

## VoIP Telephones HDE Setup & Configuration



P6995 HDE-11-VoIP



P6996 HDE-1100-VoIP



P6997 HDE-12-VoIP



P6998 HDE-1200-VoIP

**VoIP Setup & Configuration P007451 Rev. D (applicable to firmware V3.0.0 and higher)****COPYRIGHT NOTICE:**

© 2017, Guardian Telecom Inc., ALL RIGHTS RESERVED.

This manual and related material is the copyrighted property of Guardian Telecom Inc. No part of this manual or related materials may be reproduced or transmitted, in any form or by any means (except for internal use by licensed customers), without prior express written permission of Guardian Telecom Inc.. This manual, and the products, software, firmware, and/or hardware described in this manual are the property of Guardian Telecom Inc. provided under the terms of an agreement between Guardian Telecom Inc. and the recipient of this manual, and their use is subject to that agreement and its terms.

**DISCLAIMER:** Except as expressly and specifically stated in a written agreement executed by Guardian Telecom Inc., Guardian Telecom Inc. makes no representation or warranty, express or implied, including any warranty or merchantability or fitness for any purpose, with respect to this manual or the products, software, firmware, and/or hardware described herein, and Guardian Telecom Inc. assumes no liability for damages or claims resulting from any use of this manual or such products, software, firmware, and/or hardware. Guardian Telecom Inc. reserves the right to make changes, without notice, to this manual and to any such product, software, firmware, and/or hardware.

**OPEN SOURCE STATEMENT:** Certain software components included in Guardian products are subject to the GNU General Public License (GPL) and Lesser GNU General Public License (LGPL) "open source" or "free software" licenses. Some of this Open Source Software may be owned by third parties. Open Source Software is not subject to the terms and conditions of the Guardian COPYRIGHT NOTICE or software licenses. Your right to copy, modify, and distribute any Open Source Software is determined by the terms of the GPL, LGPL, or third party, according to who licenses that software.

Software or firmware provided by Guardian that is unrelated to Open Source Software is copyrighted by Guardian, subject to the terms of Guardian licenses, and may not be copied, modified, reverse-engineered, or otherwise altered without explicit written permission from Guardian Telecom Inc.

**TRADEMARK NOTICE:** Guardian Telecom Inc. and the Guardian Telecom Inc. logos are trademarks of Guardian Telecom Inc. Other product names, trademarks, and service marks may be the trademarks or registered trademarks of their respective owners.



**Toll-free 1-800-363-8010**  
**Phone (403) 258-3100**  
**Fax. (403) 253-4967**  
**[www.guardiantelecom.com](http://www.guardiantelecom.com)**  
**E-mail: [sales@guardiantelecom.com](mailto:sales@guardiantelecom.com)**

## **IMPORTANT Installation Step**

**It is important to register this VoIP product with Guardian Telecom to ensure it has the most current version of software and to receive notification of software updates.**

### **Registering Your VoIP Product**

**To register your VoIP product send an email to [info@guardiantelecom.com](mailto:info@guardiantelecom.com). Be sure to include “Guardian VoIP Registration” in the subject field of your email.**

**Include the following information:**

- Company Name of End User (Req'd)**
- Address of End User (Optional)**
- Device Model (Req'd)**
- Serial Number - Found on the Exterior Label (Req'd)**
- Date of purchase (Req'd)**
- Name of Supplier (Req'd)**
- Prime Contact name and e-mail (Req'd)**
- Secondary Contact name and e-mail (Optional)**
- Phone Info: (Optional)**

**It is very important that we receive e-mail contact information of the person responsible for maintaining the installed Guardian Equipment, in order to ensure optimum performance of the device.**

**Contact information will remain confidential and will not be used for third-party marketing purposes.**

## Important Safety Instructions

1. Read these instructions.
2. Keep these instructions.
3. Heed all warnings.
4. Follow all instructions.
5. Install in accordance with the manufacturer's instructions.
6. Do not install near any heat sources such as radiators, heat registers, stoves, or other apparatus (including amplifiers) that produce heat.
7. Only use attachments/accessories specified by the manufacturer.
8. Refer all servicing to qualified service personnel.
9. Prior to installation, consult local building and electrical code requirements.



### Warning

*Electrical Hazard:* This product should be installed by a licensed electrician according to all local electrical and building codes.



### Warning

*Electrical Hazard:* To prevent injury, this apparatus must be securely attached to the floor/wall in accordance with the installation instructions.

## Table of Contents

1.	Typical System Installation .....	7
2.	Supported Protocols .....	7
3.	Supported SIP Servers.....	7
4.	Features .....	8
5.	Getting Started .....	9
6.	RESET Switch .....	9
7.	Configure the Telephone Parameters.....	9
7.1.	Telephone Web Page Navigation.....	10
7.2.	Log in to the Configuration Home Page .....	11
7.3.	Configure the Device Parameters .....	13
7.4.	Configure the Network Parameters .....	16
7.5.	Configure the SIP Parameters .....	18
7.6.	Configure the Audio Parameters .....	22
7.6.1.	User-created Audio Files .....	24
7.7.	Configure the Event Parameters .....	26
7.8.	Configure the Autoprovisioning Parameters.....	30
7.9.	Configure Update Firmware .....	32
7.9.1.	Reboot the Telephone .....	32
8.	Setting up a TFTP Server .....	33
8.1.	In a LINUX Environment .....	33
8.2.	In a Windows Environment .....	33
9.	Operation.....	34
10.	Frequently Asked Questions.....	35
11.	Product Specifications .....	37
12.	Appendix A Time Zone Settings .....	38

## Figures

Figure 1 - Typical Installation .....	7
Figure 2 - Startup Screen.....	9
Figure 3 - Home Page.....	11
Figure 4 - Device Configuration Page .....	13
Figure 5 - Network Configuration Page .....	16
Figure 6 - SIP Configuration Page .....	18
Figure 7 - Audio Configuration Page .....	22
Figure 8 - Audacity 1 .....	24
Figure 9 - Audacity 2 .....	24
Figure 10 - WAV (Microsoft) signed 16 bit PCM.....	25
Figure 11 - Event Configuration Page .....	26
Figure 12 - Autoprovisioning Configuration Page.....	30
Figure 13 - Update Firmware Page .....	32

## Tables

Table 1 - Factory Default Settings .....	9
Table 2 - Telephone Web Page Navigation.....	10
Table 3 - Home Page Overview .....	12
Table 4 - Device Configuration Parameters .....	15
Table 5 - Network Configuration Parameters .....	17
Table 6 - SIP Configuration Parameters.....	21
Table 7 - Audio Configuration Parameters .....	23
Table 8 - Event Configuration.....	29
Table 9 - Autoprovisioning Configuration Parameters.....	31
Table 10 - Firmware Update Parameters .....	32

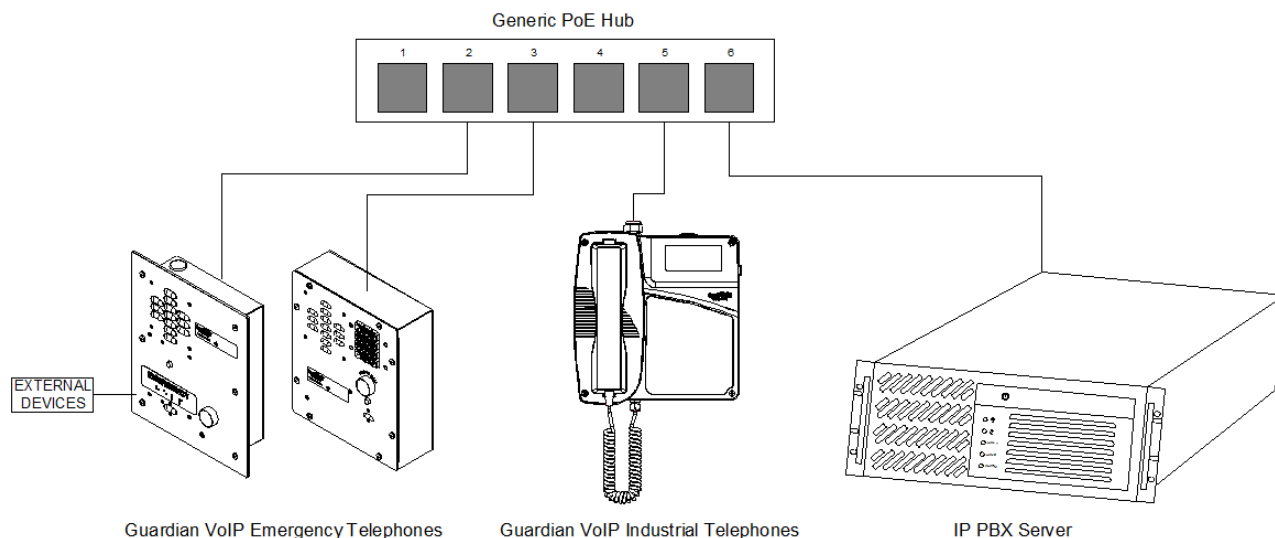
## Acronyms

DHCP Server	Dynamic Host Configuration Protocol
DHCPD	Dynamic Host Configuration Protocol Daemon
DTMF	Dual-tone Multi-frequency
HTTP	Hypertext Transfer Protocol
IP Address	Internet Protocol Address
LAN	Local Area Network
LINUX	Unix-like computer operating system
PBX	Private Branch Exchange
PC	Personal Computer
PCM	Pulse-Code Modulation
PCMA	Paired Carrier Multiple Access
PCMU	Pulse Code Modulation mu-law
PoE	Power over Ethernet
POST	Power On Self Test
RIFF	A short, repeated musical phrase
RTP	Real-time Transport Protocol
RTP Port	Real-time Transport Protocol port
AVP	Audio Video Profile
SIP	Session Initiation Protocol
TFTP	Trivial File Transfer Protocol
URL	Uniform Resource Locator
VoIP	Voice Over Internet Protocol
WAV	Waveform Audio File Format
WAVE	Waveform Audio File Format
XML File	Extensible Markup Language

## 1. Typical System Installation

The Voice-over-IP (VoIP) Telephone is a Power-over-Ethernet (PoE 802.3af) and Voice-over-IP (VoIP) two-way communications device that easily connects into existing local area networks (LANs) with a single cable connection. The telephone is compatible with most SIP-based IP PBX servers that comply with SIP RFC 3261.

Figure 1 illustrates how VoIP Telephones can be installed as part of a VoIP phone system.



**Figure 1 - Typical Installation**

## 2. Supported Protocols

The VoIP Telephone with Keypad supports:

- SIP (Session Initiation Protocol)
- HTTP Web-based configuration  
Provides an intuitive user interface for easy system configuration and verification of a VoIP Telephone.
- DHCP Client  
Dynamically assigns IP addresses in addition to the option to use static addressing.
- TFTP Client  
Facilitates hosting for the Autoprovisioning configuration file.
- RTP  
Facilitates autoprovisioning configuration values on boot
- Audio Encodings
  - PCMU (G.711 mu-law)
  - PCMA (G.711 A-law)
  - G.722.1 (SIREN 7)
  - G.722.2 (ANR-WB)
  - Packet Time 20 ms

## 3. Supported SIP Servers

As a SIP device, this product will operate with most IP PBX servers. Contact Guardian Sales to determine the interoperability of any server or visit the Guardian Website to view the Interoperability Guide.

## 4. Features

### Standard Features on All Models

- PoE 802.3af enabled (Power-over-Ethernet) or alternate power source.
- Control Relay – network configurable auxiliary relay.
- Compatible with most SIP-based IP-PBX servers that comply with SIP RFC 3261.
- Network web management.
- Guardian Discovery Utility makes it easy to detect, locate and launch the web based configuration screens.
- Product self diagnostic testing available through web interface.
- Network adjustable volume and microphone sensitivity.
- Network downloadable firmware.
- Remote programming provides network management from a central location.
- Night ringer mode.

### HDE-11-V P6995

- Recessed installation.
- Calls may be initiated from the phone or from the monitoring station.
- Single Emergency Push Button – depressing once automatically rings designated number(s) to summon help.
- Speaker – hands free operation.
- LED – provides visual confirmation connection has been made.
- Doubles as a Paging Speaker.
- ADA (American Disability Act) compliant label **HELP**.
- Night ringer mode.

### HDE-1100-V P6996

- Recessed installation.
- Calls may be initiated from the phone or from the monitoring station.
- Push-to-Call Button – depress the button once and it allows the user to dial out or it will dial a pre-programmed number.
- 12 digit teleseal keypad allows the user to manually dial numbers.
- Speaker – hands free operation.
- LED – provides visual confirmation connection has been made.
- Doubles as a Paging Speaker.

### HDE-12-V P6997

- Surface mount installation.
- Calls may be initiated from the phone or from the monitoring station.
- Emergency Push Button – depressing once automatically rings designated number(s) to summon help.
- Speaker – hands free operation.
- LED – provides visual confirmation connection has been made.
- Doubles as a Paging Speaker.
- ADA (American Disability Act) compliant label **HELP**.

### HDE-1200-V P6998

- Surface mount installation.
- Calls may be initiated from the phone or from the monitoring station.
- Push-to-Call Button – depress the button once and it allows the user to dial out or it will dial a pre-programmed number.
- 12 digit teleseal keypad allows the user to manually dial numbers.
- Emergency Push Button – depressing once automatically rings designated number(s) to summon help.
- Speaker – hands free operation.
- LED – provides visual confirmation connection has been made.
- Doubles as a Paging Speaker.
- ADA (American Disability Act) compliant label **HELP**.



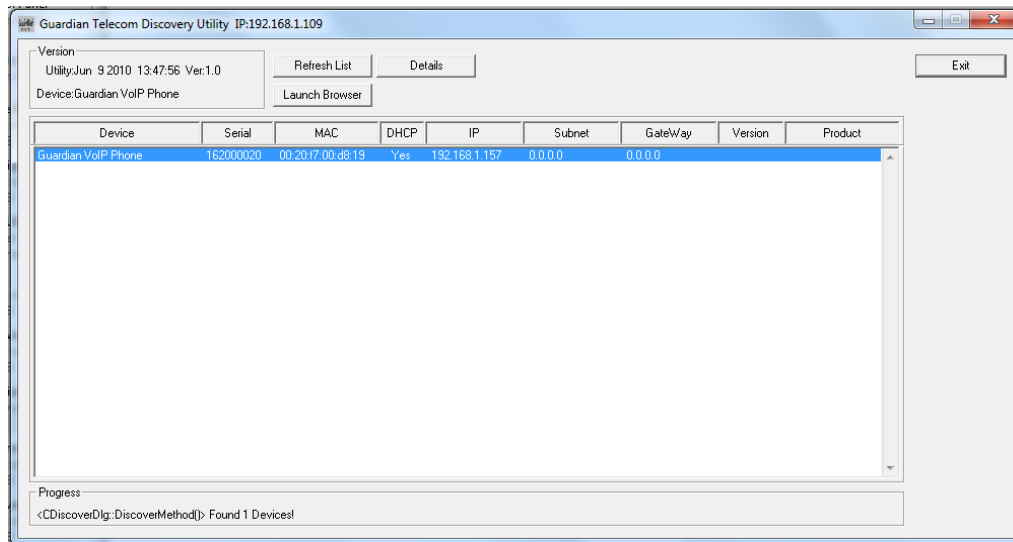
## 5. Getting Started

The Installation manual for the telephone provides information on installing and connecting the device to the server. This manual describes the steps required to customize the telephone to suit the individual's preferences.

The Discovery Utility is available on Guardian's website at <https://www.guardiantelecom.com/resources/voip-support/> and needs to be installed manually by copying the executable file to a local drive.

To access a VoIP phone for programming:

- Copy the Guardian Discovery Utility onto the network server or SIP server.
- Start the Utility by double clicking the icon.
- Click on "Refresh List".
- Click on the device to be programmed to highlight it.
- Click on "Launch Browser".



**Figure 2 - Startup Screen**

## 6. RESET Switch

- The RESET switch is used to get the IP address of the device or reset to factory defaults.
- Press and release the RESET switch within a 5 second window and it will speak the IP address through the on board speaker.
- Press and hold the RESET switch for more than 10 seconds until it indicates that it is restoring defaults and rebooting the board.

## 7. Configure the Telephone Parameters

To configure the Telephone online use a standard web browser.

All Telephones are initially configured with the following default IP settings:

Parameter	Factory Default Setting
IP Addressing	DHCP
IP Address <sup>a</sup>	10.10.10.10
Web Access Username	admin
Web Access Password	admin
Subnet Mask <sup>a</sup>	255.0.0.0
Default Gateway <sup>a</sup>	10.0.0.1









**Table 1 - Factory Default Settings**

a. Default if there is not a DHCP server present.

When configuring more than one Telephone attach the Telephones to the network and configure one at a time to avoid IP address conflicts.

## 7.1. Telephone Web Page Navigation

Table 2 shows the navigation buttons that you will see on every Telephone web page.

Web Page Item	Description
	Link to the <b>Home</b> page.
	Link to the Device Configuration page.
	Link to the <b>Networking</b> page.
	Link to the SIP Configuration page.
	Link to the Audio Configuration page.
	Link to the Event Configuration page.
	Link to the Autoprovisioning Configuration page.
	Link to the Update Firmware page.

**Table 2 - Telephone Web Page Navigation**

## 7.2. Log in to the Configuration Home Page

1. Open your browser to the Telephone's IP address. If you do not know the IP address, you can use the "Discovery Utility" to detect all VoIP devices on the network. When opened the Discovery Utility scans the network for VoIP devices, specifically Guardian VoIP devices. Individually select the device and launch the browser. Another method to obtain the IP address is to press the RESET switch for approximately three to five seconds. The phone will announce the address through the speaker. The physical location of a telephone can be determined by comparing the MAC Address, IP Address or Serial Number shown on the Discovery Utility screen with the information on the unit.

**Note:** If the network does not have access to a DHCP server, the device will default to an IP address of 10.10.10.10.

**Note:** Make sure that the PC is on the same IP network as the Telephone.

2. When prompted, use the following default **Web Access Username** and **Web Access Password** to access the **Home Page** (Figure 3):

Web Access Username: **admin**



Web Access Password: **admin (lower case)**

3. On the **Home Page**, review the setup details and navigation buttons described in Table 3.

**Note:** The Screen Captures shown are only examples; refer to the tables for definitions.

The screenshot displays the 'Guardian Telecom VoIP Phone' configuration web interface. On the left is a vertical navigation menu with buttons for 'Home', 'Device Config', 'Networking', 'SIP Config', 'Audio Config', 'Event Config', 'Autoprovisioning', and 'Update Firmware'. The 'Home' button is selected. The main content area is titled 'Home' and contains three sections: 'Device Settings', 'Current Settings', and 'Import/Export Settings'. The 'Device Settings' section includes fields for 'Device Name' (Guardian Telecom VoIP Phone), 'Change Username' (admin), 'Change Password', and 'Re-enter Password'. The 'Current Settings' section lists various parameters: Serial Number (162101341), Mac Address (00:20:f7:03:67:10), Firmware Version (v3.1.1), IP Addressing (DHCP), IP Address (10.0.0.105), Subnet Mask (255.255.255.0), Default Gateway (10.0.0.254), DNS Server 1 (10.0.0.254), DNS Server 2, Speaker Volume (4), Microphone Gain (4), SIP Mode (Enabled), Event Reporting (Disabled), Nightringer (Disabled Not registered), Primary SIP Server (Registered), Backup Server 1 (Not registered), and Backup Server 2 (Not registered). The 'Import/Export Settings' section has a text prompt 'Please specify a configuration file\*', a 'Choose File' button (showing 'No file chosen'), an 'Import Configuration' button, and an 'Export Configuration' button. At the bottom, a note states '\* You need to reboot for changes to take effect', followed by 'Save' and 'Reboot' buttons.

**Figure 3 - Home Page**

Web Page Item	Description
<b>Device Settings</b>	
Device Name:	Change the device name as required.
Change Username:	Type in this field to change the username.
Change Password:	Type in this field to change the password.
Re-enter Password:	Type the password again in this field to confirm the new password.
<b>Current Settings</b>	
Serial Number:	Shows the device serial number.
Mac Address:	Shows the device MAC address.
Firmware Version:	Shows the current firmware version.
IP Addressing:	Shows the current IP addressing setting ( <b>DHCP</b> or <b>static</b> ).
IP Address:	Shows the current IP address.
Subnet Mask:	Shows the current subnet mask address.
Default Gateway:	Shows the current default gateway address.
DNS Server 1:	Shows the current DNS Server 1 address.
DNS Server 2:	Shows the current DNS Server 2 address.
Speaker Volume:	Shows the current speaker volume level.
Microphone Gain:	Shows the current microphone gain level.
SIP Mode is:	Shows the current SIP Mode status.
Event Reporting is:	Shows the current Event Reporting status.
Nightringer is:	Ringtone broadcast when enabled and extension is called.
Primary SIP Server:	Primary SIP Server
Backup Server 1:	Redundant SIP Server "1"
Backup Server 2:	Redundant SIP Server "2"
<b>Import/Export Settings</b>	
Choose File	Select a configuration file to import to the device.
Import Configuration	Click to import the selected configuration file. The configuration file allows the user to set up a custom default configuration for the VoIP products and import into multiple devices to save time.
Export Configuration	Click to export the current configuration to a file. The export file can then be used to confirm, share or modify and re-import back into the VoIP devices.
	Click the <b>Save</b> button to save your configuration settings. <b>Note:</b> You need to reboot for changes to take effect.
	Click on the <b>Reboot</b> button to reboot the device.

**Table 3 - Home Page Overview**

### 7.3. Configure the Device Parameters

1. Click the **Device Configuration** button to open the **Device Configuration** page. See Figure 4.

Guardian Telecom Inc.

## Guardian Telecom VoIP Phone

**Device Configuration**

**Volume Settings**

Speaker Volume (0-9): 4

Microphone Gain (0-9): 4

Ringer Volume (0-9): 4

**Relay Settings**

Activate Relay with DTMF code: ☒

DTMF Activation Code: 321

DTMF Activation Duration (in seconds): 2

Activate Relay During Ring: ☐

Activate Relay During Night Ring: ☐

Pulse Relay when Ringing: ☐

Pulse Buzzer when Ringing: ☐

Activate Relay While Call Active: ☐

**NTP Settings**

NTP Server: north-america.pool.ntp.org

Posix Timezone String (see manual): PST8PDT,M3.2.0/2:00:00,M11.1.0/2:00:01

Set Time with external NTP server on boot: ☐

Periodically update with time server: ☐

Time update period (in hours): 24

Set time from NTP Server

Clock Time Format: ☒ 12 Hour ☐ 24 Hour

**Current Time**

Current Time in 24 hour format (HHMMSS): Not set

Current Date (MMDDYY): 010370

Set Clock

**Miscellaneous Settings**

Disable HTTPS (NOT recommended): ☒

Call Termination Lockout\*: ☐

Auto-Answer Incoming Calls: ☐

\* You need to reboot for changes to take effect

\* Call termination lockout only applies to red button calls

Save Reboot

Test Audio Test Microphone Test Relay Start Button Test

**Figure 4 - Device Configuration Page**

2. On the **Device Configuration** page, you may enter values for the parameters indicated in Table 4.
3. After changing the parameters, click the **Save** button followed by **Reboot** to complete.

Web Page Item	Description
Volume Settings	The volume settings describe the volume set on reboot. The user can change the volume by using the up and down arrows, but this change is temporary and the volume will be reset when the device is rebooted.
Hands Free Speaker Volume:	The speaker volume sets the initial volume of the device on boot. Valid values are 0-9. Test the speaker volume using the 'Test Audio' button below. Saving changes to the speaker volume will take effect immediately (it does not require a restart).
Hands Free Microphone Gain:	The microphone gain sets the initial input gain of the on board microphone. Valid values are 0-9. Test the speaker volume using the 'Test Microphone' button below. Saving changes to the speaker volume will take effect immediately (it does not require a restart).
Ringer Volume:	The ringing volume which is heard on the speaker of the phone. Valid values are 0-9. This setting is for units that use the speaker as a ringer. It will not affect units with a fixed buzzer.
Relay Settings	
Activate Relay with DTMF Code:	When this option is enabled, the device will activate the relay when it receives a DTMF code (rfc2833).
DTMF Activation Code:	This 25 character field can be used to set a DTMF code used to activate the relay.
DTMF Activation Duration (in seconds):	When the relay is activated with a DTMF code, it will remain active for this duration in seconds. Valid values are 1-9. <b>NOTE:</b> A DTMF activation of <b>0</b> will toggle the relay indefinitely or until the activation code is sent again.
Activate Relay During Ring:	When this option is enabled, the relay will activate when the device has received a call and is playing a ringtone.
Activate Relay During Night Ring:	When this option is enabled, the relay will activate when the device has received a call to the night ring extension.
Pulse Relay when Ringing:	When a "ring" is present the relay will pulse with a cadence of 2 seconds on, 3 seconds off.
Pulse Buzzer when Ringing:	When a "ring" is present the internal ringer will pulse with a cadence of 2 seconds on, 3 seconds off.
Activate Relay While Call Active:	When this option is enabled, the relay will activate when a call is established with another SIP device. It will remain active for the duration of the call.
NTP Settings	
NTP Server	Use this field to set the address (in IPv4 dotted decimal notation or as a canonical name) for the NTP Server. This field can accept canonical names of up to 64 characters in length.
Posix Timezone String	Allows the device to access a local network based time server or a global internet server to allow the device to auto set/update the device clock. See Appendix A.
Set Time with external NTP server on Boot	When selected, the time is set with an external the NTP server when the device restarts.
Periodically update with time server	When selected, the time is periodically updated with the NTP server at the configured interval below.
Time update period (in hours)	The time interval after which the device will contact the NTP server to update the time. Enter up to 4 digits.
Set time from NTP Server	Clicking on this button will immediately set the time.
Current Time	
Current Time in 24 hour format (HHMMSS):	Allows you to input the current time manually. (6 character limit)
Current Date (MMDDYY):	Allows you to input the current date manually. (6 character limit)
Set Clock	Clicking on this button will immediately set the clock format.

## Miscellaneous Settings



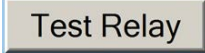

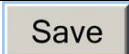

Disable HTTPS (NOT recommended):	Disables the encrypted connection to the webpage. Depending on network security enable / disable as required.
Call Termination Lockout*:	When this option is enabled, a call cannot be terminated using the Emergency Call button.
Auto-Answer Incoming Calls	When selected, the device will automatically answer incoming calls. When Auto-Answer Incoming Calls is disabled, the device will play a ring tone (corresponds to Ring Tone on the Audiofiles page) out of the speaker, or the buzzer will sound until someone presses the Call button to answer the call or the caller disconnects before the call can be answered.
	This button will play an audio test message and can be used to test the volume level.
	When this button is pressed the device will record 3 seconds of audio, beep, and then play back the recorded audio. This can be used to test the microphone gain level.
	This button will activate the relay for the DTMF activation Duration (in seconds).
	When this button is toggled, it will put the device into button test mode. In this mode, the speaker will play an audio file when a button is pressed on the device. For normal keypad input (keys 0-9) it will speak the associated file from the audio configuration page. For the other keypad keys, (*, #, any other function keys) it will play a DTMF tone while the button is depressed. The button press mode will time out after 60 seconds.
	Click the <b>Save</b> button to save your configuration settings. <b>Note:</b> You need to reboot for changes to take effect.
	Click on the <b>Reboot</b> button to reboot the device.
<ul style="list-style-type: none"> <li>• <b>You need to reboot for changes to take effect</b></li> <li>• *Call termination lock out only applies to red button calls</li> </ul>	

Table 4 - Device Configuration Parameters

## 7.4. Configure the Network Parameters

1. Click the **Networking** button to open the **Network Configuration** page (Figure 5).

Guardian Telecom Inc.

# Guardian Telecom VoIP Phone

Home Device Config **Networking** SIP Config Audio Config Event Config Autoprovisioning Update Firmware

## Network Configuration

Stored Network Settings

Addressing Mode: ☐ Static ☒ DHCP

Hostname: Guardian\_TwoButton

IP Address: 10.10.10.10

Subnet Mask: 255.0.0.0

Default Gateway: 10.0.0.1

DNS Server 1: 10.0.0.1

DNS Server 2: 10.0.0.1

DHCP Timeout in seconds\*: 60

\* A value of -1 will retry forever

VLAN Settings

VLAN ID (0-4095): 0

VLAN Priority (0-7): 0

Current Network Settings

IP Address: 10.0.0.105

Subnet Mask: 255.255.255.0

Default Gateway: 10.0.0.254

DNS Server 1: 10.0.0.254

DNS Server 2:

\* You need to reboot for changes to take effect



Save Reboot

**Figure 5 - Network Configuration Page**

2. On the **Network Configuration** page, enter values for the parameters indicated in Table 5.
3. After changing the parameters, click **Save** to store the settings before going to the other configuration pages. If no other changes are required beyond this state, click **Reboot** to reset the device for other changes to take effect. The new settings will not take effect if **Reboot** is clicked without saving first.
4. Connect the Telephone to the target network.
5. From a computer on the same network as the Telephone, open a browser with the new IP address of the Telephone.

Note: If changing from DHCP to STATIC Only: Once the reboot button has been selected, the webpage will show a countdown timer. The timer will hit zero and will reset and continue with countdown. At this point the web page can be shutdown and re-started. The web page will not automatically restart when switching from DHCP to STATIC.



Web Page Item	Description
Stored Network Settings	Shows the settings stored in non-volatile memory.
Addressing Mode:	Select either DHCP IP Addressing or Static Addressing by marking the appropriate radio button. DHCP Addressing mode is enabled on default and the device will attempt to resolve network addressing with the local DHCP server upon boot. If DHCP Addressing fails, the device will revert to the last known IP address or the factory default address if no prior DHCP lease was established. See Table 1 for factory default settings. Be sure to click <b>Save</b> and <b>Reboot</b> to store changes when configuring a Static address.
Hostname:	This is the hostname provided by the DHCP server. See the DHCP/DNS server documentation for more information. Enter up to 64 characters.
IP Address:	The IPv4 static IP address in standard dotted decimal notation.
Subnet Mask:	The IPv4 routing prefix in standard dotted decimal notation.
Default Gateway:	This is the node to go to when an IP address doesn't match any routes in the routing table. This requires standard quad-dotted decimal notation.
DNS Server 1: DNS Server 2:	The DNS server configuration is used to setup the primary and secondary name servers for the network. These use standard dotted decimal notation.
DHCP Timeout in seconds*: *A value of -1 will retry forever	Specify the desired time-out duration (in seconds) that the device will wait for a response from the DHCP server before reverting back to the stored static IP address. The stored static IP address may be the last known IP address or the factory default address if no prior DHCP lease was established. Enter up to 8 characters. A value of -1 will retry forever.
VLAN Settings	Shows the current VLAN settings.
VLAN ID (0-4095):	Specify the IEEE 802.1Q VLAN ID number. Enter up to 4 digits. <b>Note:</b> The device supports 802.1Q VLAN tagging support. The switch port connected to the device will need to be in "trunking mode" for the VLAN tags to propagate.
VLAN Priority (0-7):	Specify the IEEE 802.1p VLAN priority level. Enter 1 digit. A value of 0 may cause the VLAN ID tag to be ignored.
Current Network Settings	Shows the current network settings.
IP Address:	Shows the current Static IP address.
Subnet Mask:	Shows the current Subnet Mask address.
Default Gateway:	Shows the current Default Gateway address.
DNS Server 1:	Shows the current DNS Server 1 address.
DNS Server 2:	Shows the current DNS Server 2 address.
	Click the <b>Save</b> button to save your configuration settings. <b>Note:</b> You need to reboot for changes to take effect.
	Click on the <b>Reboot</b> button to reboot the device.

- You need to reboot for changes to take effect

Table 5 - Network Configuration Parameters

7.5. Configure the SIP Parameters

1. Click **SIP Config** to open the **SIP Configuration** page (Figure 6).

**Note:** Guardian VoIP telephones are compatible with most SIP servers.

The screenshot displays the 'SIP Configuration' page for a Guardian Telecom VoIP phone. The interface is organized into several sections:

- Navigation Bar:** Includes links for Home, Device Config, Networking, SIP Config (active), Audio Config, Event Config, Autoprovisioning, and Update Firmware.
- SIP Configuration Header:** Shows 'SIP Configuration' and 'Enable SIP operation' (checked).
- SIP Settings:**
  - ☒ Register with a SIP Server
  - ☒ Use Cisco SRST
  - Primary SIP Server: 10.0.0.253
  - Primary SIP User ID: 199
  - Primary SIP Auth ID: 199
  - Primary SIP Auth Password: \*\*\*\*\*
  - Backup SIP Server 1: [blank]
  - Backup SIP User ID 1: [blank]
  - Backup SIP Auth ID 1: [blank]
  - Backup SIP Auth Password 1: [blank]
  - Backup SIP Server 2: [blank]
  - Backup SIP User ID 2: [blank]
  - Backup SIP Auth ID 2: [blank]
  - Backup SIP Auth Password 2: [blank]
  - Remote SIP Port: 5060
  - Local SIP Port: 5060
  - Outbound Proxy: 0
  - Outbound Proxy Port: [blank]
- Dial Out Settings:**
  - ☐ Disable rport Discovery
  - Re-registration Interval (in seconds): 360
  - Unregister on Boot: ☐
  - Keep Alive Period: 10000
- Call Disconnection:**
  - Terminate Call after delay: 0
- Codec Selection:**
  - Force Selected Codec: ☐ PCMU (G.711, u-law)
  - Codec: [blank]
- RTP Settings:**
  - RTP Port (even): 10500
  - Jitter Buffer: 50
- Dial Out Settings:**
  - Blue Button Dial out Extension ID: [blank]
  - Blue Button Extension ID: [blank]
  - Red Button Dial out Extension ID: [blank]
  - Red Button Extension ID: [blank]
- Footer:** Includes a note about rebooting for changes to take effect and a warning about the blue button dial out extension being blank. It also features 'Save' and 'Reboot' buttons.

Figure 6 - SIP Configuration Page

2. On the **SIP Configuration** page, enter values for the parameters indicated in Table 6.
3. After changing the parameters, click **Save Settings**

Web Page Item	Description
Enable SIP Operation:	When this option is enabled, the device will initialize the SIP engine and try to register with a SIP server or listen for incoming SIP connections.
<b>SIP Settings</b>	
Register with a SIP Server:	When enabled, the device will attempt to register to the configured SIP Server(s) on this page. To configure the device to send and receive point-to-point SIP calls, enable <b>SIP Operation</b> and disable <b>Register with a SIP Server</b> .
Use Cisco SRST:	When enabled, the backup servers are handled according to Cisco SRST (Survivable Remote Site Telephony). It is required for use in clustered Cisco Unified Communications Manager topologies.
Primary SIP Server:	Enter the SIP server address as an IPv4 address in dotted decimal notation or a fully qualified domain name. This parameter also becomes the host portion of the SIP-URI for the device's extension on the primary SIP server. This field can accept entries of up to 255 characters in length.
Primary SIP User ID:	Specify the SIP User ID for the Primary SIP Server. This parameter becomes the user portion of the SIP-URI for the device's extension on the primary SIP server. Enter up to 64 alphanumeric characters.
Primary SIP Auth ID:	Specify the Authenticate ID for the Primary SIP Server. This parameter is required for SIP registration authentication. Enter up to 64 alphanumeric characters.
Primary SIP Auth Password:	Specify the Authenticate Password for the Primary SIP Server. This parameter is required for SIP registration authentication. Enter up to 64 alphanumeric characters.
Backup SIP Server 1:	Enter the backup SIP server address as an IPv4 address in dotted decimal notation or a fully qualified domain name. This parameter also becomes the host portion of the SIP-URI for the device's extension on the backup SIP server. This field can accept entries of up to 255 characters in length.
Backup SIP User ID 1:	Specify the SIP User ID for the first backup SIP Server. This parameter becomes the user portion of the SIP-URI for the device's extension on the first backup SIP server. Enter up to 64 alphanumeric characters.
Backup SIP Auth ID 1:	Specify the Authenticate ID for the first backup SIP server. This parameter is required for SIP registration authentication. Enter up to 64 alphanumeric characters.
Backup SIP Auth Password 1:	Specify the Authenticate Password for the first backup SIP server. This parameter is required for SIP registration authentication. Enter up to 64 alphanumeric characters.
Backup SIP Server 2:	Enter a second backup SIP server address as an IPv4 address in dotted decimal notation or a fully qualified domain name. This parameter also becomes the host portion of the SIP-URI for the device's extension on the second backup SIP server. This field can accept entries of up to 255 characters in length.
Backup SIP User ID 2:	Specify the SIP User ID for the second backup SIP Server. This parameter becomes the user portion of the SIP-URI for the device's extension on the second backup SIP server. Enter up to 64 alphanumeric characters.
Backup SIP Auth ID 2:	Specify the Authenticate ID for the second backup SIP server. This parameter is required for SIP registration authentication. Enter up to 64 alphanumeric characters.

Backup SIP Auth Password 2:	Specify the Authenticate Password for the second backup SIP server. This parameter is required for SIP registration authentication. Enter up to 64 alphanumeric characters.
Remote SIP Port:	The Remote SIP Port is the port number the device will use as the destination port when sending SIP messages. The default Remote SIP Port is 5060. The supported range is 0-65536. Enter up to 5 digits.
Local SIP Port:	The Local SIP Port is the port number the device will use to receive SIP messages. The default Local SIP Port is 5060. The supported range is 0- 65536. Enter up to 5 digits.
Outbound Proxy	Enter the Outbound Proxy address as an IPv4 address in dotted decimal notation or a fully qualified domain name (FQDN). When an IP address is configured, the device will send all SIP messages to this IP address. When an FQDN is configured, the device will run DNS NAPTR, SRV, and A queries on the FQDN to resolve an IP address to which it will send all SIP messages. This field can accept entries of up to 255 characters in length.
Outbound Proxy Port:	The Outbound Proxy Port is port number used as the destination port when sending SIP messages to the outbound proxy. A value of 0 will default to 5060. The supported range is 0-65536. Enter up to 5 digits.
Disable report Discovery:	Disabling report Discovery will prevent the device from including the public WAN IP address and port number in the contact information that is sent to the remote SIP servers. This will generally only need to be enabled when using an SBC or SIP ALG in conjunction with a remote SIP server.
Re-registration Interval (in seconds):	The SIP Re-registration interval (in seconds) is the SIP Registration lease time, also known as the expiry. The supported range is 30-3600 seconds. Enter up to 4 digits.
Unregister on Boot:	When enabled, the device will send one registration with an expiry of 0 on boot.
Keep Alive Period:	The minimum time in milliseconds between keep-alive packets sent for nat traversal. A value of 0 will disable keep alive packets.
Enable Nightringer:	When Nightringer is enabled, the device will attempt to register a second extension with the SIP server. Any calls made to this extension will play a ringtone (corresponds to <b>Night Ring</b> on the <b>Audiofiles</b> page). By design, it is not possible to answer a call to the Nightringer extension.

#### Nightringer Settings

SIP Server:	Enter the SIP server address as an IPv4 address in dotted decimal notation or a fully qualified domain name. This parameter also becomes the host portion of the SIP-URI for the device's Nightringer extension on the SIP server. This field can accept entries of up to 255 characters in length.
Remote SIP Port:	The Remote SIP Port is the port number the device will use as the destination port when sending SIP messages for the Nightringer extension. The default Remote SIP Port is 5060. The supported range is 0-65536. Enter up to 5 digits.
Local SIP Port:	The Local SIP Port is the port number the device will use to receive SIP messages for the Nightringer extension. This value cannot be the same as the <b>Local SIP Port</b> for the primary extension. The default Local SIP Port is 5061. The supported range is 0-65536. Enter up to 5 digits.
Outbound Proxy:	Enter the Outbound Proxy address as an IPv4 address in dotted decimal notation or a fully qualified domain name (FQDN). When an IP address is configured, the device will send all SIP messages to this IP address for the Nightringer extension. When an FQDN is configured, the device will run DNS NAPTR, SRV, and A queries on the FQDN to resolve an IP address to which it will send all SIP messages for the Nightringer extension. This field can accept entries of up to 255 characters in length.

Outbound Proxy Port:	The Outbound Proxy Port is port number used as the destination port when sending SIP messages to the outbound proxy for the Nightringer extension. A value of 0 will default to 5060. The supported range is 0- 65536. Enter up to 5 digits.
User ID:	Specify the SIP User ID for the SIP server. This parameter becomes the user portion of the SIP-URI for the device's Nightringer extension. Enter up to 64 alphanumeric characters.
Authenticate ID:	Specify the Authenticate ID for the SIP Server. This parameter is required for SIP registration authentication. Enter up to 64 alphanumeric characters.
Authenticate Password:	Specify the Authenticate Password for the SIP Server. This parameter is required for SIP registration authentication. Enter up to 64 alphanumeric characters. Re-registration Interval (in seconds).
Re-registration Interval (in seconds):	The SIP Re-registration Interval (in seconds) is the SIP Registration lease time, also known as the expiry. The supported range is 30-3600 seconds. Enter up to 4 digits.

**Call Disconnection**

Terminate Call after delay:	Automatically terminate an active call after a given delay in seconds. A value of 0 will disable this function. Enter up to 8 digits.
-----------------------------	---

**Codec Selection**

Force Selected Codec:	When selected, this option will allow you to force the device to negotiate for the selected codec [PCMU(G.711, u-law), PCMA(G.711, a-law), or G.722]. Otherwise, the device will perform codec negotiation using the default list of supported codecs.
Codec:	Select a desired Codec in the drop-down list.

**RTP Settings**

RTP Port (even):	Specify the port number used for the RTP stream after establishing a SIP call. This port number must be an even number and defaults to 10500. The supported range is 0-65536. Enter up to 5 digits.
Jitter Buffer:	Specify the size of the jitter buffer (in milliseconds) used for SIP calls. Valid values are 50-1000.

**Dial Out Settings**

Blue Button Dial out Extension:	Specify the extension the device will call when someone presses the blue Call button. Enter up to 64 alphanumeric characters.
Blue Button Extension ID:	A Caller identification string added to outbound calls. Enter up to 64 alphanumeric characters.
Red Button Dial out Extension:	Specify the extension the device will call when someone presses the red Emergency Call button. Enter up to 64 alphanumeric characters.
Red Button Extension ID:	A Caller identification string added to outbound calls. Enter up to 64 alphanumeric characters.

 Save

Click the **Save** button to save your configuration settings.  
**Note:** You need to reboot for changes to take effect.

 Reboot

Click on the **Reboot** button to reboot the device.

- **You need to reboot for changes to take effect**
- When the blue button dial out extension is blank it will be used as a hook switch and enable the keypad.

**Table 6 - SIP Configuration Parameters**

7.6. Configure the Audio Parameters

1. Click **Audio Config** to open the **Audio Configuration** page (Figure 7).



Figure 7 - Audio Configuration Page

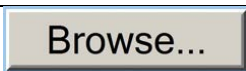
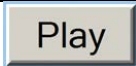




The **Audio Configuration** page is used to add custom audio to the board. User uploaded audio will take precedence over the audio files shipped with the Telephone.

2. On the **Audio Configuration** page, enter values for the parameters indicated in Table 8.

**Note:** Each entry on the **Audio Configuration** page replaces one of the stock audio files on the board. When the input box displays the word **default**, the Telephone is using the stock audio file. If that file is replaced with a user file, it will display the uploaded filename.

The character limits relate to the file names, not the contents of the file.

Web Page Item	Description
Audio Files	
0-9:	The name of the audio configuration option is the same as the spoken audio that plays on the board. '0' corresponds to the spoken word "zero." '1' corresponds to the spoken word "one." '2' corresponds to the spoken word "two." '3' corresponds to the spoken word "three." '4' corresponds to the spoken word "four." '5' corresponds to the spoken word "five." '6' corresponds to the spoken word "six." '7' corresponds to the spoken word "seven." '8' corresponds to the spoken word "eight." '9' corresponds to the spoken word "nine."
Dot:	Corresponds to the spoken word "dot." (24 character limit)
Audio test:	Corresponds to the message "Guardian_Audio.wav IP telephone test message..." (200 character limit)
Page tone:	Corresponds to a simple tone that is unused by default (24 character limit).
Your IP Address is:	Corresponds to the message "Your IP address is..." (24 character limit).
Rebooting:	Corresponds to the spoken word "Rebooting" (24 character limit).
Restoring default:	Corresponds to the message "Restoring default" (24 character limit).
Ringback Tone:	This is the ringback tone that plays when calling a remote extension (24 character limit).
Ring tone:	Tone that plays when the device is ringing.
Night Ring:	When the Nightringer is enabled, the device will attempt to register a second extension with the SIP server. Any calls made to this extension will play a ringtone.
	Click the Browse button to search for files.
	Click the Play button to hear the current message.
	Click the Delete button to empty the box.
	Click the <b>Save</b> button to save your settings.

**Table 7 - Audio Configuration Parameters**

7.6.1. User-created Audio Files

User created audio files should be saved in the following format:  
RIFF (little-endian) data, WAVE audio, Microsoft PCM, 16 bit, mono, 8000 Hz  
You can use the free utility *Audacity* to convert audio files into this format, see Figure 8 through Figure 10.  
You may download the application at <https://creativecommons.org/licenses/by/3.0/>.

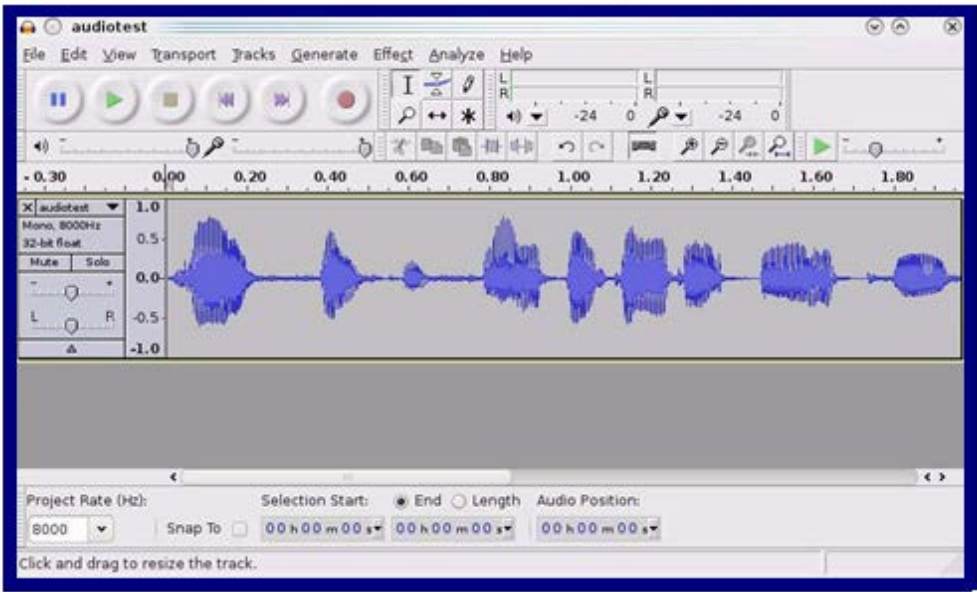


Figure 8 - Audacity 1

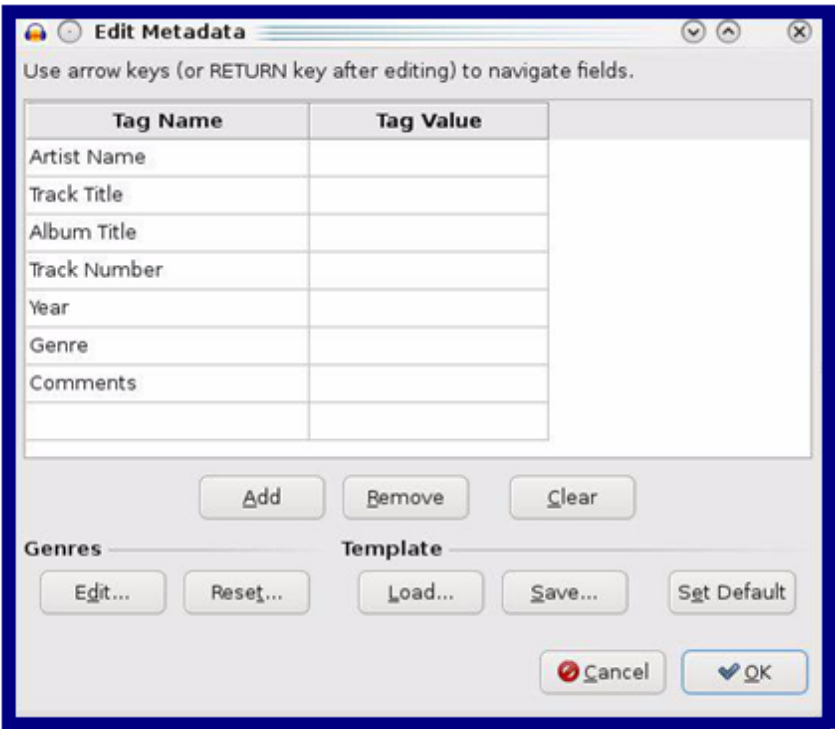
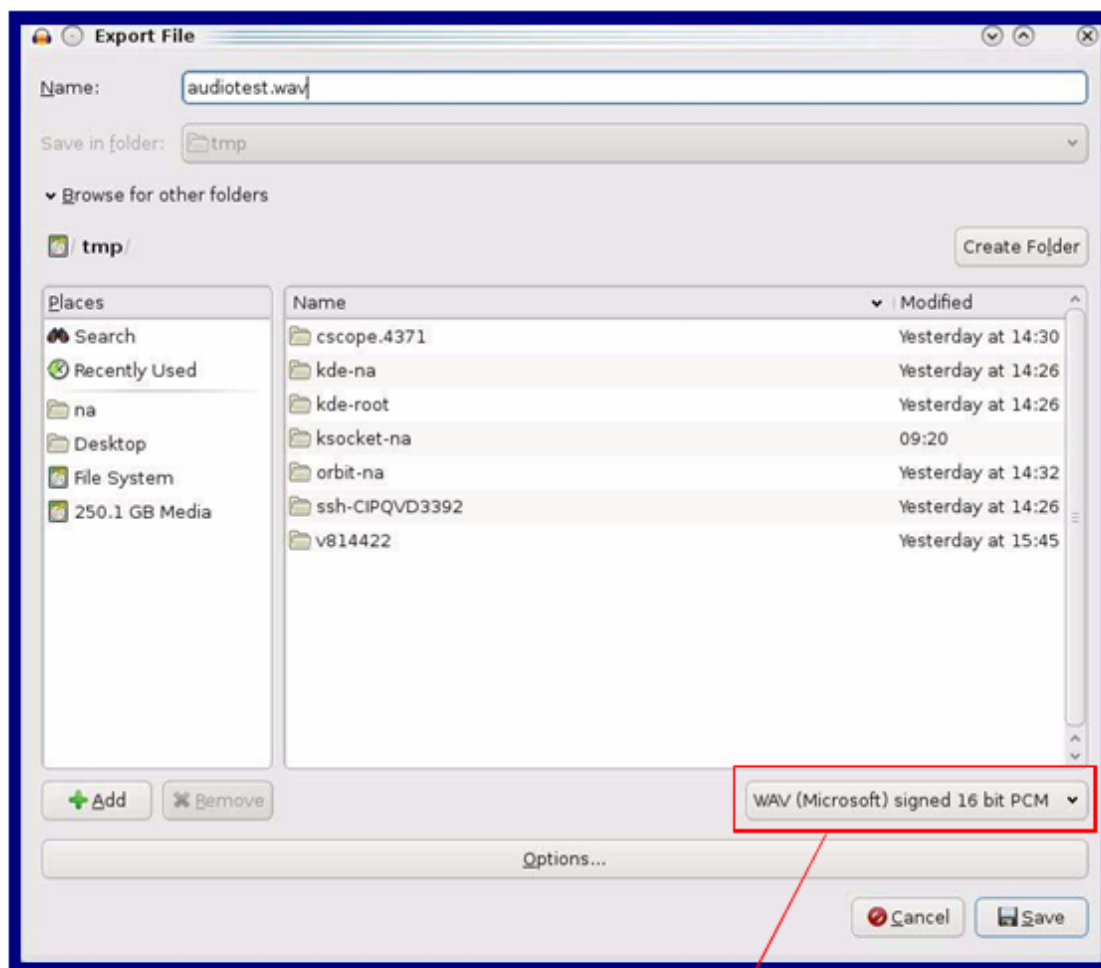


Figure 9 - Audacity 2



When you export an audio file with Audacity, save the output as:

- WAV (Microsoft) signed 16 bit PCM.



WAV (Microsoft) signed 16 bit PCM

**Figure 10 - WAV (Microsoft) signed 16 bit PCM**

7.7. Configure the Event Parameters

Click the Event Config button to open the Event Configuration page (Figure 11).

Guardian  
Telecom Inc.

Guardian Telecom VoIP Phone

Home

Device Config

Networking

SIP Config

Audio Config

Event Config

Autoprovisioning

Update Firmware

Event Configuration

Enable Event Generation: ☐

Event Server

Server IP Address: 10.0.0.250

Server Port: 8080

Server URL: xmlparse\_engine

Events

Enable Button Events: ☐

Enable Call Start Events: ☐

Enable Call Terminated Events: ☐

Enable Relay Activated Events: ☐

Enable Relay Deactivated Events: ☐

Enable Ring Events: ☐

Enable Night Ring Events: ☐

Enable Power On Events: ☐

Enable Sensor Events: ☐

Enable Remote Relay Events: ☐

Enable Security Events: ☐

Enable 60 Second Heartbeat: ☐

[Check All](#) [Uncheck All](#)

\* You need to reboot for changes to take effect

Save

Test Event

Reboot

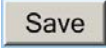

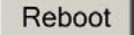
The Event Configuration page specifies a remote server that can be used to receive HTTP POST events when actions take place on the board.

Figure 11 - Event Configuration Page

Table 8 shows the web page items on the **Event Configuration** page.

Web Page Item	Description
Enable Event Generation:	When this option is selected the device will initialize the event generation engine. This mechanism can be used to send xml formatted http POST packets to an external server in response to triggers in the operation of the device.
<b>Event Server</b>	
Server IP Address:	This is the address of the remote TCP server for receiving POST events. This field can accept addresses in dotted decimal notation or canonical names of up to 64 characters in length.
Server Port:	The Remote Event Server Port is used to set the port number that the remote server is listening on.
Server URL:	POST requests have to be sent to a target script at the given IP address. This field defaults to 'xml_engine' and can accept up to 127 characters.
<b>Events</b>	<b>Examples</b>
Enable Button Events:	<p>When this option is enabled an event will be sent to the remote server every time a button is pressed.</p> <pre>POST xmlparse_engine HTTP/1.1 Host: 10.0.3.79 User-Agent: CyberData/1.0.0 Content-Length: 196 Content-Type: application/x-www-form-urlencoded &lt;?xml version="1.0" encoding="ISO-8859-1"?&gt; &lt;cyberdata NAME='GuardianVoIP Device' MAC='0020f70015b6'&gt; &lt;event&gt;BUTTON&lt;/event&gt; &lt;/cyberdata&gt;</pre>
Enable Call Start Events:	<p>When this option is enabled an event will be sent to the remote server when a call becomes active.</p> <pre>POST xmlparse_engine HTTP/1.1 Host: 10.0.3.79 User-Agent: CyberData/1.0.0 Content-Length: 201 Content-Type: application/x-www-form-urlencoded &lt;?xml version="1.0" encoding="ISO-8859-1"?&gt; &lt;cyberdata NAME='Guardian VoIP Device' MAC='0020f70015b6'&gt; &lt;event&gt;CALL_ACTIVE&lt;/event&gt; &lt;/cyberdata&gt;</pre>
Enable Call Terminated Events:	<p>When this option is enabled an event will be sent to the remote server when a call is terminated.</p> <pre>POST xmlparse_engine HTTP/1.1 Host: 10.0.3.79 User-Agent: CyberData/1.0.0 Content-Length: 205 Content-Type: application/x-www-form-urlencoded &lt;?xml version="1.0" encoding="ISO-8859-1"?&gt; &lt;cyberdata NAME='Guardian VoIP Device' MAC='0020f70015b6'&gt; &lt;event&gt;CALL_TERMINATED&lt;/event&gt; &lt;/cyberdata&gt;</pre>

Enable Relay Activated Events:	<p>When this option is enabled an event will be sent to the remote server when the relay is activated.</p> <pre>POST xmlparse_engine HTTP/1.1 Host: 10.0.3.79 User-Agent: CyberData/1.0.0 Content-Length: 234 Content-Type: application/x-www-form-urlencoded &lt;?xml version="1.0" encoding="ISO-8859-1"?&gt; &lt;cyberdata NAME='Guardian VoIP Device' MAC='0020f70015b6'&gt; &lt;event&gt;RELAY_ACTIVATED&lt;/event&gt; &lt;/cyberdata&gt;</pre>
Enable Relay Deactivated Events:	<p>When this option is enabled an event will be sent to the remote server when the relay is deactivated.</p> <pre>POST xmlparse_engine HTTP/1.1 Host: 10.0.3.79 User-Agent: CyberData/1.0.0 Content-Length: 234 Content-Type: application/x-www-form-urlencoded &lt;?xml version="1.0" encoding="ISO-8859-1"?&gt; &lt;cyberdata NAME='Guardian VoIP Device' MAC='0020f70015b6'&gt; &lt;event&gt;RELAY_DEACTIVATED&lt;/event&gt; &lt;/cyberdata&gt;</pre>
Enable Ring Events:	<p>When this option is enabled an event will be sent to the remote server when the device starts playing a ringtone.</p> <pre>POST xmlparse_engine HTTP/1.1 Host: 10.0.3.79 User-Agent: CyberData/1.0.0 Content-Length: 197 Content-Type: application/x-www-form-urlencoded &lt;?xml version="1.0" encoding="ISO-8859-1"?&gt; &lt;cyberdata NAME='Guardian VoIP Device' MAC='0020f70015b6'&gt; &lt;event&gt;RINGING&lt;/event&gt; &lt;/cyberdata&gt;</pre>
Enable Night Ring Events:	<p>When selected, there is a notification when the device receives a night ring.</p> <pre>POST xmlparse_engine HTTP/1.1 Host: 10.0.3.79 User-Agent: CyberData/1.0.0 Content-Length: 234 Content-Type: application/x-www-form-urlencoded &lt;?xml version="1.0" encoding="ISO-8859-1"?&gt; &lt;cyberdata NAME='Guardian VOIP Device' MAC='002 0f70015b6'&gt; &lt;event&gt;NIGHTRINGING&lt;/event&gt; &lt;/cyberdata&gt;</pre>
Enable Power On Events:	<p>When this option is enabled an event will be sent to the remote server when the device powers up.</p> <pre>POST xmlparse_engine HTTP/1.1 Host: 10.0.3.79 User-Agent: CyberData/1.0.0 Content-Length: 197 Content-Type: application/x-www-form-urlencoded &lt;?xml version="1.0" encoding="ISO-8859-1"?&gt; &lt;cyberdata NAME='Guardian VoIP Device' MAC='0020f70015b6'&gt; &lt;event&gt;POWER ON&lt;/event&gt; &lt;/cyberdata&gt;</pre>

Enable Sensor Events:	<p>When this option is enabled an event will be generated when a change is seen on the one of the opto-isolated input lines.</p> <pre>POST xmlparse_engine HTTP/1.1 Host: 10.0.3.79 User-Agent: CyberData/1.0.0 Content-Length: 197 Content-Type: application/x-www-form-urlencoded &lt;?xml version="1.0" encoding="ISO-8859-1"?&gt; &lt;cyberdata NAME='Guardian VoIP Device' MAC='0020f70015b6'&gt; &lt;event&gt;INPUT1&lt;/event&gt; &lt;/cyberdata&gt;</pre>
Enable Remote Relay Events:	<p>When this option is enabled an event will be generated when a change is seen on the one of the opto-isolated input lines.</p> <pre>POST xmlparse_engine HTTP/1.1 Host: 10.0.3.79 User-Agent: CyberData/1.0.0 Content-Length: 197 Content-Type: application/x-www-form-urlencoded &lt;?xml version="1.0" encoding="ISO-8859-1"?&gt; &lt;cyberdata NAME='Guardian VoIP Device' MAC='0020f70015b6'&gt; &lt;event&gt;INPUT2&lt;/event&gt; &lt;/cyberdata&gt;</pre>
Enable Security Events:	<p>When this option is enabled an event will be generated when a change is seen on the one of the opto-isolated input lines.</p> <pre>POST xmlparse_engine HTTP/1.1 Host: 10.0.3.79 User-Agent: CyberData/1.0.0 Content-Length: 197 Content-Type: application/x-www-form-urlencoded &lt;?xml version="1.0" encoding="ISO-8859-1"?&gt; &lt;cyberdata NAME='Guardian VoIP Device' MAC='0020f70015b6'&gt; &lt;event&gt;INPUT3&lt;/event&gt; &lt;/cyberdata&gt;</pre>
Enable 60 Second Heartbeat:	<p>When this option is enabled an event will be sent to the remote server every 60 seconds.</p> <pre>POST xmlparse_engine HTTP/1.1 Host: 10.0.3.79 User-Agent: CyberData/1.0.0 Content-Length: 199 Content-Type: application/x-www-form-urlencoded &lt;?xml version="1.0" encoding="ISO-8859-1"?&gt; &lt;cyberdata NAME='Guardian VoIP Device' MAC='0020f70015b6'&gt; &lt;event&gt;HEARTBEAT&lt;/event&gt; &lt;/cyberdata&gt;</pre>
	Click the <b>Save</b> button to save your configuration settings. <b>Note:</b> You need to reboot for changes to take effect.
	Click Test Event to test the event.
	Click on the <b>Reboot</b> button to reboot the device.

- You need to reboot for changes to take effect.

**Table 8 - Event Configuration**

## 7.8. Configure the Autoprovisioning Parameters

Every phone that needs auto-provisioning has to have its auto-provisioning file set up.

The auto-provisioning of a HDE-VoIP phone is done as follows:

1. Create an auto-provision template of the phone.
  - Go to Import/Export Settings near the bottom of the page on the Home Page.
  - Select Export Configuration; the device will prompt with Save options.
  - Using an appropriate editor such as Wordpad or XML modify the parameters of the file.
2. Open and edit the template to change the features of the phone.
3. Save and rename the file as desired with a file extension of .xml or .txt. For example if the phone has a MAC address of 00:20:f7:01:0f:22 the auto-provisioning file of this particular device/phone could be saved as 0020f7010f22.xml.
4. Click the Autoprovisioning button to open the Autoprovisioning Configuration page (Figure 12).

Guardian Telecom Inc.

## Guardian Telecom VoIP Phone

Home Device Config Networking SIP Config Audio Config Event Config **Autoprovisioning** Update Firmware

**Autoprovisioning**

Disable Autoprovisioning: ☐

Autoprovisioning

Autoprovisioning Server:

Autoprovisioning Filename:

Use tftp: ☒

Username:

Password:

Autoprovisioning autoupdate (in minutes):

Autoprovision at time (HHMMSS):

Autoprovision when idle (in minutes > 10):

Infinite Retries (Not Recommended): ☐



\* You need to reboot for changes to take effect

Save Reboot

**Figure 12 - Autoprovisioning Configuration Page**

On the **Autoprovisioning Configuration** page, you may enter values for the parameters indicated in Table 10.

Follow the instructions in the Guardian Discovery tool program to complete the auto-provisioning of a VoIP phone.

Web Page Item	Description
<b>Autoprovisioning</b>	
Disable Autoprovisioning:	When this option is enabled, the device will try to fetch an autoprovisioning file from a remote server.
Autoprovisioning Server:	When this option is enabled, the device will fetch its autoprovisioning file from the server specified from the DHCP server.
Autoprovisioning Filename:	Enter the Autoprovision file name that was generated with the Setup parameters.
Use tftp:	If using TFTP (instead of http) to download autoprovisioning files click the box.
Username:	The username used to authenticate with an autoprovisioning server. Leave this field blank to disable authentication.
Password:	The password used to authenticate with an autoprovisioning server. Leave this field blank to disable authentication.
Autoprovisioning autoupdate (in minutes):	<p>The reoccurring time (in minutes) the device will wait before checking for new autoprovisioning files. Enter up to 6 digits. A value of 0 will disable this option.</p> <p><b>Note:</b> To use the auto update options, enable the <b>Set Time with NTP Server on boot</b> setting on the <b>Device Configuration Page</b> page.</p>
Autoprovision at time (HHMMSS):	<p>The time of day the device will check for a new autoprovisioning file. The time must be 6 characters in length and in HHMMSS format. An empty value will disable this option.</p> <p><b>Note:</b> To use the auto update options, enable the <b>Set Time with NTP Server on boot</b> setting on the <b>Device Configuration Page</b> page.</p>
Autoprovision when idle (in minutes > 10):	<p>The idle time (in minutes greater than 10) after which the device will check for a new autoprovisioning file. Enter up to 6 digits. A value of 0 will disable this option.</p> <p><b>Note:</b> To use the auto update options, enable the <b>Set Time with NTP Server on boot</b> setting on the <b>Device Configuration Page</b> page.</p>
Infinite Retries (Not Recommended):	Selected will try to get the AutoProvisioning file from provisioning server until the file is loaded.
	Click the <b>Save</b> button to save your configuration settings. <b>Note:</b> You need to reboot for changes to take effect.
	Click on the <b>Reboot</b> button to reboot the device.

**Table 9 - Autoprovisioning Configuration Parameters**

7.9. Configure Update Firmware

Click the Update Firmware button to open the Update Firmware page (Figure 13).

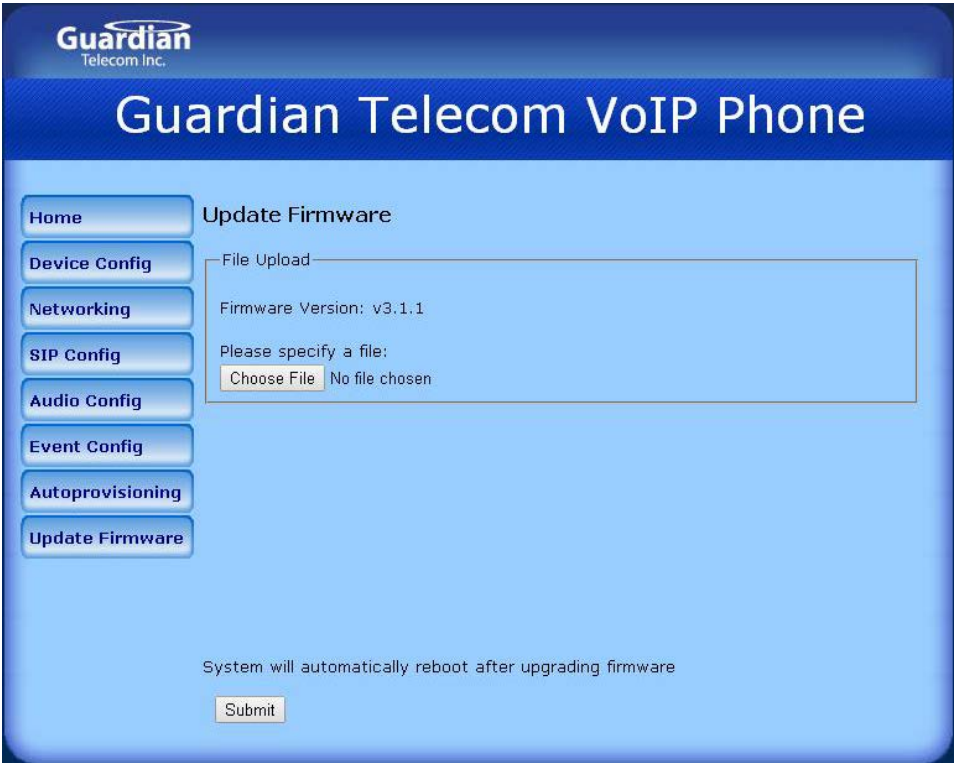


Figure 13 - Update Firmware Page

Web Page Item	Description
File Upload	
Firmware Version:	Shows the current firmware version.
Please specify a file	Select a firmware file on your system to load to the device.
<div>Submit</div>	Click on the <b>Submit</b> button to automatically upload the selected firmware, the device will automatically reboot.

Table 10 - Firmware Update Parameters

7.9.1. Reboot the Telephone

After a firmware download, the telephone will automatically initiate a reboot. To manually reboot a Telephone, log in to the web page as instructed in Section 7.2, "Log in to the Configuration Home Page". Click on reboot on any of the active pages which provide that function.



## 8. Setting up a TFTP Server

Autoprovisioning requires a TFTP server for hosting the configuration file.

### 8.1. In a LINUX Environment

To set up a TFTP server on LINUX:

1. Create a directory dedicated to the TFTP server, and move the files to be uploaded to that directory.
2. Run the following command where `/tftpboot/` is the path to the directory you created in [Step 1](#): the directory that contains the files to be uploaded. For example:

```
in.tftpd -l -s /tftpboot/your_directory_name
```

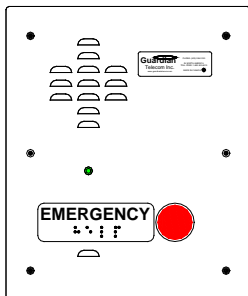
### 8.2. In a Windows Environment

You can find several options online for setting up a Windows TFTP server. This example explains how to use the Solarwinds freeware TFTP server.

To set up a TFTP server on Windows:

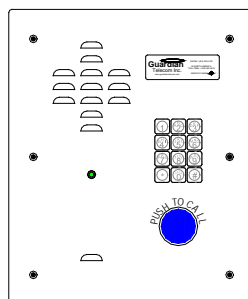
1. Install and start the software.
2. Select File/Configure/Security tab/Transmit Only.
3. Make a note of the default directory name, and then move the firmware files to be uploaded to that directory.

## 9. Operation



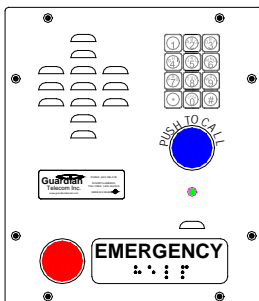
### HDE-11-VoIP & HDE-12-VoIP EMERGENCY TELEPHONES

Press the EMERGENCY button. The pre-programmed telephone number automatically dials. If the unit is equipped with an external signaling device it will also activate. The LED will indicate when a connection has been made. A conversation can now take place.



### HDE-1100-VoIP CALL TELEPHONE

**TO CALL** - Press the PUSH TO CALL button, wait for dial tone then dial the number. If a speed dial extension was programmed in the phone will automatically dial that extension. A conversation can now take place.



### HDE-1200-VoIP EMERGENCY/CALL TELEPHONE

**EMERGENCY** - Press the EMERGENCY button. The pre-programmed telephone number automatically dials. If the unit is equipped with an external signaling device option it will also activate. The LED will indicate when a connection has been made. A conversation can now take place.

**TO CALL** - Press the PUSH TO CALL button, wait for dial tone then dial the number. If a speed dial extension was programmed in the phone will automatically dial that extension. A conversation can now take place.

**NOTE:** The Emergency call always takes **priority** over a regular call. In the event the emergency button is pressed while a regular call is in progress the regular call will be disconnected and the emergency call will proceed.

## 10. Frequently Asked Questions

### 1. When I set my device network mode to “Static”, after I save and reboot the unit, the count down time continuously recycles and never exits.

Once the IP address is physically changed to static mode, the webscreen does not know to change addresses and will attempt to re-connect to the original IP address. The solution is to monitor the timer and after it hits 0:00 on the first cycle close the browser and use the discovery or open a new browser to the new IP address.

### 2. I have plugged my device into the network and it can be discovered but the web interface will not open up?

In some instances when the device is initially plugged in, some of the registers may not be configured automatically with the correct data. Press and hold the device reset button for >10sec until the device announces that it is restoring factory settings. Wait approximately 1 min for the device to complete the reboot cycle and then retry.

### 3. I have plugged in my device to my network but the Discovery tool does not see the device.

There are several possibilities:

- Verify the device is powered up. You should see the Data/Ack LED's on the LAN connector on steady or blinking.
- Ensure you wait approx 45 sec for the boot up sequence to complete.
- Confirm your IP address – Press and hold reset button for approx 2 seconds then release. The IP address will be announced through the hands free speaker or the handset earpiece.
- If IP address is returned as 10.10.10.10 then the device failed to reach the DHCP server and it reset the IP to factory default. – Verify network settings.
- Ensure multiple new devices were not added to network without pre-configuring the network settings – Static Network only!

### 4. How do I update my firmware?

1. Contact Guardian support at sales@guardiantelecom.com or at 1-800-363-8010 to obtain the most recent firmware version.
2. Extract the ulmage file to the computer that will be used to perform the update on the device.
3. With the HDE active on the network open the web interface to the device you wish to update.
4. Go to the Update Firmware update page.
5. Browse for the new ulmage file.
6. Click Select to initiate the update process. The web timer will be displayed. Once the update is completed the unit will return to the web interface.
7. Confirm version displayed on the web interface.

### 5. For additional support or answers to questions not covered on this page, who should I contact?

Contact Guardian Telecom VoIP Technical Support.

### 6. When dialling the three-digit DTMF tone on the IP phone, I can hear the DTMF-tones coming out of the speaker of the VoIP device but there is no relay action. The relay works when using the relay test-button on the configuration software. How do I fix this?

Since the relay test button is working, it seems like the problem results from interfacing with the IP phone where the DTMF tone is generated. To resolve this problem verify that the DTMF tone on the phone is set to out-of-band.

### 7. I was able to register your device with our SIP server, but when I tried to enter a DTMF tone there was no function.

Make sure your SIP phone is set to 101 for the DTMF payload type (Out of Band RFC2833).

### 8. After a period of time, my device stops working or is unreachable.

This is a common problem when the re-registration time value is not set correctly.

On a Guardian VoIP device, you need to make sure that the re-registration time value (in minutes) is **less than** that is set on the IP-PBX server.

9. On an Asterisk-based VoIP SIP PBX system, the Guardian SIP Device status is "Busy" or "Unreachable". I have set up both the Guardian VoIP SIP device and the PBX extension information for the device. I can see the device on the network, am able to PING it, and can bring up the device web page with a browser. However, when I try to call it from a phone extension, I see the word "Busy" or "Unreachable" in the Asterisk log.

In the PBX setup page for the extension of the Guardian device, find the **Qualify=** value and change it to **NO**. If the **Qualify=** value requires a numeric value, then change it to **0**.

Note that on some Asterisk systems (such as **Intuitive Voice**) this value is called the **Heartbeat=** value. Set the **Heartbeat=** value to **NO**, and then save the settings.

Also, on the product's **SIP Setup** page, make sure that the **Register Expiration (minutes)** setting is set to less than **6** minutes (**5** minutes is good) because it needs to be a value less than the Asterisk default value of 6 minutes. Save the settings after changing the **Register Expiration (minutes)** setting.

10. What type of audio files can be uploaded into the device?

RIFF (little-endian) data, WAVE audio, Microsoft PCM, 16 bit, mono 8000 Hz

You can use the free utility 'Audacity' (<http://audacity.sourceforge.net/>) to convert audio files into a format the device can recognize.

When you export an audio file with this program, you can save the output as "WAV (Microsoft) signed 16 bit PCM."


11. On the V2 products, what happens during a firmware upload if the process gets corrupted?

To guard against failed firmware upgrades, units shipped from Guardian Telecom with V.0.0.29 feature a built-in "fail safe" mechanism.

The Device will store the "TFTP Server IP" and "New Filename" entered on the "Update Firmware" web page. If, during the boot process, the Device is unable to boot the firmware, it will attempt to download the stored image from the stored TFTP server.

12. I see in the electrical connection diagram in the users guide that there is a High PIV Ultra Fast switching diode. Do I need it and if so do you have a source?

This High PIV Ultra Fast switching diode prevents CEMF kick back from an intermediary relay coil when power is cut and the coil field collapses. You could use an On-Semi MUR105 diode or an IN4007, which is readily available.

Specifications Datasheet	
Digi-Key Part Ordering	<a href="#">Ordering Page</a>

13. We have the Cisco 3550 switch and it looks like the unit is not able to negotiate the power with the switch. It keeps cycling over and over.

This happens because with default settings, the switch port is resetting power too quickly. Therefore, on the 3550 switch, on the switch port that the unit is attached to, please try adding the following CLI command:

**power inline delay shutdown 20 initial 300**

That should keep power supplied until the unit can boot up all the way.

14. The Guardian device connected to a Dell Powerconnect 3524P or 3548P port did not stay linked up. The device comes up, and then it goes down, and then it comes up cycling.

If connected to a Linksys SRW208MP switch, the Guardian device stays up.

Pantel, Cisco, or Linksys PoE endpoints all work on the Dell Powerconnect 3524P.

A user tried hard coding the switch speed/duplex and tried four different Dell Powerconnect 3524P switches. --- Check to make sure the Dell Powerconnect 3524P has flow control enabled on the port for the Guardian device to power up properly.

According to Dell:

**Flow Control Support (IEEE 802.3X):** Flow control enables lower speed devices to communicate with higher speed devices by requesting that the higher speed device refrains from sending packets. Transmissions are temporarily halted to prevent buffer overflows. For information on configuring Flow Control for ports or LAGs, see "Defining Port Configuration" or "Defining LAG Parameters" in the **Dell™ PowerConnect™ 35xx Systems User's Guide**.

**15. I am unable to connect with the unit when it is plugged into a Cisco SLM 224P switch.**

The cables or switch ports that you are connecting to are set in **switch** or **hub mode** instead of **endpoint stations**. The **MDIX** setting needs to be changed to **MDI** since Guardian VoIP products are end stations.

From the Cisco SLM 224P User Guide:

**Change to MDI:**

**MDI / MDIX** Displays the Media Dependent Interface (MDI) / Media Dependent Interface with Crossover (MDIX) status on the port. Hubs and switches are deliberately wired the opposite of the way end stations are wired, so that when a hub or switch is connected to an end station, a straight through Ethernet cable can be used, and the pairs are matched up properly. When two hubs or switches are connected to each other, or two end stations are connected to each other, a crossover cable is used to ensure that the correct pairs are connected. The possible field values are: - **MDIX** Use for hubs and switches. - **MDI** Use for end stations.

**16. I have a Cisco 6513 switch. When I plug in a Guardian VoIP device, the device constantly reboots and will not register to the SIP server.**

Adjust the switch power selection to **Power inline static**, as opposed to **Power inline dynamic**. This will allow the device to continuously receive 15.4W of power.

**17. We have registered the device on Cisco Call Manager but are unable to register the device's Nightringer using the device's MAC address. How do I configure two extensions on Call Manager for the device?**

Create a second directory number and user for the Nightringer extension. You may need to fudge a digit of the MAC Address so that Call Manager allows you to associate it to the new user. Be sure this MAC address does not match any other Guardian devices on your network.

## 11. Product Specifications

11. Product Specifications	
<b>Category</b>	
Ethernet I/F	10/100 MBPS
Protocol	SIP RFC 3261 Compatible
Power Input	802.3AF Compliant or 12-24Volts at 500mA PoE
Payload Types	G711, A-law and $\mu$ -law

## 12. Appendix A Time Zone Settings

The posix time zone string tells the internal date and time utilities how to handle daylight savings time for different time zones. Shown below are some common strings.

Common Time Zone Strings

Time Zone	Time Zone String
US Pacific Time	PST8PDT,M3.2.0/2:00:00,M11.1.0/2:00:00
US Mountain Time	MST7MDT,M3.2.0/2:00:00,M11.1.0/2:00:00
US Eastern Time	EST5EDT,M3.2.0/2:00:00,M11.1.0/2:00:00
Phoenix Arizona <sup>a</sup>	MST7
US Central Time	CST6DST,M3.2.0/2:00:00,M11.1.0/2:00:00

a. Phoenix, Arizona does not use daylight savings time.

Shown below is a breakdown of the parts that constitute the following time zone string:

- **CST6XXX,M3.2.0/2:00:00,M11.1.0/2:00:00**

Time Zone String Parts

Time Zone String	Meaning
<b>Part</b>	
<b>CST6CDT</b>	The time zone offset from GMT and three character identifiers for the time zone.
CST	Central Standard Time
6	The (hour) offset from GMT/UTC
XXX	Any three letters
<b>M3.2.0/2:00:00</b>	The date and time when daylight savings begins.
M3	The third month (March)
.2	The 2nd occurrence of the day (next item) in the month
.0	Sunday
/2:00:00	Time of day to change
<b>M11.1.0/2:00:00</b>	The date and time when daylight savings ends.
M11	The eleventh month (November)
.1	The 1st occurrence of the day (next item) in the month
.0	Sunday
/2:00:00	Time of day to change

More examples of time zone strings.

Time Zone String Examples

Time Zone	Time Zone String
Tokyo <sup>a</sup>	IST-9
Berlin <sup>b</sup>	CET-1MET,M3.5.0/1:00,M10.5.0/1:00

a. Tokyo does not use daylight savings time.

b. For Berlin, daylight savings time starts on the last Sunday in March at 01:00 UTC, and ends on the last Sunday in October at 01:00 UTC, and is one hour ahead of UTC.

**Time Zone Identifier**

A user-definable three or four character time zone identifier (such as PST, EDT, 1ST, MUT, etc) is needed at the beginning of the posix time zone string to properly set the time. However, the specific letters or numbers used for the time zone identifier are not important and can be any three or four letter or number combination that is chosen by the user. However, the time zone identifier cannot be blank.

**Three or Four Character Time Zone Identifier**

**PST8PDT,M3.2.0/2:00:00,M11.1.0/2:00:00**

Three or four character time zone identifier at the beginning of the time zone string.  
The identifier can be any three or four letters or number combination chosen by the user.

You can also use the following **URL** when a certain time zone applies daylight savings time:

<http://www.timeanddate.com/time/dst>

More information about the GMT time in various time zones.

**World GMT Table**

<b>Time Zone</b>	<b>City or Area Zone Crosses</b>
GMT-12	Eniwetok
GMT-11	Samoa
GMT-10	Hawaii
GMT-9	Alaska
GMT-8	PST, Pacific US
GMT-7	MST, Mountain US
GMT-6	CST, Central US
GMT-5	EST, Eastern US
GMT-4	Atlantic, Canada
GMT-3	Brasília, Buenos Aries
GMT-2	Mid-Atlantic
GMT-1	Cape Verdes
GMT	Greenwich Mean Time
GMT+1	Berlin, Rome
GMT+3	Israel, Cairo
GMT+4	Abu Dhabi, Muscat
GMT+5	Islamabad, Karachi
GMT+6	Almaty, Dhaka
GMT+7	Bangkok, Jakarta
GMT+8	Hong Kong, Beijing
GMT+9	Tokyo, Osaka
GMT+10	Sydney, Melbourne, Guam
GMT+11	Magadan, Soloman Is
GMT+12	Fiji, Wellington, Auckland



**Guardian Telecom Inc.**  
**Toll-free 1-800-363-8010**  
**Phone (403) 258-3100**  
**Fax. (403) 253-4967**  
**www.guardiantelecom.com**  
**E-mail: [sales@guardiantelecom.com](mailto:sales@guardiantelecom.com)**  
(Click to open message box)

***CONNECTED. PROTECTED.***

© Guardian Telecom Inc. 2017