1.3.2 Nomenclature

1.3.2.1 Acronyms

This manual often uses acronyms specific to the industry of telecommunications and data communications. Because the sections (and, to a certain extent, the subsections) can be read in any sequence, acronyms are not defined in the text. For a complete list of acronyms used in this manual, see Appendix D, starting on page 87.

1.3.2.2 Jargon

This manual often uses technical terms specific to the industry of telecommunications and data communications. Very specialized terms are sparsely used, and their meanings are clearly explained where they are used.

1.3.3 Special Paragraph Styles

The following are the notices that are used to attract special attention to certain items. They set text off from the main body of the manual. These notices also appear in other languages where required by certain regulatory bodies:

Important This notice contains special information that should not be ignored.

Caution This notice calls attention to a condition or procedure which, if not observed, could result in damage to the ZIP2 or the loss of data.

Warning This notice indicates that if a specific procedure or practice is not correctly followed, permanent damage to the ZIP2 and personal injury may result.

Danger This notice warns you of imminent hazard to yourself and others if proper procedures are not followed.

1.4 Forms of Documentation

This manual is updated with each major release of the software. The manual describes the features in that release of the software.

Between major releases of software, Zultys may issue one or more minor releases of software. These minor releases may have more capabilities than the current formal release. The features in that software (and the user interfaces to support those features) may or may not be described in this manual.

This manual is available only in PDF format, which you can download from the Zultys website at:

http://www.zultys.com

You can obtain old versions of the manual that may describe the software that you have or the latest manual that describes all the latest features of the product. You can identify the version of the manual from the title page, opposite the table of contents (page 2 of the PDF file).

When you use the PDF file, you can click on any reference in the text. This powerful feature allows you to follow the references in the text very easily. Using Acrobat, you can then return to the page you were previously reading. This is a huge benefit to you if you want to study a small area of the product.

1.5 Colophon

This document was produced on personal computers using Adobe's FrameMaker for Windows. The printed book is printed by an offset process.

The headings are set in Swiss 721, Bitstream's version of the Helvetica™ typeface; the copy is set in Zapf Calligraphic, Bitstream's version of the Palatino™ typeface; notices are set in Swiss 721 or News Gothic, Bitstream's version of the Kingsley-ATF Type Corporation typeface. The drawings were produced using Adobe Photoshop, Adobe Illustrator, and Microsoft Visio.

1.6 Documentation Feedback

Zultys appreciates any constructive feedback on all our documentation. If you have comments or error reports on any Zultys documentation, please submit your feedback to:

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Sunnyvale, California 94085 USA
techpubs@Zultys.com

Chapter 2

Receiving the ZIP2

2.1 Initial Inspection

When the shipment of your ZIP2 arrives, inspect the shipping boxes for external damages and record any discrepancies. Save the boxes and packing material in case you need to ship the phone to another facility. Always retain the packing materials if you suspect that the shipment is damaged — the carrier may need to inspect them.

Warning Do not attempt to use the ZIP2 or its accessories if it or they appear damaged. Immediately report the damage to Zultys or a local Zultys sales representative.

2.2 Package Contents

2.2.1 Unpacking the Phone

If the phone box has not been damaged in transit, unpack it carefully. Ensure that you do not discard any accessories that may be packaged in the same box as the phone.

2.2.2 Verify Contents

Upon delivery of your phone, inspect the packing list and confirm that all items listed on that note were received. Compare the packing slip with your purchase order.

Ensure that the following accessories are present in the shipment:

Carefully open the box that contains the ZIP2 and verify you have the following items:

- phone body
- handset and handset cord
- ac adapter for your country
- Ethernet cord
- User's Guide

Ensure that there are no discrepancies and then install the ZIP2 as described in chapter 3, starting on page 11.

Important If you suspect that there are discrepancies or that the equipment is not fully functional, contact Zultys or your Zultys sales representative **immediately**. Retain all packing materials and the shipping note for Zultys or its representative to inspect. ZULTYS CANNOT BE HELD RESPONSIBLE IF YOU CLAIM THAT AN ITEM IS MISSING, AND YOU HAVE NOT INFORMED ZULTYS WITHIN THREE DAYS OF RECEIPT, OR IF YOU HAVE NOT RETAINED ALL PACKING MATERIALS FOR INSPECTION.

2.2.3 Serial Numbers

Verify the serial numbers of the phone and compare it to the serial number on the packing lists. The serial number of the ZIP2 is a twelve character digit alphanumeric code printed on a white barcode label on the bottom of the phone.

2.3 In Case of Damage or Malfunction

Notify your Zultys sales or service representative under any of the following conditions:

- the shipping container or any of the contents appear damaged
- an item is missing
- there is a discrepancy between the packing slip and the equipment received
- the equipment does not function correctly

Your supplier will arrange for repair or replacement, at Zultys's discretion. In certain cases, Zultys may require a claim settlement.

2.4 Returning Items for Repair or Replacement

2.4.1 Warranty Coverage

Zultys provides a warranty only through distribution channels. If you are an end user, consult the reseller or distributor who sold you the phone for complete terms of the product that you purchased. Zultys requires that its distributors provide a standard warranty that is one year in duration and that complies with the local laws and expectations of the country in which you reside.

Before returning merchandise to Zultys for repair or replacement, you must ensure that the items are under warranty. If you are unsure about the warranty of your merchandise, call your supplier or a local Zultys sales representative for clarification. Contact your supplier for a return merchandise authorization (RMA) number before returning any merchandise; this includes equipment covered under warranty.

For merchandise not under warranty, you will be charged for a repair if the item is returned to the factory. Call your supplies for pricing on an extended warranty for your merchandise.

2.4.2 Describing the Problem

If you are returning equipment for service, attach a tag or sheet of paper to the equipment giving the following details:

- your company or institution’s name, address, and phone number
- the main person to contact, an alternative contact, and their phone numbers if different from the main phone number
- the return shipping address and any special shipping instructions
- the model number and serial number of the equipment being returned
- a description of the failure (If failure is intermittent, describe its frequency and any special conditions that initiate the failure.)
- any additional comments

2.4.3 Accessories

Do not return any of the accessories with the equipment unless you suspect that one of them is faulty. If you return an accessory, place a tag on it that clearly identifies it as yours, and briefly explain the problem.

2.4.4 Packing

Wherever possible, use the original packing materials to ship the equipment. If these are not available, containers and cushioning material similar to those originally used are available from Zultys.

If it is inconvenient to obtain supplies from Zultys, use a strong, double-walled shipping carton. Place about 70 mm (3 in) of cushioning material around all sides of the equipment.

Zultys is not responsible for any damage that occurs during shipment back to your supplier or to the factory.

2.4.5 Shipping

Obtain from your nearest Zultys sales or service representative the correct address to which you should return the equipment. Clearly mark the container with Zultys’s address and your own address. Ship the package prepaid and insured to Zultys.

The method of shipment which Zultys will return repaired merchandise back to your facility will be the same method by which you shipped the merchandise to Zultys. For example, if you shipped merchandise to Zultys by ground shipment, Zultys will return the merchandise to you by ground shipment. If the merchandise must be repaired and returned to you as soon as possible, you must arrange for the merchandise to be shipped to Zultys by overnight shipment.

2.4.6 Correspondence

In any correspondence subsequent to the return of equipment, always refer to the equipment by model number and serial number.

Chapter 3

Installation

3.1 Preparing the ZIP2 for Use

3.1.1 Handset

Connect the handset cord into the socket located on the left side panel of the phone, shown in figure 3-1.

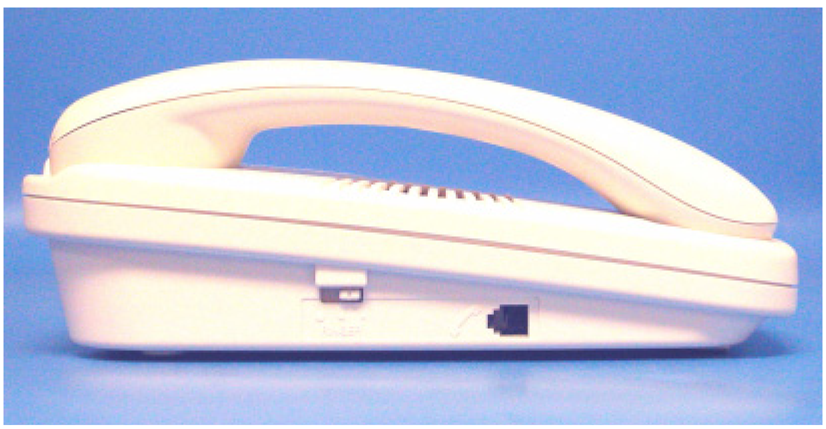

Figure 3-1 Left Panel of ZIP2

3.1.2 Power

Provide power to the ZIP2 by connecting the ac adapter into the power socket (DC In) on the top panel of the phone, as shown in figure 3-2. The ZIP2 does not have a power switch. Figure 3-1 displays the location of the DC In socket.

Important You should only use an ac adapter provided by Zultys. If you are unable to obtain this, use an adapter that has a dc output of 12 V, 600 mA. The plug should be 2.0 mm or 2.1 mm with the center positive. ***Zultys does not warrant operation of the ZIP2 with any adapters other than those supplied by Zultys for the ZIP2.***

When the ZIP2 receives power, it performs a quick power on self test. While powering up, the red and yellow LEDs flash twice.

3.1.3 Connecting to the Network

Connect the LAN socket on the top panel of your phone, shown in figure 3-2, to your Local Area Network. Figure 3-2 displays the location of the LAN socket on the phone

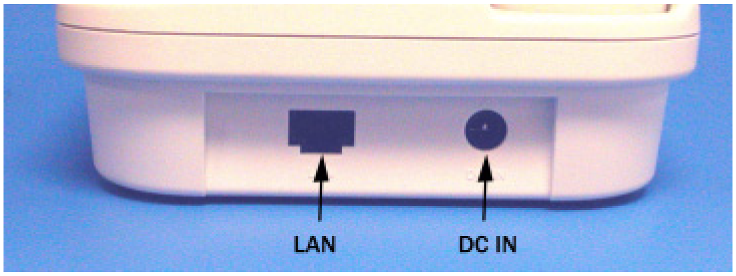

Figure 3-2 Top Panel of ZIP2

The ZIP2 Ethernet circuit has automatic sensing to determine whether the circuit should operate at 10 Mb/s or 100 Mb/s. The phone will always attempt to establish connection at the higher rate and will fall back to the lower rate only if the device to which you have connected the phone cannot operate at the higher rate.

3.2 Power On

3.2.1 Quick Flash of LEDs at Power Up

When you first provide power to it, the phone quickly flashes twice the red LED above the keypad and the yellow LED below the keypad. Both LEDs turn off a few seconds after the power stabilizes. You must ensure that the power plug is fully inserted into the phone. If the power plug is not fully inserted, the LEDs may flash as normal, but the phone will not be operational and you will not receive dial tone.

3.2.2 Dial Tone

If you hear dial tone when you pick up the phone, two conditions are true:

- the phone has power
- the phone has been assigned an IP address

If you do not hear dial tone and you have properly installed the power and Ethernet cables, ask the network administrator for assistance.

After receiving dial tone, you will continue to hear dial tone even if the network subsequently fails or if the Ethernet connection is subsequently removed. Hearing dial tone does not indicate that the phone can properly communicate with a SIP server, so you cannot make a call unless the phone can communicate with a SIP server.

3.2.3 Cannot Complete Call

If you hear dial tone, but receive a fast busy tone after you dial a number, the phone has not been able to communicate with a SIP server.

Chapter 4

Provisioning the Phone

4.1 Introduction

This chapter describes the ZIP2 telephone configuration process. You can either provision the phone to obtain its configuration from the DHCP server or manually set the IP addresses for the ZIP2 and the various network servers.

Information provided by this chapter includes factory preset DHCP and IP address settings, steps for configuring the phone, options required for full DHCP support, the boot process used by the phone, and an introduction to the Download and Configuration Web Page Utility.

4.2 Initial Provisioning

You can provision the ZIP2 phone to receive its configuration settings (IP address and LAN configuration) from a DHCP server, or you can fix the IP address and the other configuration settings.

Each ZIP2 phone is initially programmed with DHCP enabled. If your network administrator provides full DHCP and TFTP support for your ZIP2 phone, you can proceed to the boot process described in section 4.6 on page 27. Section 4.4 on page 25 describes the methods of provisioning your phone to DHCP mode or fixed IP Address mode.

When you connect the ZIP2 phone to a DHCP server, the DHCP server must be configured with the following options for your network to fully support the ZIP2:

- subnet mask (option 1)
- default gateway (option 3)
- domain name server (option 6)
- IP Address (option 50)

In addition, it is recommended that your DHCP server also provide:

- NTP Time Offset (option 2)
- domain name (option 15)
- NTP servers (option 42)
- TFTP Server (option 66)

4.3 Configuring the ZIP2 Phone

The ZIP2 phone accesses configuration settings from two sources:

- configuration files located on a TFTP server
- the Download and Configuration (Web Interface) Utility

If the Download and Configuration Utility specifies a TFTP server with ZIP2 configuration files, the settings in the configuration file will take precedence over Download and Configuration Utility settings.

4.3.1 Configuration Files

Using the configuration file process relies upon the following three sources of configuration for the phone:

- the common configuration file
- what is saved in a specific configuration file
- what is saved in the phone's memory

The common configuration file is called:

```
ZIP2_common.cfg
```

This file is stored in the root directory of the TFTP server.

The specific configuration file is called:

```
<MAC address>.cfg
```

For example,

```
0001E102C8B4.cfg
```

The format for the files is identical, and is shown in figure 4-1. This is an ASCII text file, with the name of the parameter and the value of the parameter listed on the same line. The contents of the file are case sensitive; you must enter parameter names in upper case. Comment lines are denoted with a leading pound sign (#) and have no effect on the operation of the phone.

Configuration parameters are described in Appendix B, starting on page 61.

Every time the phone restarts (either by command or by power on), the phone reads the common configuration file if it exists on the TFTP server referenced by the ZIP2. It extracts the data in the files and saves it to memory, overwriting all specified parameters that are saved in memory.

The following common configuration file instruction specifies that "sample_name" directory[1] contains the specific configuration file.

```
tftp_cfg_dir ./sample_name
```

The phone accesses the specified directory (which can be blank, or '.') and reads the specific configuration file from the specified directory. It extracts the data in the file and saves it to memory, overwriting whatever parameters were already saved in memory. The parameters specified in the specific configuration file take precedence over the same parameters specified in the common configuration file.

1. If the specific configuration file contains such a line, the phone ignores it.

```
# TFTP config file template for ZIP2s

ROMAVERSION 3.38                  # Set current Application code version

################ IP addressing
## Enable DHCP
IF0DHCP DHCP                      # Enable DHCP

############## SIP Configuration Parameters
SERVERIP   192.168.1.100          # SIP Server hostname or address.
                                  # Leave blank to search automatically
                                  # via DNS SRV
SERVERPORT 5060                   # SIP Server Port number
DOMAINNAME zultys.com             # Used as host part of SIP REGISTER AOR
SERVERREGISTER YES                # Register with SIP server

# Backup server must be used in tandem with SIP Extensions retries
#BACKUP_SERVERIP  10.1.1.100      # SIP Backup Proxy
#BACKUP_SERVERPORT 5060           # SIP Backup server port number
#BACKUP_SERVERREGISTER YES        # Register with SIP backup server

DIALPLAN     x.T|x.#|*x.T|*x.#    # local dial plan

TRANSPORT_TYPE UDP                # UDP or TCP

LINE1NUMBER 1234                  # not used
LINE1CALLERID itsme               # Call ID for display purposes
LINE1PORT 5060                    # SIP Listener port on phone
LINE1AEC YES                      # Use Acoustic Echo Cancellation
LINE1AUTHUSER 0001E1072A5F        # Authorization User ID, default MAC address
LINE1AUTHPSWD 1234                # Authorization password; leave blank when
                                  # no authorization required

#SIP_REXMT_INVITE 4               # Number of INVITE retries before failover
#SIP_REXMT_REQUEST 4              # Number of non-INVITE retries before
                                  # failover
```

Figure 4-1 Format for Configuration File

The phone continues to start, using the parameters that are now saved in its memory.

4.3.2 Accessing the Download and Configuration Utility

The ZIP2 phone provides a web interface to its Download and Configuration Utility. Through this utility, you can modify the network parameters and download the application software that controls the ZIP2 behavior.

The following sections describe the procedure for configuring the most common network settings and downloading the application code. Appendix A, starting on page 43, describes each Download and Configuration Utility panel.

You access the utility through a web browser that can access the same network as the ZIP2 being configured. Enter the IP address of the ZIP2 in the address entry box of your web browser and press the enter button. If the ZIP2 is in DHCP mode, contact your network administrator to determine the IP Address. If the ZIP2 is in Fixed IP Address mode and you have forgotten the IP address, see section 5.4.2.

The Home panel, shown in figure 4-2, will appear in your browser.

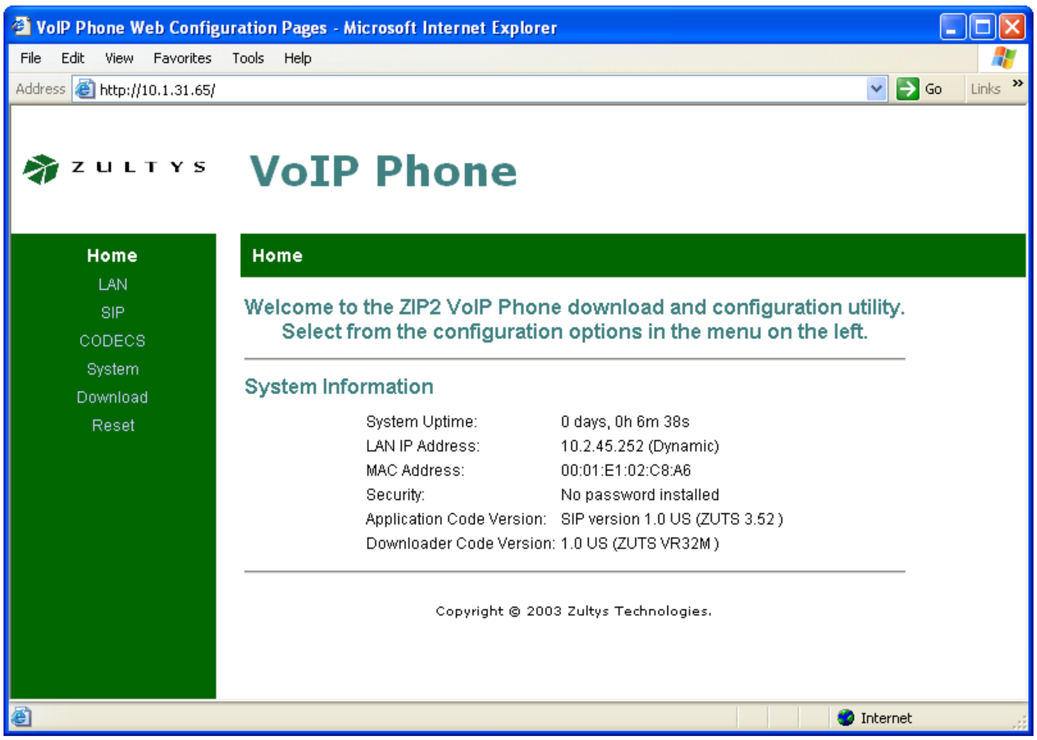

Figure 4-2 Home panel

Each panel is accessed by selecting options from two menu bars. The main menu bar runs vertically on the left side of the window. Main menu bar options include Home, LAN, SIP, CODECS, System, Download, and Reset.

The secondary menu runs horizontally, directly above the configuration parameters. Secondary menu bar options depend on the selected main menu bar item; when Home is the selected in the main menu bar, the only secondary menu bar item is also named Home, as shown in figure 4-2.

4.3.2.1 Configure the LAN Settings

To configure the LAN settings, access the LAN Configuration panel by selecting LAN | LAN Settings from the Download and Configuration Utility.

As shown in figure 4-3, you can either use the DHCP server to dynamically assign an IP address to the ZIP2 or you can specify a fixed IP address for the phone. If you dynamically assign an IP address to the phone, the web interface will also accept the IP Netmask, the IP Gateway address, the IP DNS Server address, the domain name, and the TFTP server from the DHCP server. If you fix the name of the IP address, the utility will expect you to enter all other parameters.

Figure 4-3 LAN Configuration panel

Most home firewalls and NATs provide their address as the TFTP server address in the DHCP response. This prevents auto provisioning via the true TFTP server that stores the configuration files and firmware. To enable auto provisioning in this instance, specify a different TFTP server than that provided by DHCP. Enter the address of the TFTP server in the TFTP Override Server field, as shown in figure 4-3, to solve this problem. To turn this feature off, clear the field and reset the phone.

After entering the parameters on the panel, press the **Save LAN Settings** button to load the new settings onto the ZIP2.[1]

4.3.2.2 Configure the SIP Settings

To configure the SIP settings, access the SIP Configuration panel, shown in figure 4-4, by selecting SIP | SIP from the Download and Configuration Utility.

After entering the parameters, press the **Save SIP Settings** button to load them onto the ZIP2.

1. If the configuration file and the LAN Settings panel specify different IP addressing modes (dynamic via DHCP versus Fixed IP Addressing), pressing the *Save LAN Settings* will produce unpredictable results when rebooting the phone.

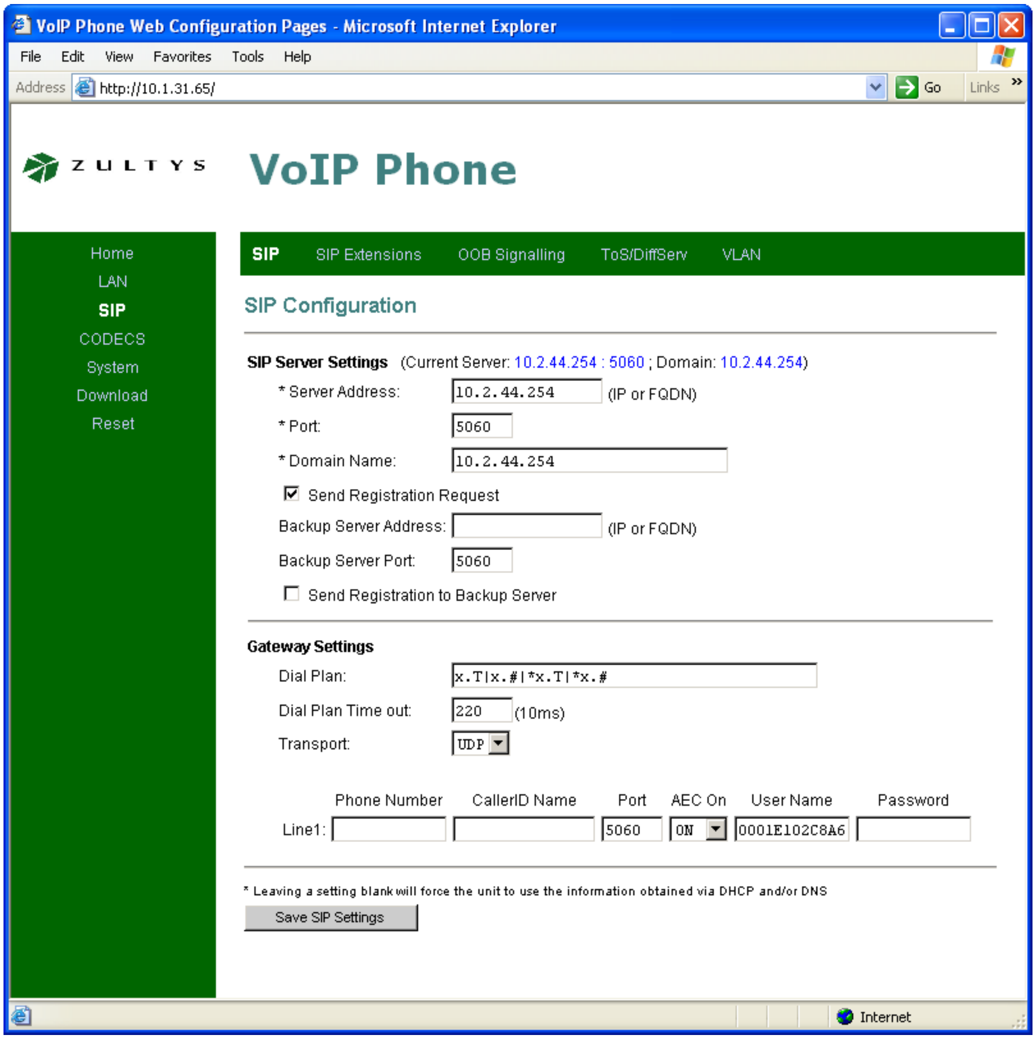

Figure 4-4 SIP Configuration panel

The top panel section, next to the **SIP Server Settings** text, lists the current server, port, and domain. You can enter values for these parameters, which are loaded to the phone when you press the Save SIP Settings button. Any parameter that you leave blank at the time you press the Save SIP Settings button will be filled on the basis of information provided via DNS-SRV.

Server Address. This is either the IP address or the Fully Qualified Domain Name (FQDN) of the SIP Server. When this parameter is left blank, the software performs a DNS-SRV lookup to resolve the SIP server for your domain.

Port. This parameter configures the transport protocol port number. The default port for both TCP and UDP transports is 5060, which is the value recommended by the SIP specification.

Domain Name. This is the domain name which will be used along with the user name to construct the address of record.

Send Registration Request. When this option is selected, the phone sends a Registration Request to the SIP Server when it is initially powered up and then once every 1800 seconds.

Backup Server Address. This is either the IP address or the FQDN of the backup SIP server. The SIP Extensions Panel, described in section A.2.6 on page 50, sets the criteria for bypassing the primary SIP server

Backup Server Port. This parameter configures the transport protocol port number for the backup SIP server. The default port for both TCP and UDP transports is 5060, which is the value recommended by the SIP specification.

Send Registration to Backup Server. When this option is selected, the phone sends a Registration Request to the backup SIP Server when the primary server is bypassed, then once every 1800 seconds.

Enter the gateway settings as follows:

Dial Plan: x.T|x.#|*x.T|*x.#. For more information, refer to Appendix C, starting on page 85.

Dial Plan Timeout: The Dial Plan Timeout specifies is an interdigit timeout. After you press a digit, the phone waits for this period before automatically dialling the call. Pressing a digit restarts the timer. This parameter is measured in centiseconds. The default value is 220 (2.2 seconds); the maximum value is 65535 (655.35 seconds).

Transport: The ZIP2 supports UDP and TCP.

Phone Number: This data entry box is not supported.

CallerID Name: This is the Caller ID text that is sent by the ZIP2 as the Caller ID banner. This Caller ID banner is the SIP display name that is placed in the From header.

Port: This is the transport protocol port; normally set to 5060.

AEC On: Acoustic Echo Cancellation.

User Name: This is the user name component of the SIP address of record.

Password: When entered, this password authenticates the phone to the SIP server to which it is sending messages.

4.3.2.3 Configuring the ZIP2 to Receive Pages

Paging parameters are located on the CODECs web interface panel, as shown in figure 4-5. To enable the ZIP2 to receive pages, complete the following paging support parameters:

- select the Enable Paging Support parameter,
- enter the IP address or FQDN of the paging server,
- enter the paging port,
- select the codec used by the paging server in the appropriate data entry boxes.

4.3.2.4 STUN

The STUN protocol specifies a scheme to determine the public IP address of an IP device. The ZIP 2 implements STUN as specified in RFC 3489, first to discover if it's behind a NAT/Firewall, then to obtain the public IP address and port number for that NAT/Firewall. If this discovery is successful, the ZIP 2 then rewrites all outgoing SIP messages (including RTP port number and source IP address) to masquerade as originating from that Public IP address and port. This is

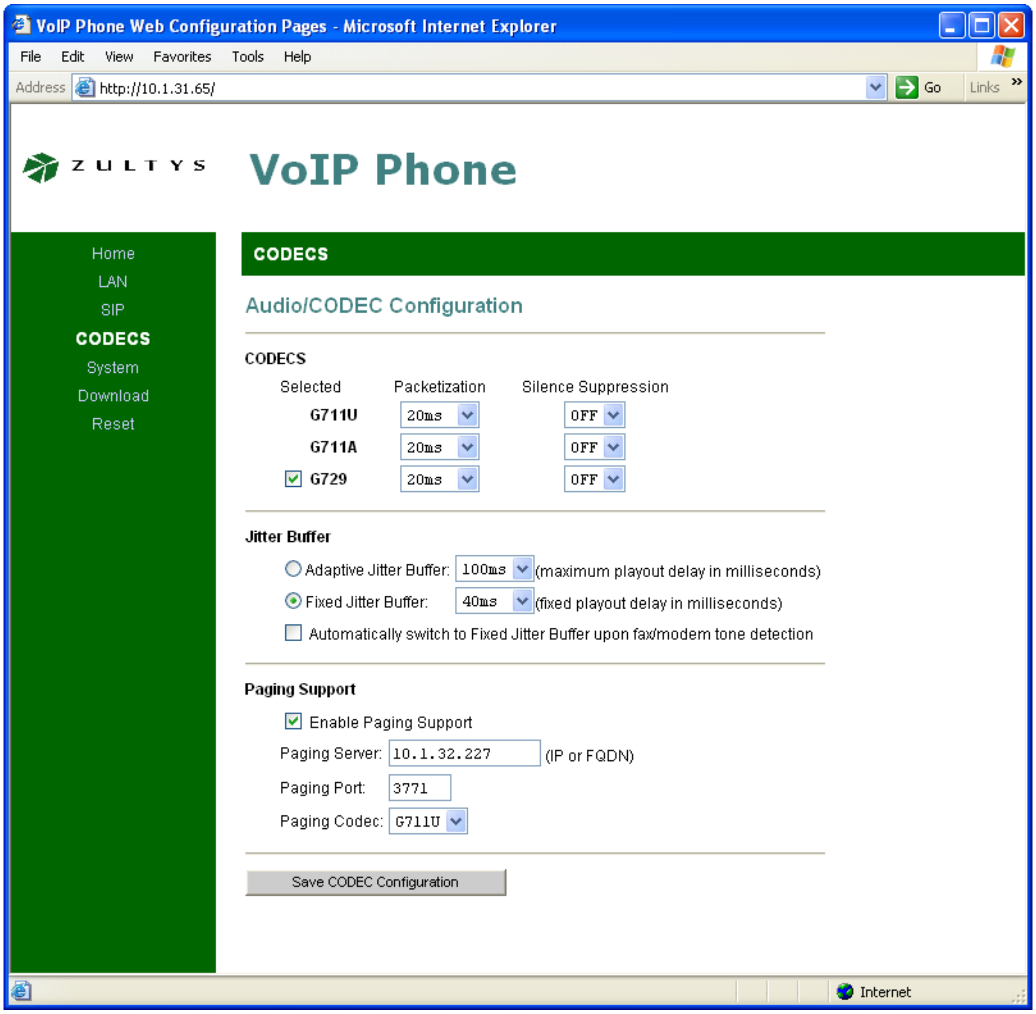

Figure 4-5 CODECs panel

required for SIP and RTP to traverse NATs, since without STUN, SIP would send explicit references to the phone's private IP address and port which is not accessible from outside the NAT/Firewall.

STUN requires a STUN server external to the NAT. This would typically be maintained by the local ISP or ITSP. STUN works across most Firewalls and NATs with the exception of a "full cone" NAT, defined as a NAT that rewrites both the IP address and port number of the phone each time it makes a connection to the outside world.

To implement STUN, specify the IP address or FQDN of the STUN server and the STUN port on the ToS/DiffServ/STUN panel, as shown in figure 4-6.

4.3.2.5 Configure other options

The Download and Configuration Utility provides many other panels for configuring ZIP2 phone parameters. A list of these parameters include:

VLAN Settings. see section A.2.4 on page 47 and section A.2.9 on page 53.

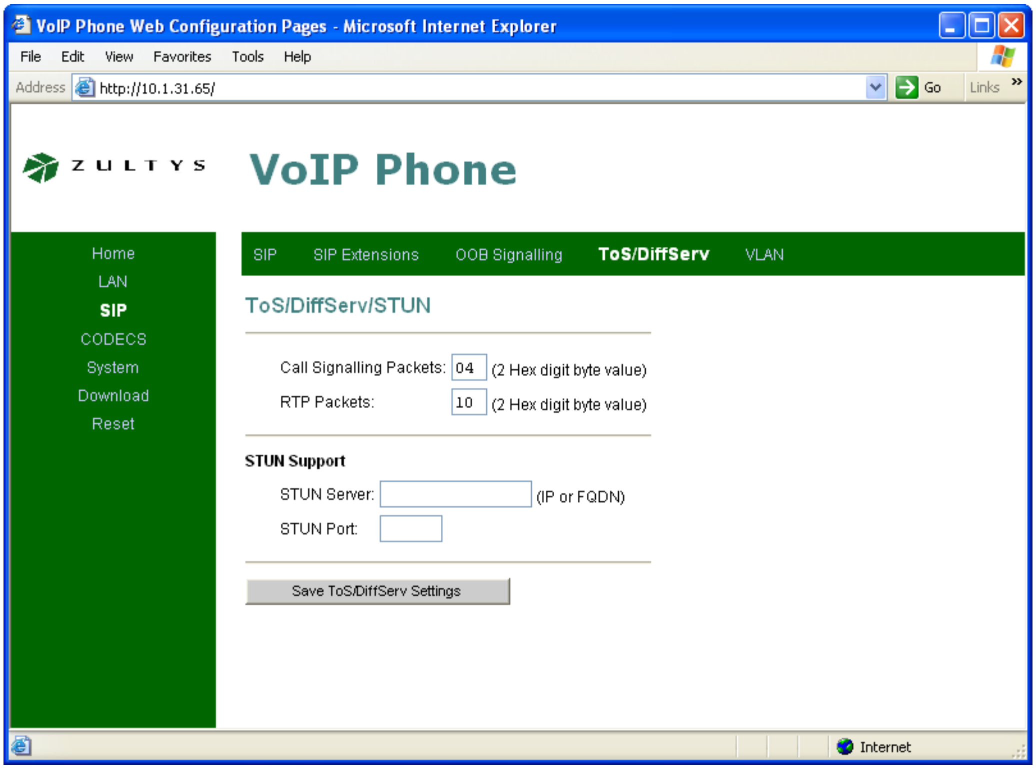

Figure 4-6 ToS / DiffServ / STUN panel

SIP Stack Extensions. see section A.2.6 on page 50.

Out of band signalling. see section A.2.7 on page 51.

ToS/DiffServ settings. see section A.2.8 on page 52.

CODECs and Jitter buffers. see section A.2.10 on page 53.

System Password. see section A.2.11 on page 55.

Local time settings. see section A.2.12 on page 55.

Ringer tone. see section A.2.13 on page 56.

SNMP settings. see section A.2.14 on page 57.

4.3.2.6 Review Settings

To access the LAN Status panel, as shown in figure 4-7, select LAN | LAN Status from the menu bars. Review the new LAN Configuration settings and return to the LAN Configuration panel if any of these settings need to be revised.

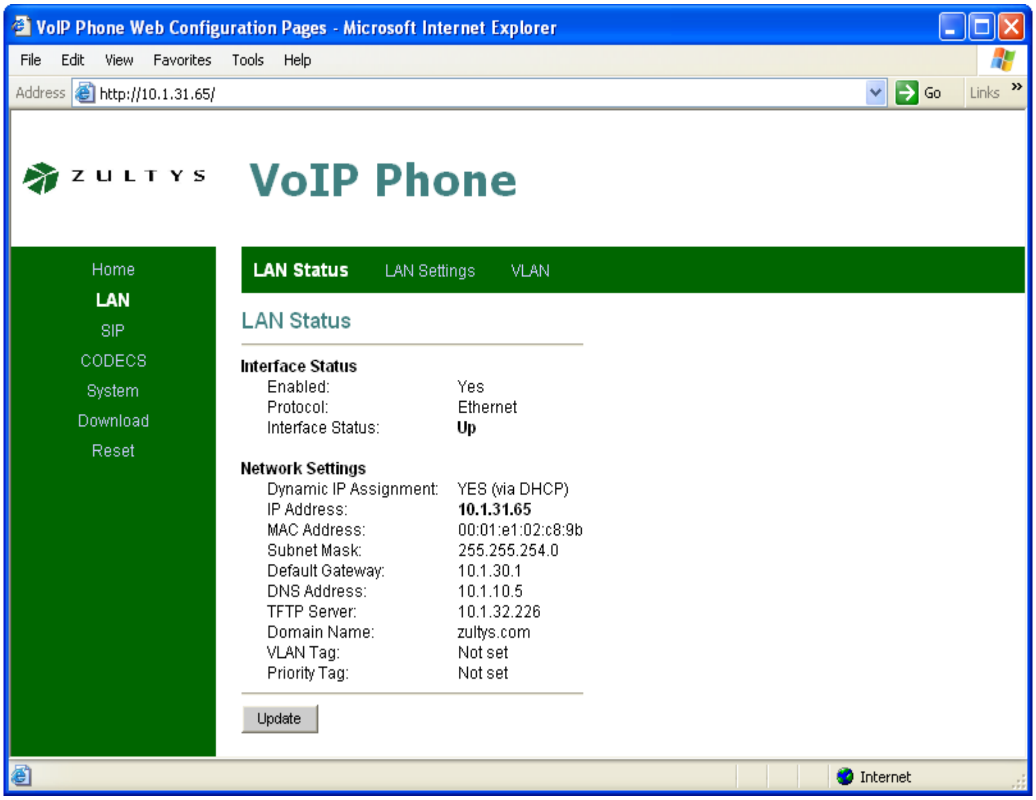

Figure 4-7 LAN Settings panel

4.3.2.7 Downloading the Boot Code and the Application Code

The boot code provides the software that initiates the ZIP2 phone upon power up. The application code provides the software that operates the ZIP2 phone. To load the application code or the boot code onto the phone, access the Download panel, shown in figure 4-8, by selecting Download from the main menu bar.

You can download the boot code and the application code file either from the TFTP server (TFTP method) or from your local computer (HTTP method). Obtain the filename and the IP address of the server, then follow the instructions on the panel.

Step 1. Enter the IP address of the TFTP server and the name of the file in the data entry box, as shown in figure 4-8.

Step 2. Enter the name of the file to be downloaded in the data entry box, as shown in figure 5-8. The name of the application code file usually takes the form v200s3.50.zts; the name of the boot code file usually takes the form up3.50.zts.

Step 3. Press the ***Start TFTP Download*** button.

Step 4. The download program generates a confirmation window (figure 4-9) to verify that you want to proceed with the download. Press the OK button.

Step 5. The program displays the Download panel. The status bar confirms that the program is booting into Downloader mode, as shown in figure 4-10.

Figure 4-8 Download panel

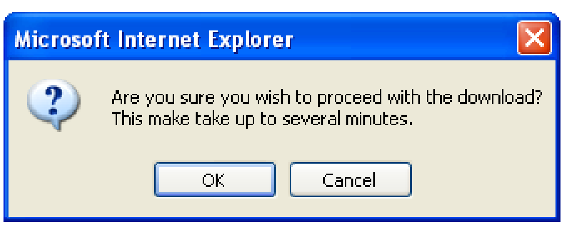

Figure 4-9 Download Confirmation panel

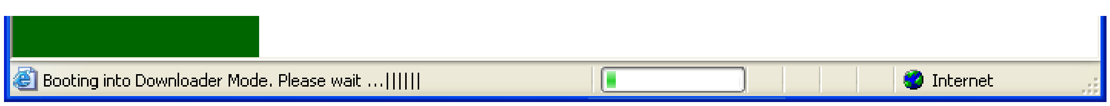

Figure 4-10 Status Bar of Download Panel – Downloader Mode

Step 6. After the program has booted into downloader mode, the status bar confirms that the program is performing the TFTP download, as displayed in figure 4-11.

Step 7. The program returns the Download Successful panel, as shown in figure 4-12, to verify that the operation is complete.

Figure 4-11 Status Bar of Download Panel – TFTP Download

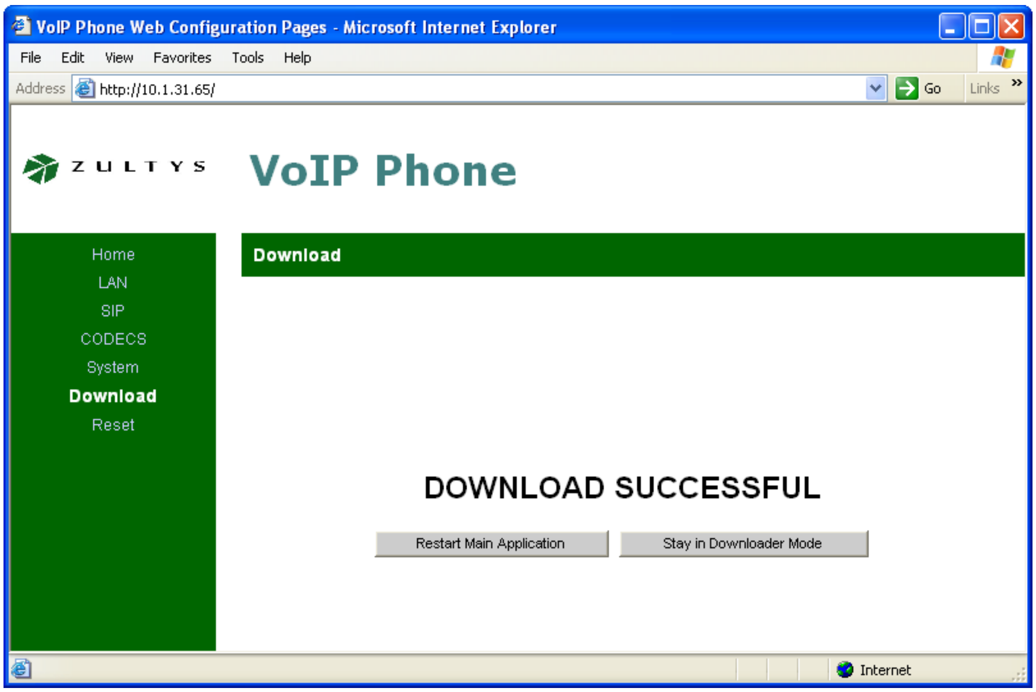

Figure 4-12 Download Panel – Success Message

Step 8. Press the Restart Application button in the Download Successful panel. The main application reboots, as explained by the panel in figure 4-13. After the phone reboots, you can verify the downloaded version number of the application code from the Home menu.

4.3.2.8 Reset the Phone

All settings stored to the phone through the Download and Configuration utility do not take effect until you power cycle or reset the phone.

- To power cycle the phone, remove power from the phone for a few seconds and then reapply power.
- To reset the phone, access the Reset panel, shown in figure 4-14, by selecting Reset from the main menu, then select ***Reset and execute Main Application***, and then press the Reset button.

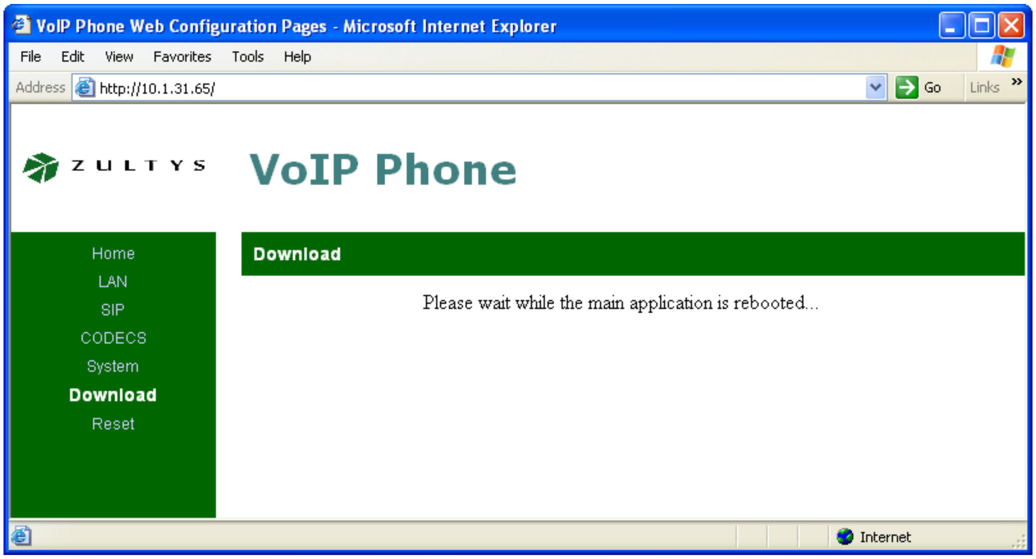

Figure 4-13 Download Panel – Wait for Reboot

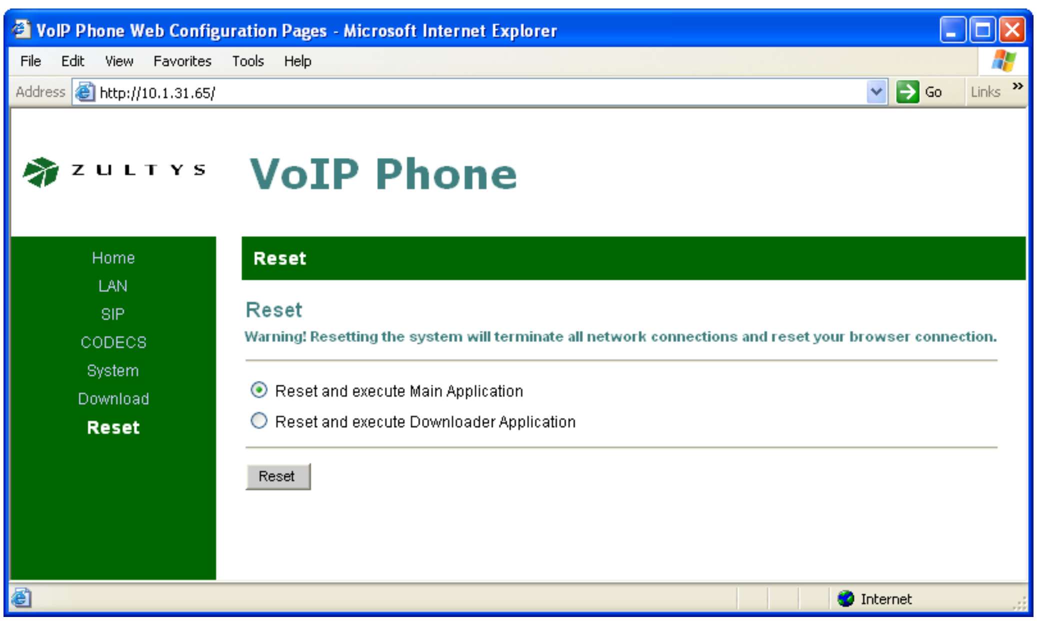

Figure 4-14 Reset panel

4.4 Setting the IP Address

You can program the ZIP2 to obtain its IP Address dynamically from the DHCP server, or you can bypass the DHCP server and specify a fixed IP address. The ZIP2 provides three methods of configuring the IP Address configuration mode:

- Select DHCP mode or specify the fixed IP address in the configuration files through the TFTP server. See section 4.3.1 on page 14.

- Select DHCP mode or specify the fixed IP address through the LAN Settings panel of the Download and Configuration Utility. See section 4.3.2.1 on page 16.
- Manually set the ZIP2 phone to DHCP mode (see section 4.4.1) or Fixed Address Mode (see section 4.4.2), then reset the phone to specify the IP Address of the phone.

The factory default IP addressing mode is DHCP mode.

4.4.1 Manually Selecting DHCP Mode

The following procedure returns the ZIP2 to DHCP mode and resets all of the factory defaults:

1. Ensure that there is no power connected to the ZIP2 (do not connect the power adapter).
2. Pick up the handset.
3. While pressing the ‘1’ button, connect power to the phone. Do not release the ‘1’ button until both the red and yellow LEDs stop flashing; this usually takes about 10 seconds.
4. Disconnect power from the ZIP2 by removing the power adapter.
5. Connect the ZIP2 to your corporate LAN using an Ethernet cable.
6. Connect power to the ZIP2 and ensure that it is powered up.
7. Verify the operation of the phone by checking for dial tone (section 3.2.2 on page 12) and completing a call (section 3.2.3 on page 12).

4.4.2 Manually Selecting Fixed IP Address Mode

Enabling fixed IP addresses disables DHCP mode and resets the IP Address of your ZIP2. You can also use this procedure to reset the IP address of the ZIP2 phone to a known address:

1. Ensure that there is no power connected to the ZIP2 (do not connect the power adapter).
2. Pick up the handset.
3. While pressing the ‘2’ button, connect power to the phone. Do not release the ‘2’ button until both the red and yellow LEDs stop flashing; this usually takes about 10 seconds.
4. Disconnect power from the ZIP2 by removing the power adapter.
5. Connect the ZIP2 to an isolated Ethernet switch or directly to a PC with an Ethernet crossover cable. Ensure that the ZIP2 is not connected to your corporate LAN.
6. Replace the power adapter onto the ZIP2 to re-connect power. Verify that the ZIP2 is fully powered up; the red LED should flash once.
7. Change your PC’s IP address to 192.168.0.10 and subnet mask to 255.255.255.0.
8. Open the web browser on your PC and access the ZIP2 web configuration at this URL: http://192.168.0.100.
9. Access SIP Configuration on the web interface, as shown in figure 4-4, enter the IP address of the SIP server, and save the settings.
10. Access LAN | LAN Settings on the web interface, as shown in figure 4-3. Set the correct IP Address for your phone and the IP subnet mask, IP default gateway, DNS server, and Domain Name for your LAN. Save the settings for future reference.
11. Turn off the ZIP2 by unplugging the power cord.

12. Connect the ZIP2 to your corporate LAN using an Ethernet cable.
13. Connect power to the ZIP2 and ensure that it is powered up.
14. Verify the operation of the phone by checking for dial tone (section 3.2.2 on page 12) and completing a call (section 3.2.3 on page 12).

4.5 Updating Settings and Software

After completing the initial configuration, you can use the Download and Configuration Utility to download new versions of the application code, modify network configuration settings, or view version and configuration information.

To view configuration and application code version information, access the Home (select Home on main menu bar) or LAN Status (LAN | LAN Status) panels.

To modify network configuration settings, access any panel listed under the LAN, SIP, CODECS, or System main menu bar option, enter the desired configuration settings, press the button at the bottom of the panel, then Reset the phone.

To download new application code, refer to section 4.3.2.7 on page 22.

To reset the phone, refer to section 4.3.2.8 on page 24.

4.6 Boot Process

4.6.1 Default

To use the ZIP2 in its usual manner, you should connect the phone to a LAN that has a DHCP server. The ZIP2 default boot up process, as summarized by the figure 4-15 flow chart, requires the following steps:

1. Connect to a DHCP Server

   Upon connecting the ZIP2 to your LAN, it searches for a DHCP server. Depending on your network, this typically requires from two to 65 seconds. If a DHCP server is found, it will respond with the Domain name, DNS server, IP address, TFTP server, and network gateway.

   A phone that is properly receiving power and has obtained a valid IP address will play a dial tone when it is off hook, as described on section 6.1.2 on page 35. The ZIP2 will not play a dial tone if it cannot locate a DHCP server.

2. Connect to the TFTP server

   The phone accesses the TFTP server to locate a configuration file that is common for all ZIP2 phones connected to your network.

   The phone reads the common configuration file, which points to the directory where the phone can find its specific configuration file. The phone then reads its specific configuration file.

3. Connect to the DNS Server

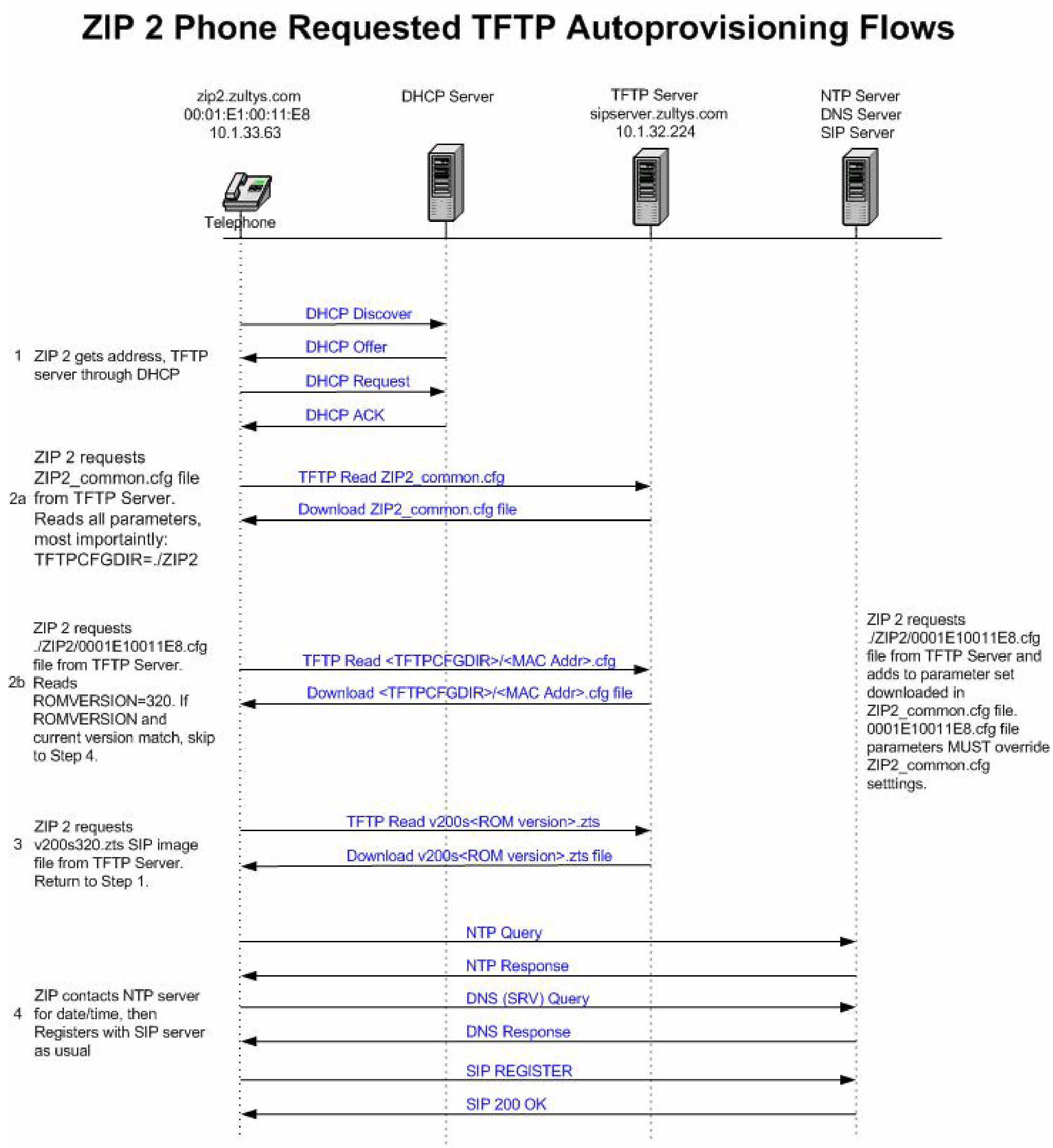

Figure 4-15 ZIP2 Phone Default Boot Process – DHCP enabled

After locating the DHCP server, the ZIP2 sends a DNS SRV query to obtain the SIP Registrar/Registration server for your DNS Domain, using the local domain name obtained from the DHCP server.

4. Connect to the SIP Registrar

Upon receiving the address of the SIP Registrar, the ZIP2 registers with the server, constructing the Address of Record from the Phone MAC address and the domain name supplied by the DHCP server.

If the phone is unable to register with the SIP Registrar, you will be unable to make any phone calls.

4.6.2 Boot up Process Variations

Although the Default process provides the preferred implementation of the ZIP2 phone, you can also disable the DHCP setting and set, as constant, such parameters as the IP addresses or the SIP server address.

Chapter 5

Interacting with the Phone

5.1 Call Appearances

The ZIP2 has two call appearances. The number of call appearances on a phone indicates the number of voice calls that the phone can handle. Each call can be individually originated or received by the phone. The person using the phone can choose to toggle between the calls or place both calls into a three-way conference.

This type of phone is different from phones that have been in common use for the past 100 years. On those older phones a physical pair (or two pairs) of wires carry a single voice call. A phone that can accommodate two voice calls connects to two such circuits, with each circuit referred to as a line. A switch on the phone selects between the lines (and therefore with whom you are talking).

The ZIP2 has a single circuit to connect to the switch. All calls take place over this same circuit, or line. Dedicated software and hardware inside the phone and the switch permit you to have two simultaneous conversations. These are referred to as two calls, not two lines.

5.1.1 First Call Appearance

When you pick up the handset on an idle ZIP2 and make a call, you are using the first call appearance. When an idle phone rings, the incoming call occupies the first call appearance.

5.1.2 Second Call Appearance

When you make a new call during an existing call, the new call uses the second call appearance. You must put the first call appearance on hold before you can make the second call. You cannot make any more calls when both call appearances are in use.

When you are on a call and the phone receives another call, the second call uses the second call appearance. The phone can receive no other call when both call appearances are in use.

5.2 Using the Keypad

The term *keypad* refers to the numeral and special purpose keys shown in figure 5-1.

5.2.1 Numeral Keys

The twelve numeral keys are labelled 0 to 9, *, and #. These keys are used for dialling phone numbers. Several numeral keys, when part of a combination keystroke, perform other phone functions. These functions are identified by the text next to the key.

Figure 5-1 ZIP2 Keypad

5.2.2 Special Purpose Keys

The keypad has four special purpose keys that are located below the numeral keys.

5.2.2.1 Function Key

The keypad has a key marked Func that serves as a function, or shift, key to select alternative uses for many of the numeral keys. To select the alternative use for a key, press and release the Func key once, then press the appropriate key to select the alternative function.

Important Do not press the Func key and another key simultaneously. The ZIP2 can read only a single key being pressed at a time.

The Func key is green and the text on that keypad that identifies the alternative function of a key is also green. For example, to initiate a the conference (conf) call, press Func, then the '7' button.

5.2.2.2 Redial key

The Redial key dials the last phone number called from the phone.

5.2.2.3 Hold key

The Hold key places a call on hold.

5.2.2.4 Speaker key

The Speaker key is used for hands free dialling and switches the destination of the call progress tones from the handset to the speaker. You can also use the speaker button to mute a call, allowing you to listen to the other party without being heard.

5.2.3 Keystroke Combinations

Several ZIP2 numeral keys can initiate other functions as part of a multiple keystroke combination.

- To forward all incoming calls, press the **Func** key, then the '1' key, and then the number of the party that will receive the forwarded calls.
- To place the phone in Do Not Disturb mode (DND), press the **Func** key, then the '3' key.
- To transfer a call, press the **Func** key, then the key marked **Transfer,** and then the number of the party that will receive the transferred call.
- To create a conference during a call, press the **Func** key, then the **Conf** key, and then dial the number of the third party.
- To toggle between the call on the first call appearance and a call on the second call appearance, press the **Func** key, and then the key marked **Toggle**.

See chapter 6, starting on page 35 for detailed instructions on performing these operations.

5.3 Handset and Speaker

5.3.1 Handset

The handset is the traditional phone apparatus through which you conduct a conversation. The handset cord socket is located on the left side panel of the phone, as shown in figure 3-1 on page 11. The handset provides the only means of talking into the phone.

5.3.2 Speaker

The speaker allows you to listen to dialling sounds without using the handset. The speaker is not part of a speakerphone, so you cannot conduct a conversation through it. The primary advantage offered by the speaker is providing hands free dialling.

5.3.3 Volume Switches

The ZIP2 provides three switches that control the output volume of phone signals. Each switch has three settings; low, medium, and high volume.

The *ringer switch*, located on the left panel, controls the volume of the call progress tones. The *speaker switch* and the *handset switch* are each located on the right side panel.

5.4 LEDs

The ZIP2 has two LEDs:

- The small yellow LED next to the speaker button indicates power on status and speaker mode.
- The large, red LED above the keypad indicates power on status, incoming calls, and the presence of voice mail messages.

Chapter 6

Using the Phone

6.1 Going Off Hook and On Hook

You can use the handset or speaker to initiate phone calls. This section describes use of the handset and speaker and defines the on hook and off hook states. Subsequent sections describe the procedures for making and terminating a call.

6.1.1 Using the Handset and Speaker

To use the handset, pick it up. When you have finished using it, replace it in the cradle.

To use the speaker for hands free dialling, press and release the Speaker key. When you are finished, press the key again. When the speaker is active, the yellow LED next to the Speaker key is on and the handset becomes inoperable. When the speaker is inactive, the phone turns off the LED.

The speaker allows you to listen to dialling signals without using the handset. The speaker is not part of a speakerphone, so you cannot conduct a conversation through it. The primary advantage offered by the speaker is providing hands free dialling.

6.1.2 Off Hook

The term *off hook* means that you do one of the following:

- pick up the handset
- press the Speaker key so that the speaker is active (LED is lit continuously yellow)

The phone is said to be off hook when you have done either of these things.

6.1.3 On Hook

The term *on hook* means that you do all of the following:

- replace the handset in the cradle
- press the Speaker key so that the speaker is inactive (speaker LED is off)

The phone is said to be on hook when both of these conditions are met.

6.2 Making a Call

You must obtain dial tone before you can dial a number.

Important This type of phone is different from phones that have been in common use for the past 100 years. On those older phones, as you press a button to dial a digit, the phone transmits the digit to the telephone exchange. On the ZIP2, the phone sends all the digits as a complete message and you therefore need to inform the phone when you have entered all the digits. The phone then assembles the complete message and sends it to the SIP server.

6.2.1 Dialling a Number

To dial a number:

1. Take the phone off hook[1] or put the current call on hold.

   The phone provides dial tone to the handset or speaker.

2. Enter the digits.

   The phone plays the DTMF tones to the handset or speaker as you type the digits.

3. When you have finished entering all the digits, do one of the following:

   a. Press the # key.

   b. Wait two to six seconds, after which time the phone assumes that you have entered all the digits.

4. The phone sends the dialled digits. It plays no sound while doing this.
5. While the call is proceeding, the phone plays ringback tone. If you receive a fast busy tone instead of ringback, see section 3.2.3 on page 12.
6. When the call is established (the called party has answered), begin your conversation.

6.2.2 Redial

To dial the name or number that was most recently dialled on the phone, press the Redial key once and then press the # key. If you have not picked up the handset, the phone enables speaker mode.

6.3 Call Proceeding and Call Answered

6.3.1 Ringback

Ringback is the tone that a caller hears after dialling a number.

1. See section 6.1.2 on page 35 for a definition of off hook.

6.3.1.1 ZIP2 to SIP Phone Calls or ZIP2 to ISDN Calls

When you call another SIP phone, the ringback that you hear is generated by your phone. This tone is determined by the country setting that you selected when you provisioned the phone.[1]

If you are making a call to the PSTN, the SIP to PSTN gateway normally opens the communication path so that you hear the sound coming from the network. Therefore, if you are calling a different country you will hear the ringback tone from that country. This tone may be different from that used in your country.

6.3.1.2 ZIP2 to CAS Calls

If you are connected to the PSTN using CAS circuits, your SIP to PSTN gateway may not know the status of the connection. It may open the speech path even though there is no ringback.

The phone behaves as if the call has been established as described in section 6.5 on page 39. Because the speech path is open, you may hear busy tone.

6.3.2 Far End Busy

If the person you are calling is busy, the phone either plays the busy tone for your country or the busy tone that is generated by the phone network at the far end. This depends whether your server has indication that the called party is busy or not.

6.3.3 Network Busy

If the network is busy, the phone either plays the fast busy tone (congestion tone) for your country or the fast busy tone that is generated by the phone network at the far end. This depends whether your server can detect whether or not the network is busy. The phone maintains this state until you go on hook.

6.3.4 Dialling an Invalid Destination

When you dial an invalid number, the phone plays the fast busy tone (congestion tone). The phone maintains this state until you go on hook.

6.4 Receiving a Call

6.4.1 LED Indication of Ringing

When the phone receives an incoming call, the red LED above the keypad will flash red during the audible portion of the ring and turn off during the silent portion of the ring.

1. See section A-12 on page 56 for information on selecting the country

6.4.2 Answering the Call

6.4.2.1 No Existing Conversation

If the phone is idle when you receive a new call, pick up the handset and begin your conversation.

6.4.2.2 Existing Conversation

If you are on a call when the phone receives a new call, you will hear a tone in the earpiece of the handset or over the speaker. To answer the new call, press the Func key and then the toggle (asterisk) key. The phone places the existing call on hold and answers the new call. You cannot receive any new calls that attempt to ring your phone after you place a call on hold.

If you have one call on hold and you have placed the phone on hook, the phone will ring until you pick up the phone to retrieve the call on hold; the call on hold will not be transferred to voice mail or anywhere else.

6.4.3 Rejecting the Call or Not Answering

6.4.3.1 Not Answering Calls

If you do not answer an incoming call, the ZIP2 will continue to ring until the calling party terminates the call attempt or until your phone system alternatively routes the call.

6.4.3.2 Do Not Disturb (DND)

When the phone is in Do Not Disturb mode, the ZIP2 immediately rejects any call that it receives. Setting DND has no effect on calls that are in progress or on hold. The phone does not make any sound when rejecting a call. Call forwarding mode overrides DND mode.

When a call is rejected, the phone behavior depends upon the system to which it is connected. Your system may route a call to voice mail or it may disconnect the caller.

To place the ZIP2 in DND mode:

1. Go off hook.

2. Press the DND key combination (Func and '3').

3. When the phone plays the busy signal, go on hook.

When the ZIP2 is in DND mode, the red LED above the keypad is lit continuously.

To cancel DND mode, repeat these three steps. The ZIP2 indicates that it is no longer in DND mode by turning off the red LED.

6.4.4 Call Forwarding

When the phone forwards a call, it redirects the incoming call to another destination. The destination can be another extension within the enterprise or an external number.

To forward all calls to another destination:

1. Go off hook.

2. Press the Call Forward key combination (Func and '1').
3. Enter the destination phone number.
4. Press the # key or wait two to six seconds.
5. When the phone plays three short tones, go on hook.

   The phone immediately forwards calls to the specified destination. When the ZIP2 is in Call Forward mode, the red LED above the keypad is lit continuously.

To cancel call forwarding, go off hook, press Func and '1', then go on hook. The ZIP 2 indicates that it is no longer in call forward mode by turning off the red LED.

6.5 During a Call

During a call, you can transfer the current conversation or place it on hold. You can create or answer a new call and you can create a conference call.

If you press one of the keys 0 to 9, *, or # while a call is active, the phone sends a message for that key and plays a DTMF tone in the earpiece or speaker simultaneously.[1]

6.5.1 Mute

If you want to hear the person with whom you are speaking, but do not want him or her to hear you, press the Speaker button. The phone disconnects the handset, which includes the microphone input. To resume a 2-way conversation, press the Speaker button again.

When you place the phone in Mute mode, you mute the entire phone, not just the current conversation. If you place the muted call on hold, and select another call, the mute function remains active and the person on the other call will not be able to hear you.

When you mute the phone and press the digit keys, the phone plays the DTMF digit into the audio path you have selected and sends the digit to the switch. You might use this feature to communicate with an IVR system.

6.5.2 Hold

6.5.2.1 Description

The Hold function allows you to maintain the state of a call but suspends the conversation. The person cannot hear you and you cannot hear the other person. When you place a call on hold, you must retrieve the call from your same phone to resume the conversation.

The ZIP2 allows you to place only one call on hold at a time. You cannot receive incoming calls when you have a call on hold.

1. The message is sent in the RTP payload according to RFC 2833.

6.5.2.2 Placing a Call on Hold

Press the Hold key to place a conversation on hold. The phone automatically activates the next call appearance and provides a dial tone. You must keep the phone off hook to keep the call on hold. If you place the phone on hook, it will ring to prompt you to continue your conversation.

6.5.2.3 Making a New Call After Placing a Call on Hold

The phone automatically switches to the new call appearance and provides you with dial tone when you have a call on hold. To make a new call, dial the number.

6.5.2.4 Resuming the Conversation

Press the Hold key to resume the call with the person on hold.

6.5.3 Call Toggling

If you have one active call and one call on hold, the toggle key combination changes the state of each call appearance: the active call is placed on hold while the other call is activated. The toggle key combination does not work during a conference call.

6.5.4 Transferring a Call

You can transfer a call you made to, or a call you answered from, another person inside or outside of the enterprise. The ZIP2 can perform unattended transfers and attended transfers.

6.5.4.1 Unattended Transfer

In an unattended call transfer, you do not speak to the person receiving the call prior to the transfer. To perform an unattended transfer:

1. Ensure that you have only one active call and that call is not on hold.
2. Press the Unattended Transfer key combination (Func and '4').

   The phone places the active call on hold, selects the other call appearance, and provides a dial tone.
3. Make the call to the phone that will receive the transferred call, as described in section 6.2 on page 36.
4. The ZIP2 plays the busy signal and is disconnected from the call. The call between the other two parties is completed normally.

6.5.4.2 Attended Transfer

In an attended call transfer, you speak to the person receiving the call prior to transferring the call. To perform an attended call transfer:

1. Ensure that you have only one active call and that call is not on hold.
2. Press the Attended Transfer key combination (Func and '6').

   The phone places the active call on hold, selects the other call appearance, and provides a dial tone.

3. Make the call to the phone that will receive the transferred call, as described in section 6.2 on page 36.
4. When the other party answers the new call, announce your intent to transfer the call,

The next step depends upon the recipient's response to receiving the transfer:

- ***If the recipient of the transfer is willing to accept the transferred call,*** hang up the ZIP2 to complete the transfer.
- ***If the recipient of the transfer is unwilling to accept the transfer,*** allow that person to hang up first. The ZIP2 then resumes your conversation with the party that was to be transferred.
- ***If the transferring call is not answered or if the line is busy,*** put the ZIP2 on hook; it will immediately ring to resume your conversation with the party that was to be transferred.
- ***If the transferring call is answered by voice mail,*** press the * button twice, then wait for the voice mail system to hang up. The ZIP2 will resume your conversation with the party that was to be transferred.

6.6 Conference Calls

The ZIP2 allows you to hold a conversation with two other people. A conversation with more than one person is called a *conference*. This feature is supported on the ZIP2 such that the function operates on any phone system that supports multiple call appearances to the phone. The ZIP2 supports only one active conference call at a time.

The ZIP2 performs the conference mixing locally on the phone.

6.6.1 Initiating a Conference Call

To initiate a conference call:

1. Make or answer the first call in the normal manner.
2. Press the Conference key combination (Func and '7').

   The phone places the first call on hold, selects the other call appearance, and provides you with dial tone.
3. Dial the third person for the conference.

   The phone reactivates the call that was on hold and joins all calls into a single call.

6.6.2 Terminating a Conference Call

6.6.2.1 Ending the Entire Conference

To end a conference call, go on hook. The phone terminates the calls on each of the call appearances.

6.6.2.2 Dropping from the Conference

When one of the two parties leaves the conference call, the call appearance that was occupied by the leaving party becomes available. If you drop out of the conference, all other calls in the conference are terminated.

The Hold key does not work during a conference call.

6.7 Receiving a Page

To enable the ZIP2 to receive pages, perform the page provisioning procedure described in section 4.3.2.3 on page 19.

Important All page parameters must be entered as configured on your system or the ZIP2 will not receive any pages.

On the receipt of an incoming paging stream, the yellow LED is continuously lit and the audio page announcement plays on the external speaker for the duration of the page.

6.8 Ending a Call

6.8.1 You Terminate the Call

You terminate a call by going on hook.[1] The phone ends the call, and silences the handset and speaker.

6.8.2 Other Party Terminates the Call

When the person that you are talking with goes on hook, your ZIP2 phone plays a busy signal until you go on hook.

1. See section 6.1.3 on page 35 for a definition of on hook.

Appendix A

Download and Configuration Utility

A.1 Introduction

This chapter describes each Download and Configuration Utility panel. Refer to section 4.3 on page 14 for instructions on using these panels to configure the ZIP2 phone.

A.1.1 Accessing the Download and Configuration Utility

You access the utility through a web browser that can access the same network as the ZIP2 being configured. Enter the IP address of the ZIP2 in the address entry box of your web browser and press the enter button. The Home panel, shown in figure A-1, will appear in your browser.

A.1.2 Interface Structure

Each panel is accessed by selecting options from two menu bars. The main menu bar runs vertically on the left side of the window. Main menu bar options include Home, LAN, SIP, CODEC, System, Download, and Reset.

The secondary menu runs horizontally, directly above the configuration parameters. Secondary menu bar options depend on the selected main menu bar item.

All panels with editable parameters provide a save button which must be pressed to download changes to the phone. After changes are saved to the phone's memory, they take effect only after power cycling the phone or performing a reset operation, as described in section A.2.16 on page 58.

A.2 Web Interface panels

A.2.1 Home

The Home panel introduces the web interface and provides basic information about the ZIP2 phone. To access this panel, as shown in figure A-1, enter the web interface or select Home in the main menu.

The Home panel reports the status and values of the following parameters:

System Uptime. The continuous time that the ZIP2 phone has been in service since the last power cycle.

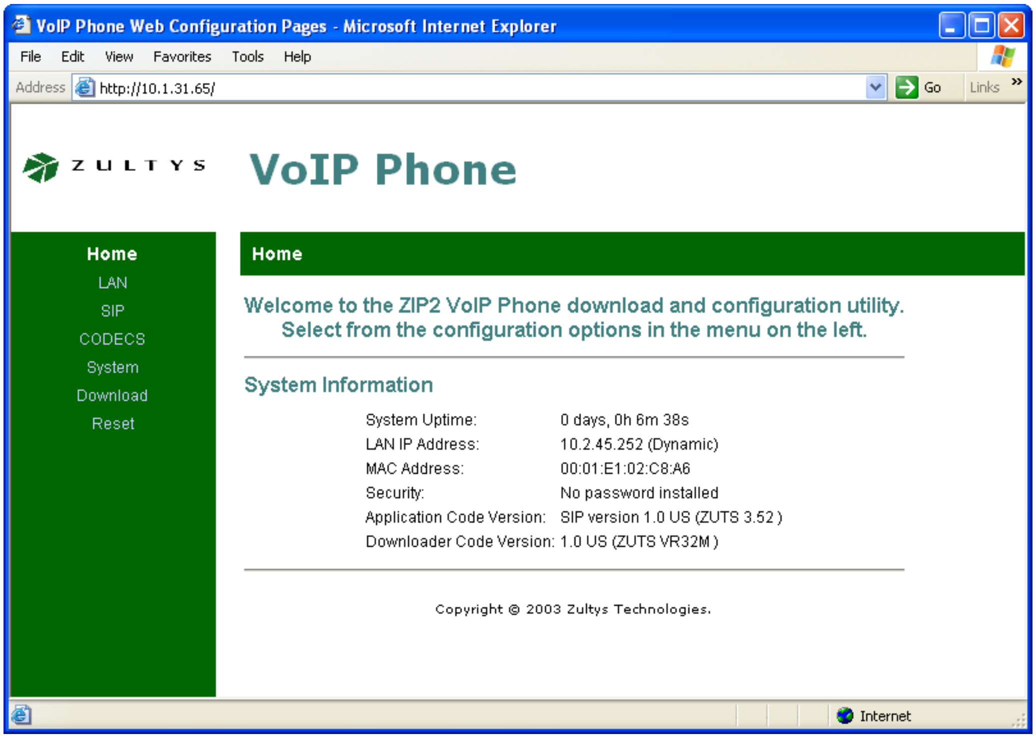

Figure A-1 Home panel

LAN IP Address. Reports the IP address and indicates whether it is static or dynamic. You change the address and allocation method from the LAN Settings panel, as described in section A.2.3.

MAC (Media Access Control) Address. The hardware number that uniquely identifies your ZIP2 phone from all other devices that can connect to an Ethernet network. The unique MAC address of any device is assigned when the device is manufactured.

Security. Indicates the presence of password protection for the web interface access to the ZIP2 configuration parameters. You set the password in the System Security window, described in section A.2.11.

Application Code Version. The software that operates the phone comprises two programs: the Application Code and the Downloader Code. The Application Code provides all ZIP2 functions. Updates to the ZIP2 phone almost always require an update to the application code.

Downloader Code Version. The downloader code is the ZIP2 software component that provides the web interface framework and basic ZIP2 configuration utilities. Once installed, the downloader code is rarely updated or modified.

A.2.2 LAN Status

The LAN Status panel reports the interface status and the value of the network settings. This panel is read only. Several parameters listed in this panel are configured in other web interface panels, while other parameters describe network conditions that are independent of the operation of the ZIP2.

To access the LAN Status panel, as shown in figure A-2, select LAN from the main menu and LAN Status from the sub-menu. To refresh panel contents, press the **Update** button.

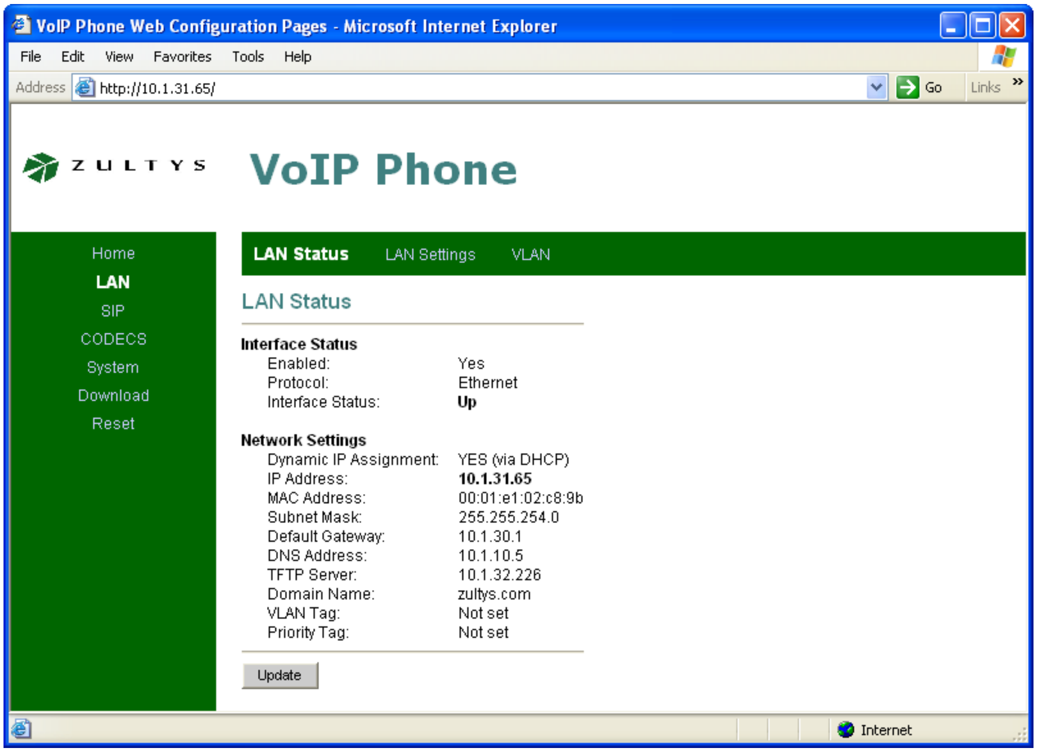

Figure A-2 LAN Settings panel

A.2.2.1 Interface Status

The interface status indicates that the phone is properly connected to an Ethernet network.

A.2.2.2 Network Settings

Network settings describes the interface between the ZIP2 and the Ethernet network to which the phone is connected.

Dynamic IP Assignment. This indicates the method by which the ZIP2 acquired its IP address. You set this parameter from the LAN Settings panel described in section A.2.3.

IP Address. This reports the IP address of the ZIP2. If the LAN IP assignment is ***Fixed***, you set this parameter from the LAN Settings panel described in section A.2.3.

MAC Address. This is the hardware number that uniquely identifies your ZIP2 phone from all other devices that can connect to an Ethernet network. The MAC address of a device is assigned when the device is manufactured.

Subnet Mask. This indicates the ZIP2 phone's subnet mask.

Default Gateway. This indicates the IP address of the default gateway.

DNS Address. This indicates the IP address of the DNS server.

TFTP Server. This indicates the IP address of the TFTP server.

Domain Name. This indicates the domain name.

VLAN Tag. This indicates the VLAN tag value, as configured in the VLAN Configuration panel described in section A.2.4

Priority Tag. This indicates the VLAN priority tag value, as configured in the VLAN Configuration panel described in section A.2.4

A.2.3 LAN Configuration

The LAN Configuration panel allows you to configure network information statically or select dynamic configuration. You access this panel as shown in figure A-3, by selecting LAN from the main menu and LAN Settings from the sub-menu.

Important Press the **Save LAN Settings** button to save the configured values to the phone reset the phone.[1]

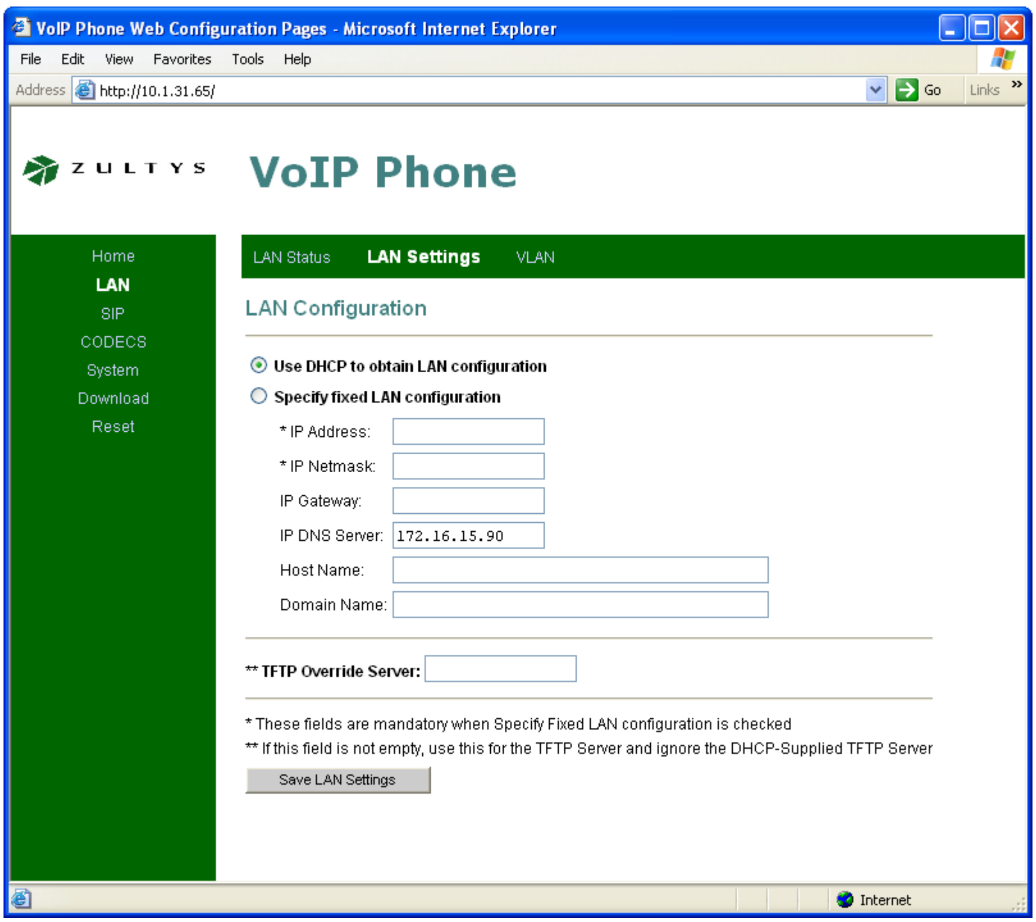

Figure A-3 LAN Configuration panel

1. If the configuration file and the LAN Settings panel specify different IP addressing modes (dynamic via DHCP versus Fixed IP Addressing), pressing the *Save LAN Settings* will produce unpredictable results when rebooting the phone.

Use DHCP to obtain LAN configuration. This option configures the phone to receive its IP address and netmask from the DHCP server. When you use DHCP to dynamically assign an IP address to the phone, the web interface will also accept the IP Netmask, the IP Gateway address, the IP DNS Server address, the domain name, and the TFTP server from the DHCP server.

The DHCP server also provides the IP address of the default gateway and DNS server, along with the host name, and the domain name.

Specify fixed LAN configuration. This option configures the phone with the fixed IP address and netmask that you specify. When selecting this option, you also provide the IP Address of the gateway, DNS server, and TFTP server along with the host and domain name of the network to which the phone connects.

TFTP Override Server. Most home firewalls and NATs provide their address as the TFTP server address in the DHCP response. This prevents auto provisioning via the TFTP server that stores the configuration files and firmware. To enable auto provisioning in this instance, specify a different TFTP server than that provided by DHCP. Enter the address of the TFTP server in the TFTP Override Server field, as shown in figure A-3 to solve this problem. To turn this feature off, clear the field and reset the phone.

A.2.4 VLAN

The VLAN Configuration panel allows you to configure VLAN and Priority tag settings. To access this panel, as shown in figure A-4, select LAN from the main menu and LAN Settings from the sub-menu.

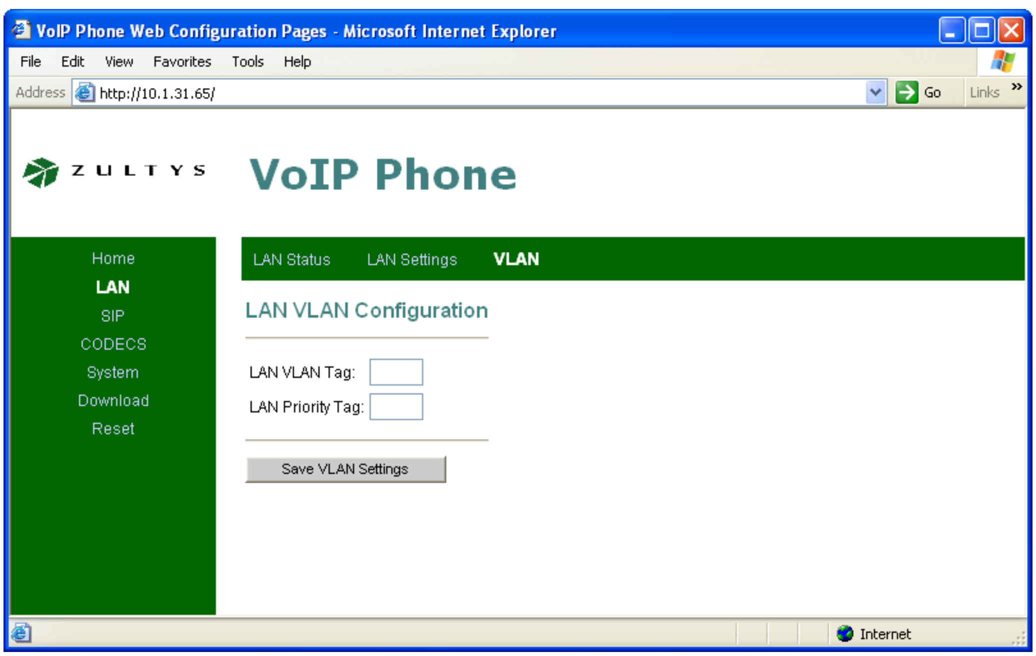

Figure A-4 VLAN Configuration panel

Important Press the **Save VLAN Settings** button to save the configured values to the phone. Settings do not take effect until you power cycle the phone or execute a Reset command (see section A.2.16 on page 58).

A.2.5 SIP Configuration

This panel comprises two sections: SIP Server Settings and Gateway Settings. To access this panel, as shown in figure A-5, select SIP from the main menu and SIP from the sub-menu.

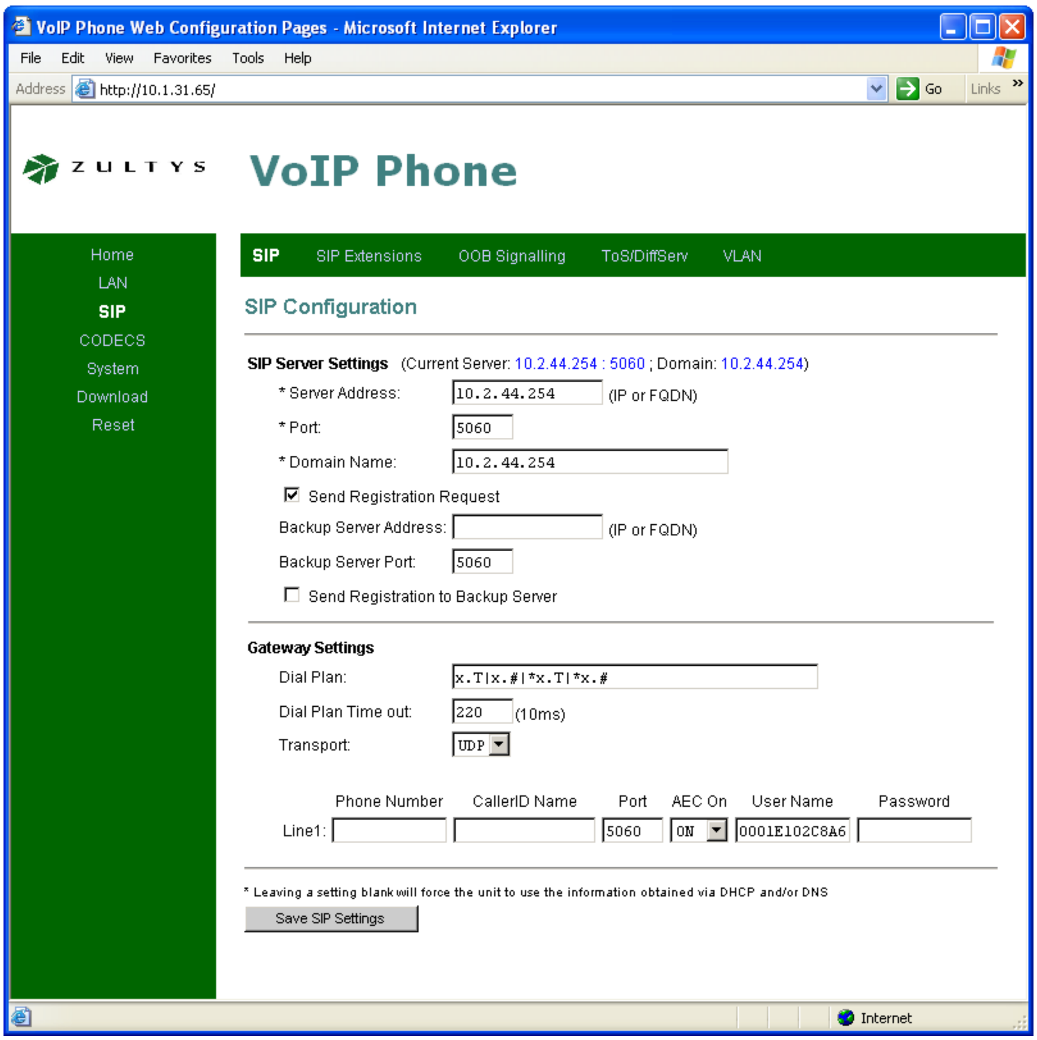

Figure A-5 SIP Configuration panel

Important Press the **Save SIP Settings** button to save the configured values to the phone. Settings do not take effect until you power cycle the phone or execute a Reset command (see section A.2.16 on page 58)

A.2.5.1 SIP Server Settings

The top panel section, next to the **SIP Server Settings** text, lists the current server, port, and domain settings. You can enter values for these parameters, which are loaded to the phone when you press the Save SIP Settings button. Any parameter that you leave blank at the time you press the Save SIP Settings button will be filled on the basis of information provided via DNS-SRV.

Server Address. This is either the IP address or the Fully Qualified Domain Name (FQDN) of the SIP Server to which the phone is connected. When this parameter is left blank, the software performs a DNS SRV lookup to resolve the SIP server for your domain.

Port. This parameter configures the transport protocol port number. The default port for both TCP and UDP transports is 5060, which is the value recommended by the SIP specification.

Domain Name. This is the domain name which makes up the domain portion of the address of record username@domain. It is also the domain used in the DNS-SRV lookup.

Send Registration Request. When this option is selected, the phone sends a Registration Request method to the SIP Server when it is initially powered up and then once every 1800 seconds.

Backup Server Address. This is either the IP address or the FQDN of the backup SIP server. The SIP Extensions Panel, described in section A.2.6, sets the criteria for bypassing the primary SIP server

Backup Server Port. This parameter configures the transport protocol port number for the backup SIP server. The default port for both TCP and UDP transports is 5060, which is the value recommended by the SIP specification.

Send Registration to Backup Server. When this option is selected, the phone sends a Registration Request to the backup SIP Server when the primary server is bypassed, then once every 1800 seconds.

A.2.5.2 Gateway Settings

Enter the following gateway settings as follows:

Dial Plan: x.T|x.#|*x.T|*x.#. For more information, refer to Appendix C on page 85.

Dial Plan Timeout: The Dial Plan Timeout specifies is an interdigit timeout. After you press a digit, the phone waits for this period before automatically dialling the call. Pressing a digit restarts the timer. This parameter is measured in centiseconds. The default value is 220 (2.2 seconds); the maximum value is 65535 (655.35 seconds).

Transport: The ZIP2 supports UDP and TCP.

Phone Number: This data entry box is not supported.

CallerID Name: This is the Caller ID text that is sent by the ZIP2 as the Caller ID banner. This Caller ID banner is the SIP display name that is placed in the From header.

Port: This is the transport protocol port; normally set to 5060.

AEC On: Acoustic Echo Cancellation; this parameter must always be set to ON.

User Name: This is the user name component of the SIP address of record.

Password: When entered, this password authenticates the phone to the SIP server to which it is sending messages using SIP Digest Authentication per RFC 3261. To disable this feature, clear the field and reset the phone.

A.2.6 SIP Stack Extensions

The ZIP2 phone supports several protocol extensions that are not defined in the base SIP RFC. This panel configures the phone to support those services required by your network implementation.

To access this panel, as shown in figure A-6, select SIP from the main menu and SIP Extensions from the sub-menu.

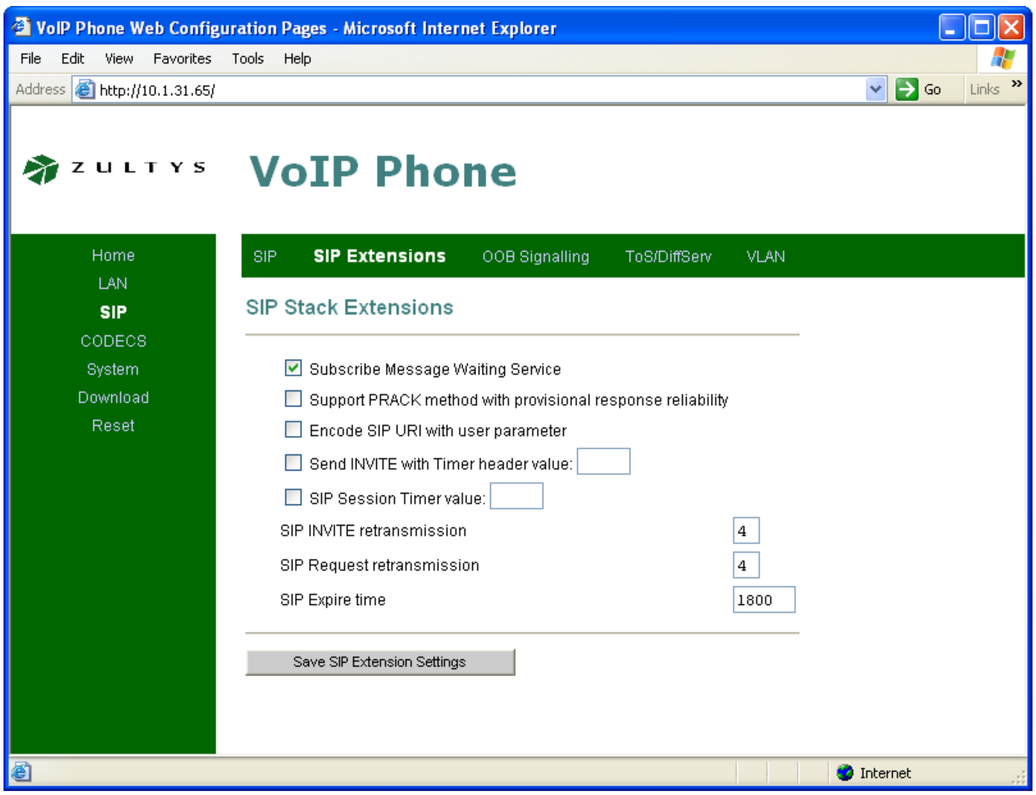

Figure A-6 SIP Stack Extensions panel

Subscribe Message Waiting Service. Implemented per draft-ietf-sipping-mwi-02.

Support PRACK method with provisional response utility. Implemented per RFC 3312.

Encode SIP URI with user parameter. Adds **User=Phone** to SIP request URIs.

Send INVITE with Timer header value. Implemented per draft-ietf-sip-session-timer-10.

SIP Session Timer value. Implemented per draft-ietf-sip-session-timer-10.

SIP INVITE retransmission. This parameter defines the number of unacknowledged INVITE methods that the ZIP2 sends before switching to the backup SIP server.

SIP Request retransmission. This parameter defines the number of unacknowledged Requests that the ZIP2 sends before switching to the backup SIP server.

SIP Expire Time. This parameter specifies the interval, in seconds, after which a Registration Expires. This value is inserted into the SIP Interval Header and SIP Expires Interval.

Important Press the **Save SIP Extension Settings** button to save the configured values to the phone. Settings do not take effect until you power cycle the phone or execute a Reset command (see section A.2.16 on page 58).

A.2.7 OOB Signalling

This panel configures out of band signalling parameters. To access this panel, as shown in figure A-7, select SIP from the main menu and OOB Signalling from the sub-menu.

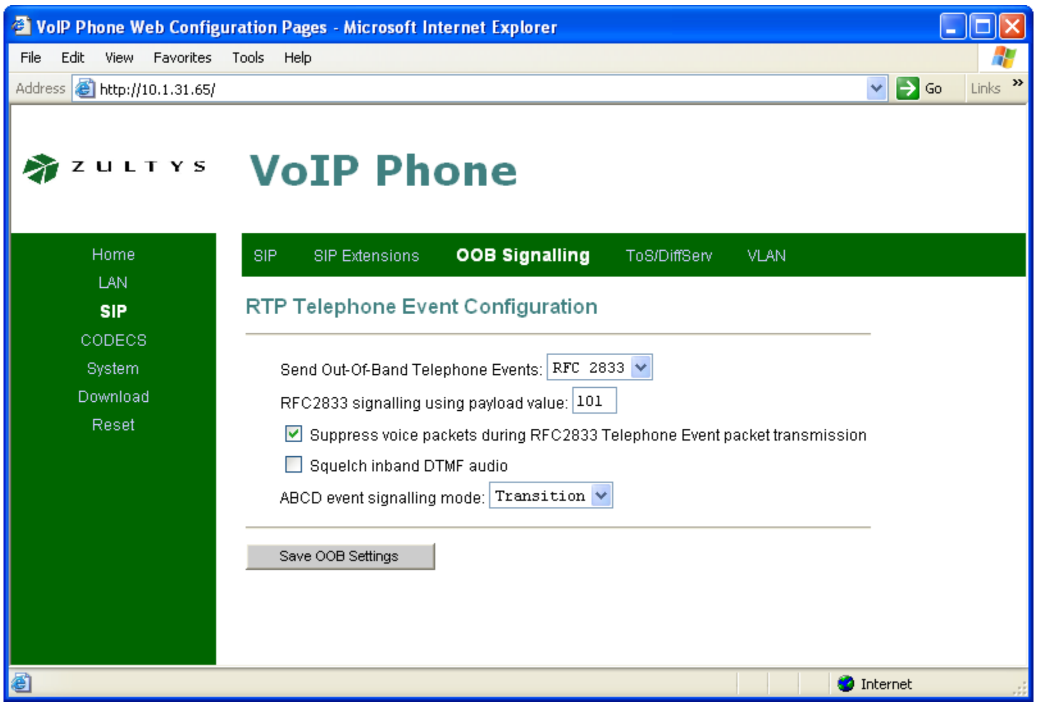

Figure A-7 RTP Telephone Event Configuration

Send Out-Of-Band Telephone Events. The ZIP2 supports the following signalling methods:

- Out of band signalling as defined by RFC 2833.

  RFC 2833 describes an RTP payload for DTMF digits, other tone signals, and telephony events.

- Out of band signalling as defined by RFC 2976 (***SIP Info*** option)
- In band signalling (***None*** option)

All other parameters are valid for systems that use RFC 2833 as the Out-Of-Band signalling protocol.

Important Press the Save OOB Settings button to save the configured values to the phone. Settings do not take effect until you power cycle the phone or execute a Reset command (see section A.2.16 on page 58).

A.2.8 ToS/DiffServ/STUN

This panel configures Layer 3 IP level quality of service parameters. To access this panel, as shown in figure A-8, select SIP in the main menu and ToS/DiffServ in the sub-menu.

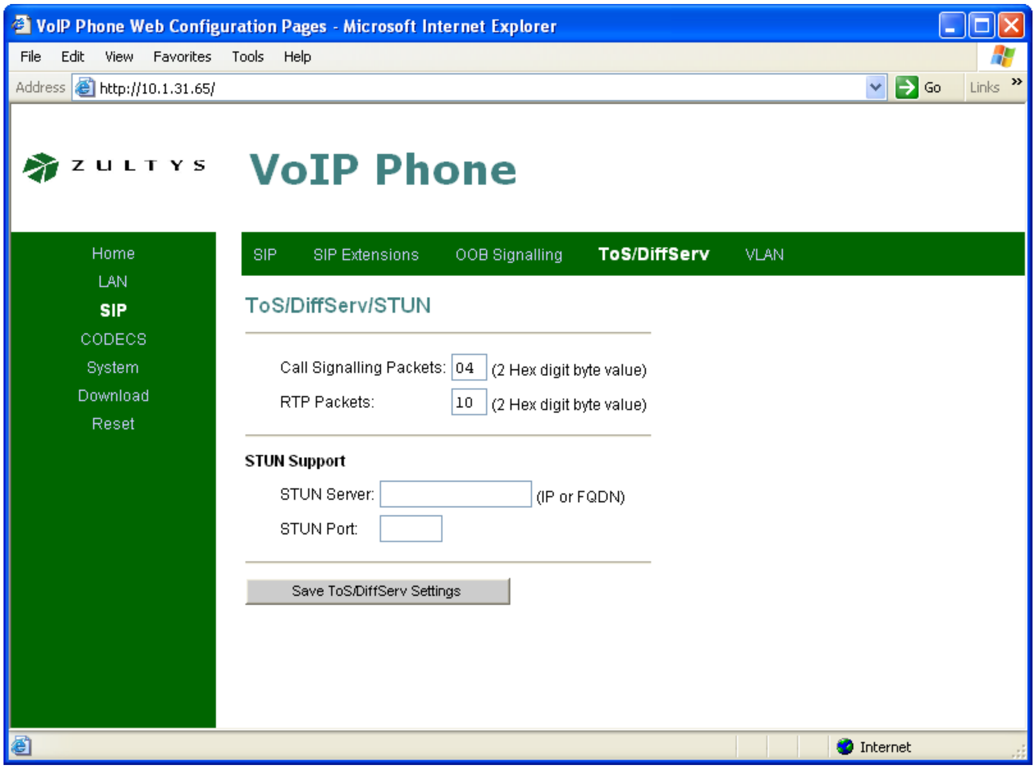

Figure A-8 ToS/DiffServ panel

Important Press the Save ToS/DiffServ Settings button to save the configured values to the phone. Settings do not take effect until you power cycle the phone or execute a Reset command (see section A.2.16 on page 58).

The STUN protocol specifies a scheme to determine the public IP address of an IP device. The ZIP 2 implements STUN as specified in RFC 3489, first to discover if it's behind a NAT/Firewall, then to obtain the public IP address and port number for that NAT/Firewall. If this discovery is successful, the ZIP 2 then rewrites all outgoing SIP messages (including RTP port number and source IP address) to masquerade as originating from that Public IP address and port. This is required for SIP and RTP to traverse NATs, since without STUN, SIP would send explicit references to the phone's private IP address and port which is not accessible from outside the NAT/Firewall.

STUN requires a STUN server external to the NAT. This would typically be maintained by the local ISP or ITSP. STUN works across most Firewalls and NATs with the exception of a "full cone" NAT, defined as a NAT that rewrites both the IP address and port number of the phone each time it makes a connection to the outside world.

To implement STUN, specify the IP address or FQDN of the STUN server and the STUN port on the ToS/DiffServ/STUN panel, as shown in figure A-8.

A.2.9 VLAN

The VoIP VLAN Configuration panel sets the VLAN tag values for the call signalling and RTP packets. Tag parameters that are not set in the panel will use the tags configured in the LAN VLAN Configuration panel, as described in section A.2.4 on page 47.

To access this panel, as shown in figure A-9, select SIP from the main menu, then select VLAN in the sub-menu.

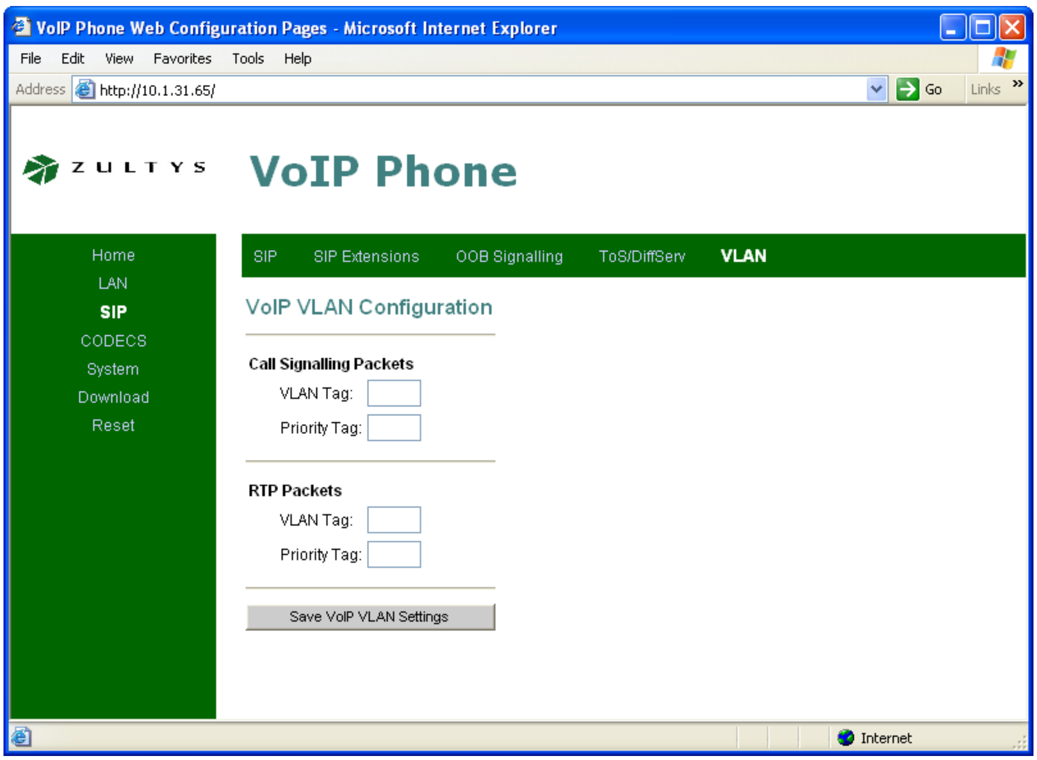

Figure A-9 VoIP VLAN Configuration panel

Important Press the **Save VoIP VLAN Settings** button to save the configured values to the phone. Settings do not take effect until you power cycle the phone or execute a Reset command (see section A.2.16 on page 58).

A.2.10 CODECS

The Audio/CODEC Configuration panel sets jitter buffer and CODEC parameters. To access this panel, as shown in figure A-10, select CODECS from the main menu.

Important Press the Save CODEC Configuration button to save the configured values to the phone. Settings do not take effect until you power cycle the phone or execute a Reset command (see section A.2.16 on page 58).

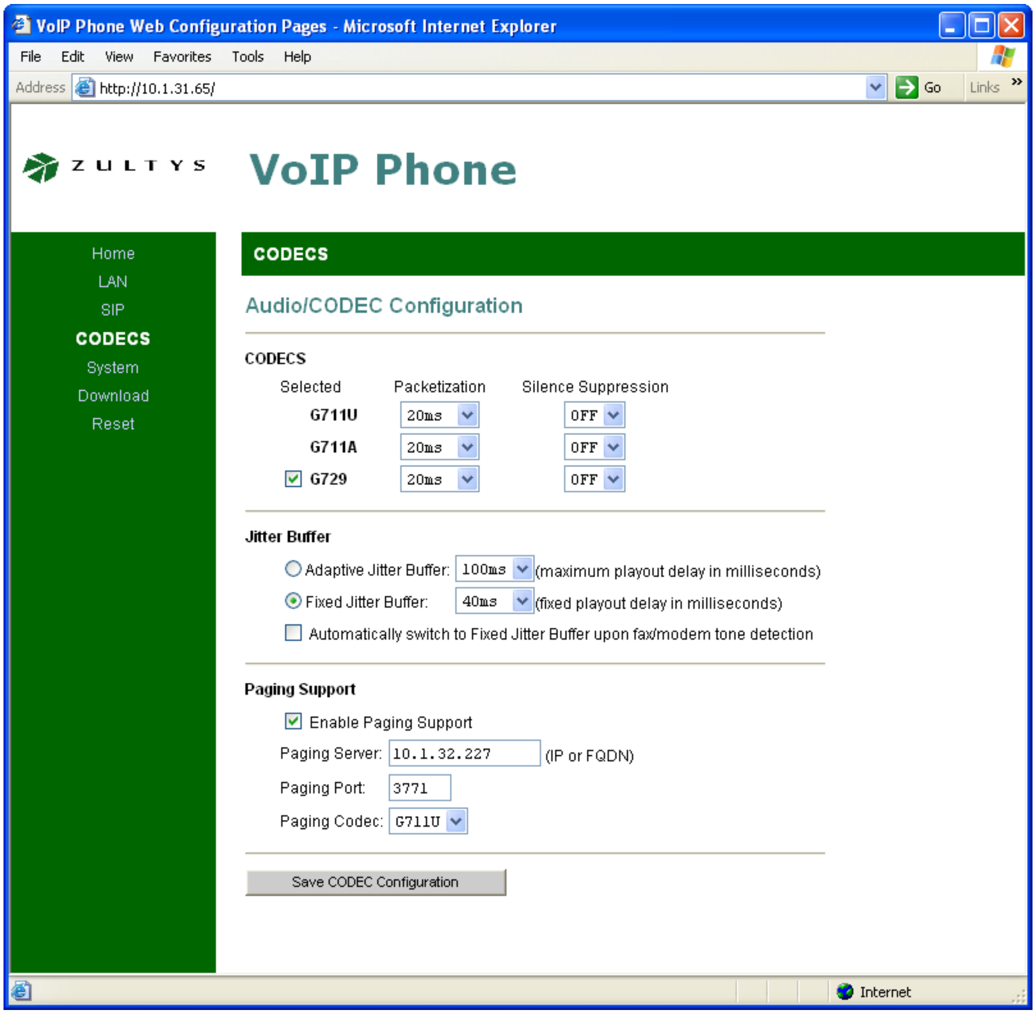

Figure A-10 Audio/CODEC Configuration panel

A.2.10.1 CODECS

These parameters configure the ZIP2 audio CODECs. The packetization and silence suppression parameters should be set to support the SIP server audio settings and capabilities.

A.2.10.2 Jitter Buffer

Jitter buffers compensate for network jitter at the receiving interface by buffering incoming packets and preparing them for playout.

A.2.10.3 Paging Support

Paging support options allow the ZIP2 to play page announcements sent from the Paging server. To enable paging on the phone, select the Enable Paging Support parameter, enter the IP address or FQDN of the paging server, the paging port, and the codec used by the paging server in the appropriate data entry boxes.

A.2.11 Security

The Set Security Password panel protects the Download and Configuration Utility from access by unauthorized users. To access this panel, as shown in figure A-11, select System from the main menu and Security from the secondary menu.

Figure A-11 Set Security Password panel

To select a new password, enter the name of the new password in both data entry boxes. Leaving both boxes blank removes password protection from the utility.

Important Press the **Save Password** button to save the configured values to the phone. Settings do not take effect until you power cycle the phone or execute a Reset command (see section A.2.16 on page 58).

A.2.12 Localization

The Localization panel configures the ZIP2 for your local time zone and allows you to specify an NTP server. To access this panel, as shown in figure A-12, select System from the main menu and Localization from the sub-menu.

Important Press the Save Localization Settings button to save the configured values to the phone. Settings do not take effect until you power cycle the phone or execute a Reset command (see section A.2.16 on page 58).

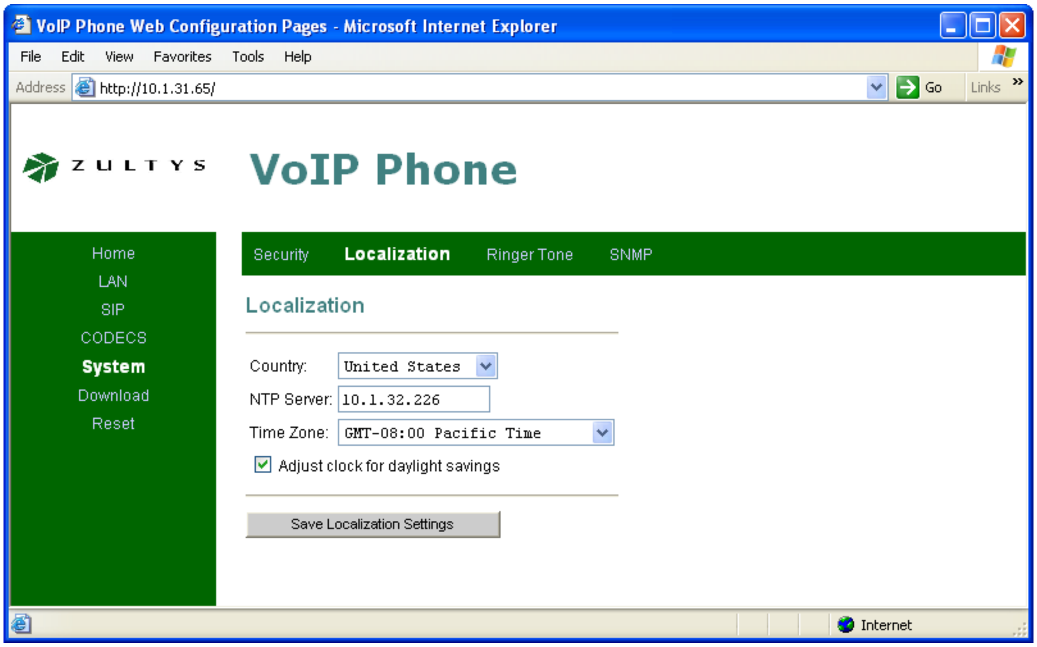

Figure A-12 Localization panel

A.2.13 Ringer Tone

The ZIP2 offers eight different ring tones to alert you of incoming calls. To access the Ring Tone Configuration panel, as shown in figure A-13, select System from the main menu and Ringer Tone from the sub-menu.

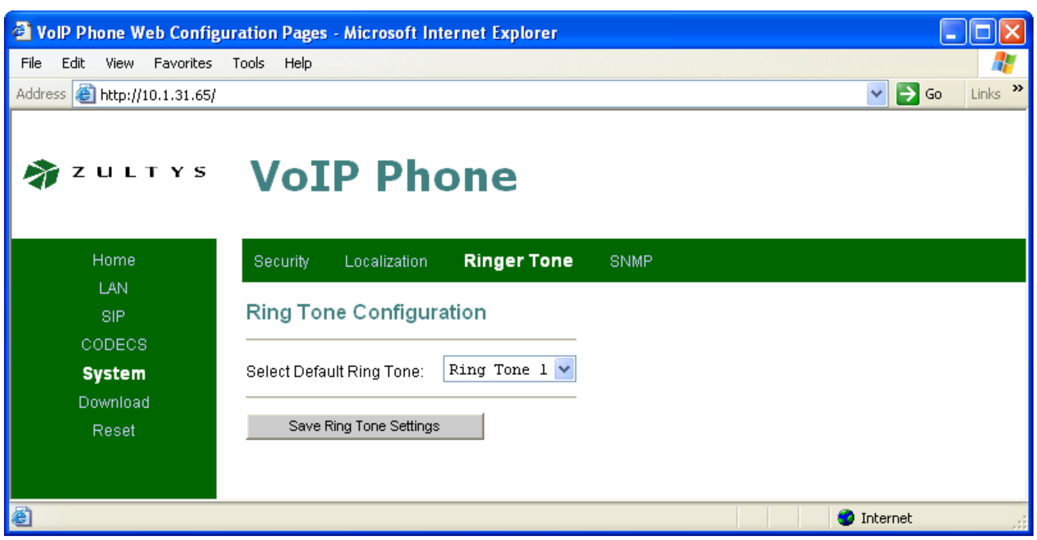

Figure A-13 Ringer Tone panel

Important Press the **Save Ring Tone Settings** button to save the configured values to the phone. Settings do not take effect until you power cycle the phone or execute a Reset command (see section A.2.16 on page 58).

A.2.14 SNMP

The Simple Network Management Protocol governs network management and the monitoring of network devices and their functions. SNMP is described formally in RFC 1157 and in a number of related RFCs.

To access this panel, as shown in figure A-14, select System in the main menu and SNMP in the sub-menu.

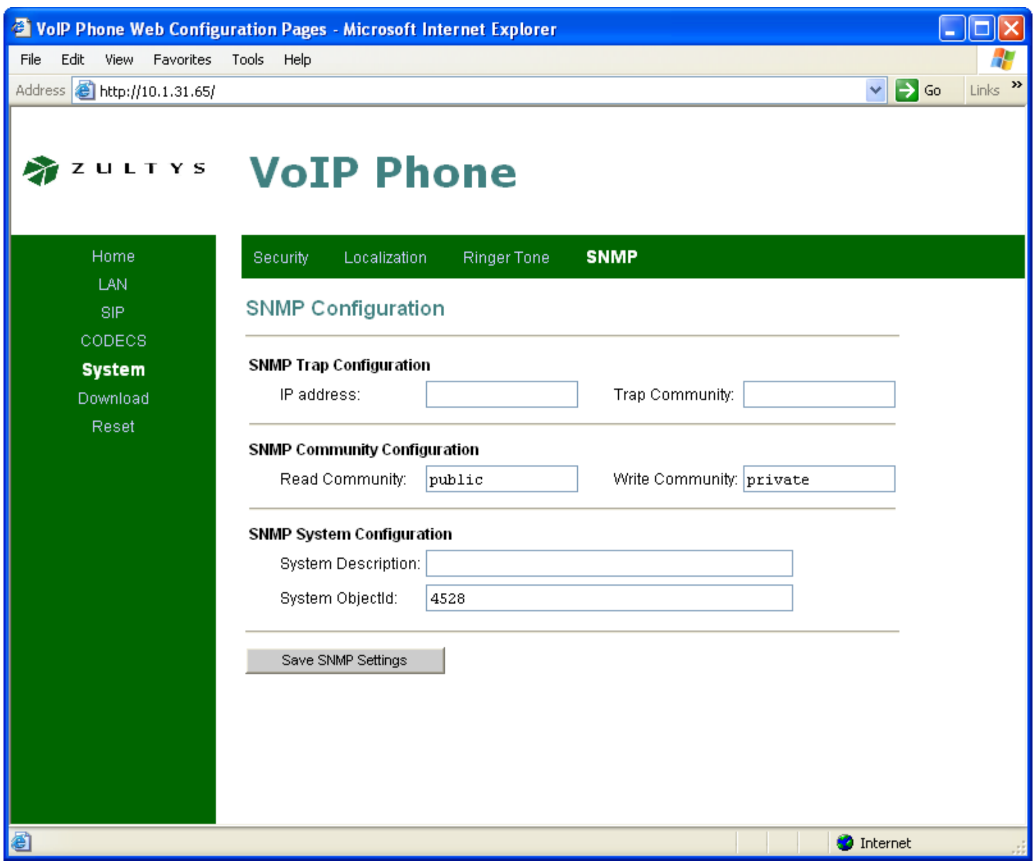

Figure A-14 SNMP Configuration panel

Important Press the **Save SNMP Settings** button to save the configured values to the phone. Settings do not take effect until you power cycle the phone or execute a Reset command (see section A.2.16 on page 58).

A.2.15 Download

The application code provides the software that operates the ZIP2 phone. To load the application code onto the phone, access the Download panel, as shown in figure A-15, by selecting Download from the main menu bar.

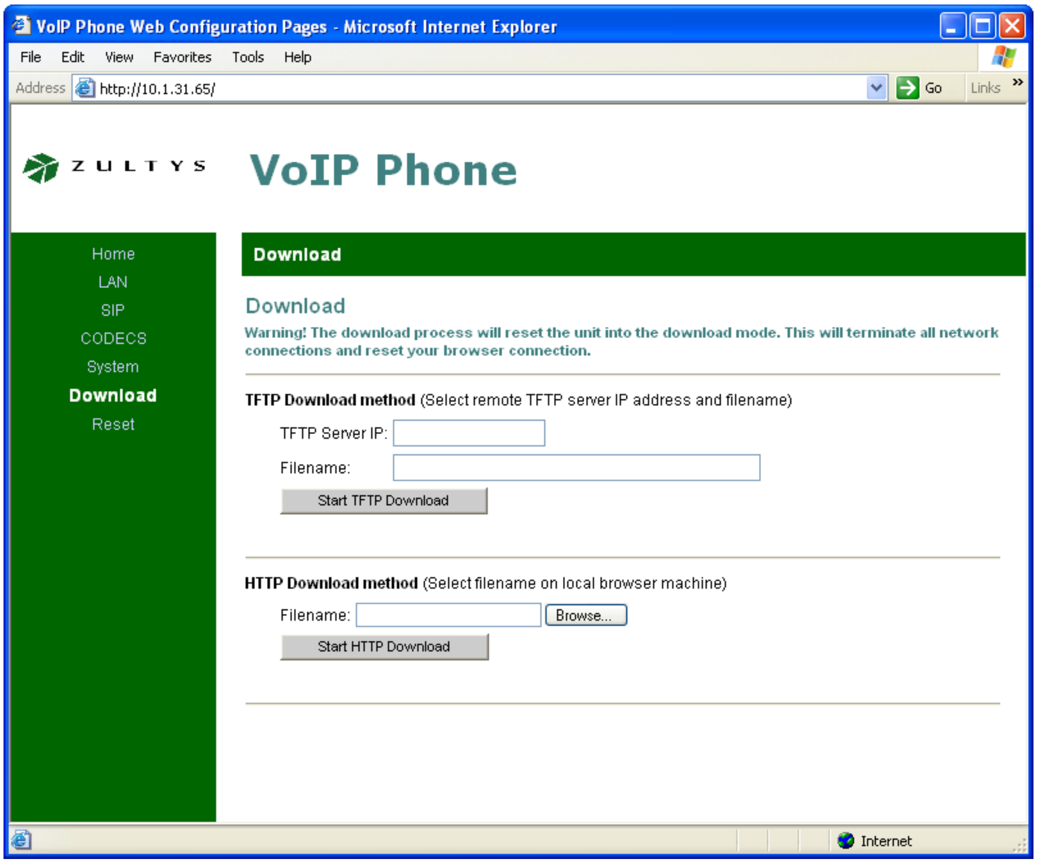

Figure A-15 Download panel

You can download the application code file from the TFTP server (using the TFTP download method) or from your local computer (using the HTTP download method). Obtain the file or the address from your system administrator, then follow the instructions on the panel.

The Home menu, as shown in section A.2.1 on page 43, displays the version number of the application code loaded on the ZIP2 phone.

A.2.16 Reset

All settings stored to the phone through the Download and Configuration utility do not take effect until you power cycle or reset the phone. To access the Download panel, as shown in figure A-16, by selecting Download from the main menu bar

- To power cycle the phone, remove power from the phone for a few seconds.
- To reset the phone, access the Reset panel, shown in figure 4, by selecting Reset from the main menu, then select *Reset and execute Main Application*, and then press the Reset button.

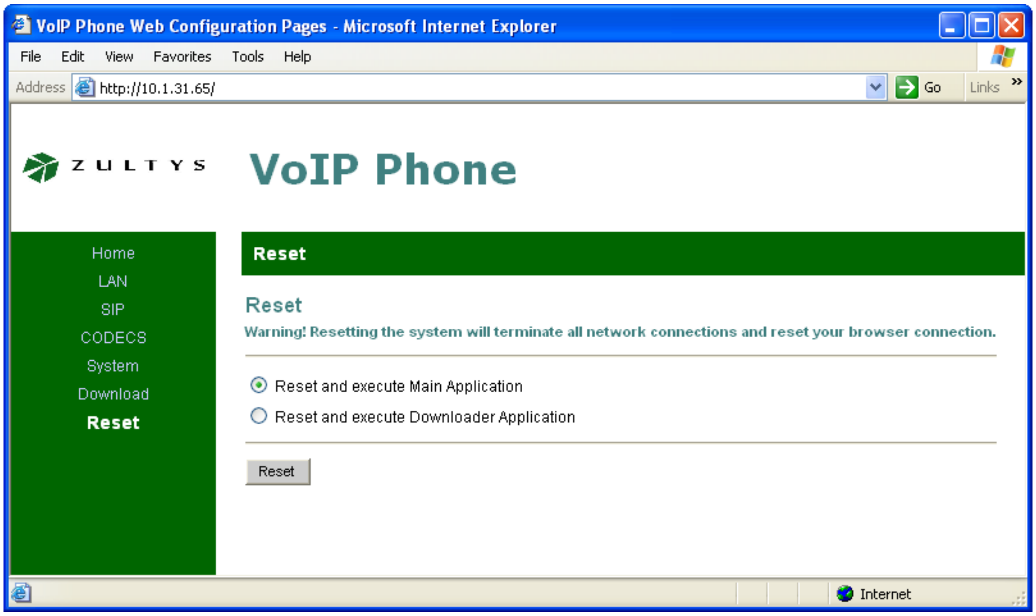

Figure A-16 Reset panel

Appendix B

Configuration Files

B.1 Introduction

The ZIP2 phone obtains its configuration from three possible sources:

- what is saved in the phone's memory
- what is saved in a common configuration file
- what is saved in a specific configuration file

This appendix describes the function, composition, and implementation of ZIP2 configuration files.

B.2 Configuration File Types

The ZIP2 phone obtains its configuration from two separate files: the common configuration file and the specific configuration file.

B.2.1 Common Configuration File

The common configuration file sets parameters on all phones within an enterprise that are to have the same values. Parameters that are normally set within a common configuration file define the network configuration, SIP server interface characteristics, and other settings that are common among all phones in an enterprise.

The common configuration file is called:

```
ZIP2_common.cfg
```

This file must be stored in the root directory of the TFTP server.

B.2.2 Specific Configuration File

The specific configuration file sets parameters for an individual phone within an enterprise. Parameters normally set within a specific configuration file customize ZIP2 features for the person using the phone, such as the extension and the device ID.

The common configuration file identifies the location of the specific configuration files. A phone extracts configuration information from the common file first, then from its specific configuration file. Parameter settings in the specific file take precedence over settings of the same parameters in the common file.

The name of the specific configuration file is:

```
<MAC address>.cfg
```

For example,

```
0001E1072A2F.cfg
```

is the specific configuration file for a ZIP2 phone that has the MAC address 00:01:E1:07:2A:2F.

B.3 Configuration File Format

Common and specific configuration files are identical in format and composition. Configuration files are stored in ASCII format. Most configuration parameters can be set in either file. Section B.4 describes the parameters used in ZIP2 configuration files.

The order of parameters within each function section does not effect the configuration of the phone. If a parameter is defined in the common file and the specific file, the specific file setting takes precedence. Figure B-1 displays an example of parameter settings in a configuration file.

```
ROMAVERSION 3.38
TFTP_CFG_DIR ./ZIP2
```

Figure B-1 Configuration File Example

Parameter names and values are case sensitive and must be separated by white space (space or tabs). The name and value of a parameter must be on the same line. Comment lines are denoted with a leading pound sign (#) and have no effect on the configuration of the phone.

B.4 Configuration Parameters

This section provides tables that list all of the configuration parameters in each functional group. Configuration parameters can be in any order, though duplicate parameters will have their values overridden by the last occurrence of that parameter in the configuration file.

Default values provided in the parameter tables are settings assigned to parameters which are not explicitly listed in the configuration file.

B.4.1 General Configuration Parameters

General parameters configure miscellaneous phone settings. Figure B-2 lists the General configuration parameters.

B.4.1.1 Mandatory Parameter Settings

Although the default values for the following parameters may be set properly for your general configuration, it is highly recommended that the configuration files explicitly define the settings for these variables.

Parameter	Description	Equivalent web interface parameter	Version Introduced
NEWPASSWD	Specifies password required to change the protected settings. Valid passwords contain four to fifteen numeric (0-9) digits. If not specified, the phone is not password protected. Default value is an empty string.	section A.2.11 on page 55	3.35
RINGERTONE	Specifies the default, canned ringer tone to use on the phone. You can select from 1-8 different tones. Default value is 1	section A.2.3 on page 46	3.35
TFTP_CFG_DIR	TFTP directory location of the specific configuration file. Parameter value is directory name that is referenced from TFTP root directory. This parameter must be set in the common configuration file. Default value is an empty string.	NA	3.35
ROMAVERSIO N	Specifies the software version that the phone must use. If the phone is running a different version, it will attempt to download the correct version from the TFTP server. Default is an empty string.	NA	3.35

Figure B-2 General Configuration Parameters

ROMAVERSION: This parameter specifies the application code version that should be running on your phone. The ZIP2 does a character by character comparison of this value with the currently loaded application code version number to determine whether it should download a new (or old) version from the TFTP server.

TFTP_CFG_DIR: This parameter points to the TFTP server directory that stores the specific configuration file for your phone. This parameter must be set in the common configuration file in order for the phone to read and process its specific configuration file.

B.4.1.2 Sample Configuration File

Figure B-3 displays the General Configuration settings section from a sample configuration file.

```
ROMAVERSION 3.38
TFTP_CFG_DIR ZIP2
```

Figure B-3 Sample Configuration File – General Configuration Settings

B.4.2 Network Configuration Parameters

Network Configuration parameters define settings required by the phone to communicate with the network. Figure B-4 lists the Network Configuration parameters.

Parameter	Description	Equivalent web interface parameter	Version Introduced
IF0DHCP	When set to' DHCP', phone uses DHCP to configure network settings: IP address, subnet mask, domain name, default gateway, DNS servers, NTP server address, and TFTP server address. Valid settings are 'DHCP' and 'FIXED': 'DHCP' - enables DHCP 'FIXED' - disables DHCP and enables Fixed IP Addressing Default value is 'DHCP'.	section 4.3.2.1 on page 16	3.35
IF0DNSDOMAINNAME	Parameter is name of the domain in which the phone resides; used for manual configuration when IF0DHCP is set to 'FIXED' or DHCP does not return the domain (DHCP option 15). Valid settings are FQDN or IP address in dotted decimal notation. Default value is an empty string.	section 4.3.2.1 on page 16	3.35
IF0DNSHOSTNAME	Parameter is name of the FQDN of the host; used for manual configuration when IF0DHCP is set to 'FIXED' or DHCP does not return the host name (DHCP option 12). Valid setting is an FQDN. Default value is an empty string.	section 4.3.2.1 on page 16	3.35
IF0IPADDRESS	Parameter is static address assigned to the ZIP2. Used for manual configuration when IF0DHCP is set to 'FIXED' or DHCP does not return an address (DHCP option 50). Valid setting is an IP address in dotted decimal notation. Default value is an empty string.	section 4.3.2.1 on page 16	3.35
IF0IPDNS	Parameter is IP address of primary DNS Server. Used for manual configuration when IF0DHCP is set to 'FIXED' or DHCP does not return DNS Server (DHCP option 6). Valid setting is an IP address in dotted decimal notation. Default value is an empty string.	section 4.3.2.1 on page 16	3.35

Figure B-4 Network Configuration Parameters

Parameter	Description	Equivalent web interface parameter	Version Introduced
IF0IPGATEWAY	Parameter is IP address of default gateway that is used for manual configuration when IF0DHCP is set to 'FIXED' or DHCP does not provide the default gateway (DHCP option 3). Valid setting is an IP address in dotted decimal notation. Default value is an empty string.	section 4.3.2.1 on page 16	3.35
IF0IPNETMASK	Parameter is Subnet mask that is used for manual configuration of the phone when IF0DHCP is set to 'FIXED' or DHCP does not return mask (DHCP option 1). Valid setting is an IP netmask in dotted decimal notation. Default value is an empty string.	section 4.3.2.1 on page 16	3.35
TFTPSERVER	IP address of TFTP server. Parameter used only if IF0DHCP is set to 'FIXED' or DHCP does not return a TFTP server (DHCP option 66). Parameter value is an IP address in dotted decimal notation. Default value is an empty string.	section A.2.3 on page 46	3.38

Figure B-4 Network Configuration Parameters (Continued)

B.4.2.1 Mandatory Parameter Settings

Although the default values for the following parameters may be set properly for your network configuration, it is highly recommended that the configuration files explicitly set the values for these parameters.

IF0DHCP: When DHCP is enabled, the DHCP server should dynamically provide an IP address and subnet mask for the phone along with IP addresses for the DNS servers, default gateway, NTP server, and TFTP server.

If DHCP is not enabled, or if the DHCP server is unable to return addresses for any of these servers, you must specify valid IP addresses for each server or the phone will not properly configure on startup.

TFTPSERVER: This parameter may be used to override the local DHCP specified Option 66 TFTP server. Many firewalls and NATs incorrectly supply this option in a non-configurable manner, which prevents remote management of phones through a VPN. Specifying this parameter overrides the locally supplied TFTP server and directs the phone to the central management TFTP server at the company headquarters. This option can be ignored on a local LAN-only deployment.

B.4.2.2 Sample Configuration File

Figure B-5 displays the Network Configuration section from a sample configuration file.

```
ROMAVERSION 3.38
TFTP_CFG_DIR ./ZIP2
IF0DHCP FIXED  # Fixed IP
IF0IPADDRESS 10.1.1.50
IF0IPNETMASK 255.255.255.0
IF0IPGATEWAY 10.1.1.1
IF0DNS 10.1.10.5
IF0DOMAINNAME foo.com
IF0DNSHOSTNAME itsme.foo.com
TFTPSERVER 10.1.10.1
```

Figure B-5 Sample Configuration File – Network Configuration Settings

B.4.3 SIP Configuration Parameters

SIP Configuration parameters allow the ZIP2 phone to operate properly in a SIP environment. Figure B-6 lists the SIP Configuration parameters.

Parameter	Description	Equivalent web interface parameter	Version Introduced
BACKUP_SERVERIP	Backup SIP server proxy address value. If the primary proxy server fails to operate, ZIP2 attempts to switch to backup proxy. If no value is entered, a backup proxy is not specified or enabled.	section 4.3.2.2 on page 17	3.35
BACKUP_SERVERPORT	Backup SIP server proxy port value. Valid settings range from 1025 to 65535. Default value is 5060.	section 4.3.2.2 on page 17	3.35
BACKUP_SERVERREGISTER	Enables backup SIP Server registration. Valid settings are YES and NO. YES–enables backup SIP server registration NO–disables backup SIP server registration. Default value is YES.	section 4.3.2.2 on page 17	3.35
DIALPLAN	If field is not empty, the ZIP2 will attempt to match input digits against the dial plan and complete the call immediately on first match. Default value is "x.T|x.*#|*x.T|*x.#", which is equivalent to the application of no dial plan.	section 4.3.2.2 on page 17	3.35
DIALPLANTIMEOUT	Specifies the interdigit timeout. After you press a digit, the phone waits for this period before dialling the call. Pressing a digit restarts the timer. This parameter is measured in centiseconds. The default value is 220 (2.2 seconds) The maximum value is 65535 (655.35 seconds).	section 4.3.2.2 on page 17	3.51
DOMAINNAME	SIP registrar domain name. When this value is set, phone attempts to register with this domain name instead of default. Default value is DHCP-derived domain name	section 4.3.2.2 on page 17	3.35

Figure B-6 SIP Configuration Parameters

Parameter	Description	Equivalent web interface parameter	Version Introduced
LINE1AEC	Turn Acoustic Echo Cancellation on handset on or off. Valid settings are YES and NO. YES – AEC is on. NO – AEC is off. Default is NO.	section 4.3.2.2 on page 17	3.35
LINE1AUTHPSWD	SIP Digest Authentication password. If field is empty, authentication is turned off. Default value is empty.	section 4.3.2.2 on page 17	3.35
LINE1AUTHUSER	Specifies the user portion of the SIP URI and consists of a string value. If left empty, phone will use its MAC address. Default value is the MAC address of the phone (upper case).	section 4.3.2.2 on page 17	3.35
LINE1CALLERID	Specifies the string that is sent in the display name part of the SIP From: header. Default value is an empty string.	section 4.3.2.2 on page 17	3.35
LINE1NUMBER	SIP userid – unused at this time. Default is empty string.	section 4.3.2.2 on page 17	3.35
LINE1PORT	Specifies the SIP port number to which the phone listens for SIP messages. Can be different than SIP Server port. Valid settings range from 1025 to 65535. Default value is 5060.	section 4.3.2.2 on page 17	3.35
SERVERIP	Specifies the IP address of the SIP proxy server that the phone will use. Parameter value is an IP address in dotted decimal notation or a FQDN. If this field is unspecified or empty, the phone performs a DNS SRV lookup to discover the local SIP Proxy server for your DNS domain. If the query is successful, the phone will use the SIP Proxy server and port number with the values returned through DNS SRV. Default value is an empty string.	section 4.3.2.2 on page 17	3.35

Figure B-6 SIP Configuration Parameters (Continued)

Parameter	Description	Equivalent web interface parameter	Version Introduced
SERVERPORT	Specifies the SIP Proxy port number to which the phone sends SIP messages. Valid settings range from 1025 to 65535. Default value is 5060.	section 4.3.2.2 on page 17	3.35
SERVERREGISTER	Enables the phone to register with the Proxy server if specified. Valid settings are YES and NO. YES – send SIP REGISTER message NO – do not send SIP REGISTER message. Default value is YES.	section 4.3.2.2 on page 17	3.35
TRANSPORT_TYPE	SIP Transport type. Valid settings are either UDP or TCP. Default value is UDP.	section 4.3.2.2 on page 17	3.35

Figure B-6 SIP Configuration Parameters (Continued)

B.4.3.1 Mandatory Fields

The SERVERIP parameter should be set in order for the ZIP2 phone to function properly. The proxy server receives SIP requests from the phone and forwards them to the next intermediate device in the network. This parameter sets the address of the proxy server for the phone. If this field is empty, the phone will perform a DNS SRV lookup to find the SIP Proxy server for this domain.

B.4.3.2 Mandatory Parameter Settings

Although the following parameters are properly set by the default values, it is highly recommended that the configuration files explicitly define the settings for these variables.

SERVERREGISTER: This parameter enables the phone to register with the proxy server. This parameter should be set to 'YES'.

B.4.3.3 Sample Configuration File

Figure B-7 displays the SIP Configuration settings section from a sample configuration file.

```
# TFTP config file template for ZIP2s

ROMAVERSION 3.38                 # Set current Application code version

################ IP addressing
## Enable DHCP
IF0DHCP DHCP                     # Enable DHCP

############## SIP Configuration Parameters
SERVERIP   192.168.1.100         # SIP Server hostname or address.
                                 # Leave blank to search automatically
                                 # via DNS SRV
SERVERPORT 5060                  # SIP Server Port number
DOMAINNAME zultys.com            # Used as host part of SIP REGISTER AOR
SERVERREGISTER YES               # Register with SIP server

# Backup server must be used in tandem with SIP Extensions retries
#BACKUP_SERVERIP  10.1.1.100     # SIP Backup Proxy
#BACKUP_SERVERPORT 5060          # SIP Backup server port number
#BACKUP_SERVERREGISTER YES       # Register with SIP backup server

DIALPLAN     x.T|x.#|*x.T|*x.#   # local dial plan

TRANSPORT_TYPE UDP               # UDP or TCP

LINE1NUMBER 1234                 # not used
LINE1CALLERID itsme              # Call ID for display purposes
LINE1PORT 5060                   # SIP Listener port on phone
LINE1AEC YES                     # Use Acoustic Echo Cancellation
LINE1AUTHUSER 0001E1072A5F       # Authorization User ID, default MAC address
LINE1AUTHPSWD 1234               # Authorization password; leave blank when
                                 # no authorization required

#SIP_REXMT_INVITE 4              # Number of INVITE retries before failover
#SIP_REXMT_REQUEST 4             # Number of non-INVITE retries before
                                 # failover
```

Figure B-7 Sample Configuration File – SIP Configuration Settings

B.4.4 SIP Extension Parameters

SIP Extension parameters allow the ZIP2 phone to leverage advanced SIP protocol features. Figure B-8 lists the SIP Extension configuration parameters.

Parameter	Description	Equivalent web interface parameter	Version Introduced
SIP_INVITE_TIMER	Send INVITE with this Timer header value. Default value is an empty string.	section A.2.6 on page 50	3.35
SIP_MESSAGE_WAITING	Backup SIP server proxy address value. If primary proxy server fails to operate, ZIP2 attempts to switch to backup proxy. Valid settings are YES and NO. YES – subscribe to SIP message waiting service NO – do not subscribe to SIP message waiting service. Default is YES.	section A.2.6 on page 50	3.35
SIP_REXMT_INVITE	Specifies the number of unsuccessful INVITE retransmissions before phone switches to the backup proxy. Valid settings range from 1 to 10. Default value is 6.	section A.2.6 on page 50	3.35
SIP_REXMT_REQUEST	Specifies number of unsuccessful retransmissions (other than INVITE) before phone switches to backup proxy. Valid settings range from 1 to 11. Default value is 10.	section A.2.6 on page 50	3.35
SIP_SEND_PRACK	Specifies whether or not to support reliable provisional response transmission (PRACK method support). Valid settings are YES and NO. YES – enable PRACK support NO – disable PRACK support Default value is NO.	section A.2.6 on page 50	3.35
SIP_SESSION_TIMER	Specifies SIP Session Timer Value in seconds. Default value is empty string.	section A.2.6 on page 50	3.35
SIP_URI_USER_PARAM	Encode SIP URI with user parameter “user=phone”. Valid settings are YES and NO. YES – add “user=phone” to SIP URI NO – do not add “user=phone” to SIP URI Default value is NO.	section A.2.6 on page 50	3.35
SIPEXPIRES	Specifies the interval, in seconds, between SIP registrations with SIP server. Default value is 1800 seconds.	section A.2.6 on page 50	3.43

Figure B-8 SIP Extension Configuration Parameters

Parameter	Description	Equivalent web interface parameter	Version Introduced
STUN_PORT	Specifies the Port number for the STUN server. Valid settings range from 1025 to 65535. Default value is 3728		3.38
STUN_SERVER	Specifies the IP address or FQDN of the STUN server for SIP/RTP NAT Traversal. Phone will automatically check for this on boot up using DNS SRV if field is empty. Default value is empty		3.38

Figure B-8 SIP Extension Configuration Parameters (Continued)

B.4.4.1 Recommended Parameter Settings

All SIP Extensions information parameters are optional. It is, however, highly recommended that the configuration files explicitly define the settings for these variables.

SIP_MESSAGE_WAITING: This parameter enables the sending of SIP subscriptions and acceptance of SIP Voice mail notifications from servers compliant with draft-ietf-sipping-mwi-01. When set to 'YES', compliant SIP servers send a notification to the phone and trigger a blinking message waiting light on receipt of voice mail.

SIP_REXMT_INVITE and SIP_REXMT_REQUEST: In concert with the BACKUP_SERVERIP parameter, these parameters can be used to guarantee dial tone on loss of connectivity to a remote central SIP Server. Setting a low number of retransmits for SIP requests transparently fails over to the local SIP Backup gateway in the event the WAN or other connection to the main server goes down.

STUN_SERVER: STUN permits users behind firewalls and NATs to tunnel SIP and RTP traffic transparently.

SIPEXPIRES: This parameter specifies the waiting period before re-registering the device. The recommended interval is 1800 seconds.

B.4.4.2 Sample Configuration File

Figure B-9 displays the SIP Extension settings section from a sample configuration file.

```
# TFTP config file template for ZIP 2's

ROMAVERSION 3.38                  # Set current Application code version

################ IP addressing
## Enable DHCP
IF0DHCP DHCP                      # Enable DHCP

############## SIP Configuration Parameters
SERVERIP   192.168.1.100          # SIP Server host name or address.
                                  # Leave blank to search automatically
                                  # via DNS SRV
SERVERPORT 5060                   # SIP Server Port number
DOMAINNAME zultys.com             # Used as host part of SIP REGISTER AOR
SERVERREGISTER YES                # Register with SIP server

# Backup server must be used in tandem with SIP Extensions retries
BACKUP_SERVERIP  10.1.1.100       # SIP Backup Proxy
BACKUP_SERVERPORT 5060            # SIP Backup server port number
BACKUP_SERVERREGISTER YES         # Register with SIP backup server

SIP_MESSAGE_WAITING YES           # enable MWI
SIP_SEND_PRACK NO                 # support PRACK method/Early media
SIP_URI_USER_PARAM NO             # encode SIP URI with user=phone
SIP_INVITE_TIMER 1800             # send INVITE with this timer header value
SIP_SESSION_TIMER 1800            # Time between session timer refreshes

SIP_REXMT_INVITE 4                # Number of INVITE retries before failover
SIP_REXMT_REQUEST 4               # Number of non-INVITE retries before
                                  # failover
SIPEXPIRES 1800                   # Set SIP registration interval in seconds.
```

Figure B-9 Sample Configuration File – SIP Extension Settings

B.4.5 VLAN Configuration

VLAN parameters can configure the ZIP2 to match the settings in your network. Figure B-10 lists the VLAN configuration parameters.

Parameter	Description	Equivalent web interface parameter	Version Introduced
IF0PRIORITYTAG	Defines the priority tag for all outgoing packets. Valid settings range from 0 to 7. Default value is an empty string	section A.2.4 on page 47	3.35
IF0VLANTAG	Defines the VLAN tag for all outgoing packets. Valid settings range from 0 to 7. Default value is an empty string.	section A.2.4 on page 47	3.35
PRIORITYTAG_CALL	Defines the priority tag for all outgoing SIP signaling packets (overrides the IF0PRIORITYTAG for signaling packets only). Valid settings range from 0 to 7. Default value is an empty string.	section A.2.9 on page 53	3.35
PRIORITYTAG_RTP	Defines the priority tag for all outgoing RTP packets (overrides the IF0PRIORITYTAG for RTP packets only). Valid settings range from 0 to 7. Default value is an empty string.	section A.2.9 on page 53	3.35
VLANTAG_CALL	Defines the VLAN tag for all outgoing SIP signaling packets (overrides the IF0VLANTAG setting for signaling packets only). Valid settings range from 0 to 4094. Default value is an empty string.	section A.2.9 on page 53	3.35
VLANTAG_RTP	Defines the VLAN tag for all outgoing RTP packets (overrides the IF0VLANTAG setting for RTP packets only). Valid settings range from 0 to 4094. Default value is an empty string.	section A.2.9 on page 53	3.35

Figure B-10 VLAN Configuration Parameters

B.4.5.1 Programming Restrictions

You can configure the ZIP2 with up to three independent VLAN settings for SIP signalling packets, RTP packets, and all packets.

Setting ANY of these parameters will enable VLANs for the phone and insert an IEEE 802.1q header into all Ethernet frames. The result is the phone will not be able to communicate with any remote device that does not belong to the same VLAN specified in the IEEE 802.1q frame, especially if it's an arbitrary number (like 0).

Do not set any of these parameters unless you have explicit values to assign.

B.4.5.2 Sample Configuration File

Figure B-11 displays the VLAN settings section from a sample configuration file.

```
# TFTP config file template for ZIP 2's

ROMAVERSION 3.38                 # Set current Application code version


############## SIP Configuration Parameters
SERVERIP   192.168.1.100         # SIP Server host name or address.
                                 # Leave blank to search automatically
                                 # via DNS SRV
SERVERPORT 5060                  # SIP Server Port number
DOMAINNAME zultys.com            # Used as host part of SIP REGISTER AOR
SERVERREGISTER YES               # Register with SIP server
VLANTAG_CALL 1
PRIORITYTAG_CALL 1
VLANTAG_RTP 2
PRIORITYTAG_RTP 2
IF0VLANTAG 3
IF0PRIORITYTAG 3
```

Figure B-11 Sample Configuration File – VLAN Settings

B.4.6 Audio Configuration

Audio parameters configure codecs and paging extensions. Figure B-12 lists the Audio configuration parameters.

Parameter	Description	Equivalent web interface parameter	Version Introduced
AJB_MAXDELAY	Specifies the Maximum playout delay for the Adaptive jitter buffer. Default value is 100 (milliseconds).	section A.2.10.2 on page 54	3.35
AUTO_JB_SWITCH	Specifies whether or not to automatically switch from an adaptive jitter buffer to a fixed jitter buffer upon fax/modem tone detection. Default value is NO.	section A.2.10.2 on page 54	3.35
ENABLE_PAGING_SWITCH	Enables Paging capabilities: YES – Enable NO – Disable Default Value is YES.	section A.2.10.3 on page 54	3.35
FJB_DELAY	Specifies the playout delay on the Fixed jitter buffer. Default value is 40 (milliseconds)	section A.2.10.2 on page 54	3.35
G711AON	Specifies whether to offer G.711 A-Law as the speech encoding method: Default value is YES.	section A.2.10.1 on page 54	3.35
G711APACK	Specifies the packetization interval for G.711 A-Law packets, where 20 indicates each RTP packet will carry 20 milliseconds worth of audio input. Supported intervals are 10, 20, 30, 40, 50, 60, 70, 80, 90 and 100 msecs, respectively. Default value is 20 (msecs).	section A.2.10.1 on page 54	3.35
G711ASS	Specifies whether to enable Silence Suppression for G.711 A-Law packets. Default value is NO.	section A.2.10.1 on page 54	3.35
G711UON	Specifies whether to offer G.711 u-Law as the speech encoding method: Default value is YES.	section A.2.10.1 on page 54	3.35
G711UPACK	Specifies the packetization interval for G.711 u-Law packets, where 20 indicates each RTP packet will carry 20 milliseconds worth of audio input. Supported intervals are 10, 20, 30, 40, 50, 60, 70, 80, 90 and 100 msecs, respectively. Default value is 20 (msecs).	section A.2.10.1 on page 54	3.35

Figure B-12 Audio Configuration Parameters

Parameter	Description	Equivalent web interface parameter	Version Introduced
G711USS	Specifies whether to enable Silence Suppression for G.711 u-Law packets. Default value is NO.	section A.2.10.1 on page 54	3.35
G729ASS	Specifies whether to enable Silence Suppression for G.729 packets. Default value is NO.	section A.2.10.1 on page 54	3.35
G729ON	Specifies whether to offer G.729 as the speech encoding method. Default value is YES.	section A.2.10.1 on page 54	3.35
G729PACK	Specifies the packetization interval for G.729 packets, where 20 indicates each RTP packet will carry 20 milliseconds worth of audio input. Supported intervals are 10, 20, 30, 40, 50, 60, 70, 80, 90 and 100 msecs, respectively. Default value is 20 (msecs).	section A.2.10.1 on page 54	3.35
JB_TYPE	Specifies the type of jitter buffer that the device uses: ADAPTIVE: Uses adaptive jitter buffer FIXED: Uses fixed jitter buffer Default value is ADAPTIVE	section A.2.10 on page 53	3.38
PAGINGCODEC	Specifies the Paging stream Codec the phone will accept: G711U – G.711 U-law G711A – G.711 a-law G.729 – G.729 Default value is G.711U	section A.2.10.3 on page 54	3.38
PAGINGPORT	Specifies the Paging Port number the phone will listen on for Paging stream. Valid settings range from 1025 to 65535 Default value is 3771	section A.2.10.3 on page 54	3.38
PAGINGSERVER	Specifies the IP Address of the Paging server. Phones will only accept Paging stream from the server specified here. This requires a paging server the follows the proprietary Zultys paging specification. Default value is an empty string.	section A.2.10.3 on page 54	3.38

Figure B-12 Audio Configuration Parameters (Continued)

All Audio parameters are optional. Figure B-13 displays the Audio settings section from a sample configuration file.

```
# TFTP config file template for ZIP 2's

ROMAVERSION 3.38                 # Set current Application code version


############## SIP Configuration Parameters
SERVERIP   192.168.1.100         # SIP Server host name or address.
                                 # Leave blank to search automatically
                                 # via DNS SRV
SERVERPORT 5060                  # SIP Server Port number
DOMAINNAME zultys.com            # Used as host part of SIP REGISTER AOR
SERVERREGISTER YES               # Register with SIP server
G711UON YES
G711UPACK 20
G711USS NO
G711AON YES
G711APACK 20
G711ASS NO
G729ON YES
G729PACK 20
G729SS NO
AJB_MAXDELAY 100
FJB_DELAY 40
AUTO_JB_SWITCH NO
ENABLE_PAGING_SWITCH YES
PAGINGSERVER 10.1.10.5
PAGINGPORT 3771
PAGINGCODEC G711U
```

Figure B-13 Sample Configuration File – Audio Settings

B.4.7 Out-of-band Signalling Information

Out-of-band Signalling parameters configure the phones' mode of signalling DTMF digits to a remote endpoint. Figure B-14 lists the Out-of-band Signalling configuration parameters.

Parameter	Description	Equivalent web interface parameter	Version Introduced
ABCDMODE	Valid parameter options are: TRANSITION - ABCD signalling is sent only at ABCD event changes (such as a hook status change). CONTINUOUS - ABCD signalling is sent continuously for the duration of a state (such as on hook), and is not sent for the inverse of the state (off hook). Default value is TRANSITION.	section A.2.7 on page 51	3.35
DROPVOICE	Specifies whether or not to drop voice packets during transmission of RFC 2833 telephone event RTP packets. YES – drop voice packets when sending RFC 2833 packets. NO – voice and RFC 2833 packets are mixed Default value is YES	section A.2.7 on page 51	3.35

Figure B-14 Out of Band Signalling Parameters

Parameter	Description	Equivalent web interface parameter	Version Introduced
OOBTELEVENTS	Specifies the OOB signalling method. The ZIP2 supports the following DTMF signalling methods: OOB_NONE – in-band DTMF signalling in the raw audio stream OOB_RFC2833 – out-of-band DTMF signalling using encoded digits per RFC 2833 OOB_SIP_INFO – out-of-band DTMF signalling using digits passed in SIP INFO packets per RFC 2976. Default value is OOB_RFC2833.	section A.2.7 on page 51	3.35
SQUELCHDTMF	Specify whether or not to squelch DTMF audio (avoid transmitting DTMF tones inband in the audio stream). YES – Suppress sending inband DTMF audio NO – Do not suppress sending inband DTMF audio Default value is NO.	section A.2.7 on page 51	3.35
TELEVENTPAYLOAD	Specifies the dynamic payload type "codec" designator to use for DTMF signalling. This is an arbitrary number between 96-128 per RFC 1890-RTP payload spec. Can cause interoperability problems with vendors that require it to exactly match their dynamic payload number. Default value is 101.	section A.2.7 on page 51	3.35

Figure B-14 Out of Band Signalling Parameters (Continued)

All Out-of-band signalling parameters are optional. Figure B-15 displays the Out-of-band signalling settings section from a sample configuration file.

```
# TFTP config file template for ZIP 2's

ROMAVERSION 3.38                 # Set current Application code version


############## SIP Configuration Parameters
SERVERIP   192.168.1.100         # SIP Server host name or address.
                                 # Leave blank to search automatically
                                 # via DNS SRV
SERVERPORT 5060                  # SIP Server Port number
DOMAINNAME zultys.com            # Used as host part of SIP REGISTER AOR
SERVERREGISTER YES               # Register with SIP server
OOBTELEVENT OOB_RFC2833
TELEVENTPAYLOAD 101
DROPVOICE YES
SQUELCHDTMF NO
ABCDMODE TRANSITION
```

Figure B-15 Sample Configuration File – Out-of-band Signalling Settings

B.4.8 Localization Parameters

Localization parameters configure date and time settings. Figure B-16 lists the Localization Information configuration parameters.

Parameter	Description	Equivalent web interface parameter	Version Introduced
DST	Specifies whether Daylight Savings Time applies. YES – Daylight Savings Time offsets are enabled. NO – Daylight Savings Time offsets are disabled. Default value is YES.	section A.2.12 on page 55	3.35
NTPSERVERIP	Parameter is IP address of NTP server. Used for manual configuration when IF0DHCP is set to 'FIXED' or DHCP does not return NTP server (DHCP option 42). Parameter value is an IP address in dotted decimal notation. Default value is an empty string.	section A.2.12 on page 55	3.35
TIMEZONE	Specifies the time zone location of the phone. Parameter value is the offset from GMT in minutes; valid settings range from -720 to +720. Default value is -480, which denote Pacific Standard Time.	section A.2.12 on page 55	3.35

Figure B-16 Localization Parameters

All Localization Information parameters are optional. Figure B-17 displays the Localization Information settings section from a sample configuration file.

```
# TFTP config file template for ZIP 2's

ROMAVERSION 3.38                 # Set current Application code version


############## SIP Configuration Parameters
SERVERIP   192.168.1.100         # SIP Server host name or address.
                                 # Leave blank to search automatically
                                 # via DNS SRV
SERVERPORT 5060                  # SIP Server Port number
DOMAINNAME zultys.com            # Used as host part of SIP REGISTER AOR
SERVERREGISTER YES               # Register with SIP server
NTPSERVER 10.1.2.1
TIMEZONE -480 # Offset from GMT in minutes
DST YES
```

Figure B-17 Sample Configuration File – Localization Settings

B.4.9 IP QoS Parameters

IP/QoS parameters configure IP QoS phone settings. Figure B-18 lists the IP QoS configuration parameters.

Parameter	Description	Equivalent web interface parameter	Version Introduced
CALLSIG_TOS	Configures DiffServ (layer 3 QoS) setting for SIP. All SIP signalling packets leaving the phone will have the ToS byte in the IP header set to this value. Valid settings range from 0 to FF, hex format. Default value is 04.	section A.2.8 on page 52	3.35
RTP_TOS	Configures DiffServ (layer 3 QoS) setting for RTP. All voice packets (RTP) leaving the phone will have the ToS byte in the IP header set to this value. Valid settings range from 0 to FF, hex format. Default value is 10.	section A.2.8 on page 52	3.35

Figure B-18 IP QoS Information Parameters

All IP QoS parameters are optional. Figure B-19 displays the IP QoS settings section from a sample configuration file.

```
# TFTP config file template for ZIP 2's

ROMAVERSION 3.38                 # Set current Application code version


############## SIP Configuration Parameters
SERVERIP   192.168.1.100         # SIP Server host name or address.
                                 # Leave blank to search automatically
                                 # via DNS SRV
SERVERPORT 5060                  # SIP Server Port number
DOMAINNAME zultys.com            # Used as host part of SIP REGISTER AOR
SERVERREGISTER YES               # Register with SIP server
CALLSIG_TOS 04
RTP_TOS 10
```

Figure B-19 Sample Configuration File – IP QoS Settings

B.4.10 SNMP Parameters

SNMP parameters configure SNMP phone settings. Figure B-20 lists the SNMP configuration parameters.

Parameter	Description	Equivalent web interface parameter	Version Introduced
SNMPREADCOMMUNITY	Specifies the SNMP read community access string. Default value is the string ‘public’.	section A.2.14 on page 57	3.35
SNMPSYSDESC	Specifies the SNMP system descriptor for the phone Default is an empty string.	section A.2.14 on page 57	3.35
SNMPSYSOBJECTID	Specifies the SNMP base Object ID. Default value is 4528	section A.2.14 on page 57	3.35
SNMPWRITECOMMUNITY	Specifies the SNMP write community access string. Default value is the string ‘private’.	section A.2.14 on page 57	3.35
TRAPHOSTCOMMUNITY	Specifies the SNMP Trap community string. Default value is an empty string.	section A.2.14 on page 57	3.35
TRAPHOSTIPADDRESS	Parameter is IP address of SNMP TRAP server. Parameter value is 32 bit IP address in dotted decimal notation. Default value is an empty string.	section A.2.14 on page 57	3.35

Figure B-20 SNMP Parameters

All SNMP parameters are optional. Figure B-21 displays the SNMP settings section from a sample configuration file.

```
# TFTP config file template for ZIP 2's

ROMAVERSION 3.38                 # Set current Application code version


############## SIP Configuration Parameters
SERVERIP   192.168.1.100         # SIP Server host name or address.
                                 # Leave blank to search automatically
                                 # via DNS SRV
SERVERPORT 5060                  # SIP Server Port number
DOMAINNAME zultys.com            # Used as host part of SIP REGISTER AOR
SERVERREGISTER YES               # Register with SIP server
SNMPREADCOMMUNITY public
SNMPWRITECOMMUNITY private
SNMPSYSDESC “ZIP2 phone”
SNMPOBJECTID 4528
TRAPHOSTCOMMUNITY opensezme
TRAPHOSTIPADDRESS 10.1.10.5
clear_settings=2
```

Figure B-21 Sample Configuration File – SNMP Settings

Appendix C

Dial Plan

C.1 Introduction

The ZIP2 phone allows provisioning of the dial plan through the web interface. A dial plan gives the ZIP2 a map to determine when a complete number has been entered and should be passed to the server for resolution into an IP address.

C.2 Syntax

The first step in applying a dialling rule is comparing the pattern to the dialled number. Each character within a dialled number must be one of the following: the characters 0 to 9, *, #, X, x, –, [,], and the period (.). These characters are interpreted by the pattern as follows:

- <Digit> “0” | “1” | “2” | “3” | “4” | “5” | “6” | “7” | “8” | “9”

  A digit is a character between zero and nine.

- <Timer> “T” | “t”

  The timer is activated after a two second delay if all other elements of the dial plan are satisfied. For instance, a dial plan of (xxxT|xxxxx) will immediately match 5 digits and will also match an entry of 3 digits after a four second delay.

- <Letter> <digit> | <timer> | “*” | “#”

  A letter is either a digit, the timer, a * sign, or a # sign)

- <Letters> <Letter><Letter>...<Letter> | <Digit> “-” <Digit>

  <Letter><Letter>...<Letter> refers to a string of letters, such as 359 or 158T.

  <Digit> “-” <Digit> refers to a range of digits, such as 3-6.

- <Range> “X” | “x” | “[”<Letters>”]”

  X and x matches any digit - a dial plan of “xxx” matches any three digit combination.

  [<Letters>] matches any letter that specified in the brackets.

- <Position> <Letter> | <Range>
- <StringElement> <Position> | <Position> “.”

  Position matches any occurrence of a position character

  Position. matches an arbitrary number of position occurrences, including none.

- <String> <StringElement> | <StringElement><String>
- <StringList> <String> | <String> "|" <StringList>
- <DialPlan> <String> | "("<StringList>")"

A dial plan, according to this syntax, is defined either by a string or by a list of strings. Regardless of the syntax, a timer (T) is allowed only if it appears in the last position of a string; 34T9 is not valid. Each string is an alternate numbering scheme.

The ZIP2 processes the dial plan by comparing the current dial string against the dial plan. If the result is under-qualified (partial matches at least one entry), then the phone will do nothing further. If the result matches or is over-qualified (no further digits could possibly produce a match), the phone sends the string to the server and clears the dial string.

C.3 Sample Dial Plans

The following sections provide examples of valid ZIP2 dial plans.

C.3.1 Simple Dial Plans

0T|xxxxxxx	Matches any seven digit string (5559876) or an operator on 0.
[2-4,8]xxx	Matches any four digit string that begins with a 2, 3, 4, or 8.
800xxxxxxx	Matches any ten digit string that begins with 800.
x.#	Matches any string of digits the precedes the # sign.

C.3.2 Non-dialled Dial Plan

x.	The ZIP2 contacts the server as soon as the handset is lifted. Dial plan matches 0 or more digits

C.3.3 Complex Dial Plan

(0T|00T|[3-5]xxx|8xxxxxxx|*xx|91xxxxxxxxxx|9011.T)

This dial plan contacts the local operator on 0, long distance operator on 00, four digit local extension that starts with 3, 4, or 5, seven digit local numbers that are prefixed by an 8, two digit star services (such as *69), ten digit long distance numbers prefixed by 91, and international numbers starting with 9011that contain a variable number of digits.

Appendix D

Acronyms

ACD	automatic call distributor
AEC	acoustic echo cancellation
CAS	channel associated signalling
CoS	class of service
DHCP	dynamic host configuration protocol
DND	do not disturb
DNS	domain name service
DTMF	dual tone multi-frequency
FQDN	fully qualified domain name
GMT	Greenwich Mean Time
HTTP	Hypertext Transfer Protocol
ICMP	Internet control message protocol
IEEE	Institute of Electrical and Electronic Engineers
IP	Internet protocol
LAN	local area network
MAC	media access control
MDI	media dependent interface
NTP	network time protocol
PBX	private branch exchange
PCM	pulse code modulation
PHB	per hop behavior
PIN	personal information number
PSTN	public switched telephone network
QoS	quality of service
RTCP	real time transport protocol control protocol
RTP	real time transport protocol
SIP	session initiation protocol
SNMP	simple network management protocol
SNTP	simple network time protocol
TCP	transmission control protocol
TFTP	thin file transfer protocol

UDP	user datagram protocol
URI	uniform resource identifier
URL	universal reference locator
VLAN	virtual local area network

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