Pub. 42004-548A

GAI-TRONICS®
A HUBBELL COMPANY

# VoIP Telephone
# Basic Configuration Guide

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# Basic Configuration Guide

## Confidentiality Notice

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## Introduction

This guide provides information for basic configuration and programming of GAI-Tronics' second generation VoIP telephones. Pub. 502-20-0171-001 Iss. 4 covers the advanced features of these devices. Select the *web support* link on the unit's embedded webpage to access the document. This document is also located on GAI-Tronics' website at http://www.gai-tronics.com.

**NOTE:** All references to the telephones in this document are for GAI-Tronics RED ALERT®, SMART Industrial, or VoIP/WiFi telephones.

For questions about configuring VoIP telephones, please contact:

**Service Group**
**GAI-Tronics**
**3030 Kutztown Road**
**Reading, PA 19605**
**800-492-1212 (8 a.m. to 5 p.m. EST) 610-777-1374 outside the USA**

## Set up and Configuration

Configure each VoIP telephone for use on the intended network before installation. Most models have memory-dial locations that require set up. Configure the telephones using one of two methods:

- web page (the simplest and quickest method for configuring an individual phone)
- configuration file download (the most efficient method for multiple updates)

**NOTE:** Both of the above access methods require the unit's username and password. Securely record the password once set or changed.

This basic configuration guide provides information to configure GAI-Tronics' VoIP telephones for basic operation using the embedded webserver. For complete information on configuring GAI-Tronics VoIP telephones using the embedded webserver or by configuration file download, please see Pub. 502-20-0171-001 Iss. 4 (see the Reference Documentation section).

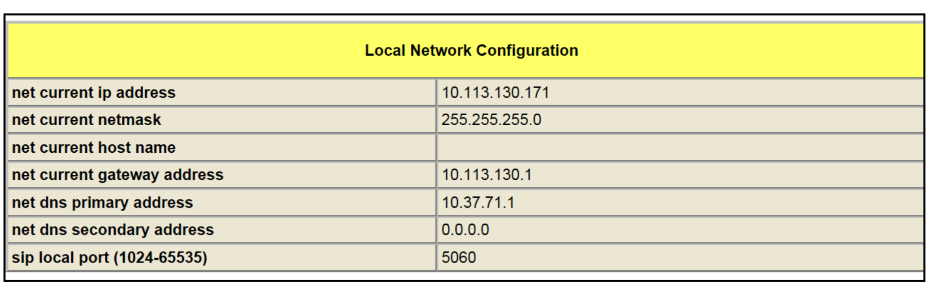
# Web Page Structure

The webpages for GAI-Tronics' VoIP telephones have a common header section showing the name of the device, the current user name, links to HOME, CONFIGURATION, and STATUS, and controls for the update process (UPDATE, SAVE, RESTORE, REBOOT, HELP, and ADVANCED).


Figure 1. VoIP Telephone Webpage Header

Under the update controls, the website shows the current location in the configuration. The HOME page shows information about the four possible VoIP accounts (SIP servers) (see Figure 2) and local network information about the telephone (see Figure 3):


Figure 2. Home Page Showing VoIP Accounts



| Local Network Configuration | |
|---|---|
| net current ip address | 10.113.130.171 |
| net current netmask | 255.255.255.0 |
| net current host name | |
| net current gateway address | 10.113.130.1 |
| net dns primary address | 10.37.71.1 |
| net dns secondary address | 0.0.0.0 |
| sip local port (1024-65535) | 5060 |

Figure 3. Home Page—Local Network Configuration

Select CONFIGURATION to access the configuration pages (shown in a bar across the top) and the various sub-pages (shown down the left-hand side of each page). The list of sub-pages down the left side changes according to the page selected from the main navigation bar. The current position in the configuration structure appears below the main page bar under the update controls.

The ADVANCED button toggles the lists of main and sub-pages between a basic set and an advanced set containing complete detail.

**NOTE:** Basic mode hides the advanced parameters located on individual sub-pages.

## Web Page Controls

The following controls appear on every web page:

- UPDATE—commits changes to any parameter(s) on the current page. Navigating to a different page without clicking update loses any changes made. Update changes the parameter immediately but does not permanently save the change (i.e. rebooting or power cycling the unit before clicking SAVE loses all changes).
- SAVE—saves the current configuration to flash memory.
- RESTORE—restores the telephone to its last saved configuration.
- REBOOT—performs a soft reboot of the telephone.
- HELP—loads the help document.
- BASIC/ADVANCED—toggles the main menu bar between a basic set of options and the advanced options containing all settings for the device.

The web page structure within the CONFIGURATION section is below (see Figure 4). Highlighted pages are only visible in advanced mode.

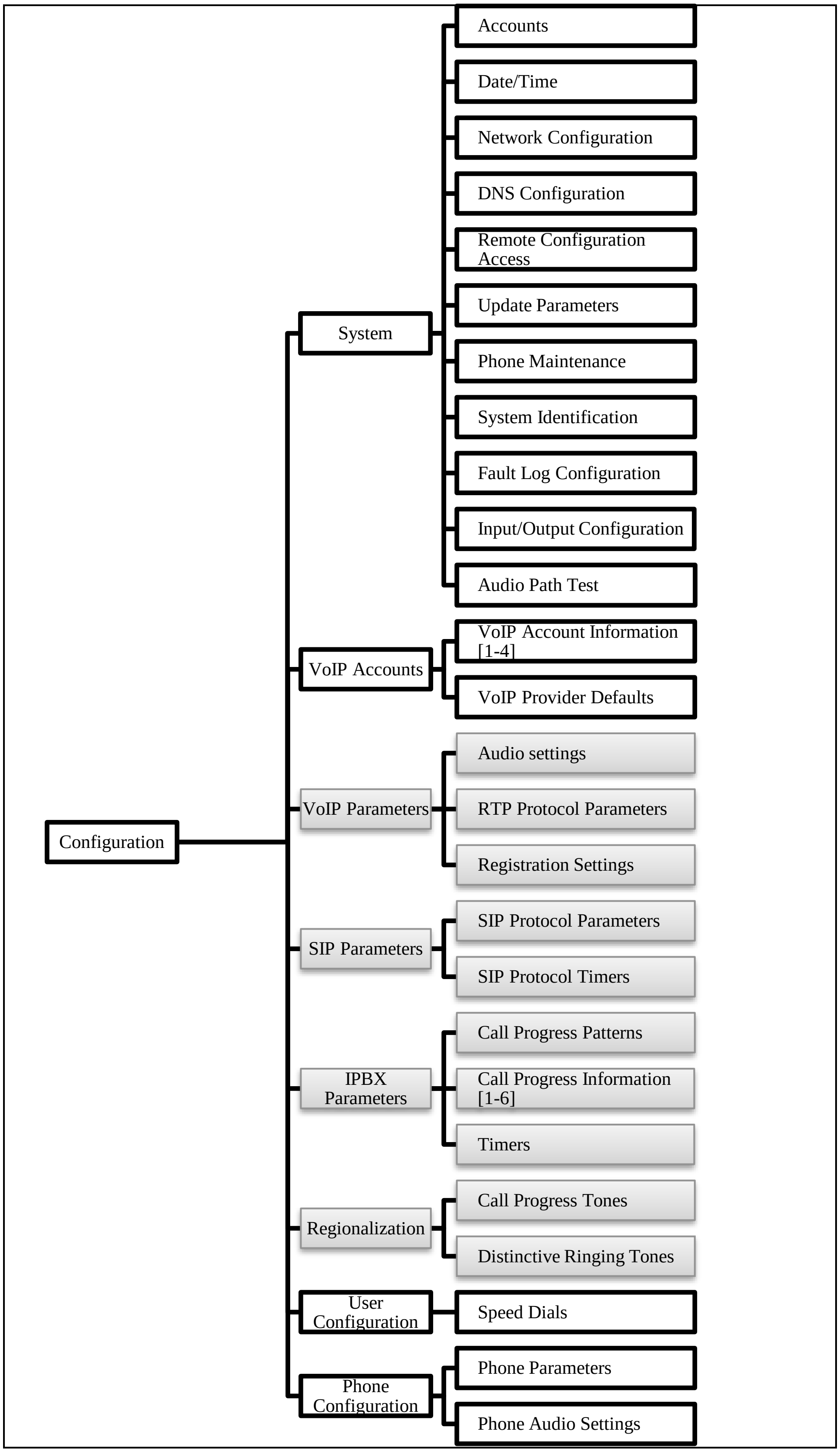



Figure 4. Configuration Pages Hierarchy

# Quick Start

The factory default settings are appropriate for most installations. Complete the following steps for default operation:

1. Provide an Ethernet connection and power (either 24–48 V dc or PoE).

   GAI-Tronics' VoIP Telephones are factory configured for DHCP with a failover to a static IP address set to 192.168.1.2 with subnet mask set to 255.255.255.0 for networks without DHCP.

2. Ensure that a DHCP server is available on the network and the telephone is accessible at the IP address allocated to it. If DHCP fails (or there is no DHCP server) the telephone reverts to the default static IP address of 192.168.1.2 after a few minutes

3. Use a web browser to access the device's configuration website at the VoIP Telephone's IP address.

4. Enter the username and password (Defaults: ***user*** & ***password***).

5. Click the CONFIGURATION link at the top of the home page and then click the VOIP ACCOUNTS link to access the VOIP ACCOUNT 1 INFORMATION page.

6. Enter the following parameters on the VOIP ACCOUNT 1 INFORMATION sub-page: (see the VoIP Accounts section).

   - **username**—Set to the extension assigned on the SIP server.
   - **domain name, proxy domain name, and register domain name**—Set these three parameters to the IP address of the SIP Server.
   - **Auth user password**—Enter the authentication password configured for the extension on the SIP server.
   - **Provider and Register**—Set to ***Enable*** and click SAVE.

   **NOTE:** Click the UPDATE or SAVE button before leaving any web page to keep the changes.

7. Verify the registration state changes to REGISTERED (only when unit connects to the network with the SIP server).

8. *If not using DHCP*:

   1. Click the CONFIGURATION link, at the top of the page.
   2. Click the SYSTEM link.
   3. Click the NETWORK CONFIGURATION button on the left side to access the NETWORK CONFIGURATION page (see the Network Configuration section).
   4. Set the IP address, subnet mask, and gateway address.
   5. Set the NET ISP DHCP ENABLE setting to ***Disable.***

9. *For an Autodial Telephone: (354-711 series or similar):*

   1. Click the CONFIGURATION link, at the top of the page.
   2. Click the USER CONFIGURATION link, to access the SPEED DIALS page.
   3. Enter the desired IPBX SPEED DIAL 1 field destination of an existing (user name/ID) on the PBX/SIP server

The telephone can make and receive calls with these basic steps. To show the telephone registration and what happens during calls:

1. Click the STATUS link, at the top of the page.
2. Click on the GENERAL STATUS or SIP STATUS links, on the left side of the page.

The information displayed helps diagnose problems.

**NOTE:** *If not using DHCP*: Configure each unit with a basic configuration before installing it. All units have identical settings as factory defaults, so configure each one individually to give it a unique identity on the network. This may be difficult to do after unit installation.

# Configuration

Configure parameters by modifying the value for the field. Click the UPDATE button before leaving a web page to keep the current changes and/or click the SAVE button to make the change permanent.

## System

### Accounts

The ACCOUNTS page contains the parameters used by the VoIP telephone to access its administrative website.

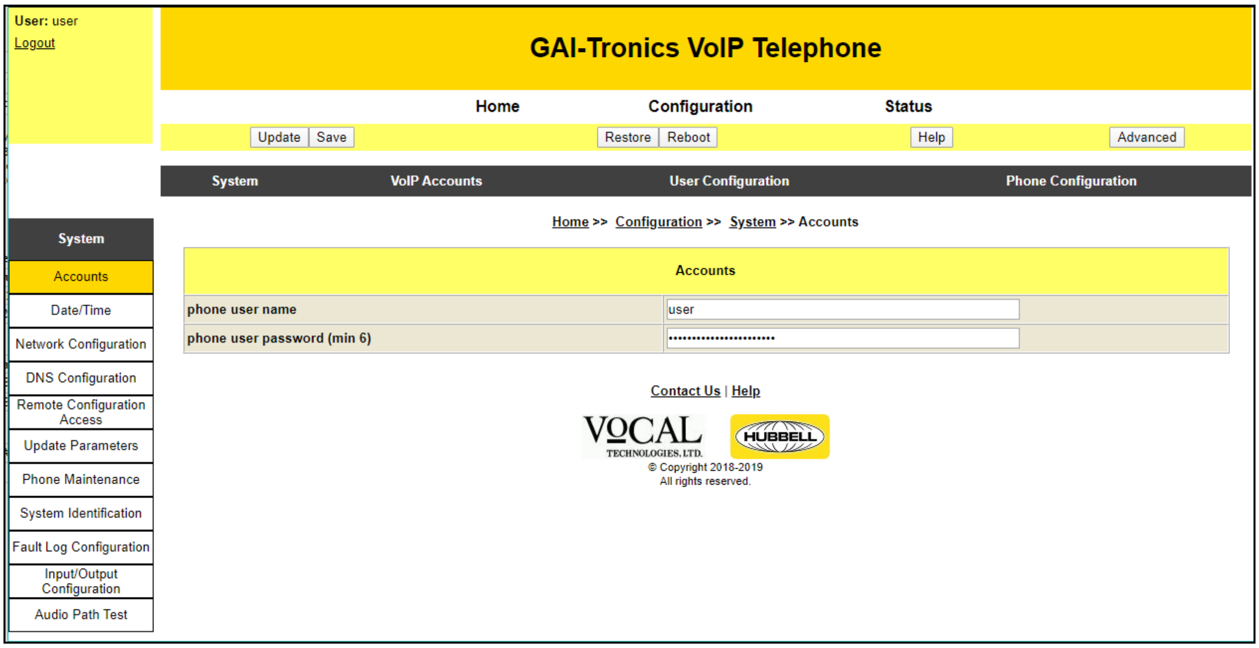



Figure 5. Accounts Page

Table 1. Accounts Parameters

| **Parameter** | **Function** |
|---|---|
| **phone user name** | The user ID used to access the VoIP telephone's built in configuration website. Default: ***user*** |
| **phone user password (min 6)** | The password required to authenticate the user account to access the configuration website on the VoIP telephone (minimum password length is six alphanumeric digits. Default: ***password*** |

## Date/Time

The DATE/TIME page contains the parameters to configure the time and date settings for the VoIP telephone. The telephones do not include a battery-backed real time clock but will keep time based on updates from an SNTP server.

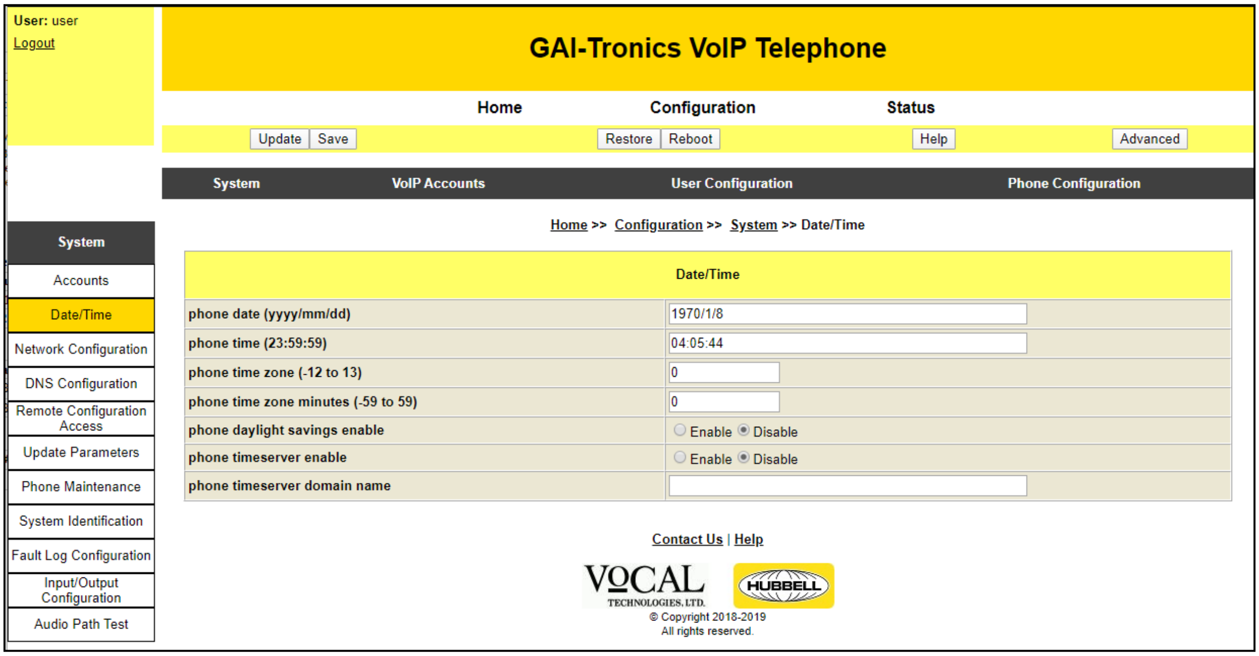



Figure 6. Date/Time Page

Table 2. Date/Time Parameters

| **Parameter** | **Function** |
|---|---|
| **phone date (yyyy/mm/dd)** | The current date on the telephone. Default: ***1970/1/8*** |
| **phone time (23:59:59)** | The current time on the VoIP telephone. Defaults to the current time. |
| **phone time zone (-12 to 13)** | Enter the time zone relative to GMT. |
| **phone time zone minutes (-59 to 59)** | Enter the offset minutes from the time zone selected. Default: ***0*** |
| **phone daylight savings enable** | Enable \| ***Disable*** |
| **phone timeserver enable** | Enable \| ***Disable*** |
| **phone timeserver domain name** | The IP address or FQDN of the SNTP server that this VoIP telephone will query for accurate time. Default: ***blank*** |

## Network Configuration

Use the NETWORK CONFIGURATION page to display or change the settings for the IP network connection. Configure the settings described below as the first step in completing the basic configuration (see Table 3).

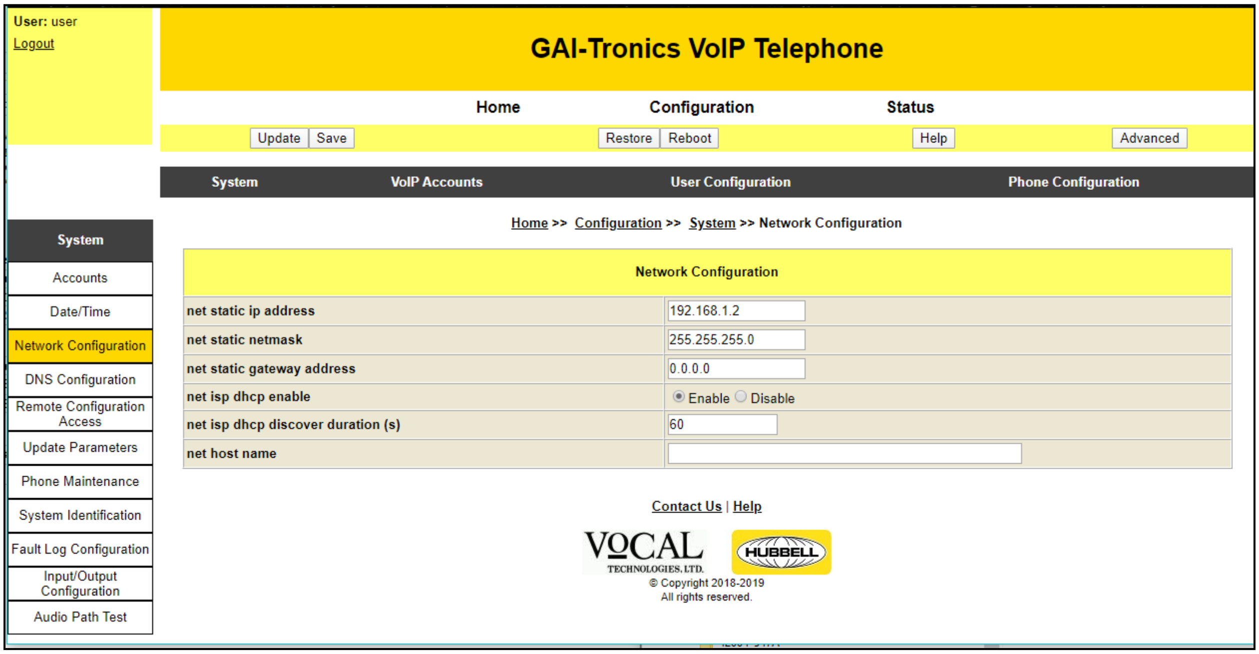



Figure 7. Network Configuration Page

Table 3. Network Configuration Parameters

| **Parameter** | **Function** |
|---|---|
| **net static ip address** | The static IP address the unit assumes on a network when not configured for DHCP or after the DHCP discovery times out.<br>Default value: ***192.168.1.2*** |
| **net static netmask** | The subnet mask for the network when static IP addressing is used.<br>Default value: **255.255.255.0** |
| **net static gateway address** | The IP address for the default router when static IP addressing is used.<br>Default value: **0.0.0.0** |
| **net isp dhcp enable** | Enables or disables the use of DHCP for the assignment of IP parameters. If this value is set to DISABLE, the telephone will use the Static IP values.<br>Values available: **Enable** or **Disable** Default value: ***Enable*** |
| **net isp dhcp discover duration (s)** | The time allowed for the device to obtain an IP address from a DHCP server before applying the **net static ip address**. |
| **net host name** | The DNS host name for the device. Default value: **BLANK**. |

## DNS Configuration

Enter the IP addresses for the primary and secondary DNS servers. DNS servers provide name to IP address resolution on an IP network. Telephones configured by DHCP ignore these settings.

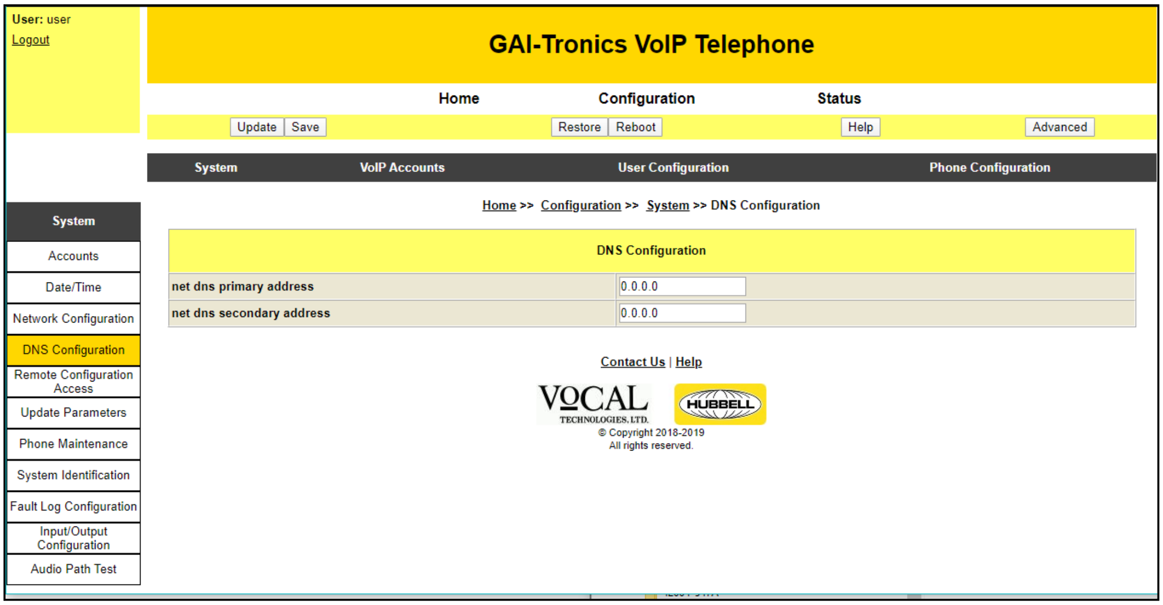



Figure 8. DNS Configuration Page

Table 4. DNS Configuration Parameters

| Parameter | Function |
|---|---|
| **net dns primary address** | Sets the IP address of the primary static DNS server. The telephone does not use the DNS server setting if DHCP is enabled. Default value: ***0.0.0.0*** |
| **net dns secondary address** | Sets the IP address of the secondary static DNS server for redundancy. The telephone does not use the DNS server setting if DHCP is enabled. Default value: ***0.0.0.0*** |

## Input/Output Configuration

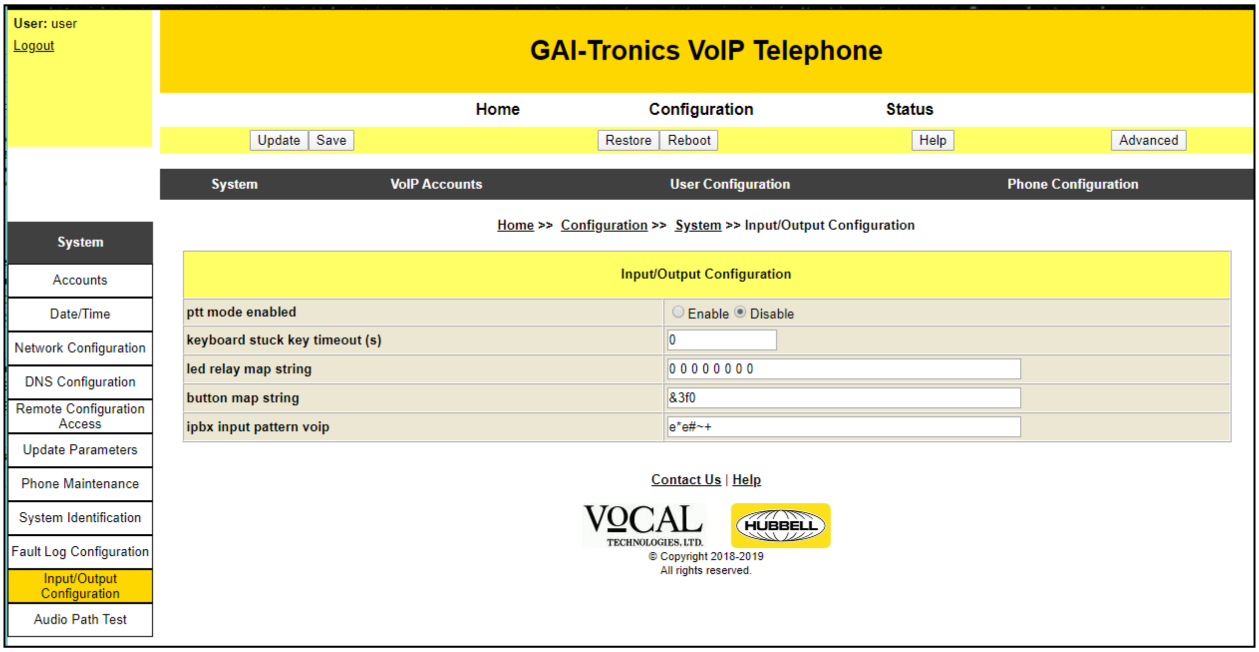



Figure 9. Input/Output Configuration Page

Table 5. Input/Output Configuration Parameters

| **Parameter** | **Function** |
|---|---|
| **ptt mode enabled** | The push to talk button functions as a push to talk/release to listen when enabled. The push to talk button functions as a microphone mute when disabled. Enable \| ***Disable*** |
| **keyboard stuck key timeout (s)** | 0 |
| **led relay map string** | 0 1 0 0 0 0 0 0 |
| **button map string** | 2m1p1l 7 1c3 0 1c2 1c6 1c1 1c5 1c9 1c4 1c8 1c# 1c7 1c0 0 1c* 0 0 0 0 0 0 0 |
| **ipbx input pattern voip** | e*e#~+ |

### LED Relay Map String

LED relay map strings control the function of the eight possible outputs: 4 LEDs, 2 relays, and 2 logical outputs. The string is a series of eight codes separated by spaces. Each code represents the function of one of the outputs. The string position of each code determines the output it controls.

**NOTE:** Outputs vary by device model—refer to product specifications for details.

Table 6. Output String Position

| Position | Output | Notes |
|---|---|---|
| 1 | LED0 | not used |
| 2 | LED1 | normally present in all products |
| 3 | LED2 | not normally present |
| 4 | LED3 | not used |
| 5 | RELAY ONE | typical in all products |
| 6 | RELAY TWO | typical in all products |
| 7 | POWER SELECT LED | normally set to OFF |
| 8 | AUDIO PRESENT | normally set to OFF |

A single numeric character defines the function of each output:

Table 7. Output Function Definitions

| Text Character | Output Function |
|---|---|
| 0 | Always Off |
| 1 | Call progress 1 |
| 2 | Call progress 2 |
| 3 | Call progress 3 |
| 4 | Call progress 4 |
| 5 | Call progress 5 |
| 6 | Call progress 6 |
| 9 | Always On |

Each output function, *Call Progress X,* is defined on the appropriate *call progress information page* (see the IPBX Parameters section).

*Example: 0 1 0 0 2 3 0 0*—LED0 is off, the CALL PROGRESS 1 information page controls LED1, LED2 and LED3 are off, the CALL PROGRESS 2 information page controls relay one, the CALL PROGRESS 3 information page controls relay two, and the POWER SELECT LED and AUDIO PRESENT outputs are off.

### Button Map String

The button map string defines the functions of a telephone’s pushbuttons, keypad keys, 4 logic inputs, and the hookswitch. The strings consist of 23 codes separated by spaces. Each of the 23 codes represents one of the inputs that can exist for a telephone; 18 pushbuttons or keypad keys, four inputs, and the hookswitch.

The position of each code within the string maps it to a button or input. i.e. The first code defines the function of *button one*, the second code defines *button two*, and so on. The final (23rd) code defines the function of the hookswitch.

Table 8. Telephone Inputs

| String Position | Description |
|---|---|
| 1 | Emergency Button (P2) |
| 2 | Call Button (P3) |
| 3 | Keypad Button 3 |
| 4 | Second Call Button (P4) |
| 5 | Keypad Button 2 |
| 6 | Keypad Button 6 |
| 7 | Keypad Button 1 |
| 8 | Keypad Button 5 |
| 9 | Keypad Button 9 |
| 10 | Keypad Button 4 |
| 11 | Keypad Button 8 |
| 12 | Keypad Button # |

| String Position | Description |
|---|---|
| 13 | Keypad Button 7 |
| 14 | Keypad Button 0 |
| 15 | Not used |
| 16 | Keypad Button * |
| 17 | Aux Input (TB1) |
| 18 | Not used |
| 19 | External Input 1 |
| 20 | External Input 2 |
| 21 | External Input 3 |
| 22 | External input 4 |
| 23 | Hookswitch |

#### Input/Button

Each assigned code consists of one or more of the following text characters:

Table 9. Input/Button Function Characters

| Function Number | Description |
|---|---|
| 0 | None |
| 1 | Digit |
| 2 | Memory Dial |
| 3 | PTT/Mute |
| 4 | Redial |
| 5 | Volume |
| 6 | Hook |
| 7 | Hook HF |
| 8 | Memory Hook |

#### Input/Button Modifiers

Apply the following modifiers to appropriate buttons and/or input function codes:

**m#** (where # is a number between 1 and 20)—sets the memory number for button/input function 2 or 8. For example, 2m1 sets it to memory 1.

A memory is defined as a group of one or more speed dials; memory numbers refer to entries in the speed dial group array (see the Speed dial group array section).

**p#** (where # is a number between 0 and 3)—sets the call priority for a memory button. For example, 2m1p1 will set a button to be memory 1 with a priority of 1. Higher numbers have higher priority. Default (if left out) is zero. Any memory button set with a priority >0 can activate the *emergency* call progress state (see the Call Progress Priorities section).

**l**—Prevent local disconnect—prevents the caller from hanging up while in a call by pressing the button again. For example, 2m1l will set a button to start a call by dialing memory 1 but pressing the button a second time will not terminate the call.

**NOTE:** This parameter is a lower-case L. Do not confuse it with the number 1 that also appears in the button map strings.

**c*x*** (where *x* is a digit between 0 and 9, or a # or * character)—sets the digit dialed for the button function 1. For example, 1c9 will cause a button to dial digit 9; 1c* will make it dial a * character.

**NOTE :** *button function 3* (PTT | mute) is a press-and-hold function that has two modes of operation, depending on the setting of the PTT MODE ENABLED parameter. The button operates as a push-to-talk button (press and hold the button to activate the microphone)/release to listen when PTT is enabled. The button functions as a mute button (press and hold to mute the microphone) when disabled.

Code unused buttons or inputs as 0:

Example: 398-711 Handsfree phone with emergency button, call button, and keypad:

2m1p1l 7 1c3 0 1c2 1c6 1c1 1c5 1c9 1c4 1c8 1c# 1c7 1c0 0 1c* 0 0 0 0 0 0 0

**NOTE:** Button map strings vary by device model number.

Example: 246-710: Handset phone with keypad:

0 0 1c3 0 1c2 1c6 1c1 1c5 1c9 1c4 1c8 1c# 1c7 1c0 0 1c* 0 0 0 0 0 0 6

⚠ **CAUTION** ⚠ —Be careful when modifying button mappings. These values are set per device model and in most cases do not require adjustment from the factory settings. Incorrect button map strings may cause abnormal telephone behavior.

### IPBX Input Pattern

Use the input pattern string to set up rules to govern the numbers and sequences dialed using the numeric keypad. Use this to limit the number of digits entered, to prohibit dialing certain numbers, or to enable a termination character. The default setting is **e*e#~+**, meaning that the user can enter up to 255 digits, terminated with either # or *. The full set of characters used to build input pattern rules is set out below (see Table 10).

Table 10. Input Pattern Parameters

<table>
<tr><th>Parameter</th><th>Description</th></tr>
<tr><td>e</td><td>Specify the ending termination digit that follows (usually * or #).<br>**NOTE:** This parameter must occur first in the rule pattern when used.</td></tr>
<tr><td>t</td><td>Set digit timeout to default for current pattern.</td></tr>
<tr><td>x</td><td>Match any numerical digit (0-9).<br>**NOTE:** Include *X* in the rule pattern to abort dialing by dialing ***** or **#**.</td></tr>
<tr><td>~</td><td>Matches any digit (0-9, A-D, *, #) excluding any specified terminators.</td></tr>
<tr><td>r</td><td>Repeat by following a number (1-9), letter (a-z for 10 to 35 times) or *, +, or . to mean any number of times (255 times)</td></tr>
<tr><td>.</td><td>Repeat previous digit any number of times (0 to 255).</td></tr>
<tr><td>+</td><td>Repeat previous digit any number of times (0 to 255).</td></tr>
<tr><td>!</td><td>Disallows pattern.</td></tr>
<tr><td>$</td><td>Indicates secondary dialing to follow—used only by fixed dial strings.</td></tr>
<tr><td><></td><td>Replace group to replace left digit(s) with right digit(s).</td></tr>
<tr><td>[]</td><td>Selection group of candidate digits.</td></tr>
<tr><td>[^]</td><td>Exclusion group of digits.</td></tr>
<tr><td>[0-9]</td><td>Selection range of candidate numerical digits.</td></tr>
<tr><td>[a-d]</td><td>Selection range of candidate letter digits.</td></tr>
<tr><td>s</td><td>Seize on string as only candidate if match to this point.</td></tr>
<tr><td>f</td><td>Pause timeout causes failure instead of dial.</td></tr>
<tr><td>p</td><td>Set digit pause to number of seconds which follow (1–9) for current pattern.</td></tr>
<tr><td>-</td><td>Human readable spacing which is ignored.</td></tr>
<tr><td></td><td>Human readable spacing which is ignored.</td></tr>
<tr><td>|</td><td>Separates different possible rule patterns.</td></tr>
</table>

The input pattern string consists of several different rules, separated by the **|** character, example:

**6xr4|60600!**—allows any 5-digit number starting with 6, except 60600.

**NOTE:** Including "~+" (which allows up to 255 unrestricted digits), in conjunction with any other rule that restricts the number or type of digits may cause a conflict with unexpected results.

Input pattern rules do not apply to memory dials.

# VoIP Accounts

Use the VOIP ACCOUNTS pages to view or change the parameters specific to the SIP signaling protocol. GAI-Tronics' VoIP telephones access up to four SIP proxies. This enables call roll-over to the next SIP server in a prioritized sequence if the telephone is unable to register or make a call.

There is a VOIP ACCOUNT INFORMATION page for each of the four possible endpoints. The four endpoint pages contain the same set of parameters to configure the associated SIP server (see Figure 10). Only one VoIP account needs to be configured for basic operation (see Table 11 for the parameters required for basic operation).

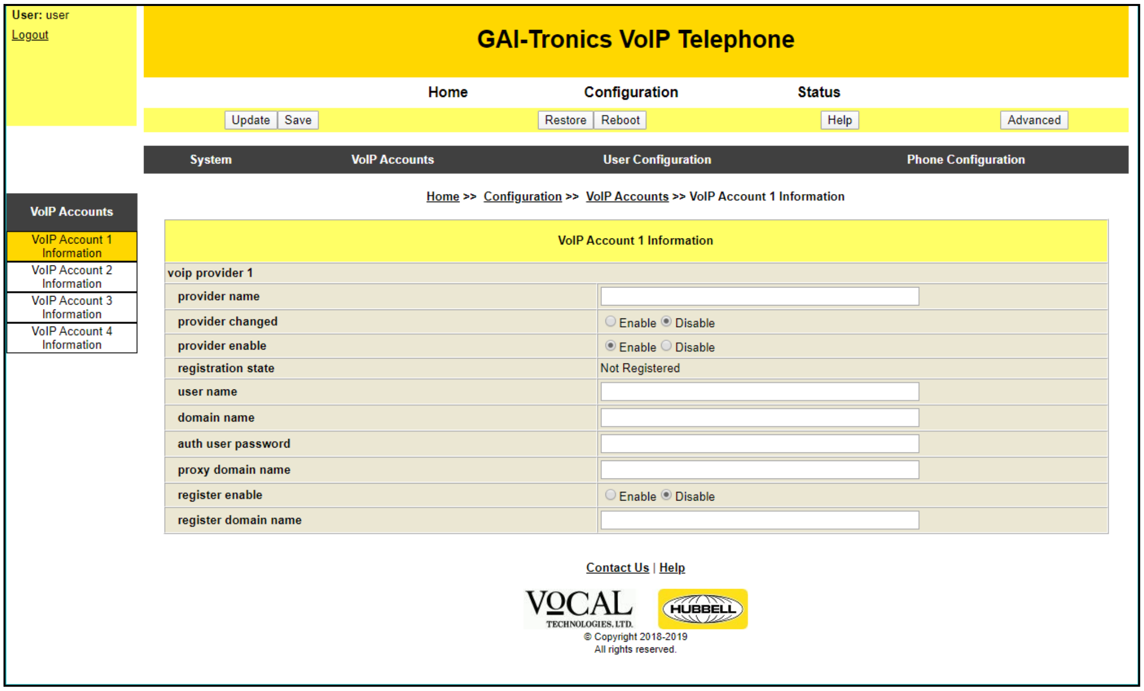



Figure 10. VoIP Account 1 Information Page

Table 11. VoIP Account Parameters

| **Parameter** | **Function** |
|---|---|
| **provider name** | Provider name used for identification purposes. Not used by SIP. |
| **provider changed** | Enable \| ***Disable*** |
| **provider enable** | ***Enable*** \| Disable |
| **registration state** | Display only field—displays either **REGISTERED** or **NOT REGISTERED** |
| **user name** | Extension number on the SIP server<br>Default value: **BLANK** |
| **domain name** | Sets the address of the SIP server, either as an IP address or FQDN. The domain name, registrar address, and proxy may or may not be the same, but the address for the SIP server must be set here.<br>Default value: **BLANK** |
| **auth user password** | Sets the password for the registrar authorization realm.<br>Default value: **BLANK** |

| Parameter | Function |
|---|---|
| **proxy domain name** | Sets the IP address or the FQDN of the SIP proxy server used for incoming/outgoing calls. The domain name, registrar address, and proxy may or may not be the same, but the address for the proxy server must be set here.<br>Default value: **BLANK** |
| **register enable** | Enable \| ***Disable*** |
| **register domain name** | Sets the address of the registrar, either as an IP address or FQDN. The domain name, registrar address, and proxy may or may not be the same, but the address for registration must be set here.<br>Default value: **BLANK** |

# User Configuration

## Speed Dials

The SPEED DIALS page stores the parameters for 20 speed dial entries and the speed dial group array. Speed dial parameters contain strings used by the VoIP telephone to automatically dial frequently dialed numbers. Each VoIP telephone can store 20 speed dial numbers and the IPBX SPEED DIAL GROUP ARRAY parameter.

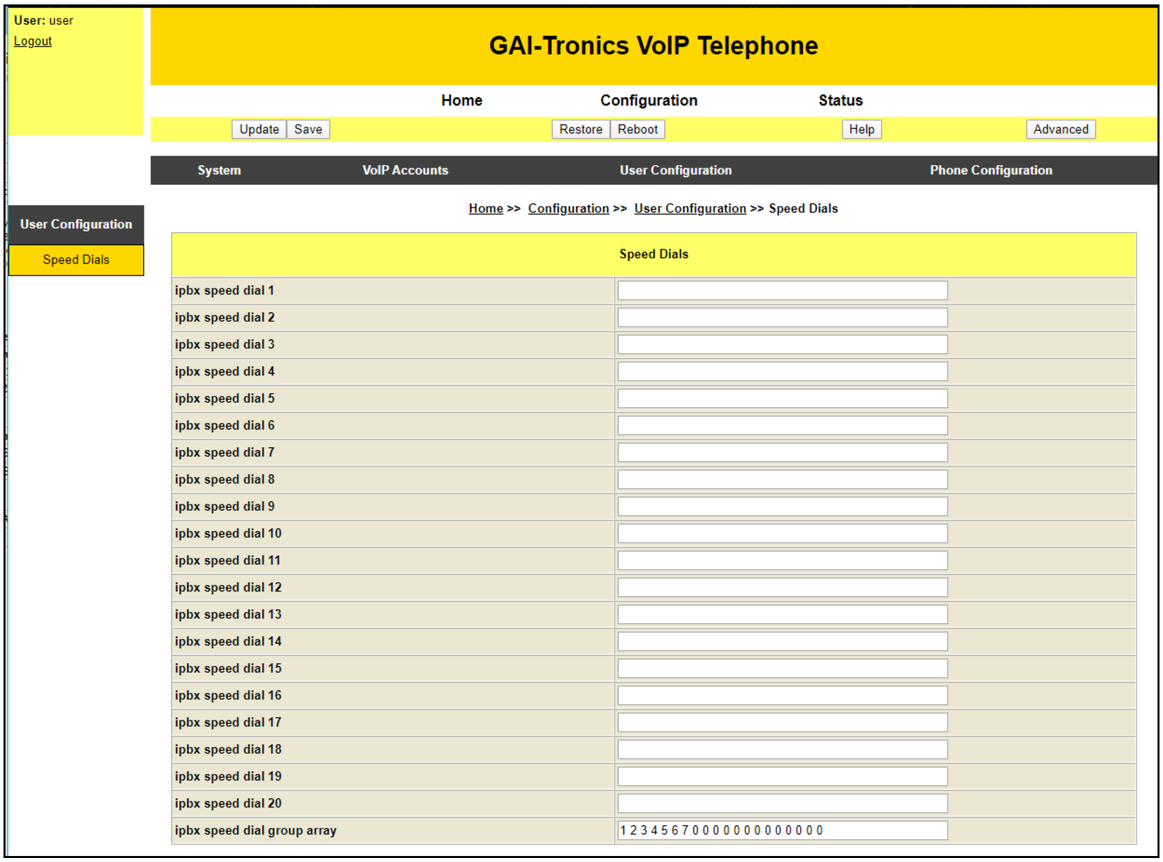



Figure 11. Speed Dials Page

### Speed dial group array

The *speed-dial group array* controls the speed dial entries invoked by each memory. It is a list of 20 numbers separated by spaces. The position of each number in the list maps it to the corresponding *speed dial entry*—i.e. the first number in the list corresponds to speed dial one, the second to speed dial two, and so on up to the 20th speed dial. Each number in the list represents the memory that invokes that speed dial.

A memory is a group of one or more speed dials. Assign memories using the *m* modifier in the button map string (see the Button Map String section). For example, 2m1 assigns a button or input to be memory 1.

Enter up to 20 speed dial entries on the speed dials page.

For example, the following array:

1 2 3 3 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0.

Note that the number 3 occurs twice. The first number in the array is 1, meaning that memory 1 will dial speed dial 1. The second number is 2—memory 2 will dial speed dial 2. The third number is 3, meaning that memory 3 will dial speed dial 3. The fourth number is also 3, meaning that memory 3 can also dial speed dial 4. When activating memory 3, the phone will first dial speed dial 3; if the call fails, it will then dial speed dial 4.

The telephone works through the speed dial group array from left to right upon memory activation. If it finds a match for the memory number, it will attempt to dial the speed dial corresponding to the position in the array. If the call fails, it will continue along the array to find another match and attempt to call that speed dial, and so on, until either the call succeeds or there are no more matches.

Set unused speed dials to 0 in the array.

**NOTE:** Because the telephone reads the array from left to right, using speed dials in a failover sequence as described above requires that the speed dial entries be in the same order as the sequence.

Example: 1 1 1 2 2 2 3 3 3 4 4 4 4 4 0 0 0 0 0 0

- a button (or input) assigned as memory 1, will first attempt to call *speed dial 1*. If that fails, it will attempt *speed dial 2*; if that fails *speed dial 3* and if that fails it will abandon the call.
- A button assigned as memory 2 will act similarly using speed dials 4, 5 and 6.
- A button assigned as memory 3 will act similarly using speed dials 7, 8 and 9.
- A button assigned as memory 4 will act similarly, but using five speed dials: 10, 11, 12, 13, and 14.
- In this example, speed dials 15–20 are unassigned.

## IPBX Parameters

Click the ADVANCED button to display all configuration headings. Click the IPBX PARAMETERS link.

VoIP telephones use call progress patterns and priorities to activate LEDs and relays during various stages of a telephone call's progress. For example, setting an LED to indicate call connection, or setting a relay activate when an incoming call rings.

It is possible to use a single output to indicate various stages of a call's progress. For example, the same LED can flash when a call is ringing and then illuminate steadily when the call connects.

Two groups of settings control these options (*patterns* and *priorities*):

- **Patterns**—Assign a pattern, or cadence to each stage of a call's progress on the IPBX PARAMETERS/CALL PROGRESS PATTERNS page.
- **Priorities**—Assign a pattern to an output with a priority sequence on the IPBX PARAMETERS/CALL PROGRESS *N* INFORMATION pages. This allows a single output to indicate more than one call progress stage or status; the priority setting determines the pattern that takes precedence. For example; an LED can be set to indicate both *in use* and *emergency*, with different flashing patterns for each. The priority is set so that the *emergency* pattern takes precedence.

## Call Progress Patterns

Call progress and telephone status combine to activate outputs. Configure the progress/status states on the IPBX PARAMETERS/CALL PROGRESS PATTERNS page:

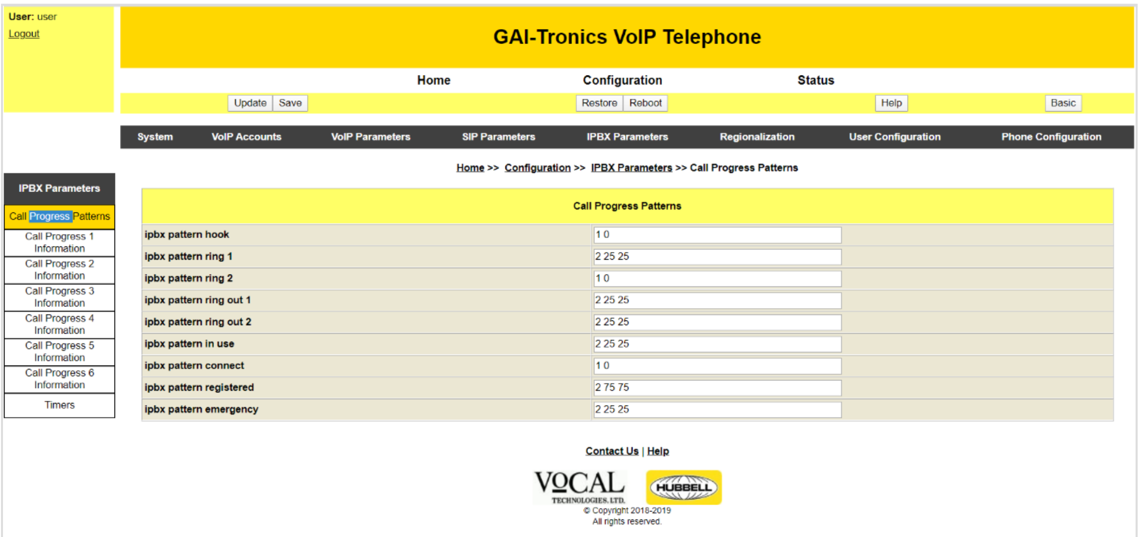



Figure 12. IPBX Call Progress Patterns

The available call progress and telephone status parameters are:

- **Hook**—activates when the telephone is off hook—either when preparing to start a call, when an outgoing call is ringing or when a call is active.
- **Ring 1/Ring 2**—when an incoming call is ringing but not yet answered. There are 2 patterns available to allow different patterns to be set up for different outputs. For example, an LED could be set to flash and a relay could be set to activate continuously during ringing.
- **Ring out 1/Ring out 2**—when an outgoing call is ringing but not yet answered. There are 2 patterns available to allow different patterns to be set up for different outputs. For example, an LED could be set to flash and a relay could be set to activate continuously during ringing.
- **In use**—activates when the telephone is either ringing or in a call
- **Connect**—activates only while a call is connected.
- **Registered**—activates when the telephone is registered with at least one provider.
- **Emergency**—activates when an outgoing emergency call is either ringing or connected. An emergency call is defined as a call initiated by a button set with a priority >0 (see the Button Map String section).

**Call progress patterns** are a list of values indicating the number of on/off transitions and display on/off times (in 10ms periods) according to the following format:

$N\ ON_1\ OFF_1\ ON_2\ OFF_2 \ldots ON_N\ OFF_N$

Where N is the number of on and off transitions counted individually in the pattern and $ON_x$ and $OFF_x$ are interleaved on and off durations in milliseconds.

- Separate values with spaces.
- N may be zero for a permanently off or unused status.
- A value of zero for an on time indicates continuously on.
- A value of zero for an off time turns the output continuously off.
- The maximum number of on and off times counted individually is 9.

For example, to flash an output on and off twice, turning on, then turning off each half-second, the pattern would be 2 50 50.

## Call Progress Priorities

Set *priorities* on one of the six CALL PROGRESS *N* INFORMATION pages (see Figure 13 and Figure 14).

Assign a call PROGRESS INFORMATION PAGE to one or more outputs (see the LED Relay Map String section).

Within each call progress information page, turn each function on by assigning a non-zero priority (i.e. disable functions by setting their priority to zero).

- To activate a single function, set its value to 1 and all the others to zero.
- To give an output multiple functions, give each function a non-zero priority number where 1 is the highest priority, 2 the next highest, and so on.

The example below shows an output set to activate while the telephone is ringing (pattern 1) only:

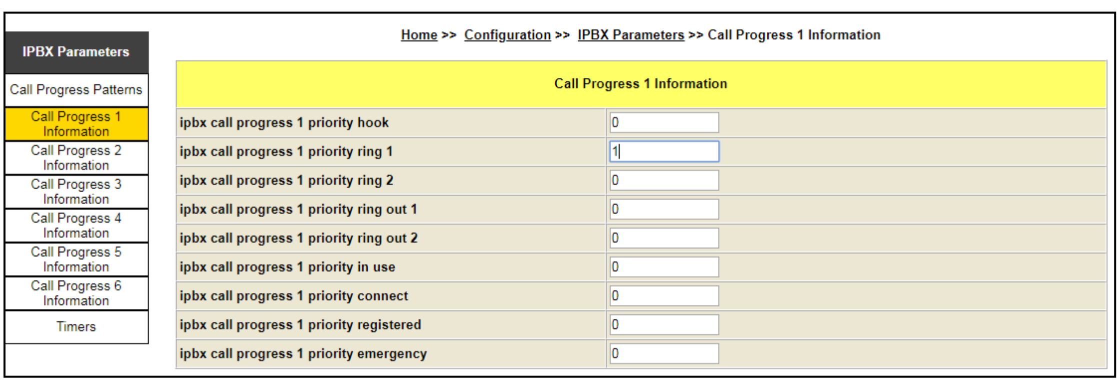



Figure 13. Call Progress 1 Information

The example below shows an output set to activate during ring (pattern 2), when a call is connected, or when the telephone is registered.

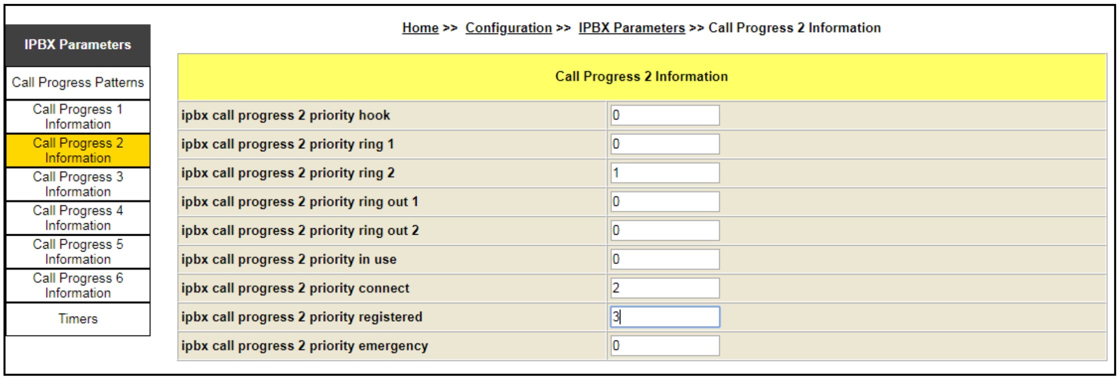



Figure 14. Call Progress 2 Information

In this example, if the phone is ringing, then the pattern assigned for *ring 2* has priority. If the phone is not ringing but the call is connected, then the pattern for *connect* is active. If the phone is nether ringing nor in a connected call, but is still registered, the pattern for *registered* is used. By setting distinctive patterns, the telephone uses a single output to indicate multiple states.

Assign each function a different priority number when using an output to indicate multiple states. The telephone automatically prioritizes functions with the same priority in the order that they appear on the web page, from top to bottom.

# Phone Configuration

## Phone Parameters

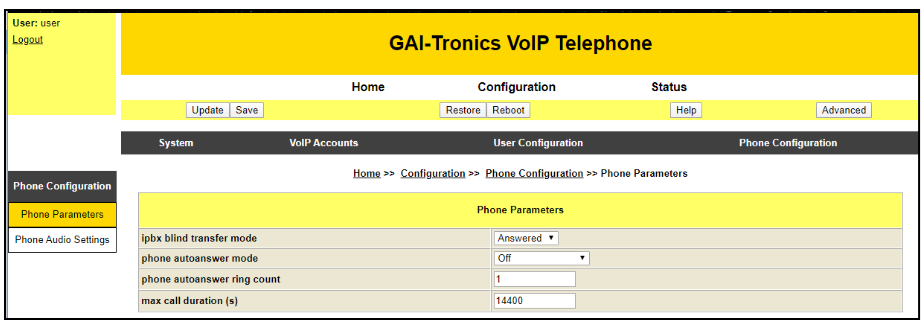



Figure 15. Phone Parameters

Table 12. Phone Configuration Parameters

| Parameter | Function |
|---|---|
| **ipbx blind transfer mode** | Sets the blind transfer mode<br>0—Immediate \| 1—Ringback \| ***2—Answered*** |
| **phone autoanswer mode** | ***Off*** \| Preceding Tone \| Silent Answer \| Babyphone mode \| Page mode |
| **phone autoanswer ring count** | 1 |
| **max call duration (s)** | 14400 |

## Phone Autoanswer Mode:

Preceding Tone:

- The telephone rings (using its default ring tone) before automatically starting a normal 2-way voice call.
- The number of rings is set by the parameter phone_autoanswer_ring_count.
- If the ring tone is continuous (i.e. has no OFF duration), the tone will play for a fixed period equal to phone_autoanswer_ring_count ×10 ms.

Silent Answer:

- The telephone will start a normal 2-way voice call immediately with no preceding tone.

Babyphone Mode:

- The telephone will start a listen-only call immediately with no preceding tone and no voice reception from the calling party.
- The telephone's speaker is muted.

Page Mode:

- The telephone rings (using its default ring tone) before automatically starting a receive-only voice call.
- The telephone's microphone is muted, and the speaker volume is set to maximum.
- The number of rings is set by the parameter phone_autoanswer_ring_count.
- If the ring tone is continuous (i.e. has no OFF duration), the tone will play for a fixed period equal to phone_autoanswer_ring_count × 10 ms.

Off:

The telephone will not auto answer incoming calls.

## Phone Audio Settings

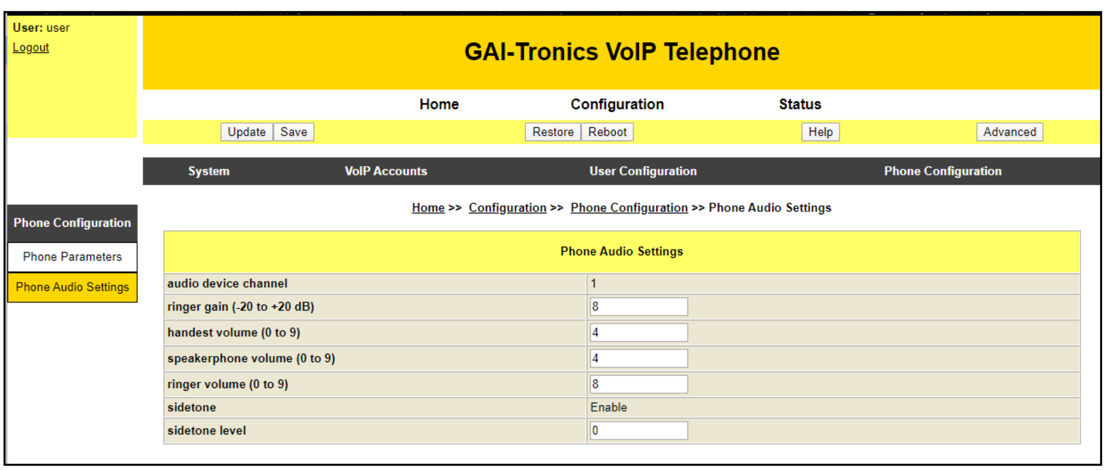



Figure 16. Phone Audio Settings

Table 13. Phone Audio Settings

| Parameter | Function |
|---|---|
| **audio device channel** | handsfree: ***0***, handset: 1 (Factory set for the phones hardware) |
| **handset volume (0 to 9)** | *Handset telephones*: modifies the earpiece level in 10 steps, each step is approximately 2 dB.<br>*Products with volume step control*: sets the starting point for each new call. Default ***4*** |
| **speaker volume (0 to 9)** | *Handsfree products*: modifies the speaker output level in 10 steps, each step is approximately 2 dB.<br>*Products with volume step control*: sets the starting point for each new call. Default ***4*** |
| **ringer volume (0 to 9)** | Modifies the ringer level in 10 steps, each step is approximately 2 dB. Default ***8*** |

# Troubleshooting

With these basic parameters configured, the telephone should operate normally. If additional functionality is needed, please visit GAI-Tronics website (www.gai-tronics.com/products/manuals_specs.htm). The following is a list of the more common problems and solutions. If your problem is not shown, check the website for more recent updates or contact GAI-Tronics for support.

| Problem | Possible Solution |
|---|---|
| **Is the device powered?** | Look for four LEDs on the main circuit board:<br>• power LED (ON)—continuously illuminated once power is applied<br>• heartbeat LED (HB)—flashes slowly once the firmware is running—usually within 40 seconds after power is applied<br>• speed LED (SP)—continuously illuminated when connected at 100 Mbps<br>• link LED (LNK)—flashes intermittently when a network connection is present.<br>Check the power supply if the power LED does not illuminate. The heartbeat LED must be flashing before the unit will function. |
| **Cannot access device web pages** | Confirm the unit's IP address using a serial connection to the USB port on the VoIP PCBA.<br>The PC must have the correct FDTI driver installed to use the USB connection. Visit www.fdtichip.com for the latest virtual COM port (VCP) driver. Make a connection using a serial terminal program such as PuTTY, using a speed of 115200 bps. Select the correct COM port. The PC may assign a different COM port number every time a different telephone is connected to the same PC. Check the PC's device manager to verify. Once connected, diagnostic information displays at various stages in the phone's operation. Cycle power to the telephone while connected to view its current IP address. Use this to verify the IP address of the configuration web page.<br>Information that is normally only useful to GAI-Tronics technical support personnel may also be displayed. |

| Problem | Possible Solution |
|---|---|
| **Cannot make calls** | Call connection problems are usually due to proxy or registration issues if the unit can be pinged by its intended call destination,<br>• Check that the SIP server listing on GAI-Tronics website to verify its operation with GAI-Tronics telephones.<br>• Check that the proxy settings are correct and that both end points are properly registered.<br>• Check that the user name and password matches between the telephone and the server.<br>• Check the unit's home page for registration status and the STATUS/SIP STATUS page as the call is being set up, refreshing frequently to see changes. The current status of the call is displayed, usually revealing where the problem lies.<br>Note that GAI-Tronics VoIP units are SIP only—calls will not connect using H.323, SCCP, or other VoIP call connection protocols. |
| **Calls connect but there is no speech (or sound is garbled)** | Audio problems are usually due to codec issues.<br>Check that both end points use the same codec and that nothing is preventing them from negotiating correctly. Reduce the number of choices in the codec list if necessary (on the VOIP PARAMETERS/AUDIO SETTINGS page) or change the preference order.<br>Also, particularly where bandwidth is limited, the network should be set to provide Quality of Service (QoS) and/or to assign a high priority to voice traffic. It may be necessary to adjust the RTP TOS value field on the VOIP PARAMETERS/RTP PROTOCOL PARAMETERS page. |

# Reference Documentation

GAI-Tronics product documentation is located on the GAI-Tronics website at https://www.gai-tronics.com.

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