





**GAI-TRONICS®**  
A HUBBELL COMPANY

# VoIP Telephone Basic Programming Guide

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## Introduction

This guide provides information on the basic configuration and programming of GAI-Tronics' VoIP Telephones. Advanced features are addressed by Pub. 42004-396, which can be accessed via the unit's embedded browser home page by selecting the "web support" link. This document can also be found at the GAI-Tronics website ([www.gai-tronics.com/products/manuals\\_specs.htm](http://www.gai-tronics.com/products/manuals_specs.htm)).

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

For questions about configuring VoIP Telephones, please contact:

### Service Group

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# Set-up & Configuration

## Quick Start

The factory defaults will generally be sufficient in most cases, but the following steps must be taken as a minimum:

- Provide an Ethernet connection and power (either 24–48 V dc or PoE).
- Using a web browser, browse to the default IP address (192.168.1.2).
- Enter a user name and password (Defaults: *user & password*).
- Set an IP address and net mask (or set DHCP) on the **IP Settings** page.
- From the **General SIP Info** page (see [Figure 2](#)) select the **SIP1 Info** sub-page, and check that the ENDPOINT parameter is **ENABLED**.
- On the **SIP1 Info** sub-page, give the telephone a LOCALID (usually its extension number).
- On the **SIP1 Info** sub-page, set the DOMAIN, PROXY and REGISTRAR parameters to the address of the SIP server. If registrar authentication is in use, also set a user name and password.
- Program any dial memories using the **Dialing & Memories** pages.

**For an Autodial Telephone:** (354-701 series or similar)

On the **Memories and Comfort Strings** sub-page (see [Figure 6](#)), edit the MEMORY #1 (and COMFORT #1) parameters to set them to the desired destination: an existing <*user name/ID*> within the IP PBX/SIP server.

With these basic steps, the telephone will be able to make and receive calls in most cases. Check the Current Status page to help diagnose problems. This will show whether or not the telephone is registered and what is happening during calls.

**NOTE:** Make sure each unit is given at least a basic configuration before installing it. All units have identical settings as factory defaults, so each one must be individually configured to give it a unique identity on the network. This may be difficult to do after the units are installed.

## IP Settings

The IP Settings page is used to display or change various settings for connection to the IP network. Complete the IP Settings described below as the first step in completing the basic configuration.

Configuration - Microsoft Internet Explorer

Address: http://192.168.1.2/config/module.html

Network

- [IP settings](#)
- [SIP settings](#)
- [Unit settings](#)
- [Access settings](#)
- [Serial settings](#)
- [Email and Syslog settings](#)
- [Clock settings](#)

Phone functions

- [Dialing & Memories](#)
- [Key mapping](#)
- [Current status](#)

Signals and Audio

- [Audio settings](#)
- [Alarm settings](#)
- [Tone settings](#)
- [LED settings](#)
- [Logic settings](#)

Module: IP settings

Current Active IP Status:

Static IP settings:

DHCP: OFF

ADDRESS: 192.168.1.2

MASK: 255.255.0.0

GATEWAY: 0.0.0.0

DNS1: 0.0.0.0

DNS2: 0.0.0.0

LOCALDOMAIN:

WEB: ON

WEBPORT: 80

TELNET: ON

TELNETPORT: 23

SYSLOG:

SYSLOGPORT: 514

SYSLOG2:

SYSLOGPORT2: 514

SYSLOGFACILITY: 14

SYSLOGSEVERITY: 5

STUN:

Apply & Save Reset Cancel

Figure 1. IP Settings Page

Table 1 lists the parameters to be completed for the basic configuration. The parameters shown in Figure 1 but not listed in the table are for advanced configurations. Click on the EDIT button to begin making changes. Click on the APPLY & SAVE button to save your changes.

Table 1. IP Settings

<b>Parameter</b>	<b>Function</b>
<b>DHCP</b>	Enables or disables the use of DHCP for the assignment of IP parameters. If this value is set to OFF the telephone will use the Static IP values. Values available: <b>ON</b> or <b>OFF</b> Default value: <b>OFF</b>
<b>ADDRESS</b>	Sets the static IP Address of the unit. Default value: <b>192.168.1.2</b> <b>Do not enter a value here if DHCP is set to ON.</b>
<b>MASK</b>	Sets the static sub-net mask. Default value: <b>255.255.0.0</b> <b>Do not enter a value here if DHCP is set to ON.</b>
<b>GATEWAY</b>	Sets the static default gateway address. Default value: <b>0.0.0.0</b>
<b>DNS1</b>	Sets the IP address of the primary static DNS server. If DHCP is enabled then this DNS server will not be used. Default value: <b>0.0.0.0</b>
<b>DNS2</b>	Sets the IP address of the secondary static DNS server for redundancy. If DHCP is enabled then this DNS server will not be used. Default value: <b>0.0.0.0</b>
<b>LOCALDOMAIN</b>	Sets the domain name of the telephone on the network, as used by DNS. May be assigned by DHCP.
<b>WEB</b>	Enables or disables access to the web server. Values available: <b>ON</b> or <b>OFF</b> Default value: <b>ON</b>
<b>STUN</b>	Sets the IP address or URL for the STUN server that will be used to resolve STUN requests. Leaving this field blank will disable the STUN facility. Default value: <b>BLANK</b>

Next, under the **Network** heading, click on the **SIP SETTINGS** link to navigate to the **SIP Settings** page.

## SIP Settings Page

The **SIP Settings** page is used to view or change parameters specific to the SIP signaling protocol. GAI-Tronics VoIP telephones can hold details of up to four SIP proxies. If the telephone is unable to register or make a call, it can roll over to the next in a prioritized sequence.

There is a **SIP Info** page for each of the four possible endpoints, and a **General SIP Info** page containing details common to them all. The four endpoint pages are sub-pages of the **General SIP Info** page (shown in [Figure 2](#)) below.

For basic configurations, leave the SIP GENERAL PROXY PARAMETERS on the **General SIP Info** page to the configured defaults.

The screenshot shows the GAI-Tronics web interface. The top left features the GAI-TRONICS logo and the text 'A Hubbell Company'. The top right has a 'Configuration' header. Below this, there are navigation links for 'Home', 'Network', 'Phone functions', and 'Signals and Audio'. The 'Network' section is expanded, showing a list of settings: IP settings, SIP settings, Unit settings, Access settings, Serial settings, Email settings, and Clock settings. An 'Edit' button is next to the 'SIP settings' link. The 'SIP settings' section is further expanded to show 'SIP General Proxy Parameters' with the following values:

LOCALPORT:	5060	
PROXYFAILOVERSTATUSES:	5xx,6xx,49x,403,404,406,9xx	
DONTSTARTMEDIAATRING:	OFF	
SENDDTMFLAST:	OFF	
RTPTOS:	46	
SINGLEPTIME:	0	
SENDMULTIPARTMIME:	OFF	
NEWBRANCHONAUTHBYE:	ON	
<b>General registration settings</b>		
MODE:	SERIAL	
REGTIMEOUT:	3600	seconds
REREGTIMEOUT:	0	seconds (0 for REGTIMEOUT)

At the top right of the settings area, there is a breadcrumb trail: 'Page 1 - General SIP Info, 2 - SIP 1 Info, 3 - SIP 2 Info, 4 - SIP 3 Info, 5 - SIP 4 Info'. The bottom of the page shows a status bar with an 'Internet' icon.

Figure 2. General SIP Info Page

Click on the **SIP 1 INFO** link to navigate to the **SIP 1 Info** sub-page.

**GAI-TRONICS**  
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Configuration

Home

Network

[IP settings](#)  
[SIP settings](#)  
[Unit settings](#)  
[Access settings](#)  
[Serial settings](#)  
[Email settings](#)  
[Clock settings](#)

Module: SIP settings

SIP 1 Parameters

LOCALID	12345
DOMAIN	mydomain.com
PROXY	
PROXYPORT	5060
PRIORITY	1
REGISTRAR	
REGISTRARPORT	5060
USERNAME	
PASSWORD	
ENDPOINT	ENABLED

Page 1 - General SIP Info,  
 2 - SIP 1 Info, 3 - SIP 2 Info,  
 4 - SIP 3 Info, 5 - SIP 4 Info

Phone functions  
[Dialing & Memories](#)  
[Key mapping](#)  
[Current status](#)

Signals and Audio  
[Audio settings](#)  
[Alarm settings](#)  
[Tone settings](#)  
[LED settings](#)  
[Logic settings](#)

Figure 3. SIP 1 Settings Sub-Page

The four SIP sub-pages are identical, and are used to set parameters for each of four possible proxies. For the basic configuration, only the first proxy needs to be configured. The parameters needed for basic configuration are listed in [Table 2](#).

Click the **EDIT** button to begin making changes. Click the **APPLY & SAVE** button to save your changes when complete.

Table 2. SIP 1 Parameters (for Basic Configuration)

<b>Parameter</b>	<b>Function</b>
<b>LOCALID &amp; DOMAIN</b>	Together, these set the URI (uniform resource identifier) of the telephone. In the example shown in <a href="#">Figure 3</a> , the URI would be <b>sip:12345@mydomain.com</b> These values are used in the To:, From:, and Contact: headers, and also in the registration process with a registrar. Any alphanumeric string is accepted. Default values (both): <b>BLANK</b>
<b>PROXY</b>	Sets the IP address or the FQDN of the SIP proxy server to be used for incoming/outgoing calls. Default value: <b>BLANK</b>
<b>PROXYPORT</b>	Sets the port number on the proxy used for SIP protocol signaling. Default value: <b>5060</b>
<b>PRIORITY</b>	Sets the failover sequence among the four SIP sub-pages. Set to <b>1</b> .
<b>REGISTRAR</b>	Sets the address of the Registrar, either as an IP address or FQDN. The registrar address and the proxy may or may not be the same, but the address for registration must be set here. Default value: <b>BLANK</b>
<b>REGISTRARPORT</b>	Sets the port number where the requests will be sent. Default value: <b>5060</b> or unspecified.
<b>USERNAME</b>	Sets the username for the registrar authorization realm. Default value: <b>BLANK</b>
<b>PASSWORD</b>	Sets the password for the registrar authorization realm. Default value: <b>BLANK</b>
<b>ENDPOINT</b>	Sets whether the sub-page is <b>ENABLED</b> or <b>DISABLED</b> . Default value: <b>ENABLED</b> for SIP1 (All others: <b>DISABLED</b> ).

Next, under the **Network** heading, click the **UNIT SETTINGS** link to navigate to the **Unit Settings** page.



## Unit Settings Page

The **Unit Settings** page is used to set parameters for how the unit interfaces to the network, including configuration file updates.

**GAI-TRONICS**  
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[Home](#)

Apply & Save Reset Cancel

**Network**

[IP settings](#)  
[SIP settings](#)  
[Unit settings](#)  
[Access settings](#)  
[Serial settings](#)  
[Email settings](#)  
[Clock settings](#)

Module: Unit settings

**Unit settings:**

HOSTNAME: gtc294al702

UPDATE: SERVER 192.168.1.230

FILE gai/v3010/updategit.nfo

INTERVAL 0

**Phone functions**

[Dialing & Memories](#)  
[Key mapping](#)  
[Current status](#)

HELPSERVER: http://www.gai-tronics.com/42004-396A.pdf

LAN: SPEED AUTO

DUPLEX FULL

**Signals and Audio**

[Audio settings](#)  
[Alarm settings](#)  
[Tone settings](#)  
[LED settings](#)  
[Logic settings](#)  
[Multicast settings](#)

CONFIGID: autogen

ANI:

DEFAULT\_ANS\_MODE: PICK-UP

ANSMODE1: aa1

ANSMODE2: aa2

PAGEMODE: aa3

**Audio Path Test:**

APTENABLE: ON

APPTIME: 0:0,24

APTCOUNT: 1

APTOKCOUNT: 1

APTREPORT: OFF

Figure 4. Unit Settings Page

Table 3 lists the parameters to be completed for the basic configuration. The parameters shown in Figure 4 but not listed in the table are for advanced configurations.

Click on the **EDIT** button to begin making changes. Click on the **APPLY & SAVE** button to save your changes.

Table 3. Unit Settings Parameters (for Basic Configuration)

Parameter	Function
<b>HOST NAME</b>	<p>Sets the unit host name and consists of a maximum 15 alpha-numeric characters (a–z, A–Z, 0–9). The host name identifies the unit on the network, and is also used in email and syslog messages to identify the source of the message. If using DHCP, this field must be kept unique for each telephone on the system.</p> <p>Default value: A unique string starting with “GT” and followed by the serial number of the main circuit board inside the telephone (referred to as the “Board serial” on the home page).</p>
<b>DEFAULT_ANS_MODE</b>	<p>Sets the default answer mode. This mode will be used to answer a call when ANSMODE1, ANSMODE2 and PAGEMODE are <u>not</u> triggered. Values available are RING, PICK-UP, PAGE and STEALTH.</p> <p>RING is normal telephone operation, where a button must be pressed or handset lifted to answer an incoming call.</p> <p>PICK-UP is intercom auto-answer mode, where the telephone auto-answers and provides normal duplex audio, preceded by an announcement tone.</p> <p>PAGE is where the unit auto-answers and disables the microphone. A “splash” tone (tone 9) is emitted from the speaker to alert those nearby of an impending page announcement. The output level of the speaker is increased to its maximum level.</p> <p>STEALTH is where the telephone provides no indication of the incoming call and immediately auto-answers the call. The speaker is muted, and the microphone gain is enhanced.</p> <p>Default value: <b>PICK-UP</b> is set as the factory default.</p>

**Audio Path Test (APT) Parameters** (On the Unit Settings Page)

The purpose of APT is to send a specific audio tone from the earpiece or speaker of a telephone and then check that it is correctly received by the microphone. This will then verify that both microphone and speaker are functioning. APT is enabled as a factory default but must be turned on to function.

APT appears as an alarm on the **ALARMS** page, and can be set to report via Syslog and/or email like any other alarm, with some differences as listed below. The test can be set to run automatically or triggered manually using the parameter controls listed in [Table 4](#).

Table 4. APT Parameters

<b>Parameter</b>	<b>Function</b>
<b>APTENABLE</b>	Sets whether APT is <b>ON</b> or <b>OFF</b> .
<b>APPTIME</b>	<p>Sets a start time (24-hour clock) and test interval (in hours). This field should contain: first, the time in hours and minutes separated by a colon (:), followed by a comma (,) followed by the interval in hours (range 1–24).</p> <p>Automatic testing will start at the specified time and repeat every specified interval until 00:00 midnight the next day.</p> <p>The cycle will then repeat the next day and so on.</p> <p>Default: <b>00:00,24</b> meaning that the test will perform once per day at midnight.</p>
<b>APTCOUNT</b>	<p>Sets the number of tests that will be performed at each interval.</p> <p>Range is 1–10.</p> <p>Default: <b>1</b> (but it can be increased to repeat the test at each interval.)</p>
<b>APTOKCOUNT</b>	<p>Sets the number of tests that must pass at each interval to be classed as a successful test. APTOKCOUNT must always be <math>\leq</math> APTCOUNT.</p> <p>For example, if APTCOUNT were set to 3 and APTOKCOUNT to 2, the test would be deemed to have passed if 2 pass readings out of 3 were recorded.</p> <p>This feature is to allow for potential disruption in areas of high ambient noise.</p> <p>Default value: <b>1</b></p>
<b>APTREPORT</b>	<p>Sets whether or not APT will send reports every time the test passes.</p> <p>Normal alarms only report if they change state. Setting APTREPORT to ON will cause the telephone to send a regular report confirming that its acoustic components are healthy.</p> <p>By inference, this report also confirms that the telephone is powered, running and connected to the network, so it also provides a useful general health check.</p> <p>If the test fails, the telephone will not send repeated reports until at least APTOKCOUNT tests pass again.</p>
<b>APTNOW</b>	APT now will start an APT test within 60 seconds. This button will only start a test if APTENABLE is set to <b>ON</b> .

Next, under the **Network** heading, click the **CLOCK SETTINGS** link to navigate to the **Clock Settings** page.

## Clock Settings Page

The telephones do not include a battery-backed real time clock, but will keep time based on updates from an SNTP server. Adjustments for daylight savings time can be made by setting DST start and end dates and times. The **Clock Settings** page is used to set the required parameters.

The screenshot shows the 'Clock settings' configuration page. The left sidebar contains navigation links for Network, Phone functions, and Signals and Audio. The main area is titled 'Module: Clock settings' and includes the following parameters:

- SNTP:** 192.168.1.108
- SNTPINTERVAL:** 60 minutes
- TIMEZONE:** -05:00: EST Eastern/CDT Central Daylight/NYC New York City
- FORMAT:** DD/MM
- DST:**
  - ADJUST: ON
  - OFFSET: +01:00
  - STARTDAY: 0
  - STARTDOW: 1 (1 = Sunday)
  - STARTMONTH: 3
  - STARTWOM: 2
  - STARTTIME: 02:00 (24Hr clock)
  - ENDDAY: 0
  - ENDDOW: 1 (1 = Sunday)
  - ENDMONTH: 11
  - ENDWOM: 1
  - ENDTIME: 02:00 (24Hr clock)

Figure 5. Clock Setting Page

Table 5 lists the parameters to be completed for the basic configuration. Click the **EDIT** button to begin making changes.

Click the **APPLY & SAVE** button to save your changes when complete.

Table 5. Clock Setting Parameters (for Basic Configuration)

Parameter	Function
<b>SNTP</b>	Sets the address for the SNTP server to be used, as an IP address or a FQDN.
<b>SNTPINTERVAL</b>	Sets the interval, in minutes, between SNTP update requests. Default value: <b>60</b>
<b>TIMEZONE</b>	Sets the current time zone for local time from a dropdown list.
<b>FORMAT</b>	Sets the date format to either US (MM/DD) or UK (DD/MM) style.

The remaining parameters on this page set the behavior of the internal clock for daylight savings time (DST), but these are not required for the basic configuration.

Next, under the **Phone Functions** heading, click the **DIALING & MEMORIES** link to navigate to the first **Dialing and Memory** page.

## Dialing & Memories Pages

The **Dialing & Memories** pages are used to set various “dialing” actions, i.e., how the telephone initiates calls. Depending on the model, the telephone front panel may have a numeric keypad, memory buttons, both, or neither. The numeric telephone keypad is used to enter a number (user ID) one digit at a time, whereas memory buttons or the hookswitch of an Autodial Telephone refer to complete, predetermined numbers/users.

Each memory button is assigned a memory list, consisting of one or more memories. Calls initiated by pressing that memory button automatically divert to the next number in the list if the call fails to be answered.

### Memories and Comfort Strings Sub-page

The telephone can store 20 call destinations, as shown on the first **Dialing & Memories** page (shown in [Figure 6](#)).

The screenshot displays the configuration page for 'Memories & Comfort Strings'. At the top left is the GAI-TRONICS logo (A Hubbell Company). The page title is 'Configuration'. A navigation menu on the left lists various settings categories: Home, Network (IP, SIP, Unit, Access, Serial, Email, Clock settings), Phone functions (Dialing & Memories, Key mapping, Current status), and Signals and Audio (Audio, Alarm, Tone, LED, Logic, Multicast settings). The main content area shows 'Module: Dialing & Memories' and a table with 20 rows. The columns are 'MEMORY' and 'COMFORT'. The first row's 'COMFORT' cell is highlighted in yellow.

	MEMORY	COMFORT
1		
2		
3		
4		
5		
6		
7		
8		
9		
10		
11		
12		
13		
14		
15		
16		
17		
18		
19		
20		

Figure 6. Memories & Comfort Strings Sub-page

The telephone auto-dial memory buttons are pre-configured from the factory. Only the destination telephone numbers or addresses (Memory) must be entered. The Comfort String can also be configured, if desired.

Click the **EDIT** button to begin making changes. Click the **APPLY & SAVE** button to save your changes when complete.

Table 6. Autodial Destination Parameters

<b>Parameter</b>	<b>Function</b>
<b>MEMORY</b>	Each of the 20 possible entries is either a <ol style="list-style-type: none"> <li>1. <i>&lt;username&gt;/&lt;user ID&gt;</i> of an IP PBX/SIP server that is generally set as a sequence of digits; or</li> <li>2. A SIP URI for a per-to-peer connection.</li> </ol>
<b>COMFORT</b>	Each of the 20 possible entries can also be assigned a COMFORT string, which is a string of digits that will be played back to the user as DTMF when the call is being set up. This simulates the dialing digit tones heard on a normal telephone. If these comfort digits are desired, the comfort string must be entered.

**NOTE:** These memories (telephone numbers or addresses) are not assigned directly to the hookswitch or EMERGENCY button. They must be set up in the **Memory Lists** sub-page. Click on the **Memory Lists** link to navigate to the **Memory Lists** sub-page.

### **Memory Lists Sub-page**

Memory Lists are strings of Memories (telephone numbers or addresses assigned on the Memories and Comfort Strings Sub-page) that the telephone will call in sequence when the associated button is pressed.

The telephone can hold up to 11 memory lists (0–10). List 0 is the Emergency List and is mapped to a button designated as EMERGENCY, if the telephone is so equipped.

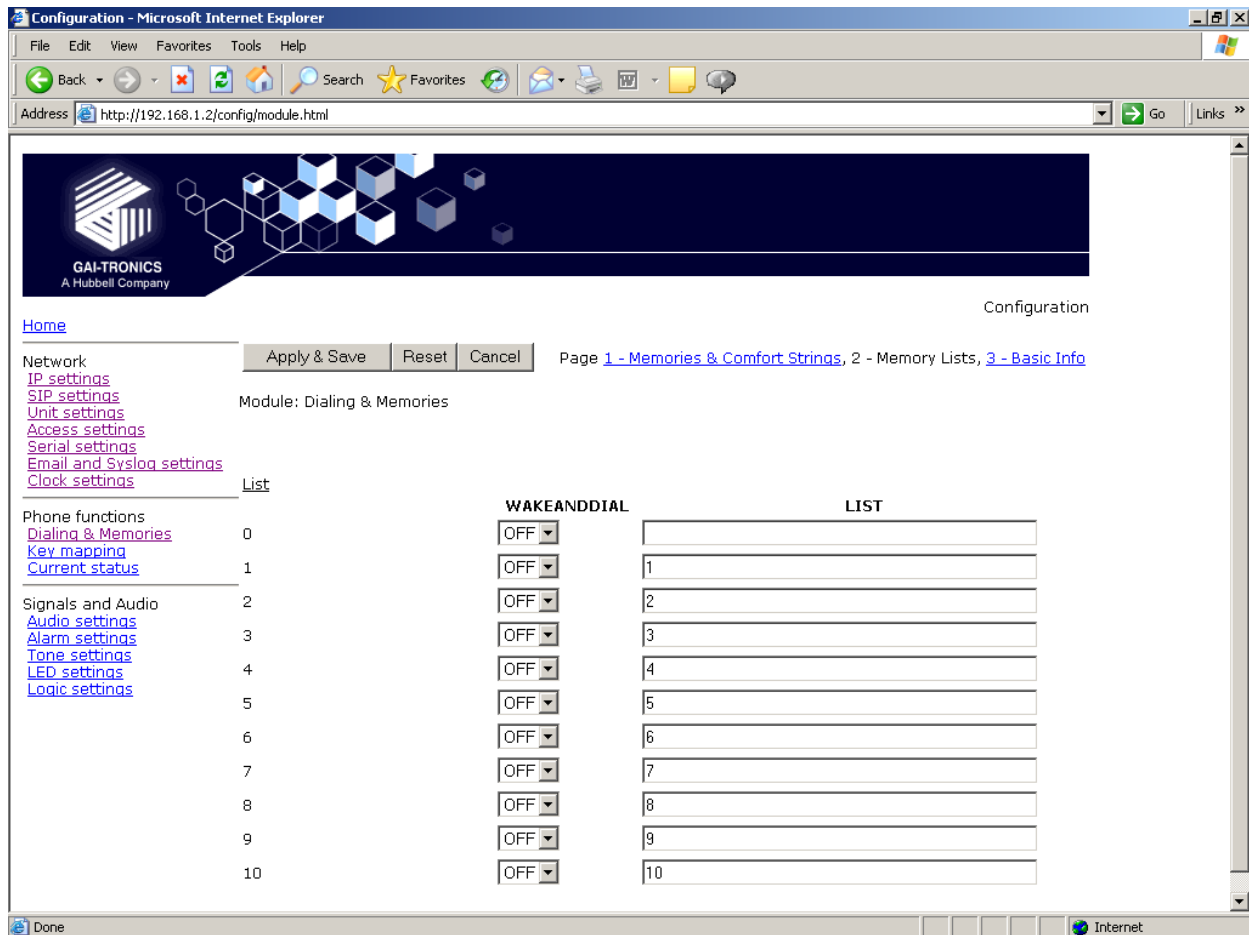


Figure 7. Memories Lists Sub-Page

A list can also be set to activate as soon as the handset is lifted if the telephone is to have auto-dial capability(See the Basic Info Sub-page section for further information).

Click the **EDIT** button to begin making changes. Click the **APPLY & SAVE** button to save your changes when complete.

Table 7. Dialing and Memories Parameters

Parameter	Function
<b>LIST</b>	<p>Each list can contain up to 20 memory entries, separated by commas.</p> <p>For example, if you wanted the <b>MEM1</b> button to call the memory 1 (usually a telephone number or address), if that failed to then call memory 5, and if that failed call memory 10, you would enter “1, 5, 10” in the List box for List 1.</p> <p>When a memory list is invoked, the telephone will attempt to place a call to each memory in the list in sequence until a call is successful or it reaches the end of the list.</p> <p>Each memory can appear in more than one list.</p>
<b>WAKEANDDIAL</b>	<p>When set to <b>ON</b>, the telephone will come off hook and start to process the list as soon as the appropriate button is pressed. This is factory set for hands-free telephones, but can be set for handset telephones if required.</p>

Next, click on the **BASIC INFO** link to navigate to the **Basic Info** sub-page.

## Basic Info Sub-page

This page is used to edit an additional parameter (**OFFHOOK**) for the dialing configuration. This parameter is factory configured for most of the telephone models covered by this manual and does not require modification.

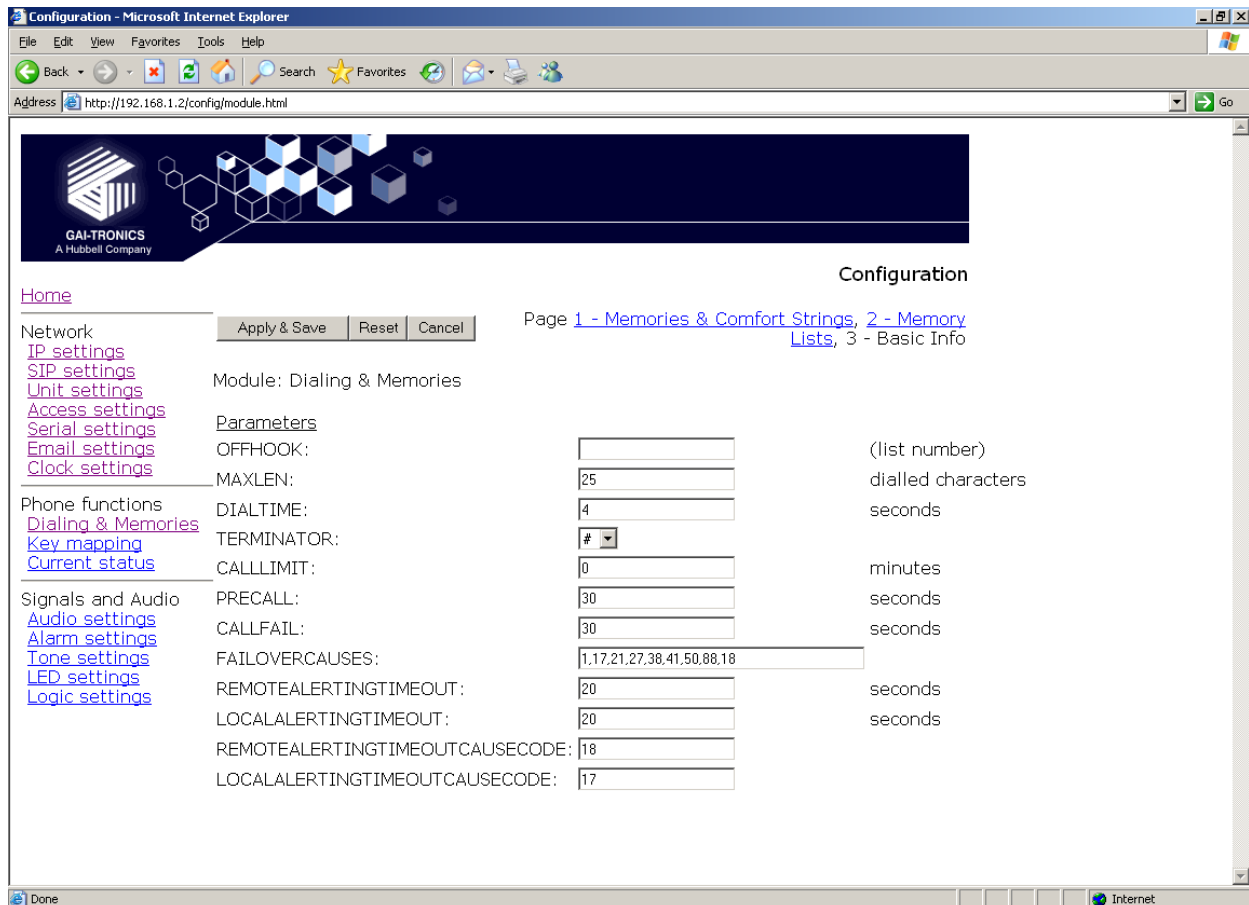


Figure 8. Basic Info Sub-page

The remaining parameters shown in [Figure 8](#), but not listed in [Table 8](#), are for advanced configurations and should NOT be changed from the factory defaults.

Click on the **EDIT** button to begin making a change. Click on the **APPLY & SAVE** button to save your change.

Table 8. Dialing and Memories Parameter (for Basic Configuration)

Parameter	Function
<b>OFFHOOK</b>	Sets a memory list number to be invoked when the handset is taken off hook (in a handset model).  In most cases, this parameter is factory set for the specific telephone model and will not need to be changed. <b>For an Autodial Telephone, this is set to a memory list number.</b>



Next, under the **Signals & Audio** heading, click on the **AUDIO SETTINGS** link to navigate to the **Audio Settings** page.

## Audio Settings Page

This page sets various audio parameters within the telephone CODEC.

Configuration

[Home](#)   

**Network**  
[IP settings](#)  
[SIP settings](#)  
[Unit settings](#)  
[Access settings](#)  
[Serial settings](#)  
[Email settings](#)  
[Clock settings](#)

Module: Audio settings  
**Audio & Codec Parameters**

**CODEC:**

**Phone functions**  
[Dialing & Memories](#)  
[Key mapping](#)  
[Current status](#)

**Signals and Audio**  
[Audio settings](#)  
[Alarm settings](#)  
[Tone settings](#)  
[LED settings](#)  
[Logic settings](#)  
[Multicast settings](#)

**SAMPLE:**  
 G711  ms  
 G722  ms  
 G729  ms

**FRAMES:**  
 G711  fpp  
 G722  fpp  
 G729  fpp  
 G7231  fpp

**VAD:**

**DTMF:**

**DTMFPT:**

**DTMFPLAYBACK:**

**HANDSETVOLUME:**

**HANDSFREEVOLUME:**

**RINGERVOLUME:**

**LINEVOLUME:**

**HANDSETGAIN:**

**HANDSFREEGAIN:**

**LINEGAIN:**

**JITTERMIN:**

**JITTERMAX:**

**SIDETONE:**

**SIDETONELEVEL:**

1=G711A  
 2=G711u  
 3=G722  
 4=G729  
 5=G7231-6.3  
 6=G7231-5.3

96 -> 127  
 96 For RFC2833 Sect 3.14  
 101 for CISCO compatability

Figure 9. Audio Settings Page

Table 9 lists the parameters to be completed for the basic configuration. The parameters shown in Figure 9 but not listed in the table are for advanced configurations.

Click on the **EDIT** button to begin making changes. Click on the **APPLY & SAVE** button to save your changes.

Table 9. Audio Settings Parameters (for Basic Configuration)

<b>Parameter</b>	<b>Function</b>
<b>HANDSETVOLUME</b>	If the telephone is a handset model, this parameter sets the handset earpiece volume. The range is 1–9. <b>NOTE:</b> If the telephone is a hands-free model, this setting has no effect.
<b>HANDSFREEVOLUME</b>	If the telephone is a hands-free model, this parameter sets the speaker volume. The range is 1–12. <b>NOTE:</b> If the telephone is a handset model, this setting has no effect.
<b>HANDSETGAIN</b>	If the telephone is a handset model, this parameter sets the handset microphone gain. The range is 1–8. <b>NOTE:</b> If the telephone is a hands-free model, this setting has no effect.
<b>HANDSFREEGAIN</b>	If the telephone is a hands-free model, this parameter sets the microphone gain. The range is 1–8. <b>NOTE:</b> If the telephone is a handset model, this setting has no effect.

Next, under the **Signals & Audio** heading, click on the **MULTICAST SETTINGS** link to navigate to the **Multicast Settings** page or the **Logic Settings** page. Accessing both pages may be required, depending on the desired operation.

## Multicast Settings Page

Multicast allows a single audio stream to be sent to multiple endpoints simultaneously to achieve multi-point broadcasts or public address functionality over IP. This is only applicable to telephones with a loud-speaking capability, such as hands-free units.

### NOTES:

- Multicast requires the use of a SIP server that specifically supports it, and each endpoint (i.e., telephone) must be individually configured to receive multicast packets.
- When making a multicast call, the SIP server will send a paging request to a specific IP address and expect multiple phones to accept and play the subsequent audio.
- The GAI-Tronics telephone can be programmed with up to eight multicast IP addresses to allow the receipt of multicasts from different sources or to enable zoning of multicasts.
- Each multicast address can be assigned a priority to define which can override which.
- Although multicast is factory enabled, a telephone can still make and receive normal calls. Normal calls can also be assigned a priority level, defining whether calls can override multicasts or vice versa.

	ADDRESS FILTER	PRIORITY	OUTPUT1	OUTPUT2	TONE
1	0.0.0.0:255.255.255.255	0	DISABLED	DISABLED	DISABLED
2	0.0.0.0:255.255.255.255	0	DISABLED	DISABLED	DISABLED
3	0.0.0.0:255.255.255.255	0	DISABLED	DISABLED	DISABLED
4	0.0.0.0:255.255.255.255	0	DISABLED	DISABLED	DISABLED
5	0.0.0.0:255.255.255.255	0	DISABLED	DISABLED	DISABLED
6	0.0.0.0:255.255.255.255	0	DISABLED	DISABLED	DISABLED
7	0.0.0.0:255.255.255.255	0	DISABLED	DISABLED	DISABLED
8	0.0.0.0:255.255.255.255	0	DISABLED	DISABLED	DISABLED

Figure 10. Multicast Settings Page

Click the **EDIT** button to begin making changes. Click the **APPLY & SAVE** button to save your changes when complete.

Table 10. Multicast Page Settings

<b>Parameter</b>	<b>Function</b>
<b>TIMEOUT</b>	Sets an enforced delay (in seconds) between one multicast session ending and another beginning. The range is 1–120. Default value: <b>120</b>
<b>SPEAKERVOLUME</b>	Sets the speaker volume during a multicast. Volume will revert to the setting on the AUDIO page when the multicast session has ended. The range is 1–10, Default value: <b>3</b>
<b>Override level</b>	Sets the override level (between 0 and 8) for normal telephone calls with respect to the priority level set against multicast calls defined below. 1 is highest priority, 8 is lowest. 0 means no priority and will not override any multicast. For example, if override level is set to 5, a voice call will override a multicast having a priority of 6, but not one having a priority of 4. If a voice call and a multicast have the same priority level the multicast will take precedence. If an incoming call is made to a telephone while a higher priority multicast is in progress, the caller may hear the multicast audio but a speech call will not be connected to the telephone until the multicast has ended. Note: Emergency calls started from the telephone (i.e., using a button designated as an <b>Emergency</b> button) will always override any normal or multicast call, regardless of priority or override level.
<b>ADDRESS</b>	The telephone will accept multicast calls sent to this address by the SIP server. Must be an IP address complete with port, e.g., 242.0.1.75:5000. Reserved addresses for multicast channels are normally in the range 224.0.0.0 to 224.0.0.255.
<b>FILTER</b>	Sets a range of acceptable multicast source IP addresses. The telephone will only accept a multicast if the source is within this IP address range. The format is two IP addresses separated by a colon. Default value: <b>0.0.0.0:255.255.255.255</b>
<b>PRIORITY</b>	Sets a priority level for this multicast with respect to other multicasts and normal telephone calls. The range is 0 to 8, with 1 being the highest priority, 8 being the lowest and 0 having no priority. If a higher priority event (multicast or telephone call) occurs during an existing multicast, it will be interrupted and resume after the higher priority event has finished.
<b>OUTPUT1</b>	Sets whether <b>OUTPUT1</b> is <b>ENABLED</b> or <b>DISABLED</b> during this multicast. If <b>ENABLED</b> it will be in a permanently energized state, with no timing or cadence control. After the multicast has ended it will revert to its function as defined on the <b>LOGIC</b> page.
<b>OUTPUT2</b>	Sets whether <b>OUTPUT2</b> is <b>ENABLED</b> or <b>DISABLED</b> during this multicast. If <b>ENABLED</b> it will be in a permanently energized state, with no timing or cadence control. After the multicast has ended it will revert to its function as defined on the <b>LOGIC</b> page.
<b>TONE</b>	Sets whether <b>tone 10</b> (TONES page) is <b>ENABLED</b> or <b>DISABLED</b> during this multicast.

# Logic Settings Page

The Logic Settings page sets the operation of the four auxiliary inputs and the two volt-free contact outputs.

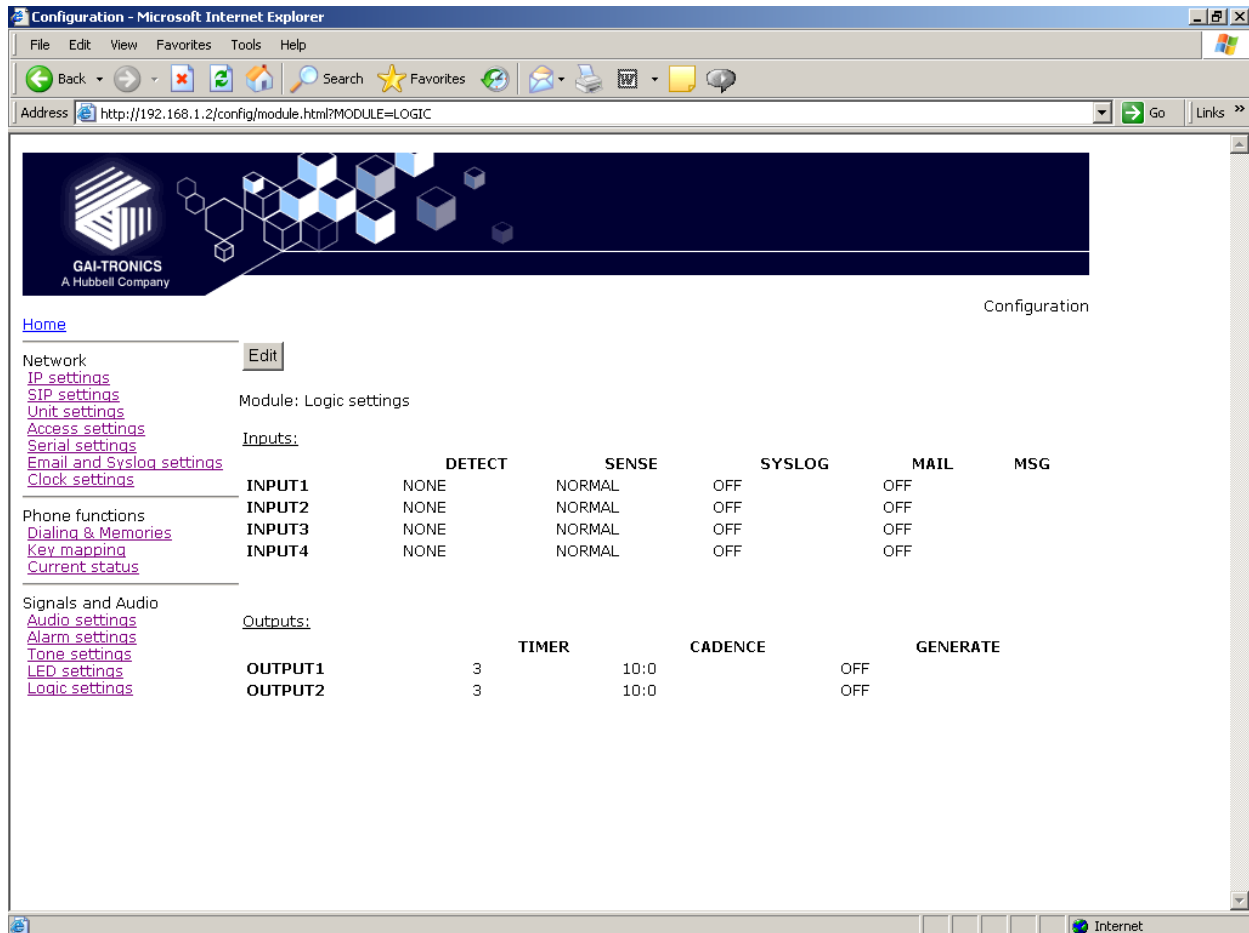


Figure 11. Logic Setting Page

## Inputs

The four auxiliary inputs are activated by connecting the relevant input terminal to a common terminal via a volt-free contact. See the appropriate installation guide for connection details and electrical specifications. If the contact is open, the input is normally deemed to be ON, and if the contact is closed it is deemed to be OFF. The sense can be inverted, see [Table 11](#) below:

Table 11. Input Contact Open/Close Sense Configuration

External Contact	Sense Normal	Sense Invert
Open	ON	OFF
Closed	OFF	ON

The auxiliary inputs can be configured to report their status to a remote site using two methods:

- Syslog output over TCP
- SMTP mail message

For each input, the following parameters can be set:

Table 12. Input Contact Configuration

<b>Parameter</b>	<b>Function</b>
<b>DETECT</b>	Specifies whether an input will report being set to its ON condition only (ON), its OFF condition only (OFF), on either event (ON+OFF), or not at all (NONE). The ON and OFF states are affected by the SENSE setting below.
<b>SENSE</b>	If set to <b>NORMAL</b> , a contact closure will report as OFF. If set to <b>INVERT</b> , a contact closure will report as ON. Default value: <b>NORMAL</b>
<b>SYSLOG</b>	Enables or disables <b>SYSLOG</b> reporting for the selected input. Syslog settings are on the IP Setting page. Refer to Pub. 42004-396 for further information on syslog reporting, if required.
<b>MAIL</b>	Enables or disables <b>SMTP</b> reporting for the selected input. SMTP settings are on the Email Settings page. Refer to Pub. 42004-396 for further information if required.
<b>MSG</b>	Replaces the default text message Aux_in <input_number> with the text entered (maximum 40 characters). The status <on/off> is appended to the end of the text. If the MSG value is blank, the default message is reinstated.  The message sent (for both mail and syslog reports), takes the form:  HOSTNAME COUNT TIME MSG ON/OFF  Where: HOSTNAME is from the Unit Settings page COUNT is a volatile event counter (rolls over at 10000) TIME is the event time and date from the unit's clock MSG is the message set by the MSG field above. If no message has been set, the default is "Aux_in x".  ON/OFF is either the word ON or OFF according to the state of the input, taking account of the SENSE setting.

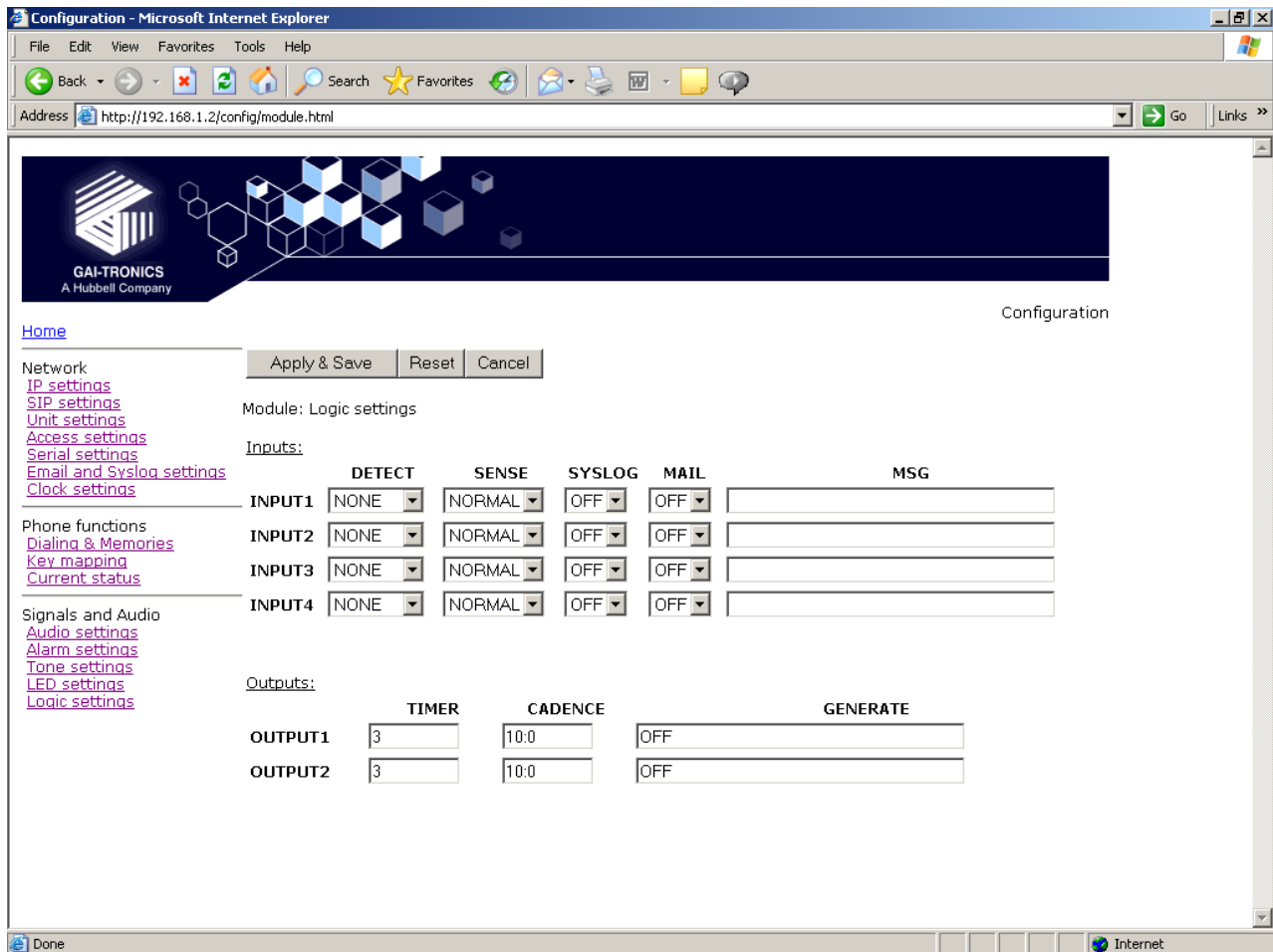


Figure 12. Logic Settings Page

The **Logic Settings** page is used to program the two volt-free contacts only if multicast is not utilized; or if the unit being programmed will be capable of receiving both SIP calls and multicast calls.

## Outputs

The two outputs are both volt-free contacts, but their ratings differ. See the appropriate installation guide for connection details and electrical specifications.

Table 13. Output Contact Configuration

Parameter	Function	
<b>GENERATE</b>	The GENERATE field sets the function of the output by use of the following keywords:	
	<b>“Generate” Keyword</b>	<b>Function</b>
	<b>ON</b>	Sets the output permanently on.
	<b>OFF</b>	Sets the output permanently off.
	<b>PULSE +</b>	Sets the output to activate once only for the period defined by the TIMER field, on receipt of a Recall signal from a remote telephone.
	<b>MUTE +</b>	Sets the output to indicate if the audio input is muted.
	<b>RING +</b>	Sets the output to pulse when an incoming call is ringing. The pulsing on /off periods are set by the CADENCE field.
	<b>CALL +</b>	Sets the output to pulse when an outgoing call is active. The pulsing on /off periods are set by the CADENCE field.
	<b>CONNECT +</b>	Sets the output on when a call is connected.
	<b>HOOK +</b>	Sets the output on when the telephone is off hook, and off when it is back on hook.
	<b>INUSE +</b>	Sets the output on when an incoming call arrives or when the user goes off hook for an outgoing call, and off when the call ends.
	<b>RINGCADENCE +</b>	Causes the output to pulse in time with the ring tone cadence.
	<b>RINGOUT +</b>	Sets the output to pulse when an outgoing call is ringing (but not yet connected). The pulsing on /off periods are set by the CADENCE field.
<b>PAGE +</b>	Sets the output to pulse when a call is present that has been signaled as a PAGEMODE call. See UNIT page and refer to Pub. 42004-396 for further information, if required. The pulsing on /off periods are set by the CADENCE field.	



Parameter	Function	
	<b>REGISTERED +</b>	Sets the output to pulse when the telephone is registered with at least one SIP server. Can be used as a “phone available” indicator. The pulsing on /off periods are set by the CADENCE field.
	<b>EMERGENCY +</b>	Sets the output to pulse whenever there is an outgoing call present that has been initiated by an EMERGENCY button. The pulsing on /off periods are set by the CADENCE field.
	<b>NOTE:</b> The ON and OFF keywords must be used on their own. The other keywords (indicated by a + symbol), can be combined and entered in any order, separated by the plus (+) character. For example, to set an output to pulse when an incoming call is ringing, and be on steadily when the call is connected enter RING+CONNECT.	
<b>TIMER</b>	Sets the timer value for the PULSE command in seconds. Default value is <b>3</b> . The minimum is 0 and the maximum is 3600.	
<b>CADENCE</b>	Sets the cadence for those keyword commands that require it. The cadence is entered as two numbers separated by a colon (:) character, representing the on and off times in tenths of a second. For example, to set a cadence of 1 second on, half a second off, enter 10:5.	

## 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](http://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 here check the website for more recent updates, or contact GAI-Tronics for support.

Problem	Possible Solution
<b>Is the unit powered up?</b>	<p>Look for two LEDs on the main circuit board—there is a power LED and a heartbeat LED. The power LED lights continuously as soon as power is applied, the heartbeat flashes slowly once the firmware is running - usually within 40 seconds after power is applied.</p> <p>If the power LED doesn't light, check the power supply to the unit. Once power is restored the unit will not function until the heartbeat LED is flashing.</p>
<b>I can't access the web pages.</b>	<p>If the unit is correctly powered up, but you cannot browse to its web pages over the network, you will usually need to make a serial connection to the unit and check the following using the Command Line Interface:</p> <ul style="list-style-type: none"> <li>• Are the IP and UNIT settings correct?</li> <li>• Is the web server enabled?</li> <li>• Can the telephone ping other destinations on the network? The IP module has PING and TRACEROUTE functions to help troubleshoot routing problems.</li> <li>• Some switches may not auto-negotiate speed correctly - try changing the LAN speed (UNIT module) from AUTO to 10.</li> </ul>
<b>I can't make calls</b>	<p>If the unit can ping (and be pinged by) its intended call destination, call connection problems are usually due to proxy or registration issues.</p> <ul style="list-style-type: none"> <li>• Check that the SIP server is listed on the GAI-Tronics website as having been tested with GAI-Tronics telephones.</li> <li>• Check that the proxy settings are correct and that both end points are properly registered.</li> <li>• If the SIP server requires authentication, check that the user names and passwords match between the telephone and the server.</li> <li>• Check the current status page as the call is being set up, refreshing frequently to see changes. The current status of the call will be displayed, usually revealing where the problem lies.</li> </ul> <p>Note that GAI-Tronics VoIP units are SIP only - calls will not connect using H.323, SCCP or other VoIP call connection protocols.</p>
<b>Calls connect but there is no speech (or sound is garbled).</b>	<p>Audio problems are usually due to codec issues.</p> <p>Check that both end points can use the same codec, and that nothing will prevent them negotiating correctly. If necessary, reduce the number of choices in the codec list (on the AUDIO page) or change the preference order.</p> <p>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 RTPTOS field on the SIP page.</p> <p>The FRAMES and SAMPLE values on the Auto Settings page may need to be adjusted. Contact GAI-Tronics for instructions.</p>