Pub. 42004-482B

GAI-TRONICS®
A HUBBELL COMPANY

VoIP Speaker & Speaker Amplifier Basic Programming Guide

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Introduction

This guide provides information on the basic configuration and programming of GAI-Tronics' VoIP Speakers. 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).

NOTE: All references to "speakers" in this document are understood to be GAI-Tronics VoIP/WiFi Speakers or Speaker Amplifier products.

For questions about configuring VoIP Speakers, 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

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 Settings page (see Figure 2) select the SIP1 Info sub-page, and check that ENDPOINT is **ENABLED**.
- On the SIP1 Info sub-page, give the speaker a LOCALID (usually its extension number).
- On the SIP1 Info sub-page, set DOMAIN, PROXY and REGISTRAR all to the address of the SIP server. If registrar authentication is in use, also set a user name and password.

With these basic steps, the speaker will be able to receive calls in most cases. Check the Current Status page to help diagnose problems. This will show whether or not the speaker 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.

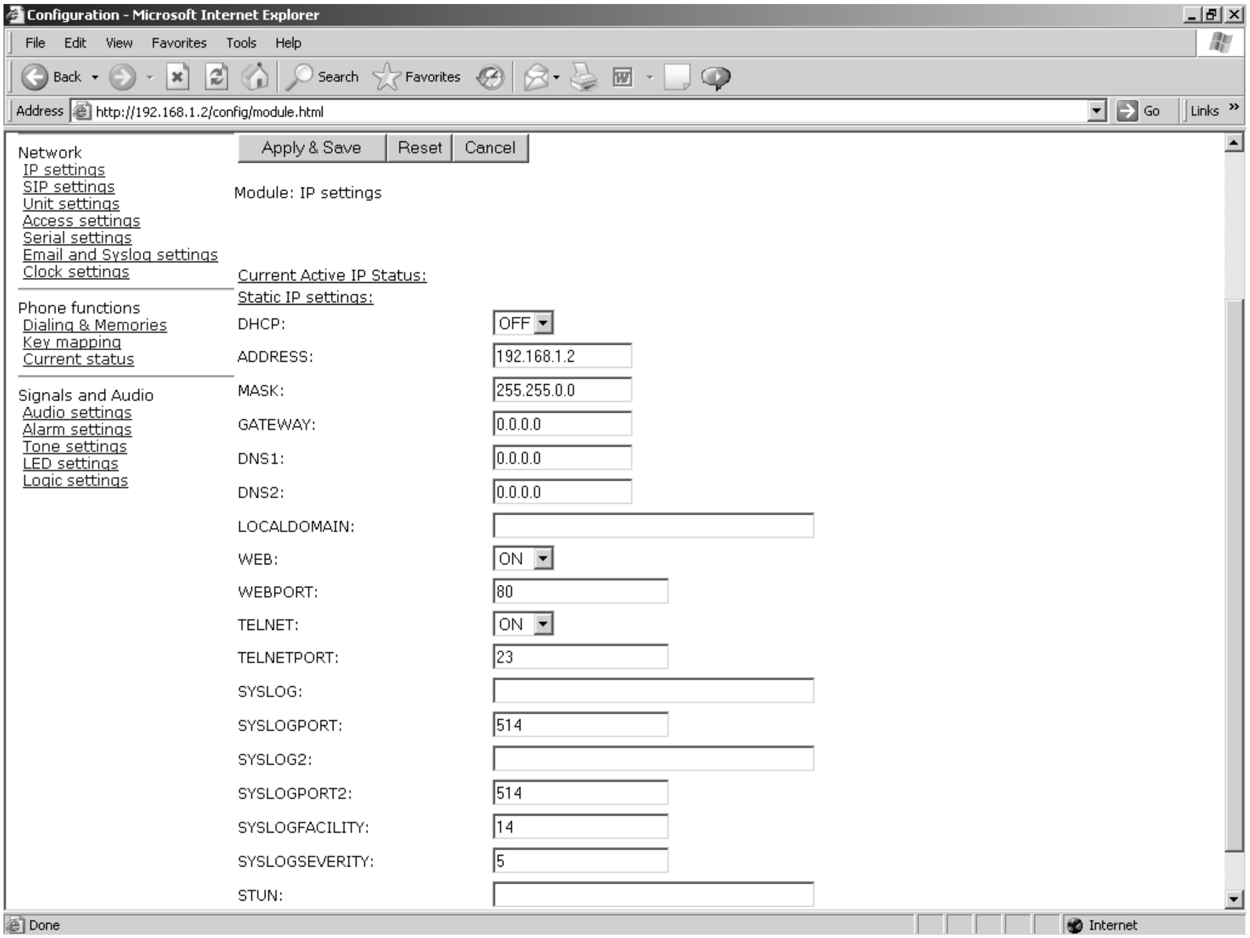

Figure 1. IP Settings Page

Table 1 lists the parameters to be completed for the basic configuration. The parameters shown in Figure 1 but not listed in Table 1 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 Setting Parameters needed for Basic Configuration

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

Next, under the Network heading, click on the link for **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 speakers can hold details of up to four SIP proxies. If the speaker is unable to register, 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 page, which is shown in Figure 2.

For basic configurations, allow the SIP General Proxy parameters on the General SIP Info page to remain at default.

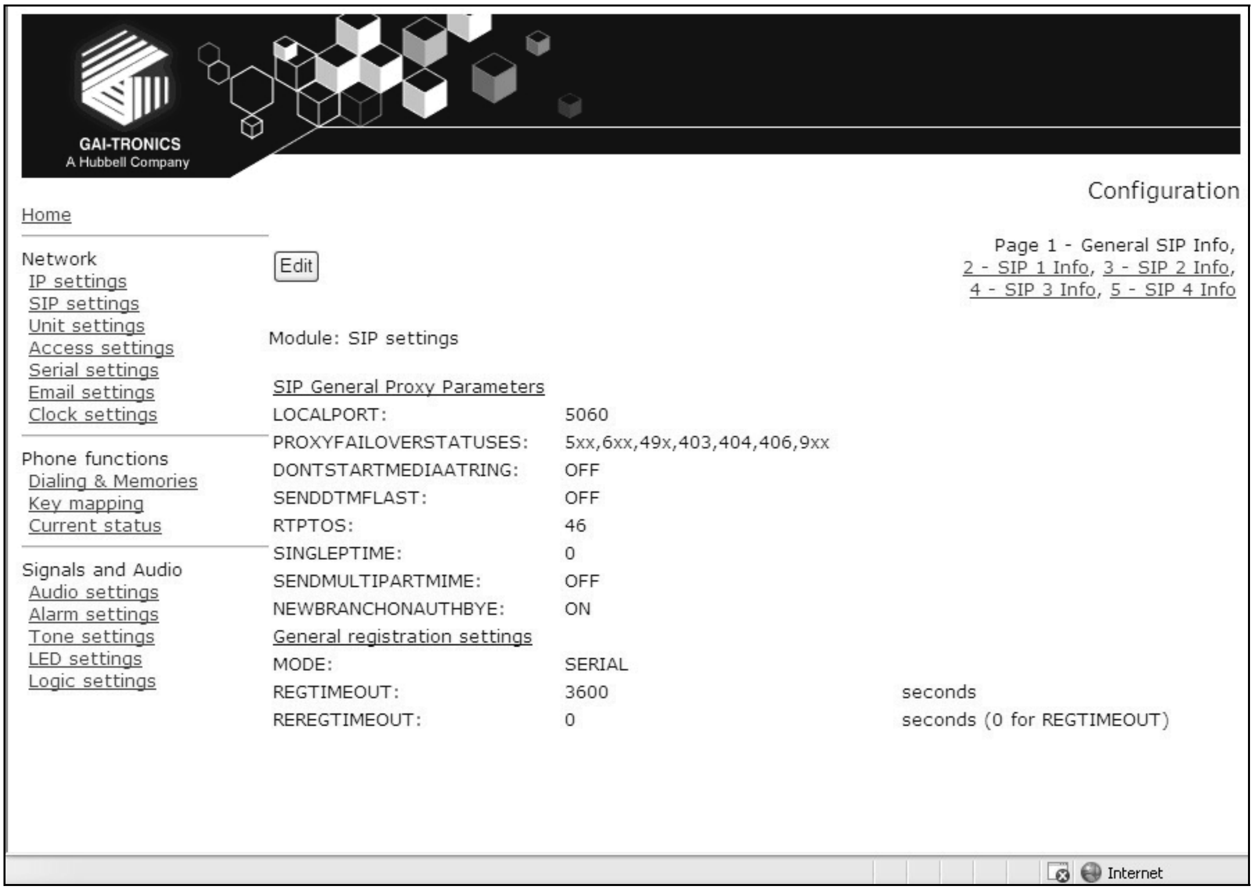

Figure 2. General SIP Info Page

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

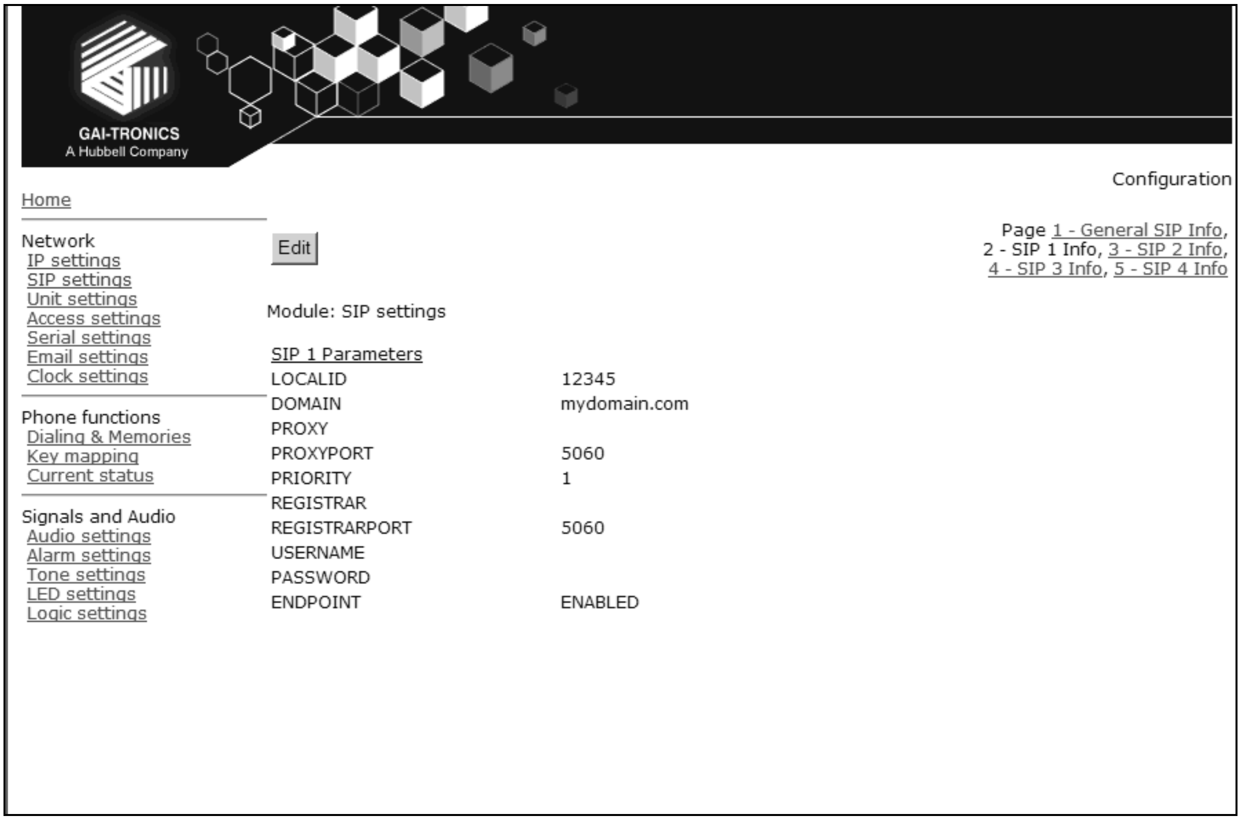

Figure 3. SIP 1 Settings Sub-Page

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 is 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 Settings Sub-Page Parameters needed for basic configuration

Parameter	Function
LOCALID & DOMAIN	Together, these set the URI (uniform resource identifier) of the speaker. In the example shown in Figure 3, the URI would be **sip:12345@mydomain.com** 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): **blank**
PROXY	Sets the IP address or the FQDN of the SIP proxy server to be used for incoming/outgoing calls. Default value: **blank**
PROXYPORT	Sets the port number on the proxy used for SIP protocol signaling. Default value: **5060**
PRIORITY	Sets the failover sequence among the four SIP sub-pages. Set to **1**.
REGISTRAR	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: **blank**
REGISTRARPORT	Sets the port number where the requests will be sent. Default value: **5060** or unspecified.
USERNAME	Sets the username for the registrar authorization realm. Default value: **blank**
PASSWORD	Sets the password for the registrar authorization realm. Default value: **blank**
ENDPOINT	Sets whether the sub-page is ENABLED or DISABLED. Default value: **ENABLED** for SIP1 (All others: DISABLED).

Next, under the Network heading, click the link for **Unit Settings** 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.

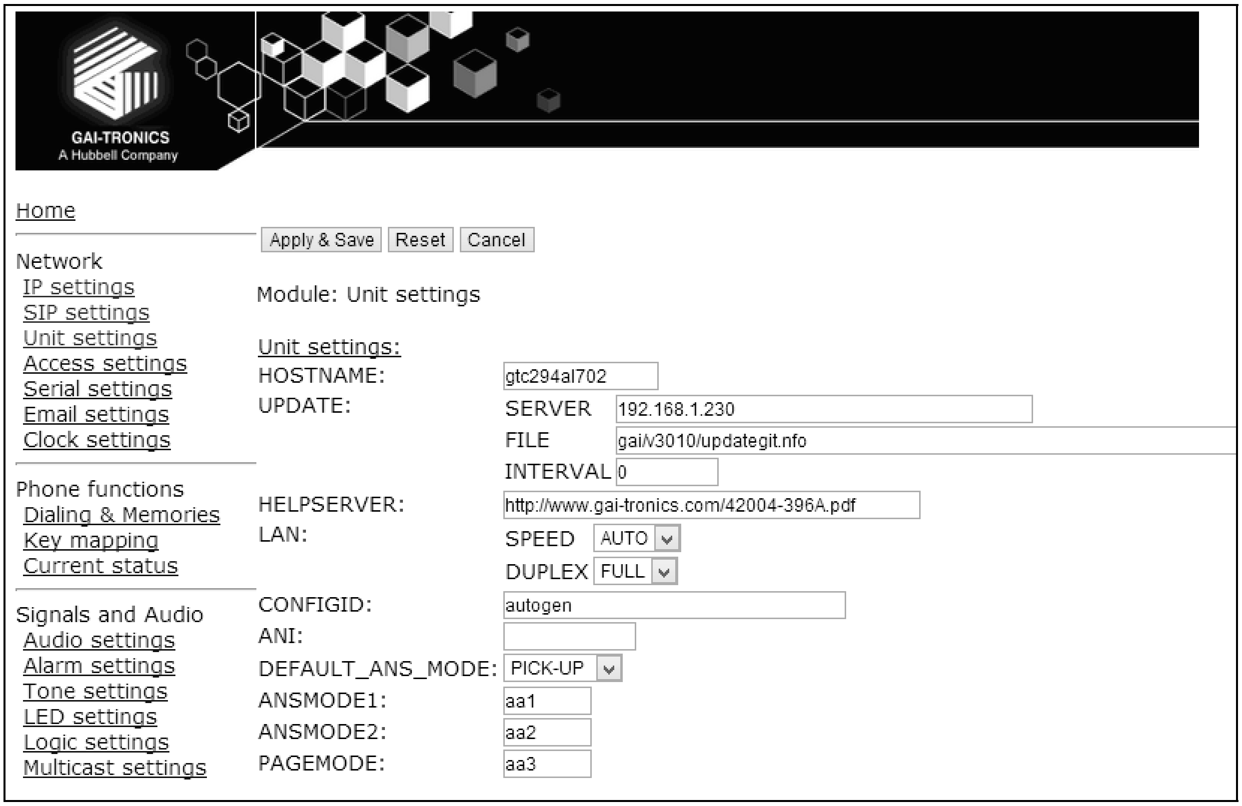

Figure 4. Unit Settings Page

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

Parameter	Function
HOST NAME	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 speaker on the system. Default value: A unique string starting with "GT" and followed by the serial number of the main circuit board inside the speaker (referred to as the "Board serial" on the home page).
DEFAULT_ANS_MODE	Sets the default answer mode. This mode will be used to answer a call when ANSMODE1, ANSMODE2 and PAGEMODE are not triggered. Values available are RING, PICK-UP, PAGE and STEALTH. RING is normal telephone operation, where a button must be pressed or handset lifted to answer an incoming call. **(Not applicable to speakers.)** PICK-UP is speaker auto-answer mode, where the speaker auto-answers, preceded by an announcement tone. PAGE is where the telephone auto-answers. 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. **(Not applicable to speakers.)** 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. **(Not applicable to speakers.)** Default value: **PICK-UP** is the factory default and should remain unchanged.

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

Clock Settings Page

The speakers 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.

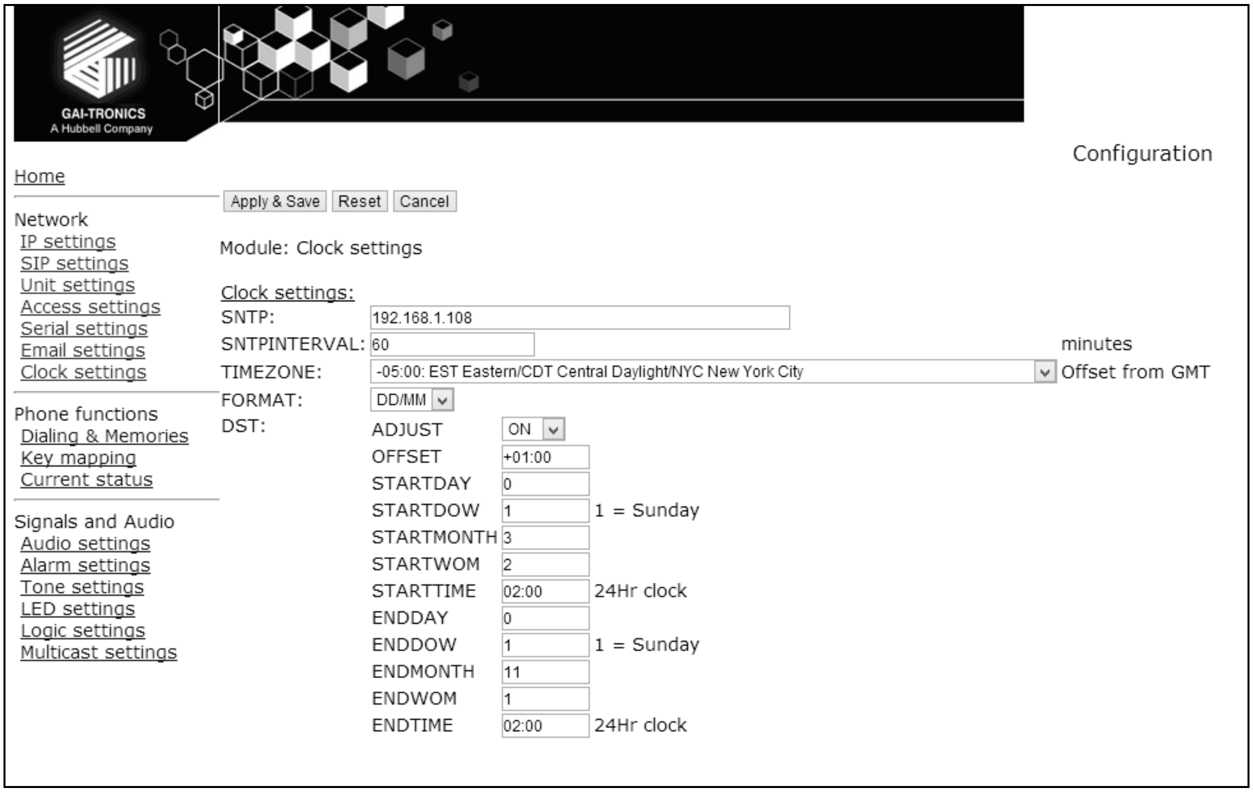

Figure 5. Clock Setting Page

Table 4 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 4. Clock Setting Parameters for basic configuration

Parameter	Function
SNTP	Sets the address for the SNTP server to be used, as an IP address or a FQDN.
SNTPINTERVAL	Sets the interval, in minutes, between SNTP update requests. Default value: **60**
TIMEZONE	Sets the current time zone for local time from a dropdown list.
FORMAT	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 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 speaker CODEC.

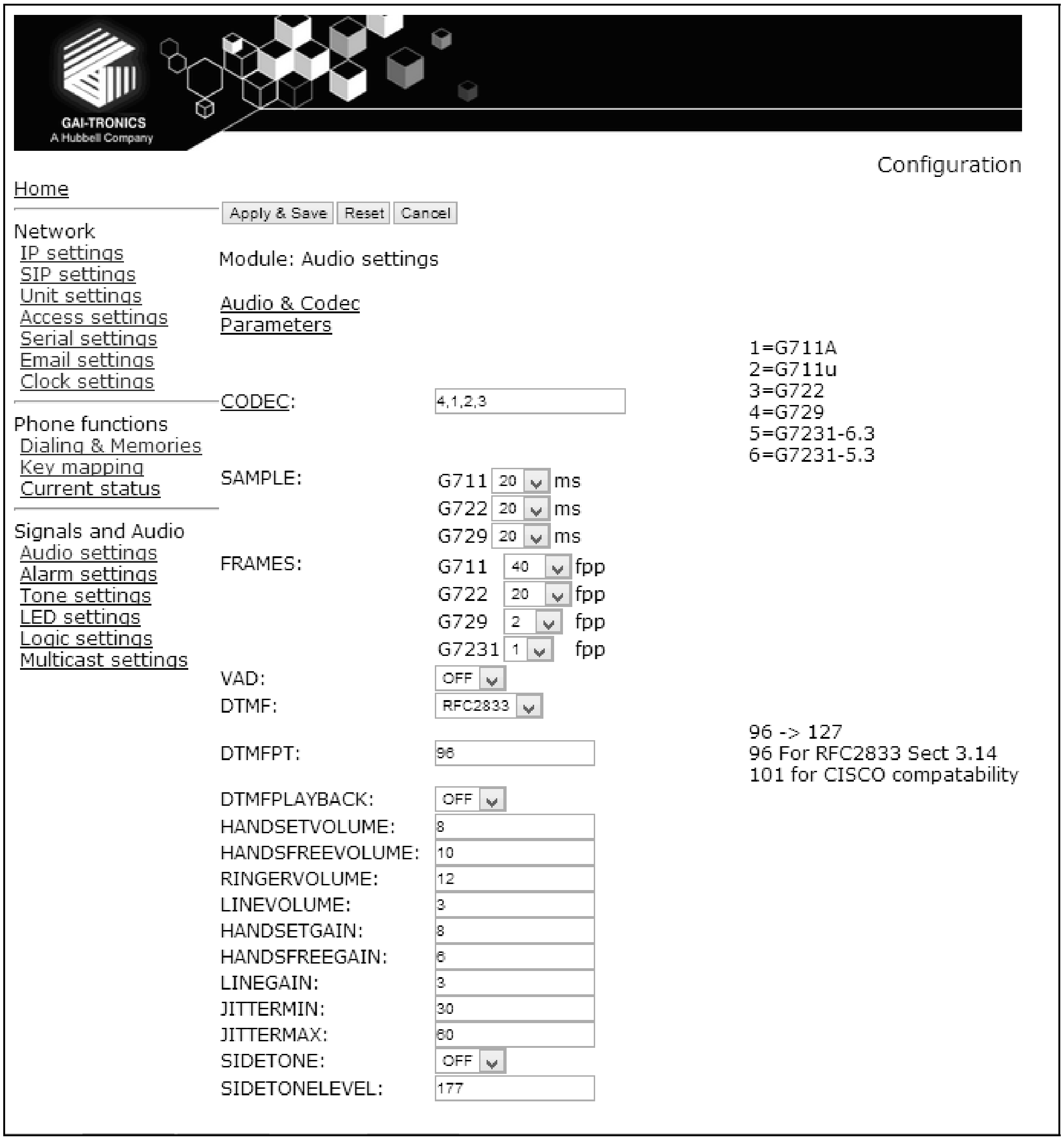

Figure 6. Audio Settings Page

Table 5 lists the parameters to be completed for the basic configuration. The parameters shown in Figure 6 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 5. Audio Settings Parameters for Basic VoIP Speaker Configuration

Parameter	Function
HANDSFREEVOLUME	This parameter sets the speaker volume. The range is 1–12.

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 desired operation.

Multicast Settings Page

Multicast allows a single audio stream to be sent to multiple endpoints simultaneously, to achieve multi-point paging or Public Address functionality over IP.

NOTES:

- Multicast requires the use of a SIP server that specifically supports it, and each endpoint (i.e., speaker) 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 speakers to accept and play the subsequent audio.
- The GAI-Tronics speaker 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 speaker can receive normal calls. Normal calls can also be assigned a priority level, defining whether calls can override multicasts or vice versa.

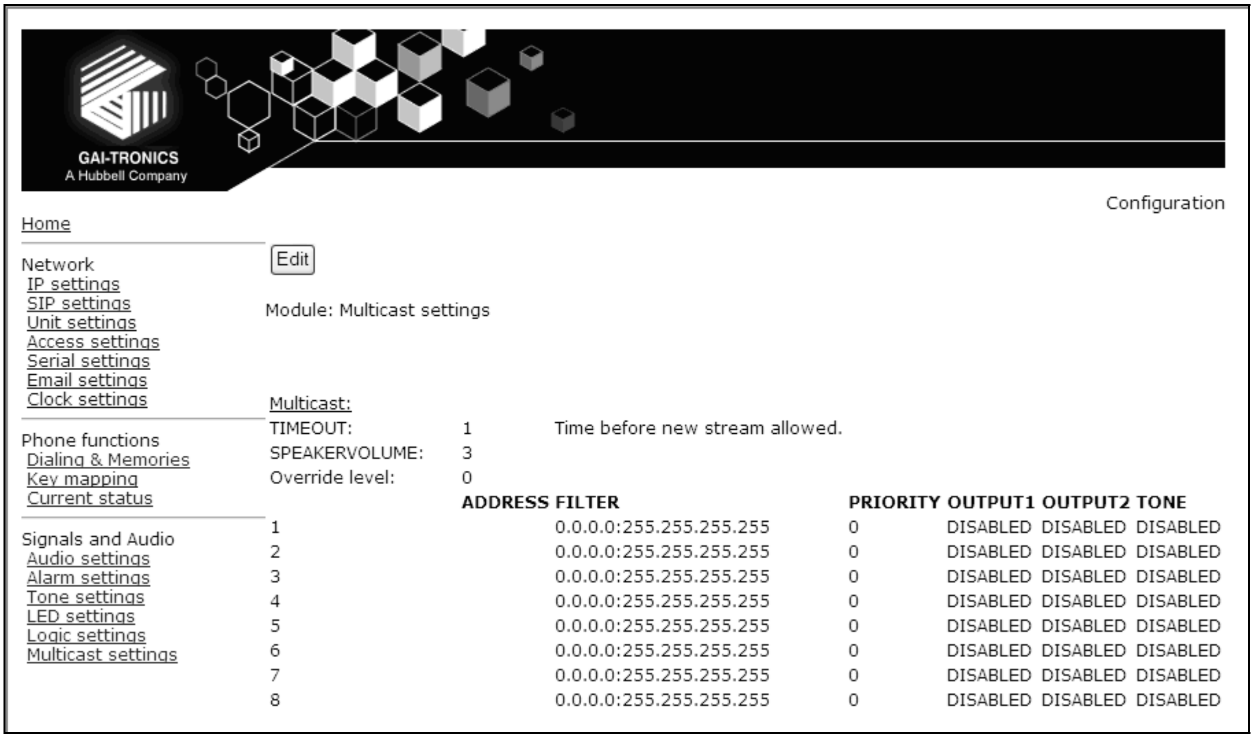

Figure 7. Multicast Settings Page

Click the **Edit** button to begin making changes. Click the **Apply & Save** button to save your changes when complete.

Table 6. Multicast Page Settings

Parameter	Function
TIMEOUT	Sets an enforced delay (in seconds) between one Multicast session ending and another beginning. The range is 1–120. Default value: **120**
SPEAKERVOLUME	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: **3**
Override level	Sets the override level (between 0 and 8) for normal 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 call will override a multicast having a priority of 6, but not one having a priority of 4. If a call and a multicast have the same priority level, the multicast will take precedence. If an incoming call is made to a speaker while a higher priority multicast is in progress, the call will not be connected to the speaker until the multicast has ended.
ADDRESS	The speaker 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.
FILTER	Sets a range of acceptable multicast source IP addresses. The speaker 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: **0.0.0.0:255.255.255.255**
PRIORITY	Sets a priority level for this multicast with respect to other multicasts and normal 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 call) occurs during an existing multicast, it will be interrupted and resume after the higher priority event has finished.
OUTPUT1	Sets whether OUTPUT1 is ENABLED or DISABLED during this multicast. If ENABLED 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 LOGIC page.
OUTPUT2	Sets whether OUTPUT2 is ENABLED or DISABLED during this multicast. If ENABLED 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 LOGIC page.
TONE	Sets whether tone 10 (TONES page) is ENABLED or DISABLED 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. This manual covers only the volt-free contact output settings. Refer to Pub. 42004-396 for input programming information.

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.

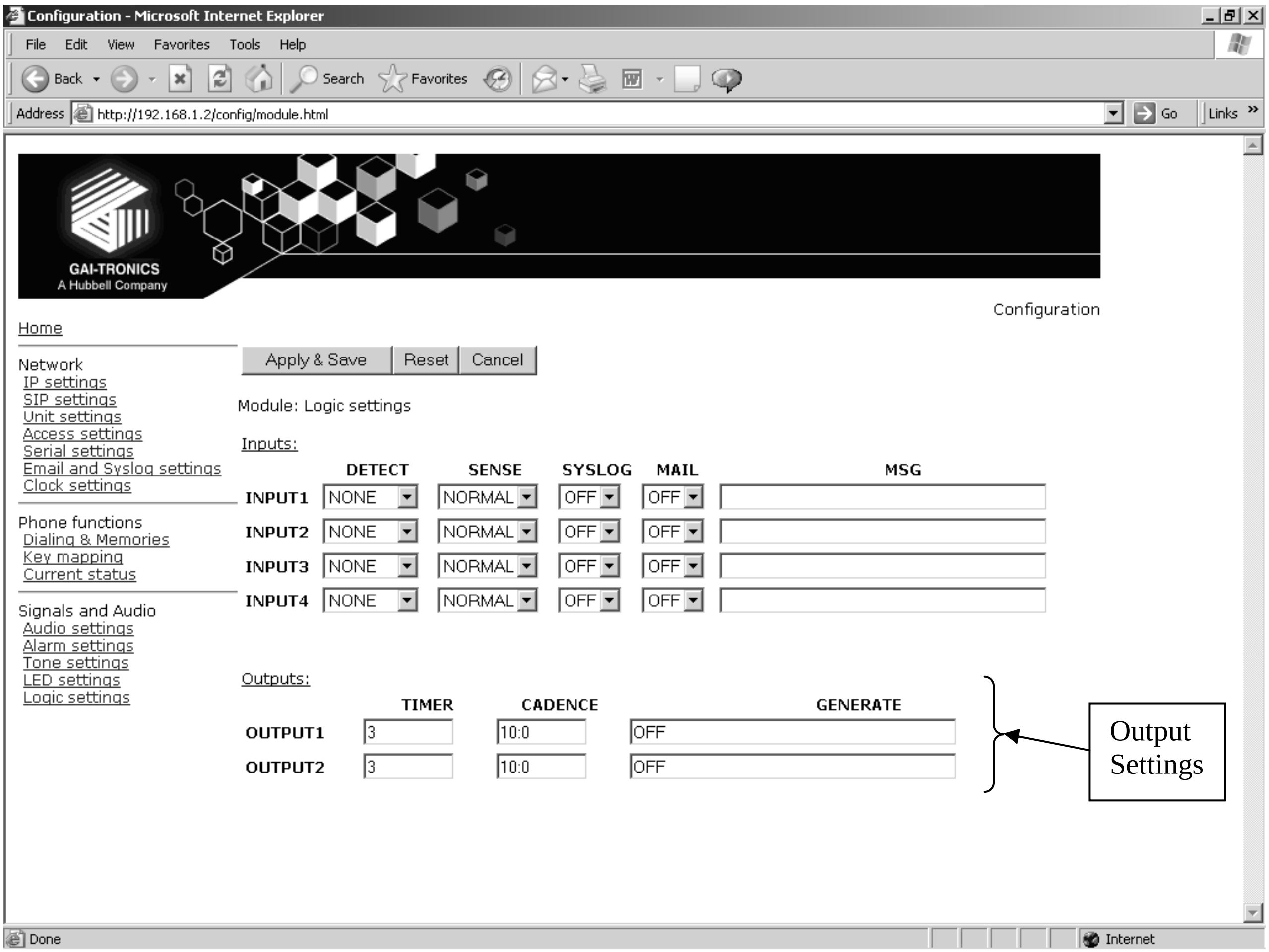

Figure 8. Logic Settings Page

Outputs

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

Table 7.

Parameter	Function	
GENERATE	The GENERATE field sets the function of the output by use of the following keywords:	
	"Generate" Keyword	Function
	ON	Sets the output permanently on.
	OFF	Sets the output permanently off.
	PULSE +	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.
	MUTE +	Sets the output to indicate if the audio input is muted.
	RING +	Sets the output to pulse when an incoming call is ringing. The pulsing on /off periods are set by the CADENCE field.
	CALL +	Sets the output to pulse when an outgoing call is active. The pulsing on /off periods are set by the CADENCE field.
	CONNECT +	Sets the output on when a call is connected.
	HOOK +	Sets the output on when the telephone is off hook, and off when it is back on hook.
	INUSE +	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.
	RINGCADENCE +	Causes the output to pulse in time with the ring tone cadence.
	RINGOUT +	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.
	PAGE +	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	
	REGISTERED +	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.
	EMERGENCY +	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.
	NOTE: 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.	
TIMER	Sets the timer value for the PULSE command in seconds. Default value is **3**. The minimum is 0 and the maximum is 3600.	
CADENCE	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 speaker should operate normally. If additional functionality is needed, please visit the 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 here check the website for more recent updates, or contact GAI-Tronics for support.

Problem	Possible Solution
Is the unit powered up?	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. 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.
I can't access the web pages.	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: • Are the IP and UNIT settings correct? • Is the web server enabled? • Can the speaker ping other destinations on the network? The IP module has PING and TRACEROUTE functions to help troubleshoot routing problems. • Some switches may not auto-negotiate speed correctly - try changing the LAN speed (UNIT module) from AUTO to 10.

Warranty

Equipment. GAI-Tronics warrants for a period of one (1) year from the date of shipment, that any GAI-Tronics equipment supplied hereunder shall be free of defects in material and workmanship, shall comply with the then-current product specifications and product literature, and if applicable, shall be fit for the purpose specified in the agreed-upon quotation or proposal document. If (a) Seller's goods prove to be defective in workmanship and/or material under normal and proper usage, or unfit for the purpose specified and agreed upon, and (b) Buyer's claim is made within the warranty period set forth above, Buyer may return such goods to GAI-Tronics' nearest depot repair facility, freight prepaid, at which time they will be repaired or replaced, at Seller's option, without charge to Buyer. Repair or replacement shall be Buyer's sole and exclusive remedy. The warranty period on any repaired or replacement equipment shall be the greater of the ninety (90) day repair warranty or one (1) year from the date the original equipment was shipped. In no event shall GAI-Tronics warranty obligations with respect to equipment exceed 100% of the total cost of the equipment supplied hereunder. Buyer may also be entitled to the manufacturer's warranty on any third-party goods supplied by GAI-Tronics hereunder. The applicability of any such third-party warranty will be determined by GAI-Tronics.

Services. Any services GAI-Tronics provides hereunder, whether directly or through subcontractors, shall be performed in accordance with the standard of care with which such services are normally provided in the industry. If the services fail to meet the applicable industry standard, GAI-Tronics will re-perform such services at no cost to buyer to correct said deficiency to Company's satisfaction provided any and all issues are identified prior to the demobilization of the Contractor's personnel from the work site. Re-performance of services shall be Buyer's sole and exclusive remedy, and in no event shall GAI-Tronics warranty obligations with respect to services exceed 100% of the total cost of the services provided hereunder.

Warranty Periods. Every claim by Buyer alleging a defect in the goods and/or services provided hereunder shall be deemed waived unless such claim is made in writing within the applicable warranty periods as set forth above. Provided, however, that if the defect complained of is latent and not discoverable within the above warranty periods, every claim arising on account of such latent defect shall be deemed waived unless it is made in writing within a reasonable time after such latent defect is or should have been discovered by Buyer.

Limitations / Exclusions. The warranties herein shall not apply to, and GAI-Tronics shall not be responsible for, any damage to the goods or failure of the services supplied hereunder, to the extent caused by Buyer's neglect, failure to follow operational and maintenance procedures provided with the equipment, or the use of technicians not specifically authorized by GAI-Tronics to maintain or service the equipment. THE WARRANTIES AND REMEDIES CONTAINED HEREIN ARE IN LIEU OF AND EXCLUDE ALL OTHER WARRANTIES AND REMEDIES, WHETHER EXPRESS OR IMPLIED BY OPERATION OF LAW OR OTHERWISE, INCLUDING ANY WARRANTIES OF MERCHANTABILITY OR FITNESS FOR A PARTICULAR PURPOSE.

Return Policy

If the equipment requires service, contact your Regional Service Center for a return authorization number (RA#). Equipment should be shipped prepaid to GAI-Tronics with a return authorization number and a purchase order number. If the equipment is under warranty, repairs or a replacement will be made in accordance with the warranty policy set forth above. Please include a written explanation of all defects to assist our technicians in their troubleshooting efforts.

Call 800-492-1212 (inside the USA) or 610-777-1374 (outside the USA) for help identifying the Regional Service Center closest to you.