

GAI-TRONICS
A division of Hubbell Ltd

VoIP Telephones



Configuration Guide: Firmware version 3

(The previous versions of this manual, for firmware versions 1 & 2, remain available)

VoIP Telephones

Configuration Guide Firmware version 3

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Contents

1. Introduction.....	5
2. What's new ?.....	5
2.1 New in Version 3.....	5
2.2 New in Version 2.....	6
3. How the product is intended to work.....	6
3.1 Operating Sequence.....	7
3.2 Dictionary of terms.....	7
4. Setting up and Configuring the telephones.....	9
4.1 Quick Start.....	9
4.2 Frequently Asked Questions (FAQs).....	9
4.2.1 What network facilities do I need to provide?.....	10
4.2.2 How do I set up dialling and memory lists?.....	10
4.2.3 Can I set the phone to make calls without a proxy (ie peer-to-peer)?.....	11
4.2.4 How do I set up Real-time alarm reporting via email or syslog?.....	11
4.2.5 How can I set up an external beacon to flash when the phone is ringing?.....	12
4.2.6 How do I set up a door-entry system?.....	13
4.2.7 How can I use the phone to make paging or PA announcements?.....	14
4.2.8 What additional features are available with CMA?.....	14
5. Web pages in detail.....	14
5.1 Login.....	15
5.2 Home Page.....	16
5.3 IP settings.....	17
5.3.1 Note about Syslog:.....	18
5.4 SIP settings.....	19
5.4.1 SIP Info sub-pages:.....	21
5.5 Unit settings.....	22
5.5.1 Audio Path Test.....	24
5.6 Access settings.....	25
5.7 Serial settings.....	26
5.8 Email settings.....	27
5.9 Clock settings.....	28
5.10 Dialling & Memories.....	29
5.10.1 Memories sub-page.....	30
5.10.2 Memory Lists sub-page.....	31
5.10.3 Basic Info sub-page.....	32
5.11 Key mapping.....	33
5.11.1 Activating relays using DTMF codes.....	35
5.12 Current status.....	36
5.13 Audio settings.....	37
5.14 Alarm settings.....	39
5.15 Tone settings.....	42
5.15.1 Suggested Tone Settings for Various Countries:.....	44
5.16 LED settings.....	45
5.17 Logic settings.....	48
5.18 Multicast settings.....	50
6. Configuration File update.....	53
6.1 Configuration File Syntax.....	55
6.2 Configuration File Commands.....	56
7. Time Zone Table.....	66
8. Example Configuration File.....	68
9. Command Line Interface.....	72
9.1 CLI Syntax.....	73
9.2 ACCESS Module Command Line Syntax.....	74
9.3 ALARMS Module Command Line Syntax.....	75
9.4 KEY Module Command Line Syntax.....	76
9.5 LED Module Command Line Syntax.....	76
9.6 DIALPLAN Module Command Line Syntax.....	77
9.8 CLOCK Module Command Line Syntax.....	78
9.9 AUDIO Module Command Line Syntax.....	79

9.10	TONES Module Command Line	79
9.11	IP Module Command Line Syntax	80
9.12	LOCAL Module Command Line Syntax	80
9.13	LOGIC Module Command Line Syntax	81
9.14	SIP Module Command Line Syntax	82
9.15	SMTP Module Command Line Syntax	83
9.16	Status Module Command Line Syntax	83
9.17	UNIT Module Command Line Syntax	84
9.18	Multicast Module Command Line Syntax	85
10.	Reset to Defaults	86
11.	Troubleshooting	87
11.1	Is the unit powered up?	87
11.2	I can't access the web pages.....	87
11.3	I can't make calls	87
11.4	Calls connect but there is no speech (or sound is garbled)	88
12.	Licensing Notices.....	88

1. Introduction

This guide provides information on the operation and configuration of GAI-Tronics' range of rugged VoIP telephones with firmware version 3, released in January 2012.

There are significant changes to some of the web pages and commands from those in previous versions. Issue 1 and issue 2 of this manual will remain available on the GAI-Tronics UK website (www.gai-tronics.co.uk/voipsupport.htm) as a reference for earlier versions.

The firmware version of each unit is displayed at the bottom of its home web page, and as part of the welcome message following login via a Telnet or serial connection. In each case the firmware version is a series of 3 numbers separated by dots (periods). The main firmware version is the first number. For example:

- 1.2.13 indicates firmware version 1
- 2.1.6 indicates firmware version 2.
- 3.0.0 indicates firmware version 3.

Upgrading to the latest version is possible in most circumstances, but please note that certain new features may not be enabled on upgraded phones - contact GAI-Tronics for details.

GAI-Tronics VoIP telephones are available in a variety of model styles, including handset and hands-free models, but the programming and configuration methods are common to all. Please note that the features may depend on the model type, and that therefore this guide may describe features not available on the particular model being configured.

Features of the GAI-Tronics range of VoIP telephones include:

- SIP compatible (RFC3261) only
- Registration with multiple SIP proxies (new in v2)
- Configurable via web pages, serial link or downloading a configuration file
- Outgoing cascading call lists
- Real-time alarm reporting via email or Syslog
- 4 auxiliary inputs, 2 volt-free contact outputs (revised in v2)
- Remote operation of contacts ("door opening" function)
- 3 "autoanswer" modes, including paging mode (revised in v2)
- Compatible with GAI-Tronics' Call Management Application (CMA)

This guide does not include information on:

- Installation, cabling and connections (see guide 502-20-0115-001 for non ATEX phones and 502-20-0133-001 for Auteldac4 VoIP)
- Setting up, configuring and operating a network for VoIP. Please ensure that the network is configured to allow VoIP communications (using the SIP protocol) between the desired locations before attempting to configure GAI-Tronics telephones.

2. What's new ?

2.1 New in Version 3

Version 3 added the following features from version 2:

Acoustic Path Testing (APT)

APT allows remote testing of handsets, microphones and speakers. APT can be used to verify that a phone is functioning acoustically. The test can be run on demand or on a scheduled basis, reporting its results via Syslog and / or email. See section 5.5.1.

Multicast

(Only applicable to hands-free products). Multicast allows a single audio stream to be received by multiple endpoints simultaneously, to achieve multi-point paging or Public Address functionality over IP. (Requires a multicast compliant SIP server). 8 definable multicast address ranges, with individual priority levels, for zoning. Assignable relay outputs and splash tones. See section 5.18.

Reset to factory defaults

The unit can be restored to factory defaults by pressing an internal reset button on the PCB. This can be used to recover a phone where the user name or password have been lost. See section 10.

Relay operation from DTMF tones

For door opening purposes, output relays can now be triggered by the receipt of a programmable DTMF tone sequence. This means that a phone's relays could potentially be triggered from any phone on the system. See section 5.11.1.

Default autoanswer mode

The unit can now be set to automatically answer an incoming call, with a choice of 3 different autoanswer modes to suit applications such as paging and intercom. See section 5.5.

2.2 New in Version 2

Version 2 firmware added the following features from version 1:

Multiple SIP proxies

The unit can now hold up to 4 alternate addresses for the SIP proxy and registrar with a prioritised failover sequence between them. This means that if it fails to register with the first server it will attempt to do so with the next and so on.

The unit can be set to automatically refresh its registration at a predetermined interval to ensure that registration is maintained at all times (or if not raise an alarm).

This provides a high degree of resilience across the network and reduces the possibility of a single point of failure jeopardising the operation of the whole system.

Additional functions for Relay outputs and LEDs

In addition to the functions in version 1, the unit can now trigger its output relays and / or LEDs on:

- PAGE (activated by PAGEMODE, see section 5.5). For example a relay could be used to activate a public address amplifier, allowing the unit to be used as a mini PA.
- EMERGENCY (if an outgoing call is designated as an emergency call), where for example a relay could be used to activate an emergency beacon, and
- REGISTERED, where for example an LED could indicate that the unit is available for use (i.e. it can make a call).

Additional LED drive

Version 2 allows 3 programmable LEDs instead of 2. Note that the number of LEDs fitted varies with model type. Some standard models have no LEDs fitted.

Page Mode

Auto-answer mode 3 is now explicitly referred to as PAGE MODE to highlight its potential use as a PA or paging system. Functionally it is unchanged, except for the LED and relay triggers described above.

3. How the product is intended to work

The VoIP telephone has been designed to mimic the behaviour of a traditional, analogue telephone, specifically based on the GAI-Tronics range of rugged telephones, to give continuity where VoIP and analogue units are used in similar situations.

Accordingly, traditional telephone terminology is used throughout the manuals and documentation, and many of the features are designed to mimic analogue telephone behaviour.

A major difference between analogue telephones and VoIP is that, with analogue units, most signalling and tones such as ringing, dial tone, busy tone etc., are provided by a telephone exchange (PABX), whereas the VoIP unit must generate these itself. The telephone provides features to change the various tones to emulate those of different countries or PABXs, to give familiar operation in its intended location.

3.1 Operating Sequence.

Typical sequences of events for various model types are explained below:

Handset models (Titan, Commander)

Placing a call

- Lift handset (off hook)
- Dial tone in receiver
- Dial number - confidence tones in receiver
- Call progress tone in receiver (e.g. ring tone)
- Call is answered by remote party
- Normal voice call
- Replace handset (on hook)
- Call terminates.

Receiving a call

- Telephone rings
- Lift handset (off hook)
- Normal voice call
- Replace handset (on hook)
- Call terminates.

Hands-free models (VR, Help Point)

Placing a call

- Press button
- Dialling confidence tones heard from speaker (wake and dial)
- Call progress tone heard from speaker (e.g. ring tone)
- Call is answered by remote party
- Normal voice call
- Call terminates. (On hook)

Receiving a call

- Ringing heard from speaker
- Press any button to answer call (off hook)
- Normal voice call
- Call terminates. (On hook)

3.2 Dictionary of terms

Busy tone

A tone played to the user to indicate that a call has failed because the called party is engaged

Call progress tone

One of a number of different tones played to the user to indicate the status of a call. Dial tone, busy tone and NU tone are all examples of call progress tones.

Confidence tones

Tones played to the user to indicate that dialling is in progress, by imitating DTMF tones used by analog telephones.

Dial tone

A tone played to the user to indicate that the telephone is ready to dial – ie it is off hook and waiting for a button to be pressed to initiate a call.

Dialling

Used to describe the process of initiating a call, usually by pressing a memory button or a series of digit buttons.

DTMF

Standing for “dual tone multi-frequency”, the dialling digit tones produced by a touch-tone phone. Commonly used for signalling in analogue systems.

Handset phone

Used to denote a telephone from the GAI-Tronics Titan or Commander product ranges, with a separate handset attached to the main telephone body by a heavy duty flexible cord. No separate loudspeaker is fitted to these models.

Hands-Free phone

Used to denote a telephone from the GAI-Tronics Help Point or Vandal Resistant product ranges, with a microphone and speaker integrated into a flat panel. No corded handset is fitted to these models.

LNR

Standing for “last number redial”, this is a button provided on some models of GAI-Tronics phone to redial the last manually dialled number.

Memory dial number

On an analogue or cellular phone, memory numbers are pre-stored digit sequences used to start calls. With VoIP these can also be URI's rather than numbers, but are still referred to in the same way.

Mute

A function to temporarily mute the microphone so that the remote party cannot hear. On GAI-Tronics telephones this function is provided by the "S" button.

NU tone

Number unobtainable tone – used to indicate that a call cannot connect due to the end point not being recognised.

Off hook

Used to denote the state of a telephone during an active call, or when a call has been initiated. For a handset phone, off hook usually means that the handset is lifted.

On hook

Used to denote a telephone in the idle state – no call started or answered. A telephone is still on hook when it is ringing on an incoming call. For a handset phone, on hook usually means the handset is not lifted. If a call is terminated whilst the handset is still lifted (for example by the CALL LIMIT timer), the telephone is placed into the on hook state. For a hands-free phone, on hook means that no ON or WAKE & DIAL button has been pressed following a terminated call or reset.

Recall

On analogue phones, the Recall button is used to activate exchange signal, usually to transfer a call. The GAI-Tronics VoIP telephone does not have a recall facility, but the “R” button (where fitted) can be used to activate an output on a remote phone, for example as a door release.

Register Fail tone

A tone played to the user initiating a call to indicate that the telephone is not currently registered with a registrar, meaning that a call cannot be made.

Ring tone

A tone played to the user initiating a call to indicate that the call has been placed but not yet answered. This usually signifies that the remote end is ringing.

Ringing

A loud alert tone made by the telephone indicating that an incoming call is ready to be answered.

Secrecy (mute)

A function to temporarily mute the microphone so that the remote party cannot hear. On GAI-Tronics telephones this function is provided by the "S" button.

Sidetone

On handset phones, part of the microphone signal is fed to the earpiece so that the user can hear his or her own voice during the call. This makes it a more natural experience, and has been a feature of analogue telephones since their invention. Not used on hands-free phones.

4. Setting up and Configuring the telephones.

Each telephone must be configured for use on the intended network. Most models have memory-dial locations, which will need to be set up. The telephone also has a range of customisable features. All of these can be set up using one of 4 different methods:

- Web pages (the simplest and quickest method for configuring an individual phone)
- Downloading configuration files (the most efficient method for multiple updates)
- Command-line commands via direct serial link
- Command-line commands via Telnet session

Note:

All the above access methods require you to know the unit's username and password.

All methods, except direct serial link, also require you to know the unit's IP address.

Please ensure these details are recorded securely once set or changed.

All of the telephone's features can be configured using any of the above methods, but the most complete description of features is contained in the web page section (Section 5).

4.1 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 24-48Vdc or PoE)^{1, 2}
- Using a web browser, browse to the default IP address **192.168.1.2**
- Enter the user name and password (Defaults: **user & password**)
- Set an IP address and net mask (or set DHCP) on the IP page
- From the SIP settings page, select the SIP1 Info sub-page, check that ENDPOINT is ENABLED
- On the SIP1 Info sub-page, give the phone 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.
- Program any dial memories using the Dialling & Memories pages

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 phone is registered and what is happening during calls (refresh the page to see changes).

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.

4.2 Frequently Asked Questions (FAQs)

Note: a more up-to date list of questions and answers may be available on the GAI-Tronics website. See www.gai-tronics.co.uk/voipsupport.htm for more details.

¹ Early models will only accept 48Vdc as an external power supply, later models will accept 24-48Vdc. Units are marked accordingly next to the power terminals - see installation guide 502-20-0115-001 (or 502-20-0133-001 for Auteldac4 VoIP) for details.

² For Auteldac4 VoIP, PoE can only be supplied on the spare pair (802.3af alternative B) not the data pairs. See installation guide 502-20-0133-001 for details.

4.2.1 What network facilities do I need to provide?

This may vary widely depending on how your network is constructed and what else it is carrying, but as a general guide you will probably need:

- A SIP proxy server (to route calls)
- A SIP registrar server (frequently combined with the proxy server) to resolve URIs to IP addresses
- A TFTP server (for downloading configuration files).
- A TCP Syslog server (for reporting alarms and external inputs)
- An SMTP server (for reporting via email)
- An STNP server (to synchronise the internal clock)

Dedicated systems, such as Gatekeepers, VoIP-enabled PABXs or soft PABXs may also provide these functions. Bear in mind that GAI-Tronics telephones only support Session Initiation Protocol (SIP) to RFC3261, as opposed to H.323 or SCCP VoIP protocols for example.

Note that the performance of VoIP telephones depends on the provision of sufficient bandwidth and prioritisation on the network to give the quality of service required.

4.2.2 How do I set up dialling and memory lists?

Let's assume you have a telephone with 2 buttons: memory 1 for information, memory 2 for emergency. You want the emergency button to call "888" only. You want the information button to call the information desk, or if that is busy the security office, or failing that the administration centre on 223344.

First set up the 4 possible user agents (end points) as memories on the memories page (it doesn't matter which end point is in which memory):

	MEMORY	COMFORT
1	888	888
2	sip:info@mydomain.com	223344
3	sip:security@mydomain.com	223344
4	223344@mydomain.com	223344
5		
6		

Note that comfort strings have been set to give the user confidence that "dialling" is taking place when the button is pressed.

Then set up 2 memory lists, one for each button:

	WAKEANDDIAL	LIST
0	OFF	
1	ON	2:3:4
2	ON	[]
3	OFF	
4	OFF	
5	OFF	

Memory list 1 relates to memory button 1, and will dial memories 2, 3 and 4 in cascade. Memory list 2 is for memory button 2, and will dial memory 1 only. Note that, in this case, WAKEANDDIAL is set for both - the normal case for help point and hands-free telephones. Refer to the Dialling & Memories pages in section 5.10 for more details.

4.2.3 Can I set the phone to make calls without a proxy (ie peer-to-peer)?

There are two ways of setting the phone to make peer to peer calls.

The first is where there is no proxy server on the system at all. In this case:

1. Set the ENDPOINT field on SIP 1 Info page to ENABLED, but make sure DOMAIN, PROXY and REGISTRAR are blank. Set ENDPOINT on SIP info 2, 3 and 4 to DISABLED.
2. Make each entry on the Memories page the address of an endpoint or phone, in the form 1@192.168.1.2. Note that the number before the "@" symbol is not normally significant³ - there just needs to be a number, followed by "@", followed by the IP address of the end point.⁴
3. Note that peer-to-peer calls can only be made by using a memory - not by manually dialling from a numeric keypad. All phones have at least one memory list (the OFFHOOK list). Refer to section 5.10 for details on setting up memories.

The second way is where one or more proxy servers are in use, but you want to be able to make a peer-to-peer call if no proxy is available. This is referred to as "failover to peer-to-peer". In this case:

1. Set the proxy address on one of the 4 SIP info pages (usually the one with the lowest priority) to be the IP address of an endpoint, in the form [192.168.1.2](#), but set the REGISTRAR address to be blank.
2. If all attempts to make calls to higher priority proxies fail, the phone will attempt to place a call to this IP address as a peer-to-peer entity, regardless of what number is dialled or what entry is selected from a memory list.

NOTE: you cannot make a peer-to-peer call by entering an IP address on a numeric keypad - peer-to-peer calls can only be made using a memory dial.

4.2.4 How do I set up Real-time alarm reporting via email or syslog?

To do this you will need to set up email and/or syslog facilities within the phone, then set up the alarm itself, using the following 3 web pages:

- Refer to the Email page to enter the required SMTP server settings for email.
- Refer to the IP settings page to set up Syslog server settings.
- Refer to the Alarms page to set which alarm events will report.

In the example shown below, a syslog message will be generated if the telephone has a cold reset (ie recovers from a power failure) or has an integrity loop fault (ie the handset has been detached). In addition, it will send an email to the security office if the handset is detached.

³ Some SIP phones may require this to be their phone number.

⁴ Note, this could also be a FQDN (fully qualified domain name) if there is a DNS available on the network



Configuration

Apply & Save Reset Cancel

Page 1 - Alarm Configuration, 2 - Alarm History

Module: Alarm settings

Alarms:

	REPORT	STATUS	ONTIME	OFFTIME	SYSLOG	MAIL	MSG
Handset Integrity Loop	ON	OFF	2	2	ON	ON	Handset detached
Configuration Error	NONE	OFF	0	0	OFF	OFF	
Cold Reset	ON	OFF	0	0	ON	OFF	Power interruption
Warm Reset	NONE	OFF	0	0	OFF	OFF	

4.2.5 How can I set up an external beacon to flash when the phone is ringing?

Traditional telephone beacons and sounders, with ring detectors, will not work on VoIP because there is no ring signal. You will therefore need a powered beacon or sounder instead, and use the telephone's volt-free contacts to activate it. These beacons or sounders must be provided with a separate power supply - they cannot be powered from the telephone. Having connected an external device to an output (say Output 1), the next step is to set the output to activate it when required.



Configuration

Apply & Save Reset Cancel

Module: Logic settings

Inputs:

	DETECT	SENSE	SYSLOG	MAIL	MSG
INPUT1	NONE	INVERT	OFF	OFF	
INPUT2	NONE	INVERT	OFF	OFF	
INPUT3	NONE	INVERT	OFF	OFF	
INPUT4	NONE	INVERT	OFF	OFF	

Outputs:

	TIMER	CADENCE	GENERATE
OUTPUT1		10:0	RING
OUTPUT2		10:0	OFF

Enter the keyword "RING" for the relevant output. The example above shows the output set with a cadence of 10:0, meaning continuously on. This would be suitable for a beacon, because beacons usually flash (once per second) when permanently energised. It might not suit a sounder, however, because it would emit a continuous tone, which might not be recognisable as a phone ringing. For a sounder on its own, the keyword "RINGCADENCE" is a better option, causing the sounder to be energised in time with the normal phone ringer. For a beacon and sounder together, it is often best to use a separate output for each as shown:

Configuration

Apply & Save Reset Cancel

Module: Logic settings

Inputs:

	DETECT	SENSE	SYSLOG	MAIL	MSG
INPUT1	NONE	INVERT	OFF	OFF	
INPUT2	NONE	INVERT	OFF	OFF	
INPUT3	NONE	INVERT	OFF	OFF	
INPUT4	NONE	INVERT	OFF	OFF	

Outputs:

	TIMER	CADENCE	GENERATE
OUTPUT1		10:0	RING
OUTPUT2		10:0	RINGCADENCE

In this example, Output 1 is set to activate a flashing beacon, whilst Output 2 is set to activate a sounder in sync with the cadence of the ring signal (set on the Tone settings page). In both cases the outputs are energised when the phone is ringing with an incoming call, and de-energised when the call is answered or disconnected. Refer to the Logic Settings page (section 5.17) and Tone settings page (section 5.15) for more details.

4.2.6 How do I set up a door-entry system?

A common application is to have a single button hands-free telephone mounted outside a door, and a 15 button Commander model at a remote security point. Visitors arriving at the door use the hands-free unit to call the security point. A security guard answering the call can release the door lock by pressing the "R" button on the Commander unit. To achieve this, connect one of the volt-free outputs on the hands-free telephone (say output 1) to the electronic door release mechanism. Using the Logic settings page, set this output to PULSE:

Configuration

Apply & Save Reset Cancel

Module: Logic settings

Inputs:

	DETECT	SENSE	SYSLOG	MAIL	MSG
INPUT1	NONE	INVERT	OFF	OFF	
INPUT2	NONE	INVERT	OFF	OFF	
INPUT3	NONE	INVERT	OFF	OFF	
INPUT4	NONE	INVERT	OFF	OFF	

Outputs:

	TIMER	CADENCE	GENERATE
OUTPUT1	3	10:0	PULSE
OUTPUT2		10:0	

Note that the TIMER is set to 3, meaning that the output will remain active for 3 seconds after being activated. To activate this output from the security office, set the RECALL setting on the Key mapping page of the Commander unit to the IP address of the hands-free unit. So, for example, if the IP address of the hands-free were 192.168.9.2, the setting would be:



Refer to the Logic settings page (section 5.17) and the Key mapping page (section 5.11) for more details.

4.2.7 How can I use the phone to make paging or PA announcements?

If you are using GAI-Tronics CMA, simply set the PAGEMODE field (on the UNIT page) to "aa3". CMA has a page button that will place a call to the unit in page mode, i.e. an announcement tone will be heard from the unit, following which the CMA operator will be able to make a page through the unit's speaker (see section 5.5).

It may be possible to activate this feature from systems other than CMA - contact GAI-Tronics for details.

Note page mode is usually implemented using handsfree models (VR and Help Point for example) but it may also be possible with other models, depending on application. The integral relays can also be set to activate during a page, and this feature could be used to trigger an external public address amplifier. Contact GAI-Tronics for details.

4.2.8 What additional features are available with CMA?

GAI-Tronics CMA is a security call centre application for Windows XP™ designed for use with GAI-Tronics analogue and VoIP telephones, providing powerful features such as:

- Automatic call answering
- Text-to-speech auto announcements
- Location ID linked to a user-definable mapping application
- Call recording and incident logging
- Call queueing

One of the system's most important functions is to give callers the reassurance that their call is being dealt with and that their location is known.

The ANI field on the UNIT page is used as an identifying token to CMA. Using this the telephone can automatically announce location information (using text-to-speech) to the user and the call centre operator when a call is made. It is also used to locate the phone on a map to help the operator identify its location and give assistance to the caller.

CMA can also activate 3 special auto-answer modes on hands-free VoIP telephones if required by using codes also entered on the UNIT page:

Stealth mode, where the operator can listen discreetly to the telephone (the ANSMODE1 field should be set to "aa1").

Intercom mode, where the operator can make a call to a telephone and start two-way voice communication immediately, without the user having to answer (the ANSMODE2 field should be set to "aa2").

Page mode, where the operator can make an announcement directly to the telephone, but not listen (the PAGEMODE field should be set to "aa3").

5. Web pages in detail

The following sections describe the embedded web pages in detail. Once past the login screen, all the pages have a similar layout.

Configuration - Microsoft Internet Explorer

Address: http://192.168.1.2/config/module.html?MODULE=SIP

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Configuration

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

Network

IP settings [Edit]

SIP settings

Unit settings

Access settings

Serial settings

Panel settings

Clock settings

Module: SIP settings

SIP General Proxy Parameters

LOCALPORT: 5060

PROXYFAILOVERSTATUSES: 5xx,6xx,49x,403,406,9xx,404

DONTSTARTMEDIAATRING: OFF

SENDTMPLAST: OFF

RTPTOS: 46

SINGLEPTIME: 20

SENDMULTIPARTMIME: OFF

Signals and Audio

Audio settings

Alarm settings

Tone settings

LED settings

Logic settings

General registration settings

MODE: MULTIPLE

REGTIMEOUT: 3600 seconds

PEREGTIMEOUT: 30 seconds (0 for REGTIMEOUT)

Navigation pane

Edit button

Module name

Links to sub-pages

Page values

The left hand navigation pane gives direct access to each of the 16 main pages, grouped by functional headings of Network, Phone functions and Signals & Audio, plus the home page. Most pages have an "Edit" button that allows the changing of parameters. Some pages have entry dialog boxes that accept certain predefined values. These values are listed in the sections below. Some pages have links to related sub pages. Each page displays its module name near the top for ease of navigation.

Note that these pages have been developed and tested on Microsoft Internet Explorer (v6). Screen layout may appear differently using other browsers.

5.1 Login

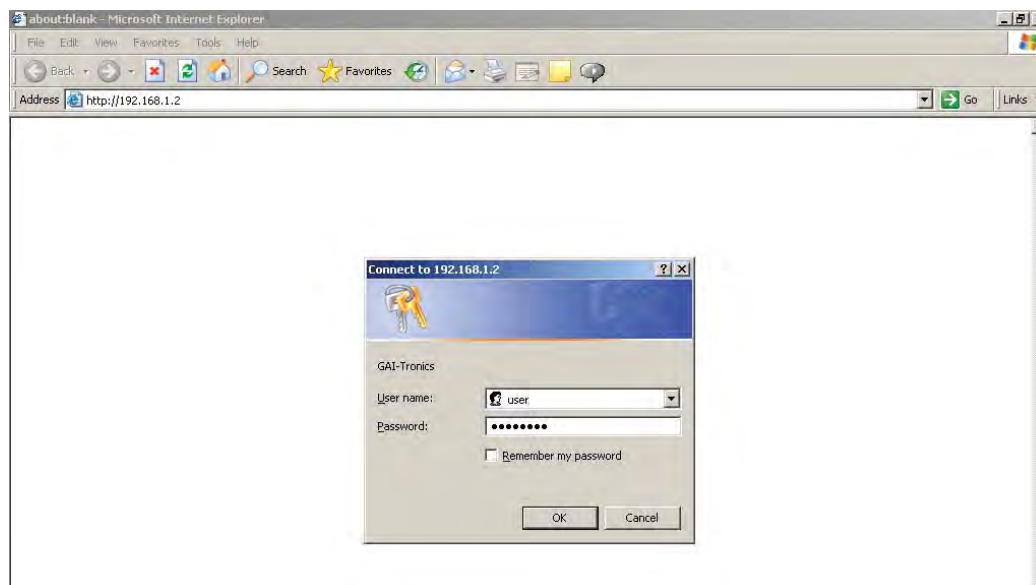
To access the web pages, navigate to the unit's IP address using a web browser such as Internet Explorer.

The factory default setting is for static IP addressing, with an address of:

192.168.1.2

Note that the unit's default subnet mask is 255.255.0.0.

The Phone will request a user name and password as shown.

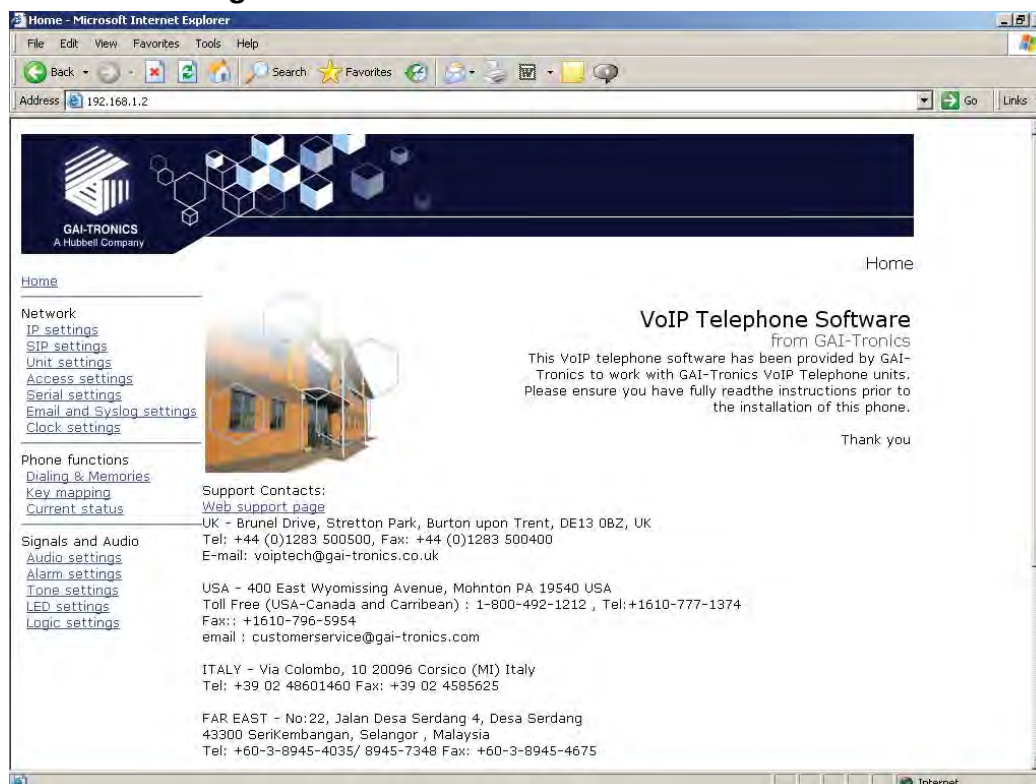


The default user name and password are

user
password
 (lower case)

The user name and password can be changed using the Access Settings page. On accepting the username and password, the phone's home page is displayed.

5.2 Home Page



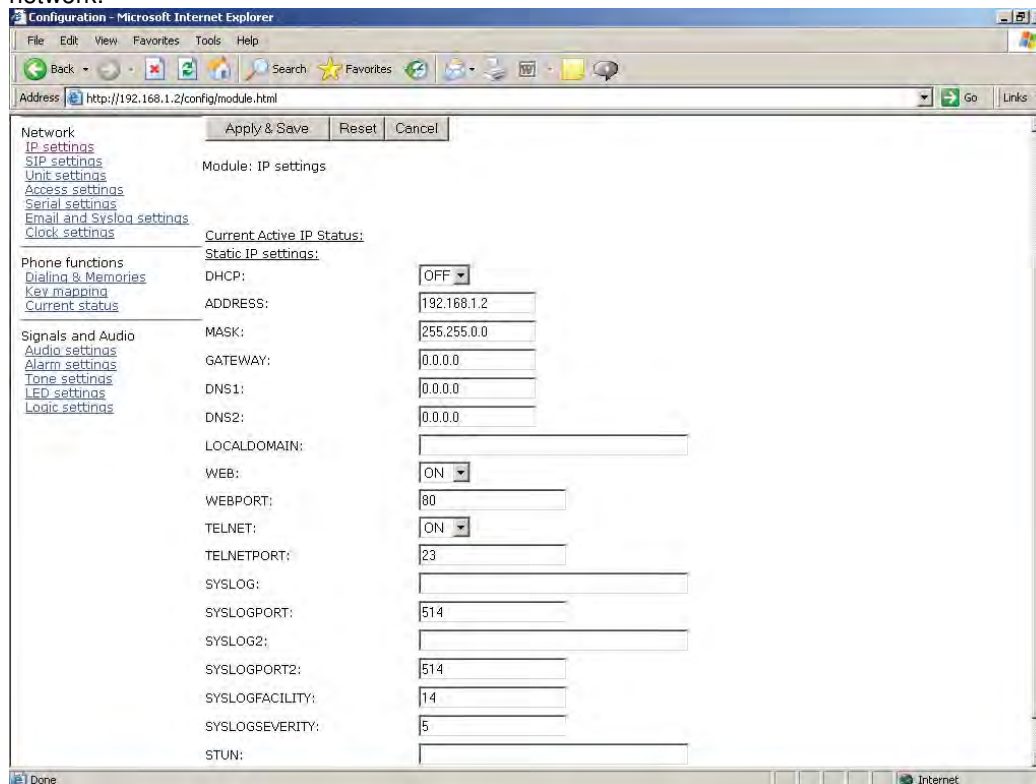
No settings can be changed directly from the home page.

The Web support page link defaults to <http://www.gai-tronics.co.uk/voipsupport.htm>, but can be changed on the Unit Settings page (section 5.5).

At the bottom of the home page (you may need to scroll down, depending on screen resolution) there is a list of information about the phone including serial numbers of the unit and its PCBs, software versions and MAC ID.

5.3 IP settings

The IP settings page is used to display or change various settings for connection to the IP network.



DHCP: 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: ON or OFF, default value is OFF)

ADDRESS: Sets the static IP Address of the unit. (Default value is 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 is 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 is 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 is 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 is 0.0.0.0)

LOCALDOMAIN: Sets the domain name of the telephone 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 is ON)

WEBPORT: Sets the TCP port through which the Telephone Web server can be accessed (Default Value is 80)

TELNET: Enables or disables access to the telnet server (Values available: ON or OFF, default value is ON)

TELNETPORT: Sets the TCP port through which the Telephones telnet server can be accessed (Default Value is 23)

SYSLOG: Sets the destination address for syslog server messages. (Valid values: IP address or FQDN. Default value: blank)

SYSLOGPORT: Sets the port number to be used for syslog messages. The default value is 514

SYSLOG2: Sets the destination address for a second syslog server for redundancy. (Valid values: IP address or FQDN. Default value: blank)

SYSLOGPORT2: Sets the port number to be used for syslog messages (second syslog server). The default value is 514

SYSLOGFACILITY: Sets the SYSLOG message facility level, as per RFC3164. (Default value: 14)

SYSLOGSEVERITY: Sets the SYSLOG message severity level, as per RFC3164. (Default value: 5)

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)

At the bottom of the IP settings page are 2 action buttons, each with an entry box. The entry boxes will accept either an IP address or FQDN. These buttons provide useful diagnostic functions:

PING: Sends an ICMP ping to the entered address, providing a results page.

TRACEROUTE: Executes a series of PING messages with varying HOP numbers in order to determine the routing used to reach the destination address. A results page is displayed.

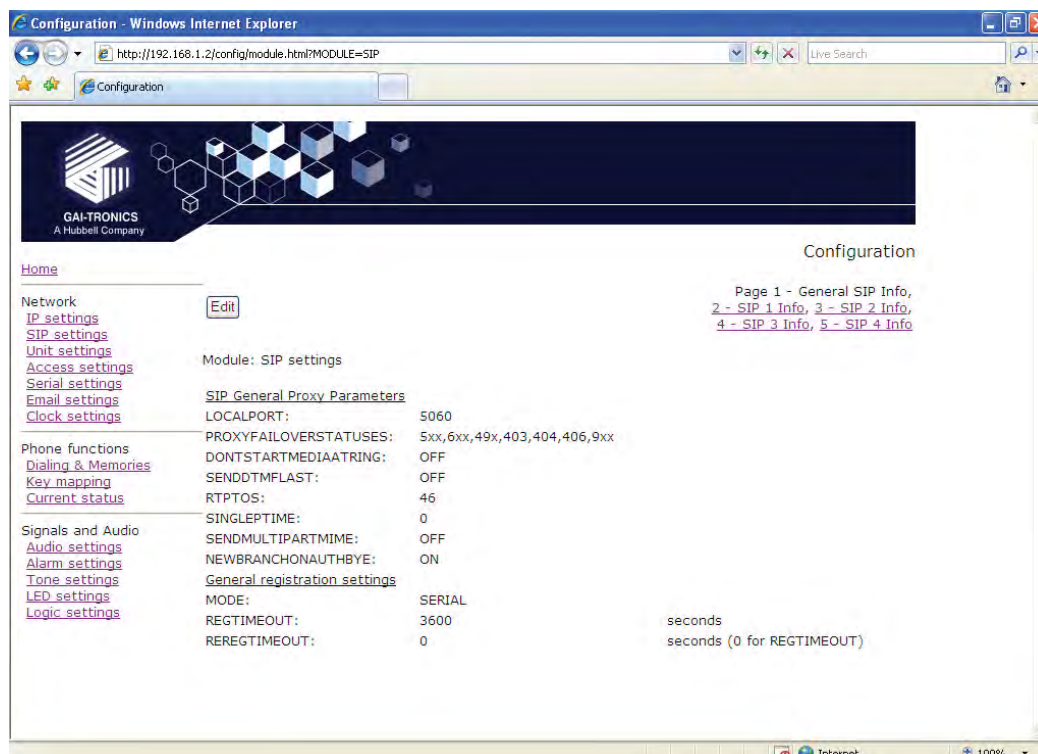
5.3.1 Note about Syslog:

GAI-Tronics VoIP products send Syslog messages using TCP (as opposed to UDP). Please make sure that Syslog servers support TCP.

SYSLOG over TCP ensures reliable delivery, and utilises port number 514 by default. Note that in the event of a TCP session failure there is no higher layer protocol acknowledging the receipt of the message, but each message has an Event Count parameter that will indicate if a previous message has been lost

5.4 SIP settings

The SIP settings page is used to view or change parameters specific to the SIP signalling protocol. GAI-Tronics VoIP phones can hold details of up to 4 SIP proxies. If the phone is unable to register or make a call it can fail over to the next in a prioritised sequence. There is a SIP Info page for each of the 4 possible endpoints, and a General SIP Info page containing details common to all. The 4 endpoint pages are sub pages of the General page shown below:



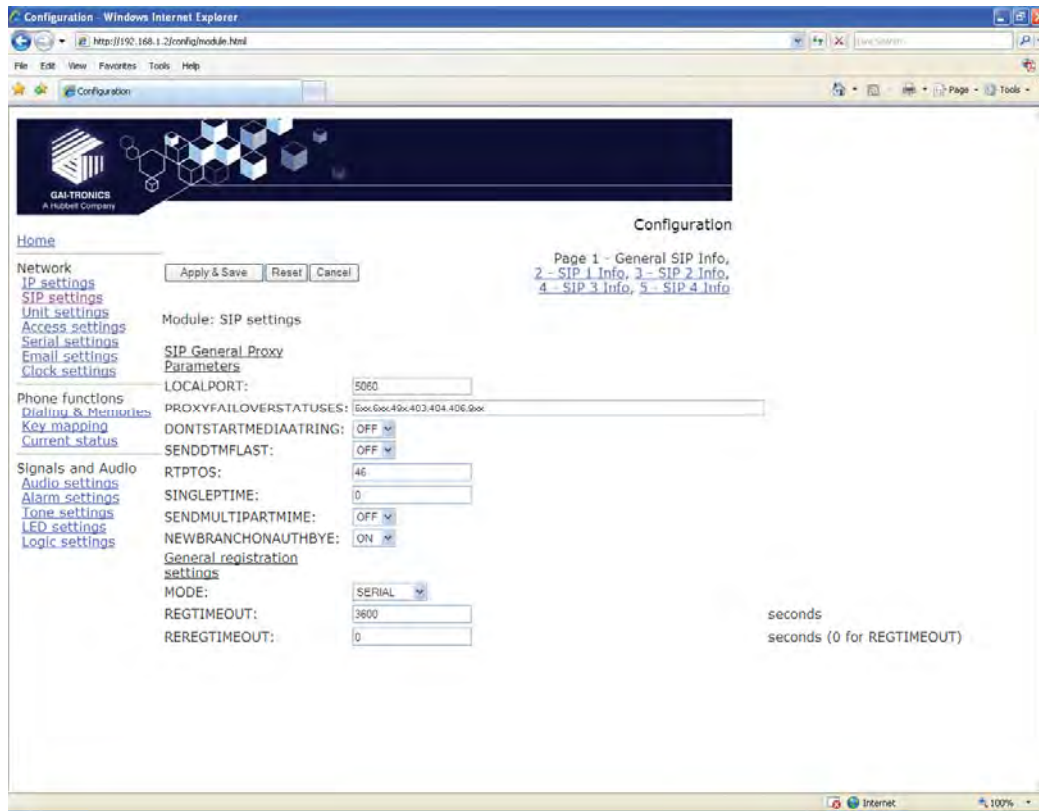
LOCALPORT: Configures the port number used for the local SIP signalling socket.

Default value: 5060

PROXYFAILOVERSTATUSES: This field contains a list of SIP error codes that will trigger a fail over from one proxy to the next. Codes are 3 digits and the wildcard character "x" can be used (ie 5xx would include any code from 500 to 599 inclusive). Codes are separated by commas. Maximum field length 79 characters, ie 20 codes. The default list is 5xx, 6xx, 49x, 403, 406, 9xx. Codes are as defined in RFC3261 except 9xx, which is defined as "time-out" and should always be included in the list.

Note that there are two failover mechanisms: one for proxies (defined here) and a second for memories (defined in section 5.10.3). If a call fails due to a proxy error, the phone will then try to place the call to the same number on the next proxy. If the call fails due to an endpoint problem (for example "busy"), the phone will try the next number in the list, on the current proxy.

DONTSTARTMEDIAATRING: This setting is not normally required. It can be used to delay the sending of media packets to end points until the call has been answered. Only required if problems are encountered with certain types of end point. Default value: OFF.



SENDDTMFLAST: This setting is not normally required. It can be used to reorder the codec sequence to end points, so that the DTMF codec is sent last. Only required if problems are encountered with certain types of end point. Default value: OFF

RTPTOS: Sets the value of the TOS/Diffserv field in the UDP packets carrying RTP data. This value prioritises traffic over the network to provide QoS (Quality of Service) for voice, see RFC2474. Valid values are 1->63 (Default value = 46)

SINGLEPTIME: Certain endpoints can only accept a single audio packet time regardless of CODEC (see AUDIO page). This field forces a single packet time to the value set in ms. Valid values are 0 to 100, where 0 disables the feature allowing codecs to use the packet times set on the AUDIO page. Default value 0.

SENDMULTIPARTMIME: This option is for future enhancement and should always be set to 'OFF'. Default value OFF.

NEWBRANCHONAUTHBYE: This is a legacy option that is no longer used, and must always be set to 'ON'. Default value 'ON'.

MODE: This field sets whether multiple proxies and registrars are used serially or concurrently. If set to SERIAL the phone will attempt to register with the next priority registrar if registration with the current one fails. If set to MULTIPLE it will attempt to maintain registration with all enabled registrars, and will use the priority sequence for outbound call failover. Default value: SERIAL. When only a single proxy / registrar is enabled, set this value to SERIAL to ensure any registration failure is detected quickly.

REGTIMEOUT: Sets the Registration timeout value (in seconds) that will be suggested by the telephone to a Registrar. Following the expiry of this timeout, the telephone will be deregistered and then automatically attempt to re-register. (Value range: 0 to $2^{32} - 1$, default value: 3600) The registration server can ignore or override this suggested time.

REREGTIMEOUT: Sets a period in seconds after which the phone will force a re-registration period and the server cannot override it. Disabled if set to zero. Default value 0. This field can be used to ensure that registration is maintained for this particular phone, regardless of the general settings on the registration server. For example, if this were an emergency phone, setting this field to 30 would force re-registration every 30 seconds even if the server normally only refreshes registration once an hour. In this way, if the proxy server fails or becomes unavailable, the phone can detect it quickly and either attempt to register with the

next server in the priority list (if MODE is set to SERIAL) or direct calls to the next priority server (if MODE is set to MULTIPLE).

Note that, if the current registrar becomes unavailable, the telephone may not be able to make a call until it re-registers with the next.

5.4.1 SIP Info sub-pages:

The screenshot shows the GAI-TRONICS configuration interface. The top navigation bar includes 'Home' and 'Configuration'. The main content area is titled 'Module: SIP settings' and displays 'SIP 1 Parameters' with the following settings:

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

On the left side, there are navigation links for various settings categories: Network (IP, SIP, Unit, Access, Serial, Email, Clock), Phone functions (Dialing & Memories, Key mapping, Current status), and Signals and Audio (Audio, Alarm, Tone, LED, Logic). On the right side, there are links for 'Page 1 - General SIP Info', '2 - SIP 1 Info, 3 - SIP 2 Info', '4 - SIP 3 Info, 5 - SIP 4 Info'.

Each of the 4 sub pages is identical, and is used to set parameters for each of 4 possible proxies.

LOCALID & DOMAIN: together these set the URI (uniform resource identifier) of the phone. In the example shown above 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.

They will accept any alphanumeric string and their default values are 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 signalling. Default value: 5060

PRIORITY: Sets the failover sequence between the 4 pages.

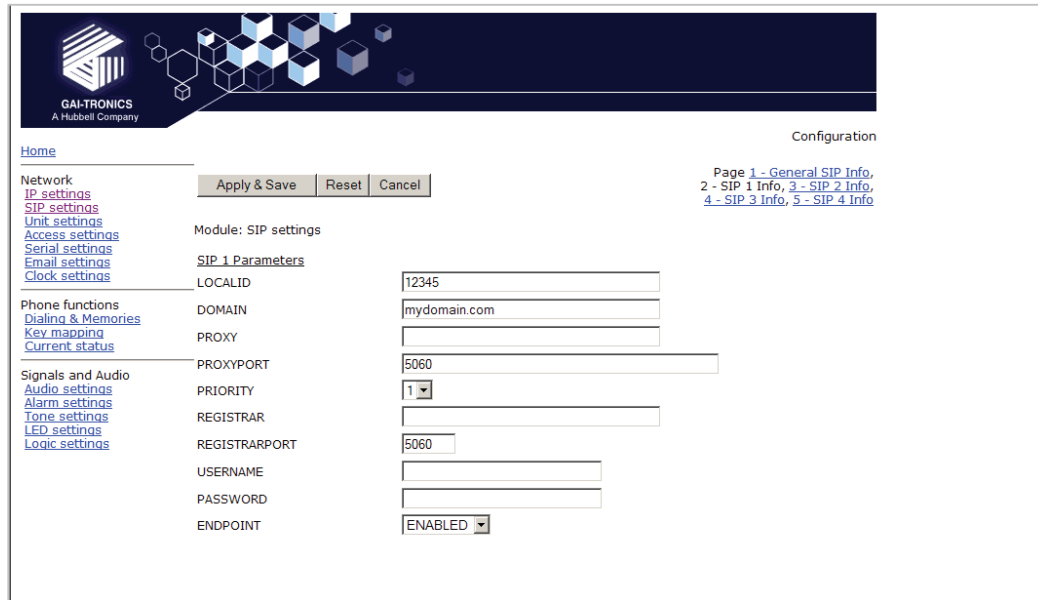
REGISTRAR: Sets the address of the Registrar, either as an IP address or FQDN. The registrar address and the proxy may or may not be the same, but the address for registration must be set here. Default value: blank

REGISTRARPORT: Sets the port number to send the requests to. Is 5060 by default or if unspecified.

USERNAME: Sets the username for the registrar authorisation realm. (Default value: blank)

PASSWORD: Sets the password for the registrar authorisation realm. (Default value: blank)

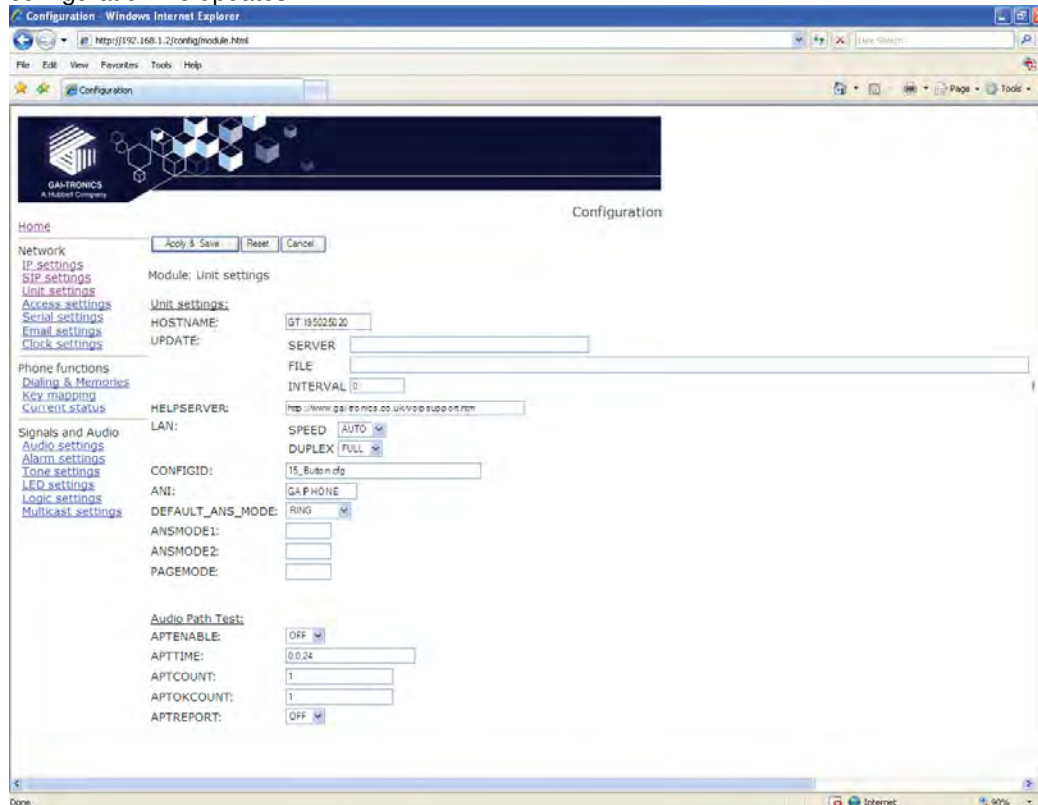
ENDPOINT: Sets whether the subpage is ENABLED or DISABLED. (Default value: ENABLED for SIP1, all others DISABLED).



Note that the Proxy address could also be that of a peer-to-peer entity, allowing the unit to make a direct peer-to-peer to connection. This can provide an extra level of resilience, allowing the unit to fall back to a peer to peer call in the event that all proxy servers become unavailable

5.5 Unit settings

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



HOSTNAME: Sets the unit host name. Maximum 15 alphanumeric characters (a-z, A-Z , 0-9). Default Value is a unique string starting with "GT" and followed by the serial number of the main circuit board inside the phone (referred to as the "Board serial" on the home page). 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 phone on the system.

UPDATE SERVER: Sets the address of the host running the TFTP server. (Valid values: IP address or FQDN. Default value: blank)

UPDATE FILE: The name of the update control file on the update server. This name may contain the macro symbols %m, %h and %i. These symbols are expanded to the MAC address, host name and IP address respectively. (Default value: blank)

UPDATE INTERVAL: Forces the unit to attempt a file download every X hours where X can be an integer value between 0 and 1000. A value of 0 disables the periodic update request. The default value is 1. Any non-zero value will cause the unit to attempt a configuration file download at boot time.

HELPSERVER: Sets the default address for the Help web page reached from the link on the home page. The default value is <http://www.gai-tronics.co.uk/voipsupport.htm>, but it can be changed to any appropriate page available on the network.

LAN SPEED: Sets the speed or auto negotiation status for the WAN Ethernet port. Valid values: 10, 100 or AUTO. Default value: AUTO. If the speed is auto negotiated the duplex setting has no effect.

LAN DUPLEX: Sets the duplex value for the WAN Ethernet port. Valid values: FULL or HALF. Default value: FULL.

CONFIGID: Used by the configuration upgrade script to determine if the local configuration is the same as the one it wants to upgrade to. If this matches the CONFIGVERSION line in the update control file, no download will take place. Default value: blank.

ANI: Used as an identifying token to GAI-Tronics CMA Call Management Application. Default value: "GAIPHONE". Maximum 12 characters.

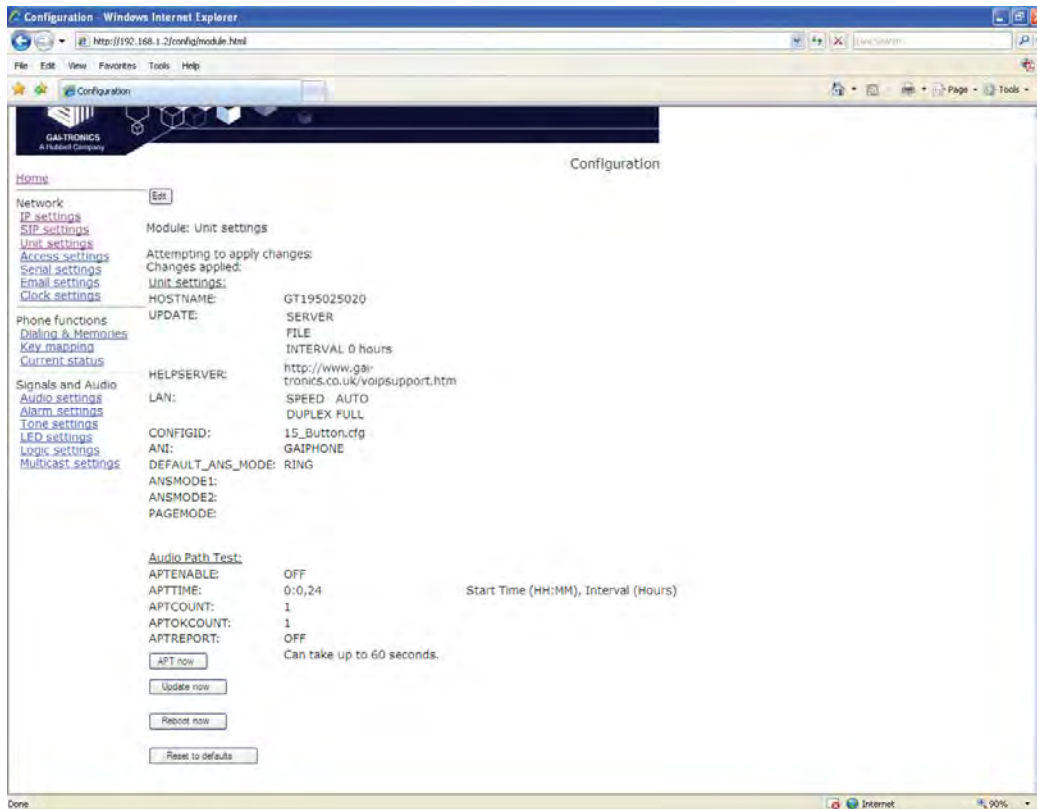
DEFAULT_ANS_MODE: Sets the default answer mode. This mode will be used to answer a call when ANSMODE1, ANSMODE2 and PAGEMODE are not triggered. Values available are RING, PICK-UP, PAGE and STEALTH. RING is normal phone operation, where a button must be pressed or handset lifted to answer an incoming call. PICK-UP is as described in ANSMODE2 below. PAGE is as described in PAGEMODE below. STEALTH is as described in ANSMODE1 below. Default value: 'RING'.

The next 3 fields set "passwords" that can be used by GAI-Tronics CMA to activate 3 special auto-answer modes, usually for hands-free telephone types.

ANSMODE1: Stealth auto-answer mode, 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. Sending a DTMF "*" during a call will change the unit to ANSMODE 2. For activation from CMA, set this field to "aa1"

ANSMODE2: Sets Intercom auto-answer mode, where the telephone auto answers and provides normal duplex audio, preceded by an announcement tone. For activation from CMA, set this field to "aa2"

PAGEMODE: 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. For activation from CMA, set this field to "aa3"



At the bottom of the UNIT page are three action buttons:

Update Now: Causes the phone to fetch the update file immediately.

Reboot Now: Causes the unit to reboot.

Reset to defaults: Causes the unit to reset to a predefined default configuration, effectively returning it to the settings it had when it left the factory. **WARNING:** this will undo any configuration changes, including returning the IP address and net mask to their default, static values of 192.168.1.2 and 255.255.255.0. It will also reset the username and password to defaults.

5.5.1 Audio Path Test

Audio Path Test (APT) is a factory option which must be specified at order time. If the phone was not ordered with APT, these controls will not appear on the UNIT 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 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 controls below:

APTENABLE sets whether APT is on or off

APPTIME sets a start time (24h clock) and test interval (in hours). The 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). Automatic testing will start at the specified time and repeat every specified interval until 00:00 midnight the next day. The cycle will then repeat the next day and so on. Default is **00:00,24** meaning that the test will perform once per day at midnight.

APTCOUNT sets the number of tests that will be performed at each interval. Range is 1-10. Default is 1, but it can be increased to repeat the test at each interval.

APTOKCOUNT sets the number of tests that must pass at each interval to be classed as a successful test. Default value is 1. APTOKCOUNT must always be \leq APTCOUNT. For example if APTCOUNT were set to 3 and APTOKCOUNT to 2, the test would be deemed to

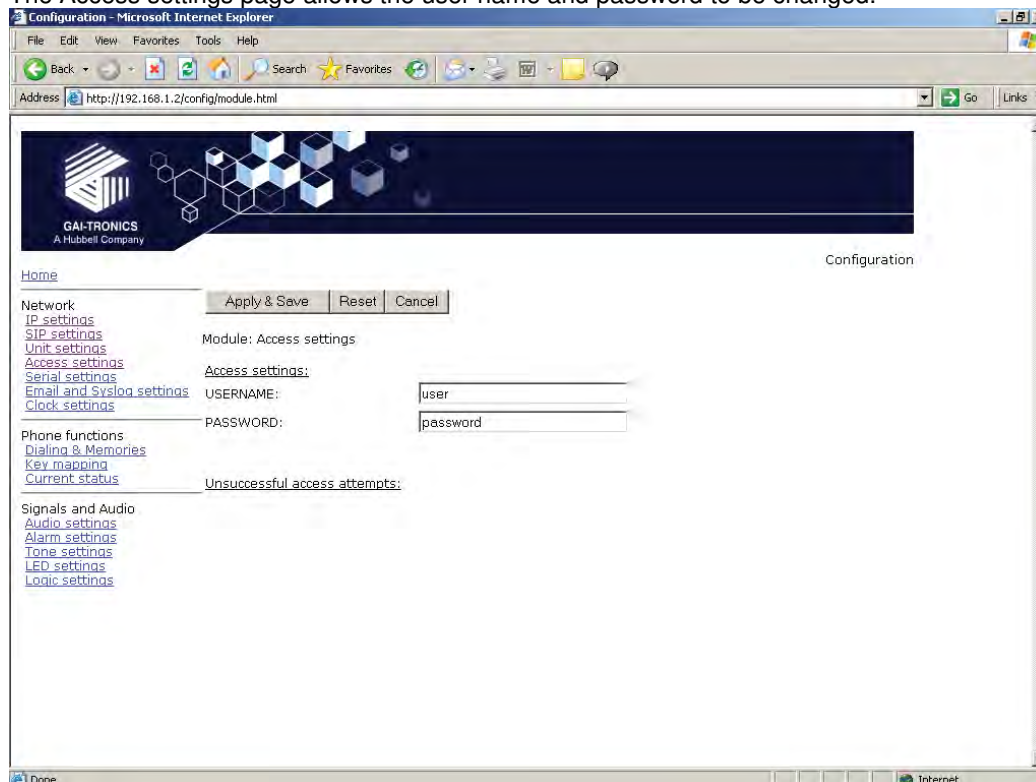
have passed if 2 pass readings out of 3 were recorded. This feature is to allow for potential disruption in areas of high ambient noise.

APTREPORT sets whether or not APT will send reports every time the test passes. Normal alarms only report if they change state; setting APTREPORT to ON will cause the phone to send a regular report confirming that it's acoustic components are healthy. By inference this report also confirms that the phone is powered, running and connected to the network so it also provides a useful general health check. If the test fails, the phone will not send repeated reports until at least APTOKCOUNT tests pass again.

APT now will start an APT test within 60 seconds. This button will only start a test if APTENABLE is set to ON.

5.6 Access settings

The Access settings page allows the user name and password to be changed.

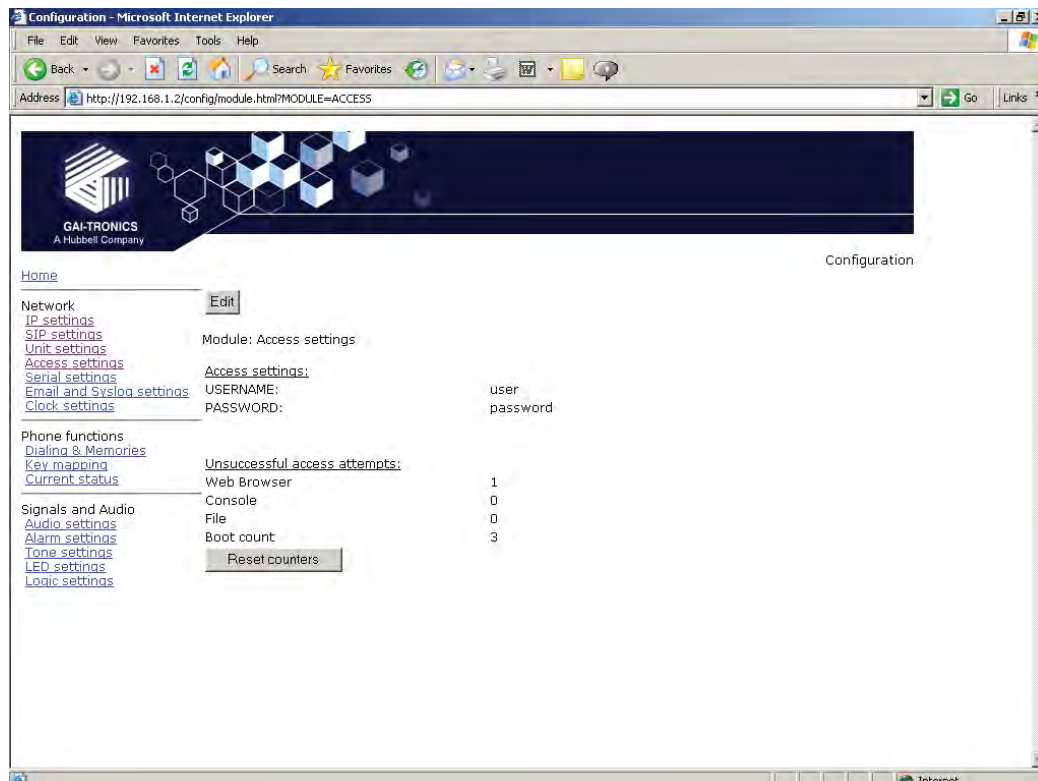


USERNAME: Can be up to 30 characters long, and can contain only the alphanumeric characters a-z, A-Z, 0-9. The default value is "user". The Username cannot be blank.

IMPORTANT: The word '**root**' is a reserved username and must not be used or assigned a password. Setting a user name of "root" will make it impossible to access the phone, and will require a reset to factory defaults.

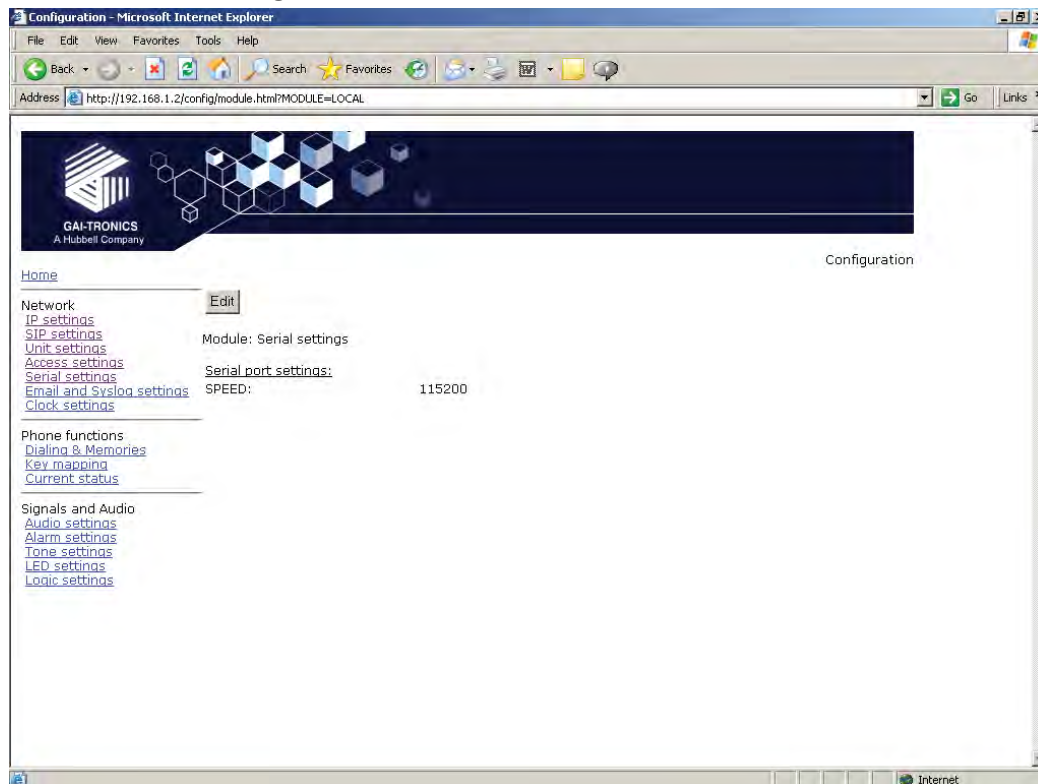
PASSWORD: Can be up to 30 characters long, and can contain only the alphanumeric characters a-z, A-Z, 0-9. The default value is "password". Password can be blank if required.

Note: please make sure to record the user name and password securely. They will be required to access the phone every time, whether by web page, command line or configuration file. In the event that the username and password are lost, the unit will need to be reset to factory defaults. This can be done by holding down a button on the main circuit board. See section 10.



At the bottom of the Access page are a series of counters showing how many unsuccessful access attempts have been made to this phone, and how many times it has been rebooted. The counters can be reset using the "Reset counters" button.

5.7 Serial settings



The Serial settings page is used to set the speed for communication on the serial port.

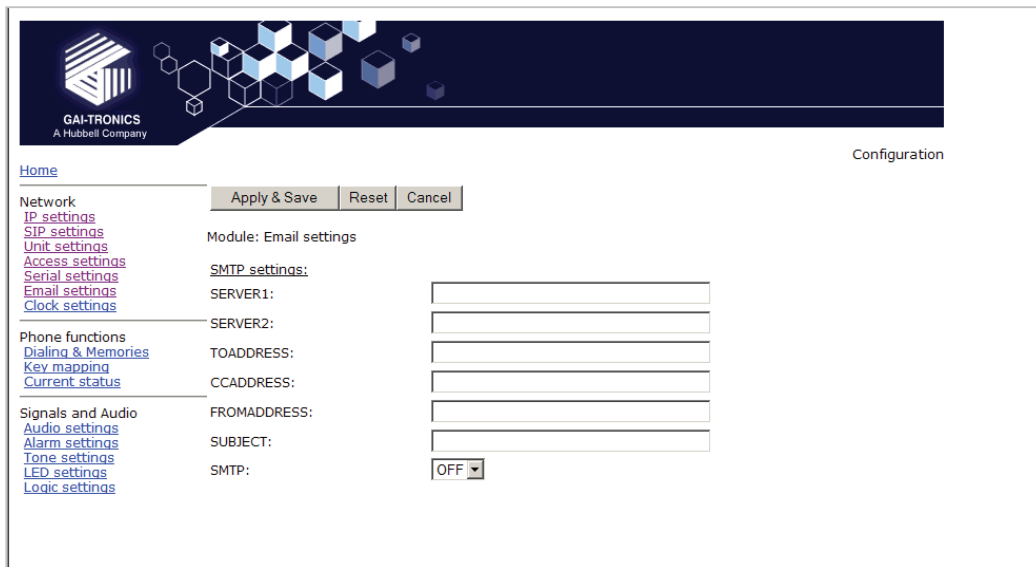
Speeds available (from a drop-down list) are: 9600, 19200, 38400, 56700 & 115200 baud.
 The default value is 115200.
 The other parameters for serial comms are: 8 data bits, 1 stop bit, no parity.

5.8 Email settings

The telephone can report various alarm and input conditions via email (see the ALARMS and LOGIC pages in sections 5.14 and 5.17). The Email settings page is used to set the parameters required.

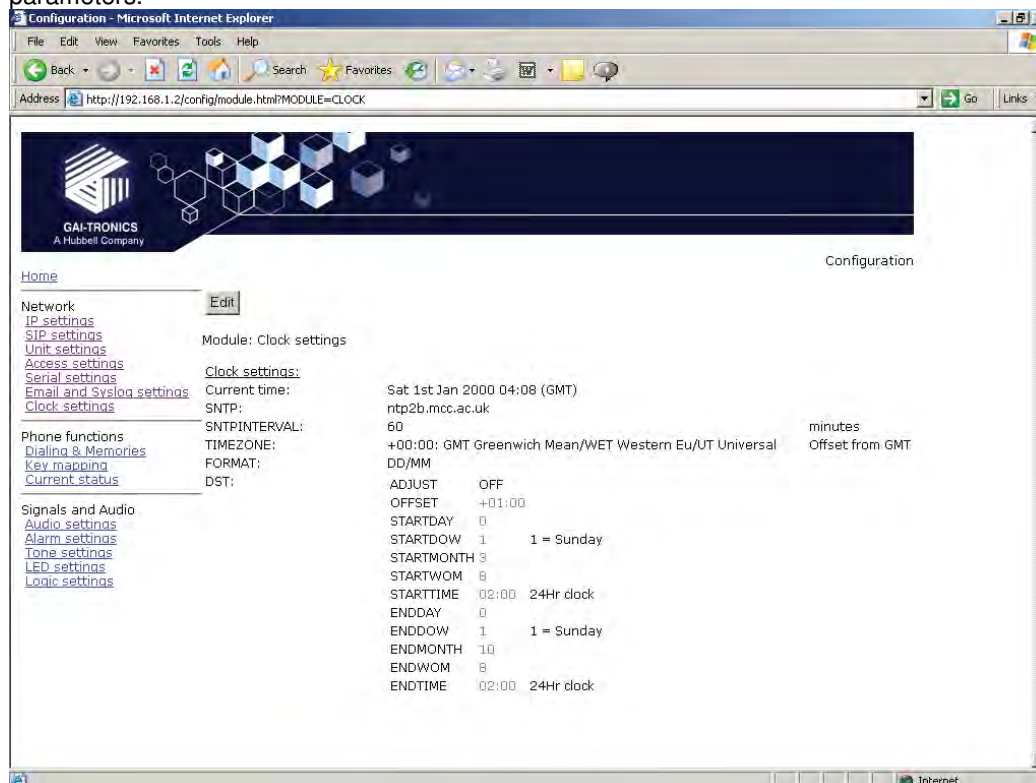


SERVER1: Sets the primary SMTP server, as an IP address or a FQDN
SERVER2: Sets the secondary SMTP server, as an IP address or a FQDN, for redundancy.
TOADDRESS, CCADDRESS & FROMADDRESS: Set the email addresses that will appear in the message. Note that the phone can send the message to two separate addresses (TO & CC) Each of these fields can contain a single email address of the form abc@xyz.com
SUBJECT: Sets the subject that will appear with each email message from this unit.
SMTP: enables or disables email.



5.9 Clock settings

The telephone does not include a battery backed real time clock, but will keep time based on updates from an SNTP server. It can also adjust for daylight savings time by setting DST start and end dates & times. The clock settings page is used to set the required parameters.



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 is 60.

TIMEZONE: Sets the current time zone for local time from a dropdown list. See section for a full list of available timezones.

FORMAT: Sets the date format to either UK (DD/MM) or US (MM/DD) style.

The remaining parameters on this page set the behaviour of the internal clock for daylight savings time (DST). The normal default is for the clock to advance by one hour between the last Sunday in March and the last Sunday in October, with the changes becoming effective at 2am on each of these days. To achieve this, the settings are:

ADJUST	ON
OFFSET	+01:00
STARTDAY	0
STARTDOW	1
STARTMONTH	3
STARTWOM	8
STARTTIME	02:00
ENDDAY	0
ENDDOW	1
ENDMONTH	10
ENDWOM	8
ENDTIME	02:00

Where:

ADJUST: Sets whether automatic Daylight Savings Time adjustment is on or off.

OFFSET: If DST is on, sets the offset. Default is +01:00

The remaining 10 parameters on this page set the start and end of the DST period:

STARTDAY: Sets the day of the month on which DST begins:

- 1 -31 for days of month
- 0 ignore this value and use STARTDOW value
- Default is 0

STARTDOW: Sets the day of the week on which DST begins (1 - Sunday, 7 - Saturday). Default value is 1.

STARTMONTH: Sets the Month in which DST will begin (Default value is 3).

STARTWOM: Sets the week of the month in which DST will begin. Valid values are 1 - 6, where 1 is the first week and each subsequent number is a subsequent week. 8 signifies the last week of the month regardless of which week the last week is (Default value is 8).

STARTTIME: Sets the hour of the day and the minute of the hour on which the unit will start to use the DST offset if enabled, in the 24-hour format. Default = 02:00.

ENDDAY: sets the day of the month on which DST ends:

- 1 -31 for days of month
- 0 ignore this value and use ENDDOW value
- Default is 0

ENDDOW: sets the day of the week on which DST ceases (1 - Sunday, 7 - Saturday). Default value is 1.

ENDMONTH: sets the Month in which DST will cease to operate (Default value is 10).

ENDWOM: sets the week of the month in which DST will cease. Valid values are 1 - 6, where 1 is the first week and each subsequent number is a subsequent week. 8 signifies the last week of the month regardless of which week the last week is (Default value is 8).

ENDTIME: sets the hour of the day and the minute of the hour on which the unit will cease to use the DST offset if enabled, in 24 hour format. Default = 02:00.

5.10 Dialling & Memories

The dialling and memory pages are used to set various "dialling" actions - ie how the telephone initiates calls.

Depending on the keypad layout (see Key mapping page, section 5.11), the telephone may have a numeric keypad, memory buttons or both.

The numeric keypad is used to enter a number one digit at a time, whereas memory buttons are used to dial complete, predetermined numbers.

Each memory button is assigned a memory list, consisting of one or more memories.

Calls started from memory buttons automatically divert to the next number in the list if the call fails, as described below.

5.10.1 Memories sub-page

The telephone can store 20 call destinations, shown on the first Dialling & Memories page.

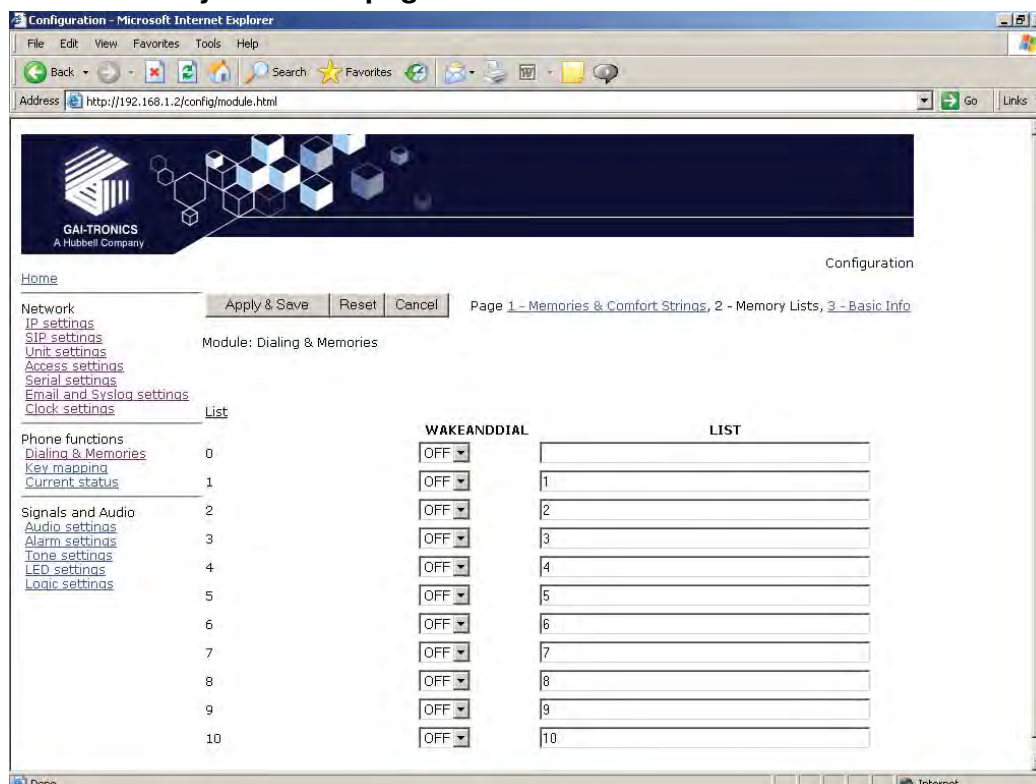
	MEMORY	COMFORT
1	<input type="text"/>	<input type="text"/>
2	<input type="text"/>	<input type="text"/>
3	<input type="text"/>	<input type="text"/>
4	<input type="text"/>	<input type="text"/>
5	<input type="text"/>	<input type="text"/>
6	<input type="text"/>	<input type="text"/>
7	<input type="text"/>	<input type="text"/>
8	<input type="text"/>	<input type="text"/>
9	<input type="text"/>	<input type="text"/>
10	<input type="text"/>	<input type="text"/>
11	<input type="text"/>	<input type="text"/>
12	<input type="text"/>	<input type="text"/>
13	<input type="text"/>	<input type="text"/>
14	<input type="text"/>	<input type="text"/>
15	<input type="text"/>	<input type="text"/>
16	<input type="text"/>	<input type="text"/>
17	<input type="text"/>	<input type="text"/>
18	<input type="text"/>	<input type="text"/>
19	<input type="text"/>	<input type="text"/>

Each entry has a **MEMORY** field, which can be a string of dialable characters or a SIP URI. Dialable characters are the digits 0-9, and the letters A,B,C and D.

Each entry 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 dialling digit tones heard on a normal telephone. If these comfort digits are required, the comfort string must be entered, even if the memory itself is a number.

Note these memories are not assigned directly to memory buttons - they must be called up in memory lists on the next page.

5.10.2 Memory Lists sub-page.



The telephone can hold up to 11 memory lists (0-10). Each list can be mapped to a button (for example if the key mapping page shows a button marked MEM1, this will use memory list 1). Refer to the Key mapping page (section 5.11) for the buttons available in this phone. List 0 is the Emergency List and is mapped to a button designated as "Emergency" if fitted. A list can also be set to activate as soon as the handset is lifted - see the "Basic Info" sub-page.

Each list can contain up to 20 memory entries, separated by commas. For example if you wanted the MEM1 button to call memory 1, 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. 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.

Each memory can appear in more than one list.
See the "Basic Info" sub-page for valid call fail causes.

Each list can also be set to "Wake and Dial". With this set to ON, the telephone will come off hook and start to process the list as soon as the appropriate button is pressed. This is normally set for hands-free telephones and help points without a separate "ON" button, but can be set for handset phones if required.

Once a call is connected, pressing a memory button will cause DTMF to be sent if the first entry in its memory list consists of dial-able characters.

5.10.3 Basic Info sub-page.

This page is used to set some additional parameters to do with dialling.

Parameters	Value	Unit
OFFHOOK:		(list number)
MAXLEN:	25	dialled characters
DIALTIME:	4	seconds
TERMINATOR:	#	
CALLLIMIT:	0	minutes
PRECALL:	30	seconds
CALLFAIL:	30	seconds
FAILOVERCAUSES:	1,17,21,27,38,41,50,88,18	
REMOTEALERTINGTIMEOUT:	20	seconds
LOCALALERTINGTIMEOUT:	20	seconds
REMOTEALERTINGTIMEOUTCAUSECODE:	18	
LOCALALERTINGTIMEOUTCAUSECODE:	17	

OFFHOOK: Sets a memory list number to be invoked when the handset is taken off hook (in a handset model) or when an "ON" button is pressed (on a hands-free model).

The next 3 parameters govern how the telephone decides whether or not the user has entered the complete number when dialling manually:

MAXLEN: Sets the maximum number of dialable characters that can be entered manually before the telephone assumes that the number is complete and starts the call. Range is 1-99, default value 25.

DIALTIME: Sets the inter-digit timeout value in seconds. Once the user has entered the off hook state, then failure to receive another digit within the timeout period will result in the call being initiated with the dialled digits received so far. A value of 0 seconds disables the use of the inter-digit timeout. The default value is 5 seconds. The maximum is 20 seconds.

TERMINATOR: Sets the dial string terminator character to be either #, * or if omitted (not used). The default value is blank (not used). If the user dials the selected character the call setup will be initiated.

CALLLIMIT: sets the maximum time allowed for a call in minutes. The range is 0 – 240 in minutes. The value 0 disables the timer. The default value is 0. The call is terminated when this timer expires.

PRECALL: Sets length of time in seconds that a phone will remain in the initial off hook state generating dial tone without a dialling key being pressed. After this delay the phone will cease dial tone and enter the on hook state even if the hook switch is off hook. The value 0 disables this timeout. The default value is 30. Maximum is 60.

CALLFAIL: Sets the length of time that the phone will play tone 1 (dial tone) after the call has ended. The default value is 30 seconds. The value 0 disables this timeout. Range is 0-30

FAILOVERCAUSES: Comma separated list of cause codes that would allow the phone to try the next entry in a list of memories. It is in no particular order. The cause codes are as defined by Q.931 - See table below. The default list is:1,17,18,21,27,38,41,50,88

Code	Cause	SIP Clearance Code	Comment
0 or 16	EndedByRemoteUser EndedByLocalUser		Call ended normally
1	EndedByNoUser	Failure_NotFound (404)	
16	EndedByCallerAbort		
17	EndedByRemoteBusy	Failure_BusyHere (486)	Causes NU tone to be played out.
18	Time out	Request time out (408)	
21	EndedByAnswerDenied	Declined	Local user refused call
21	EndedByRefusal	Default	All Others
27	EndedByConnectFail		
38	EndedByTransportFail		
41	EndedByTemporaryFailure	Failure_RequestTimeout (408)	
50	EndedBySecurityDenial	Failure_Forbidden (403)	
88	EndedByCapabilityExchange	Failure_UnsupportedMediaType (415)	

Note that there are two failover mechanisms: one for memories (defined here) and a second for proxies (defined in section 5.4). If a call fails due to a proxy problem, the phone will then try to place the call to the same number on the next proxy. If the call fails due to an endpoint problem (for example "busy"), the phone will try the next number in the list, on the current proxy.

REMOTEALERTINGTIMEOUT: sets the maximum length of time in seconds that the phone will ring on an outgoing call before timing out and returning "Number Unobtainable" (Tone 6) to the user. A value of 0 disables the timer, meaning it will ring until the caller hangs up or the remote end refuses the call. Range is 0-600. Default value 0.

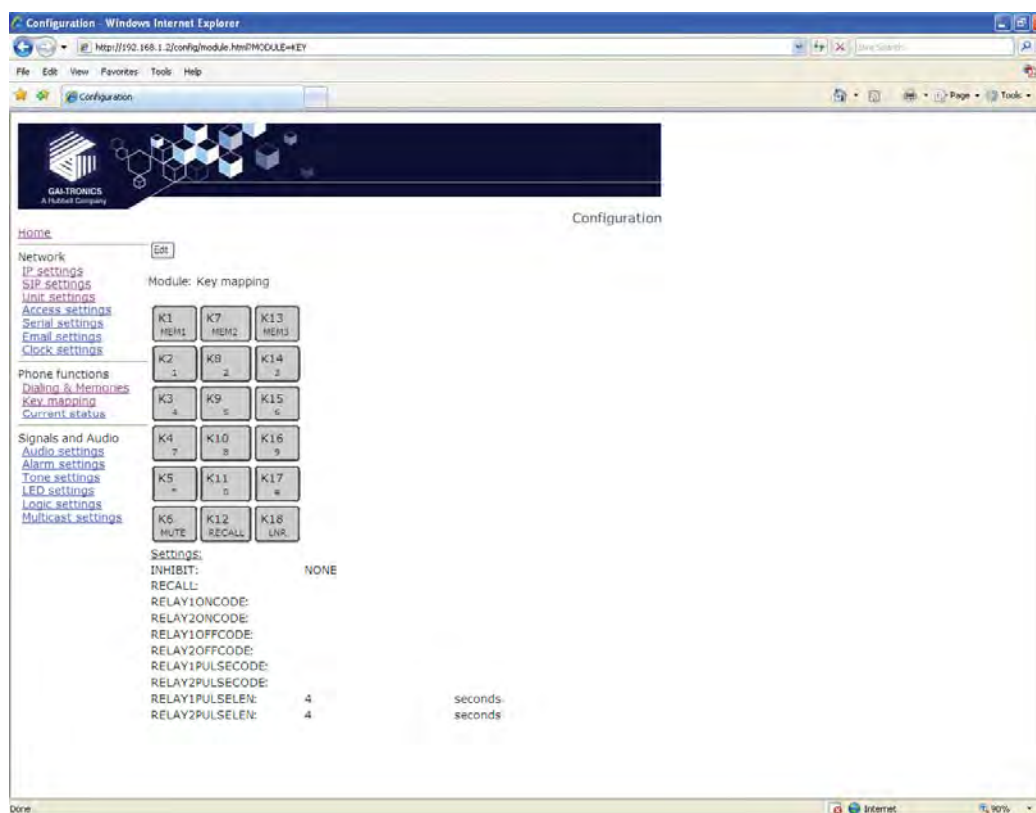
LOCALALERTINGTIMEOUT: sets the maximum length of time in seconds that the phone will ring on an incoming call before timing out and returning a "no answer" result to the caller. A value of 0 disables the timer, meaning it will ring until the caller stops ringing. Range is 0-600. Default value 0.

REMOTEALERTINGTIMEOUTCAUSECODE: sets the cause code (from the table above) that will be entered in the call description record (CDR) if an outgoing call times out without being answered. Default value 18.

LOCALALERTINGTIMEOUTCAUSECODE: sets the cause code (from the table above) that will be entered in the call description record (CDR) if an incoming call times out without being answered. It also sets the failover code that is returned to the calling party. Default value 18, which will send a failover code of 408 from the table above.

5.11 Key mapping

The Key Mapping page shows the key map of the telephone, and also controls code activation for the phone's relays. The key map is factory configured to the hardware and will vary according to the precise model supplied (an 18 button version is shown) . The keymap cannot be changed by users.



The table below lists all the possible key functions:

Key	Function
0	Dials a '0'.
1	Dials a '1'.
2	Dials a '2'.
3	Dials a '3'.
4	Dials a '4'.
5	Dials a '5'.
6	Dials a '6'.
7	Dials a '7'.
8	Dials an '8'.
9	Dials a '9'.
*	Dials a '**'.
#	Dials a '#'.
A	Dials an 'A'.
B	Dials a 'B'.
C	Dials a 'C'.
D	Dials a 'D'.
MUTE	Toggle action key to silence/enable the transmission of audio from the unit. Usually assigned to a key marked "S" (for Secrecy)
RECALL	Defined below.
LNR	Last Number Redial
ONHOOK	Clears a call and puts the phone into the on hook state. Usually assigned to a key marked "OFF"
OFFHOOK	Answers a call or puts the phone into the off hook state ready to dial. Usually assigned to a key marked "ON"
TOGGLEHOOK	Toggle action key to take the phone on and off hook.
MEM 1, MEM 2 etc., to MEM 10	Attempts to initiate a call using Memory List 1, Memory List 2, etc., to Memory List 10.

Key	Function
EMERGENCY	Overrides any existing call and attempts to initiate a call using Memory List 0. Other keys can be inhibited during an emergency call - see below.
PULSE	Activates any output configured with a "PULSE" keyword on the Logic page (section 5.17). The output(s) will remain active for the duration of the TIMER setting.
PULSE1	Activates Output 1 if it is configured with a "PULSE" keyword on the Logic page. The output will remain active for the duration of the TIMER setting.
PULSE2	Activates Output 2 if it is configured with a "PULSE" keyword on the Logic page. The output will remain active for the duration of the TIMER setting.
VOLUMEUP	Increases audio output level (either HANDSETVOLUME or HANDSFREEVOLUME as appropriate)
VOLUMEDOWN	Decreases audio output level (either HANDSETVOLUME or HANDSFREEVOLUME as appropriate)
VOLUMENEXT	Steps the audio output volume to the next level, where the levels are defined as current volume setting, midway to maximum, and maximum. A further press will loop the volume back to current. Affects either HANDSETVOLUME or HANDSFREEVOLUME as appropriate
GAINUP	Increases HANDSETGAIN or HANDSFREEGAIN as appropriate.
GAINDOWN	Decreases HANDSETGAIN or HANDSFREEGAIN as appropriate.
NOEFFECT	Key is disabled.

The keypad related fields are:

INHIBIT: If the telephone has an "emergency" button, and a call started from this button is in progress, one or more of the following buttons can be inhibited by entering keywords in this field:

DIGIT will inhibit any button capable of generating a digit

MEMORY will inhibit any memory-dial button

CLEAR will inhibit any button capable of clearing or ending a call.

The keywords can be entered in any order and must be separated by a plus (+) character.

For example to inhibit all 3, enter "DIGIT+MEMORY+CLEAR"

The keyword NONE (which must be used on its own) will disable the inhibit function, and is the default setting. The field cannot be blank.

RECALL: If the telephone has a Recall button, it can be used to activate the volt-free contact outputs or LEDs of another telephone on the network. Enter the IP address of the remote unit here. Any OUPUT or LED set with a GENERATE action of PULSE in the remote phone will be activated when the Recall button is pressed on the local phone. (See LED and LOGIC pages, sections 5.16 & 5.17)

5.11.1 Activating relays using DTMF codes

The phone's 2 relays can be activated remotely using DTMF codes. When the phone is in a voice call, if it receives a DTMF tone sequence that matches any of the fields below, the designated relay will activate accordingly. Any sequence must be between 4 and 6 digits, and is restricted to the numbers 0-9, characters A,B,C & D and the characters * and #. The default values are all blank. Note that entering duplicate sequences for different actions on the same relay may cause unpredictable results and the phone will not check for it. Care must also be taken to ensure that DTMF can be transported reliably from the remote phone to this unit, in terms of codec compatibility, in band / out of band transport etc.

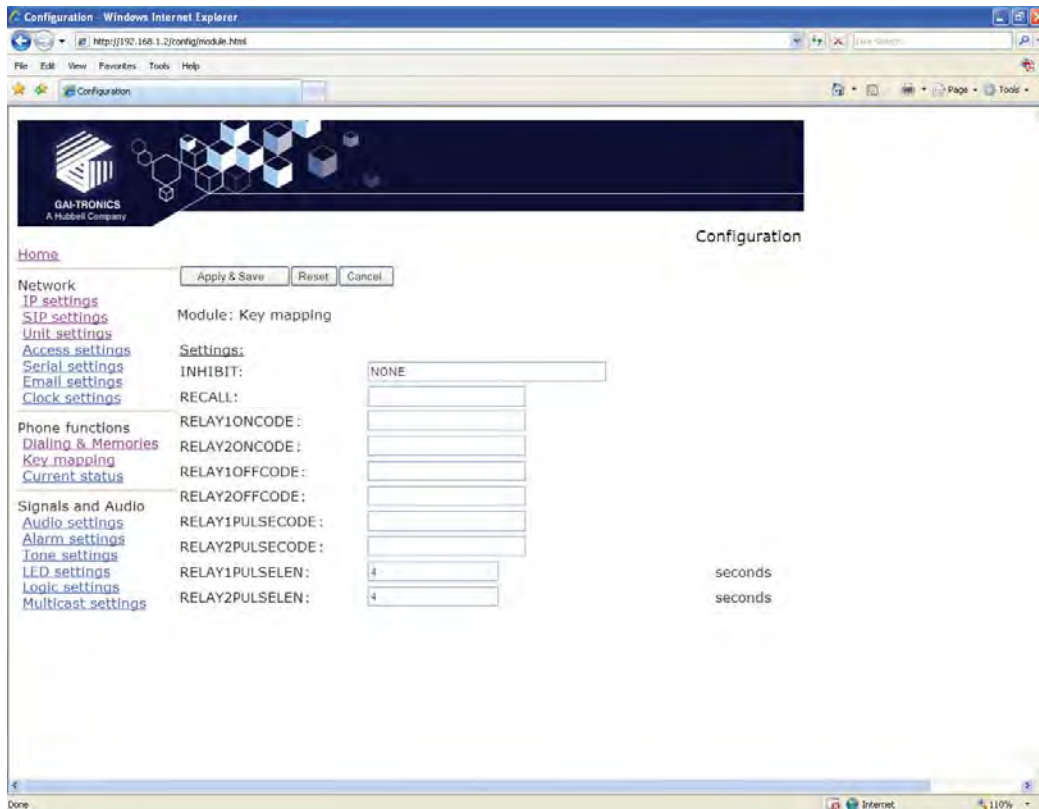
The code activation fields are:

RELAY1ONCODE, RELAY2ONCODE: matching one of these codes will cause the designated relay to turn on until instructed to turn off again or power is lost.

RELAY1OFFCODE, RELAY2OFFCODE: matching one of these codes will cause the designated relay to turn off.

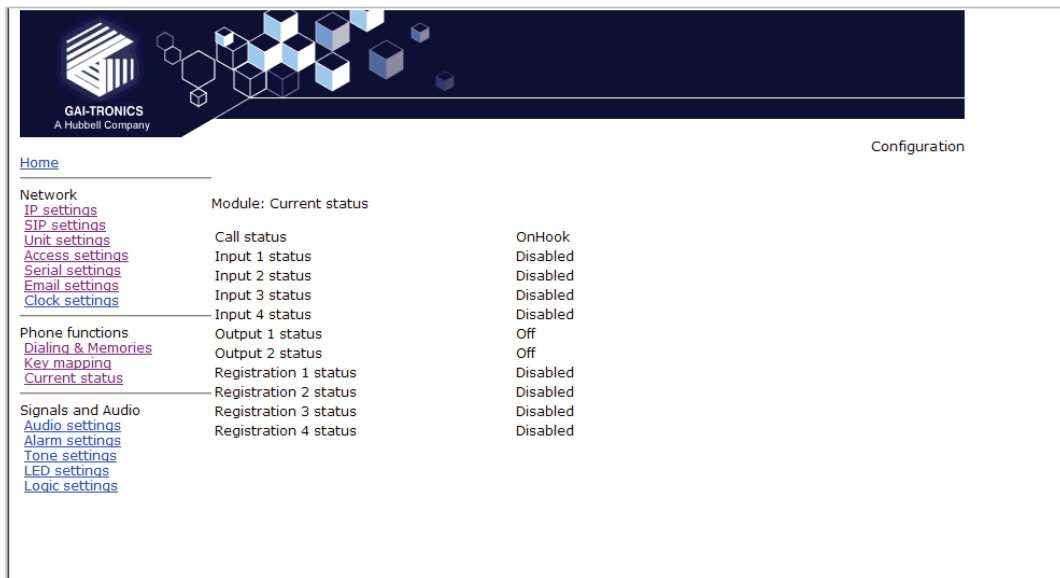
RELAY1PULSECODE, RELAY2PULSECODE: matching one of these codes will cause the designated relay to turn on for a period defined by RELAY1PULSELEN or RELAY2PULSELEN as appropriate and then turn off again.

RELAY1PULSELEN, RELAY2PULSELEN: define the period in seconds that the designated relay will turn on for. Default value is 4, range is 1-60.



5.12 Current status

The Current status page shows the status of any existing call (including "OnHook" if appropriate), the 4 inputs, 2 outputs and the registration status of the 4 proxies as configured on the SIP sub pages. There are no changeable parameters on this page.



Note that the input status reflects the settings on the Logic page (section 5.17). If the input is set to detect "NONE", the status will report as Disabled. If the input is set to detect either ON or OFF (or both), the status will report as follows:

External contact	SENSE NORMAL	SENSE INVERT
------------------	--------------	--------------

Closed	OFF	ON
Open	ON	OFF

5.13 Audio settings

This page sets various audio parameters within the telephone

CODEC: This setting chooses the CODEC order of preference that will be used by the phone. It is made up of a list of values from 1 to 6, separated by commas.

The values have the following meanings:

- | | |
|-----------------|------------------------------|
| 1 = G.711 A-law | 4 = G.729 a/b |
| 2 = G.711 u-law | 5 = G.723.1 MP-MLQ (6.3kbps) |
| 3 = G.722 | 6 = G.723.1 ACELP (5.3kpbs) |

Example: 6,5,4 would set the order of preference to be G.723.1 ACELP followed by G.723.1 MP-MLQ followed by G.729. None of the other codecs would be included.

NOTE: If codecs 5 & 6 are both used, they must be next to one another in the list.

SAMPLE: Sets the sample period for the G711, G722 and G 729 codecs to be either 10 or 20ms (individually). Default setting is 20ms.

NOTE: the sample size cannot be bigger than the packet size (packet size = frames per packet x frame period). Normally the packet size will be at least 20ms, but if you have set a low packet size (see below), you may need to set the sample period to 10ms.

FRAMES: sets the number of audio sample periods or "frames" per IP packet.

Default values:

G.723.1 = 1. Each frame is 30ms (20 or 24 bytes), range is 1-4 frames

G.729 = 2. Each frame is 10ms (10 bytes), range is 2-10 frames.

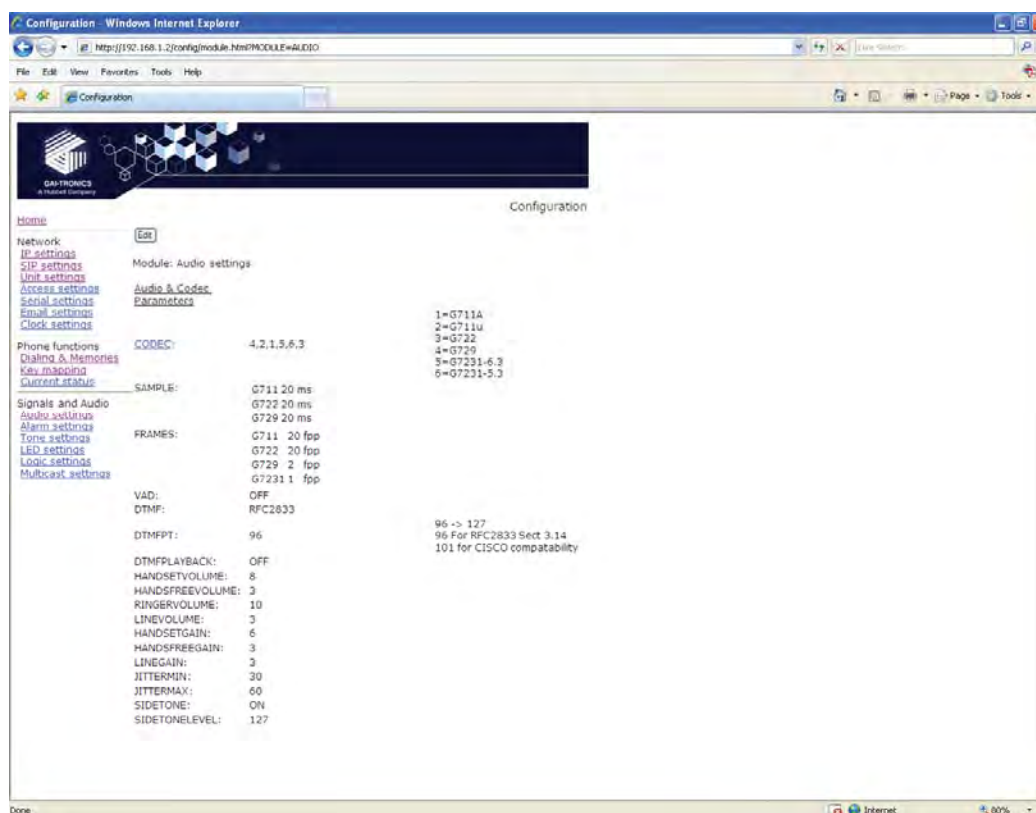
G.711 = 20. Each frame is 1ms (8 bytes), range is 10-100 frames.

Increasing the number of frames per packet allows the bandwidth used on the IP connection to be minimised, but increases transmission delay.

Decreasing the number of frames per packet reduces transmission delay but increases the bandwidth used.

Note: the packet size (frame size x frames per packet) must be greater than the sample size (see above). Make sure the SAMPLE size and FRAMES value for each codec are set accordingly.

VAD: Enables or disables the use of Voice Activity Detection. This command is only valid for G723 and G729 Codec settings. The default value is OFF. Note that when using the G729 codec, VAD must be set to on.



DTMF: Sets the transmission of DTMF digits to be either in band or out of band. The default setting is out of band, when DTMF is transmitted using RFC 2833.

DTMFPT: Sets the payload type parameter in the RTP packets when sending DTMF tones 'out-of-band' according to RFC2833. The default value is 96, but should be changed to 101 when using Cisco CallManager™.

DTMFPLAYBACK: sets whether DTMF tones are heard in the earpiece when digit buttons 0-9, * or # are pressed. Default value OFF.

HANDSETVOLUME: If the telephone is a handset model, this parameter sets the handset earpiece volume. The range is 1-9 and the default value is 8. If the telephone is a hands-free model, this setting has no effect.

HANDSFREEVOLUME: If the telephone is a hands-free model, this parameter sets the speaker volume. The range is 1-12 and the default value is 3. If the telephone is a handset model, this setting has no effect.

LINEVOLUME: This parameter is for future enhancements and has no effect.

Note: these volume settings set the starting volume within the available range. If the telephone has a volume control button or buttons, these will only act up to the extents of the range. In other words if the volume is set to its maximum on the web page, a "VOLUMEUP" button will have no effect.

RINGERVOLUME: This parameter sets the ringer volume for both handset and hands-free models. The range is 1-12 and the default value is 10.

HANDSETGAIN: If the telephone is a handset model, this parameter sets the handset microphone gain. The range is 1-8 and the default value is 6. If the telephone is a hands-free model, this setting has no effect.

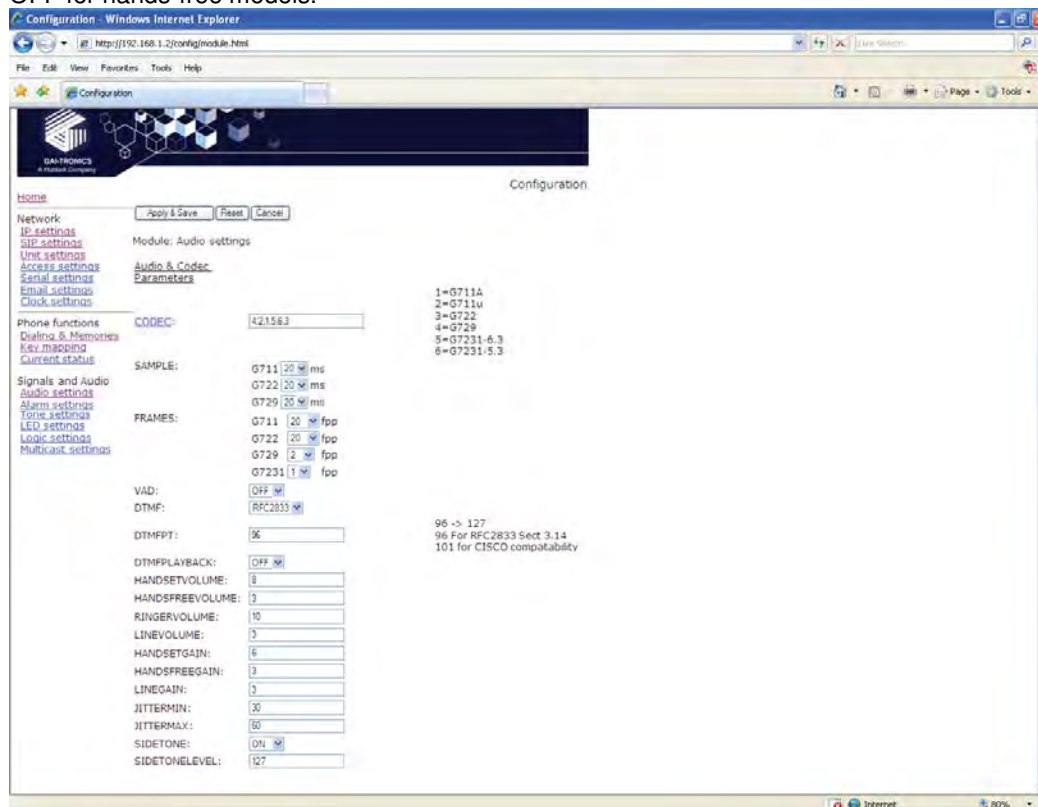
HANDSFREEGAIN: If the telephone is a hands-free model, this parameter sets the microphone gain. The range is 1-8 and the default value is 3. If the telephone is a handset model, this setting has no effect.

LINEGAIN: This parameter is for future enhancements and has no effect.

JITTERMIN: Minimum size of dynamic jitter buffer. Range is 30-120. Default value is 30.

JITTERMAX: Maximum size of dynamic jitter buffer. Range is 30-120. Default value is 60.

SIDETONE: Sets whether sidetone is on or off. Default setting is ON for handset models, OFF for hands-free models.

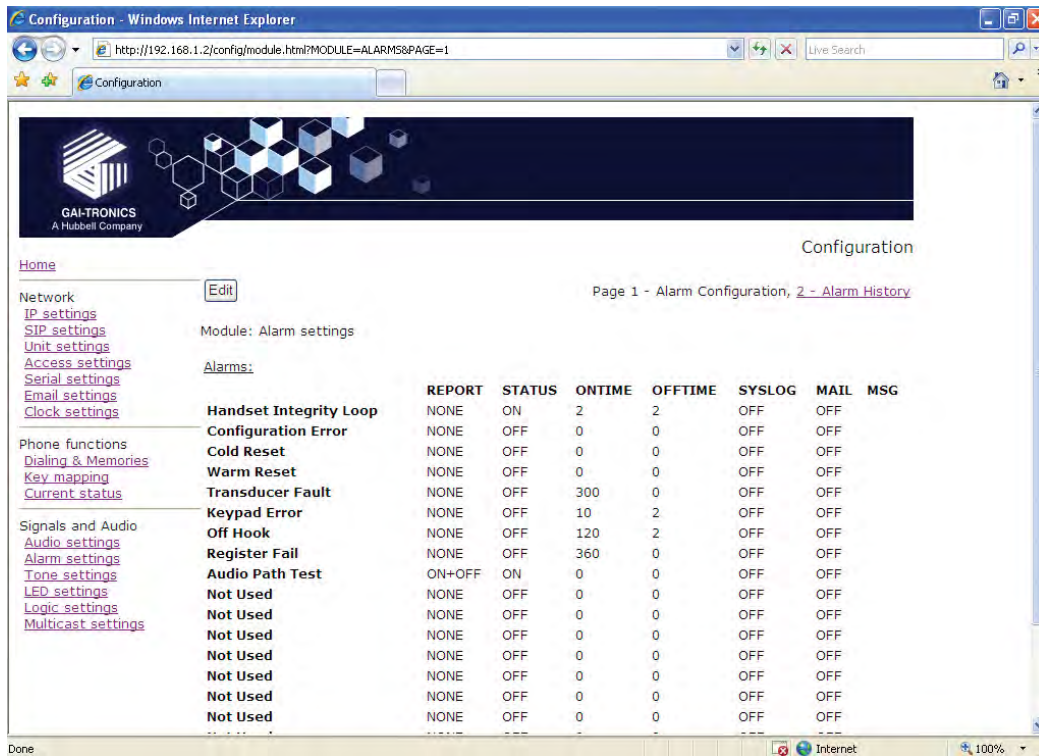


SIDETONELEVEL: If sidetone is set to ON, this parameter sets its level. Range is 0-255, default value 127. Take care when setting this level to ensure it is neither too high nor too low for safe and acceptable performance.

5.14 Alarm settings

The telephone can recognise and generate several hardware and configuration fault condition alarms. These alarms can be signalled to a remote site using two methods: -

- Syslog output over TCP
- SMTP mail message



The available alarms are:

#	Description	Function	Defaults
1	Handset Integrity Loop	Signals a broken handset loop.	Default ON and OFF times 5s
2	Configuration Error	Signals that the configuration file currently used by the unit has one or more entry errors which have been ignored.	Default ON time is 0, OFF time is not applicable
3	Cold Reset	Signals that the unit has reset due to a power cycle.	Default ON time is 0, OFF time is not applicable.
4	Warm Reset	Signals that the unit has reset due to an internal software command or error.	Default ON time is 0, OFF time is not applicable.
5	Transducer Fault	Not implemented.	
6	Keypad Error	Signals that a key has remained pressed for the entire ONTIME period.	Default ON and OFF times 6s.
7	Key Hook	Signals an off hook condition in excess of the ONTIME when not in a call.	Default ON time 30s, OFF 2s.
8	Register Fail	Signals a failure to register with all enabled proxy servers for a period in excess of the ONTIME. This alarm will not occur if registration is maintained with at least one of the enabled registrars on the SIP pages.	Default ON time is 360s, OFF time is not applicable.
9	Audio Path Test	Signals a failure of an audio path test as configured on the UNIT page. Note that a status of "ON" means that the test has failed.	ONTIME and OFFTIME are not applicable.

For each alarm, the following parameters can be set:

REPORT: specifies if an alarm will be generated on assertion of an alarm condition only (ON), on removal of the alarm condition only (OFF), on either event (ON+OFF) or not at all (NONE)

STATUS: This field shows the current status of the alarm (not a changeable parameter).

ONTIME: assigns alarm activation De-bounce Period to a specific alarm number. The alarm event must be present at the start and at the end of the de-bounce Period before the alarm status will be signalled using e-mail or syslog messaging (If enabled).

The period is specified in seconds and can take a value of 0 – 30,000. A value of 0 indicates that there is no de-bounce period for this alarm type and a message will be generated immediately the alarm event is detected.

OFFTIME: assigns an alarm removal De-bounce Period to a specific alarm number. The alarm event must be absent at the start and at the end of the de-bounce period before the alarm clearance will be signalled using e-mail or syslog messaging (if enabled).

The period is specified in seconds and can take a value of 0 – 30,000. A value of 0 indicates that there is no de-bounce period for this alarm type and a message will be generated immediately the alarm event is detected.

SYSLOG: enables or disables SYSLOG reporting for the selected alarm. Syslog settings are on the IP setting page (section 5.3).

MAIL: enables or disables SMTP reporting for the selected alarm number. SMTP settings are on the Email settings page (section 5.8).

MSG: Replaces the default text message ALARM <alarm_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

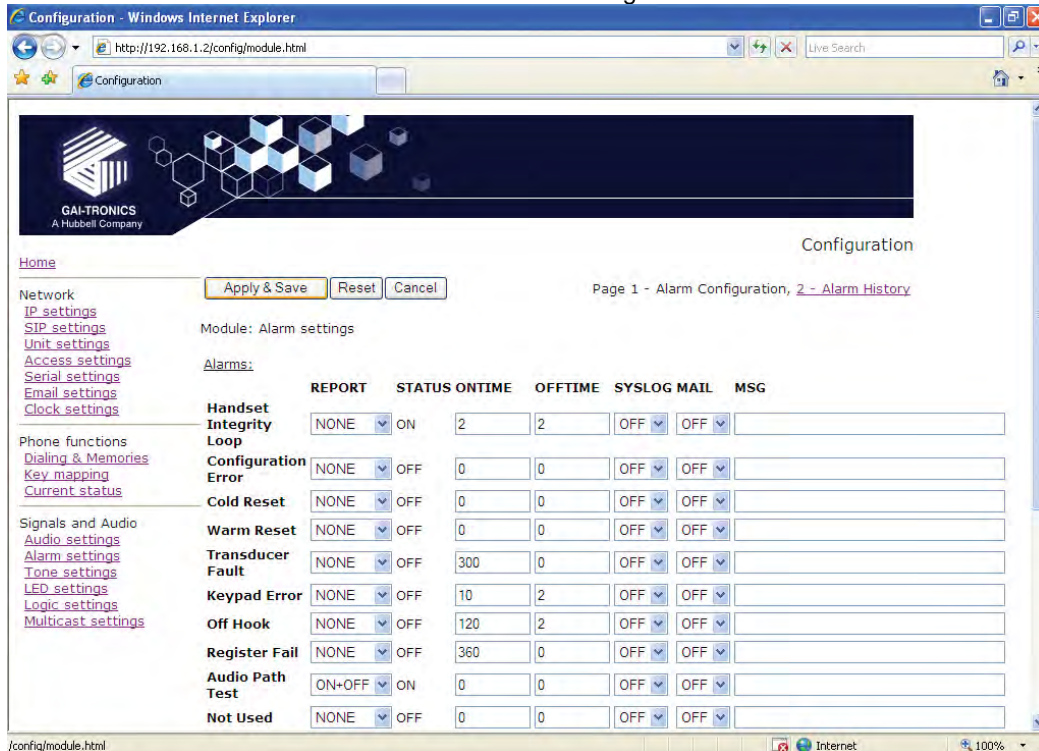
HOSTNAME is from the Unit settings page (section 5.5).

COUNT is a volatile event counter (modulus 10000)

TIME is the event time and date from the unit's clock

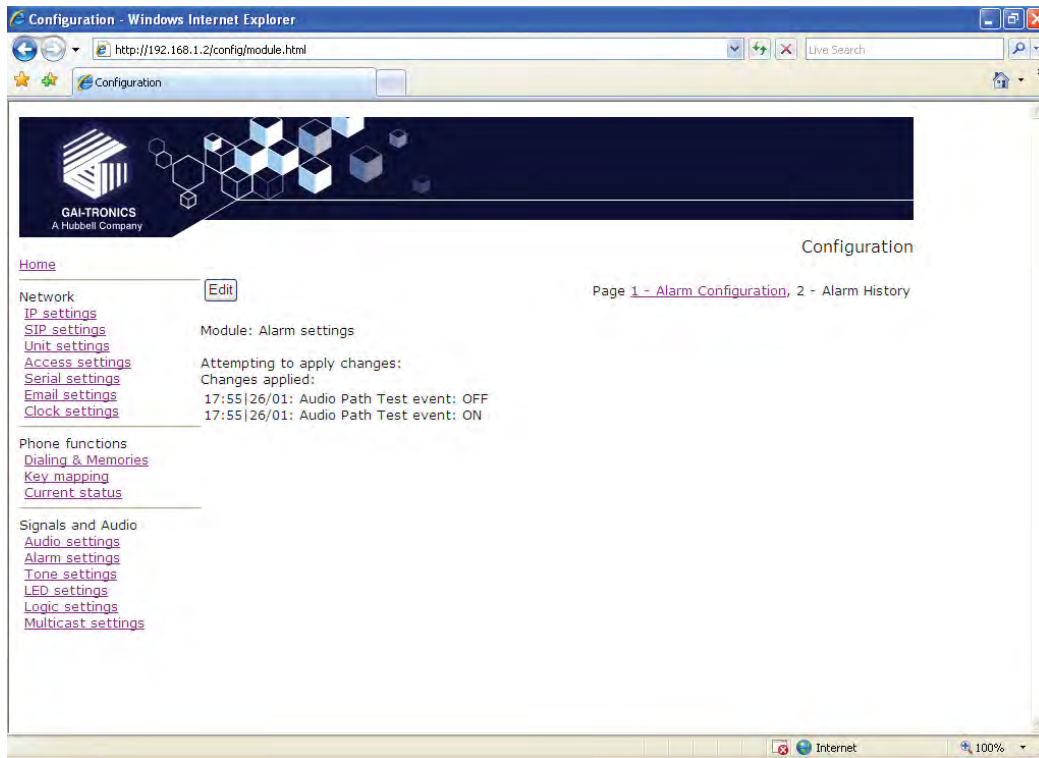
MSG is the message set by the MSG field above. If no message has been set, the default is "ALARM x" where x is the number shown against the alarms below.

ON/OFF is either the word ON or OFF according to the state of the alarm.



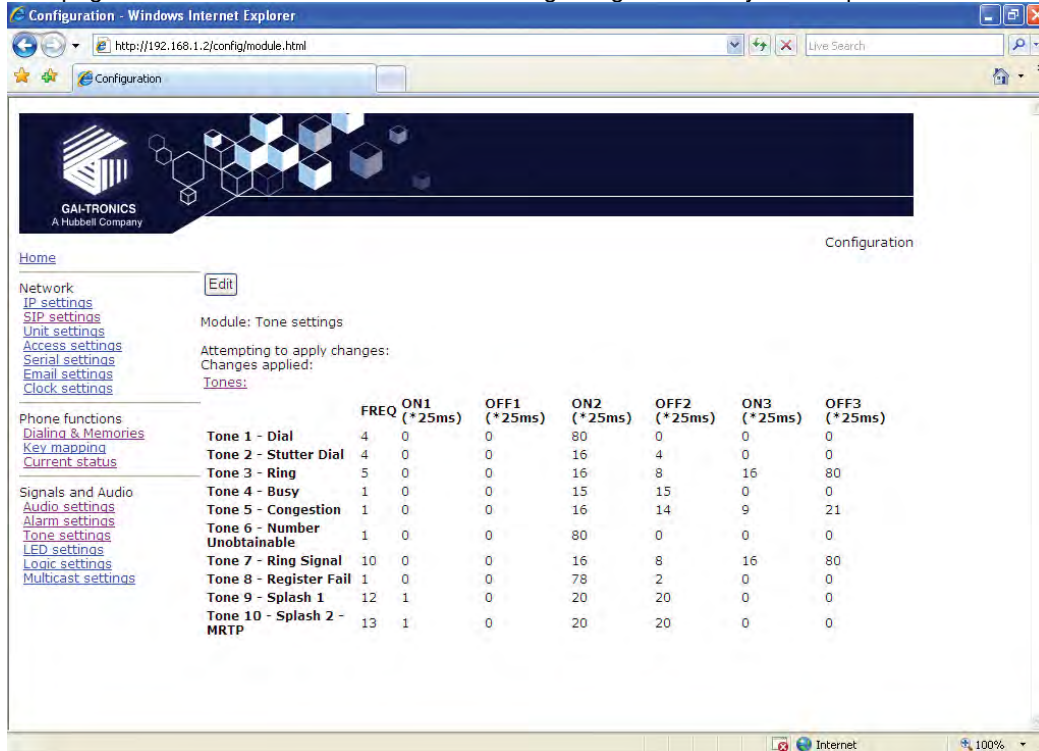
The History page displays a list of recent alarms (since last reset).

The "Edit" button on this page has no function.



5.15 Tone settings

This page is used to set the various tones and signals generated by the telephone.



The telephone can generate 8 tone signals, usually set to emulate those used by normal analogue phones:

- Tone 1 - Dial: after taking the phone off hook but before dialling
- Tone 2 - Stutter Dial: reserved for future use
- Tone 3 - Ring: when a call has been placed but not yet answered
- Tone 4 - Busy: when the called party is engaged

Tone 5 - Congestion: when the call cannot connect due to network congestion

Tone 6 - Number Unobtainable: when the call cannot connect due to the endpoint not being recognised

Tone 7 - Ring Signal: announcing an incoming call.

Tone 8 – Register Fail: When a call cannot be made due to registration failure

Tone 9 - Splash 1: Announcing an incoming PAGE call

Tone 10 - Splash 2 MRTP: Announcing an incoming multicast

Each tone can be configured by setting a tone frequency (ie the note), and the cadence (ie the timing pattern). These are normally set to simulate exchange tones common to the phone's location, but may be configured for any purpose, for example to give distinctive ring tones to differentiate between phones mounted close together. A table of typical tones used in various countries is included below, and the make up of the tones is explained as follows:

Frequency

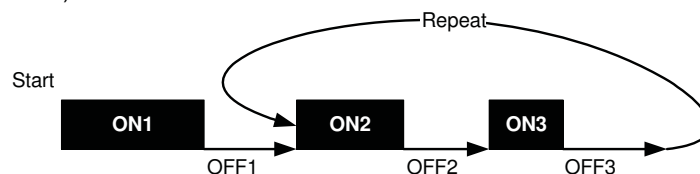
Frequency No.	Tone.
1	400 Hz
2	425 Hz
3	440 Hz
4	350 Hz + 450 Hz
5	400 Hz + 450 Hz
6	480 Hz + 620 Hz
7	20 Hz + 675 Hz
8	20 Hz + 1000 Hz
9	20 Hz + 1350 Hz
10	30 Hz + 2575 Hz
11	2970 Hz alternating with 3380 Hz at 30 Hz and maxed gains
12	220 Hz
13	440 Hz

Frequencies 1 to 6 are commonly used for call progress, whilst 7 to 10 are usually used for ring signals.

For example dial tone in the UK is a compound tone of 350+450 Hz, corresponding to frequency No.4.

Cadence

The telephone sets the cadence of a tone using ON and OFF times. To allow for most regional tone patterns there are 3 pairs of ON and OFF times - an initial pair, which is played once only, and 2 subsequent pairs that are repeated one after the other until the tone stops. (See diagram below).



ON and OFF times are entered in units of 25ms (ie 1s is entered as 40) and are in the range 0 - 600.

To create a continuous tone, set any one of the ON times to a value (say 80), and all the other ON and OFF times to 0.

5.15.1 Suggested Tone Settings for Various Countries:

UK	No.	Freq	On 1	Off 1	On 2	Off 2	On 3	Off 3
Dial	1	4	80	0	0	0	0	0
Stutter Dial	2	4	0	0	30	30	0	0
Ring	3	5	0	0	16	8	16	80
Busy	4	1	0	0	15	15	0	0
Congestion	5	1	0	0	16	14	9	21
NU	6	1	0	0	80	0	0	0
Ring signal	7	10	0	0	16	8	16	80
Register Fail	8	4	0	0	78	2	0	0

US	No.	Freq	On 1	Off 1	On 2	Off 2	On 3	Off 3
Dial	1	4	0	0	80	0	0	0
Stutter Dial	2	4	0	0	4	4	0	0
Ringback	3	3	0	0	80	160	0	0
Busy	4	6	0	0	20	20	0	0
Congestion	5	6	0	0	10	10	0	0
NU	6	1	0	0	80	0	0	0
Ring signal	7	10	0	0	80	160	0	0
Register Fail	8	4	0	0	78	2	0	0

France	No.	Freq	On 1	Off 1	On 2	Off 2	On 3	Off 3
Dial	1	3	0	0	80	0	0	0
Stutter Dial	2	4	0	0	80	0	0	0
Ringback	3	3	0	0	60	140	0	0
Busy	4	3	0	0	20	20	0	0
Congestion	5	3	0	0	20	20	0	0
NU	6	3	0	0	20	4	0	0
Ring signal	7	10	0	0	60	140	0	0
Register Fail	8	4	0	0	78	2	0	0

Netherlands	No.	Freq	On 1	Off 1	On 2	Off 2	On 3	Off 3
Dial	1	2	0	0	80	0	0	0
Stutter Dial	2	2	0	0	20	2	0	0
Ringback	3	2	24	176	40	160	0	0
Busy	4	2	0	0	19	19	0	0
Congestion	5	2	0	0	10	10	0	0
NU	6	2	0	0	40	4	0	0

Ring signal	7	10	0	0	40	160	0	0
Register Fail	8	4	0	0	78	2	0	0

Portugal	No.	Freq	On 1	Off 1	On 2	Off 2	On 3	Off 3
Dial	1	1	0	0	80	0	0	0
Stutter Dial	2	1	0	0	40	8	0	0
Ringback	3	1	0	0	40	200	0	0
Busy	4	1	0	0	20	20	0	0
Congestion	5	1	0	0	20	20	0	0
NU	6	2	0	0	8	8	0	0
Ring signal	7	10	0	0	40	200	0	0
Register Fail	8	4	0	0	78	2	0	0

Norway	No.	Freq	On 1	Off 1	On 2	Off 2	On 3	Off 3
Dial	1	2	80	0	0	0	0	0
Stutter Dial	2	2	0	0	20	2	0	0
Ringback	3	2	0	0	40	160	0	0
Busy	4	2	0	0	20	20	0	0
Congestion	5	2	0	0	10	10	0	0
NU	6	2	0	0	80	0	0	0
Ring signal	7	10	0	0	80	160	0	0
Register Fail	8	4	0	0	78	2	0	0

Note that the ring signal frequency is not specified by any regulations or customs. Frequency 10 is shown in the examples above, but any could be used according to preference.

Splash tones are not country specific so are not shown above.

5.16 LED settings

The telephone, depending on model, can have up to 3 programmable LEDs, LED1, LED2 and LED3, which can be configured using the LED Settings page.

The screenshot shows the GAI-TRONICS Configuration page for LED settings. The page header includes the GAI-TRONICS logo and the text 'A Hubbell Company'. The page title is 'Configuration'. The main content area is titled 'Module: LED settings' and displays configuration options for three LEDs. The configuration is as follows:

LED	GENERATE	TIMER	CADENCE
LED1:	GENERATE RING+HOOK	3	2:1
LED2:	GENERATE OFF	3	10:0
LED3:	GENERATE OFF	3	10:0

At the bottom of the configuration area, there are buttons for 'HELPPPOINT' and 'MODE'.

Each LED has 3 parameter entry fields:

GENERATE: This field sets the function of the LED by the use of the following keywords:

ON	Sets the LED permanently on.
OFF	Sets the LED permanently off.
PULSE +	Sets the LED to illuminate once only for the period defined by the TIMER field, on receipt of a Recall signal from a remote phone.
MUTE +	Sets the LED to indicate if the audio input is muted.
RING +	Sets the LED to flash when an incoming call is ringing. The flashing on /off periods are set by the CADENCE field.
CALL +	Sets the LED to flash when an outgoing call is active. The flashing on /off periods are set by the CADENCE field.
CONNECT +	Sets the LED on when a call is connected.
HOOK +	Sets the LED on when the telephone is off hook, and off when it is back on hook.
INUSE +	Sets the LED 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 LED to flash in time with the incoming ring signal cadence. This cadence is set by the parameters for Tone 7 (TONES page, section 5.15).
RINGOUT +	Sets the LED to flash when an outgoing call is ringing (but not yet connected). The flashing on /off periods are set by the CADENCE field.
PAGE +	Sets the LED to flash when a call is present that has been signalled as a PAGEMODE call (see UNIT page, section 5.5). The flashing on /off periods are set by the CADENCE field.
REGISTERED +	Sets the LED to flash when the phone is registered with at least one SIP server. Can be used as a "phone available" indicator. The flashing on /off periods are set by the CADENCE field.
EMERGENCY +	Sets the LED to flash whenever there is an outgoing call present that has been initiated by an EMERGENCY button. The flashing on /off periods are set by the CADENCE field.

The ON and OFF keywords must be used on their own. The other keywords (indicated by a + symbol in the table above), can be combined and entered in any order, separated by a plus (+) character. For example to set an LED to flash when an incoming call is ringing, and illuminate 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 & 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.

Configuration

[Home](#)

Apply & Save | Reset | Cancel

Network
[IP settings](#)
[SIP settings](#)
[Unit settings](#)
[Access settings](#)
[Serial settings](#)
[Email 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: LED settings

LEDs:

LED1:

GENERATE

TIMER

CADENCE

LED2:

GENERATE

TIMER

CADENCE

LED3:

GENERATE

TIMER

CADENCE

At the bottom of the page is a mode entry box with a **MODE** button. The box offers 3 choices from a drop-down menu. These preset the functions of LED1 and LED2 to mimic existing analogue telephone models:

HELPPPOINT: Sets LED1 to RING+HOOK, LED2 to OFF

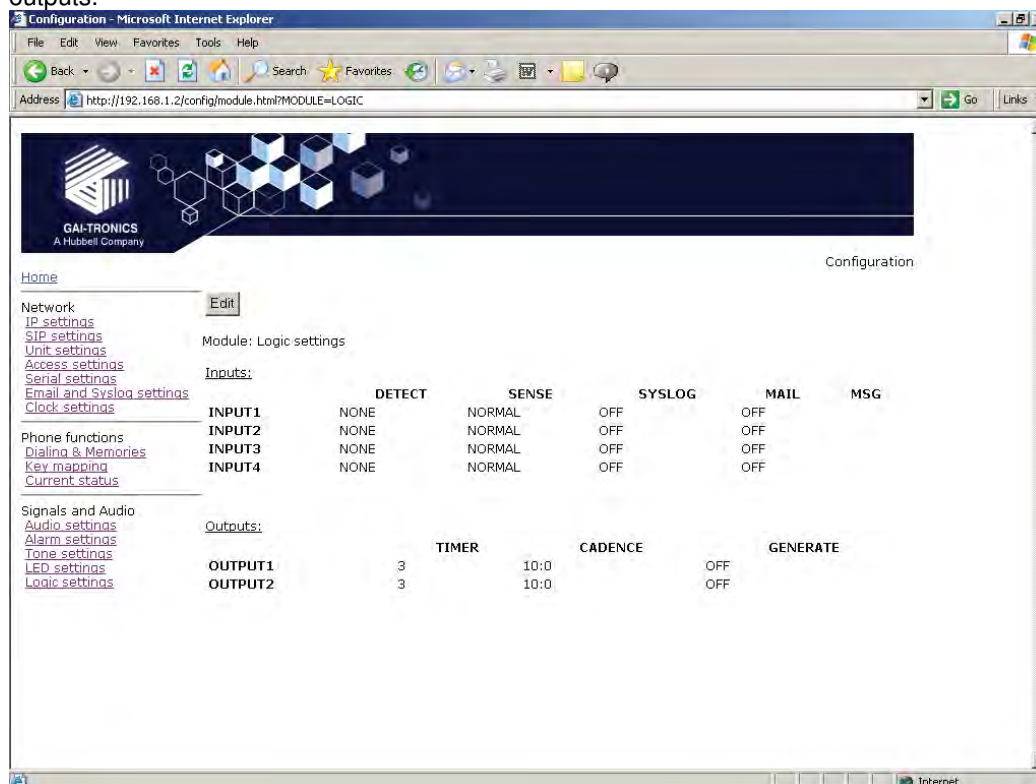
DDA: Sets LED1 to HOOK+RINGOUT, and LED2 to CONNECT+RING.

OFF: Sets LED1 & LED2 both OFF

Clicking the MODE button has the effect of applying and saving the mode settings. TIMER and CADENCE settings are not affected by the MODE (ie they must be set independently).

5.17 Logic settings

The Logic settings page sets the operation of the 4 auxiliary inputs and the 2 volt-free contact outputs.



Inputs

The 4 auxiliary inputs are activated by connecting the relevant input terminal to a common terminal via a volt-free contact. See appropriate installation guide for connection details and electrical limits. 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 below:

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:

DETECT: Specifies if 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.

SENSE: If set to NORMAL, a contact closure will report as OFF. If set to INVERT, a contact closure will report as ON. Default is NORMAL

SYSLOG: enables or disables SYSLOG reporting for the selected input. Syslog settings are on the IP setting page (section 5.3).

MAIL: enables or disables SMTP reporting for the selected input. SMTP settings are on the Email settings page (section 5.8).

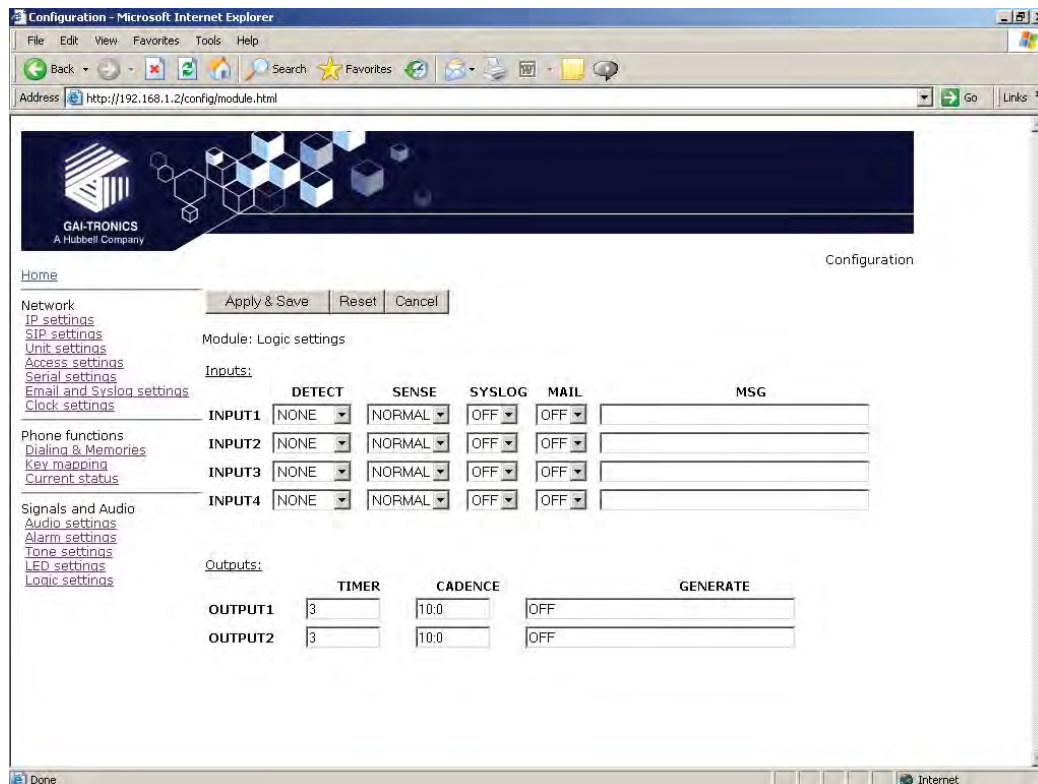
MSG: Replaces the default text message Aux_in <input_number> with the text entered (maximum 40 characters). The status <on/off> is appended to the end of the text. If the MSG value is blank, the default message is reinstated.

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 (section 5.5)
 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.



Outputs

The 2 outputs are both volt-free contacts, but their ratings differ. See appropriate installation guide for connection details and electrical limits.

Each output has 3 parameter entry fields:

GENERATE: This field sets the function of the output by the use of the following keywords:

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 phone.
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 signalled as a PAGEMODE call (see UNIT page, section 5.5). The pulsing on /off periods are set by the CADENCE field.
REGISTERED +	Sets the output to pulse when the phone 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.

The ON and OFF keywords must be used on their own. The other keywords (indicated by a + symbol in the table above), can be combined and entered in any order, separated by a 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 & 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.

5.18 Multicast settings

Multicast is a factory option which must be specified at order time. If the phone was not ordered with Multicast, this page will not appear.

Multicast allows a single audio stream to be sent to multiple endpoints simultaneously, to achieve multi-point paging or Public Address functionality over IP. As such it is only applicable to telephones with a loudspeaking capability such as handsfree units and Help Points

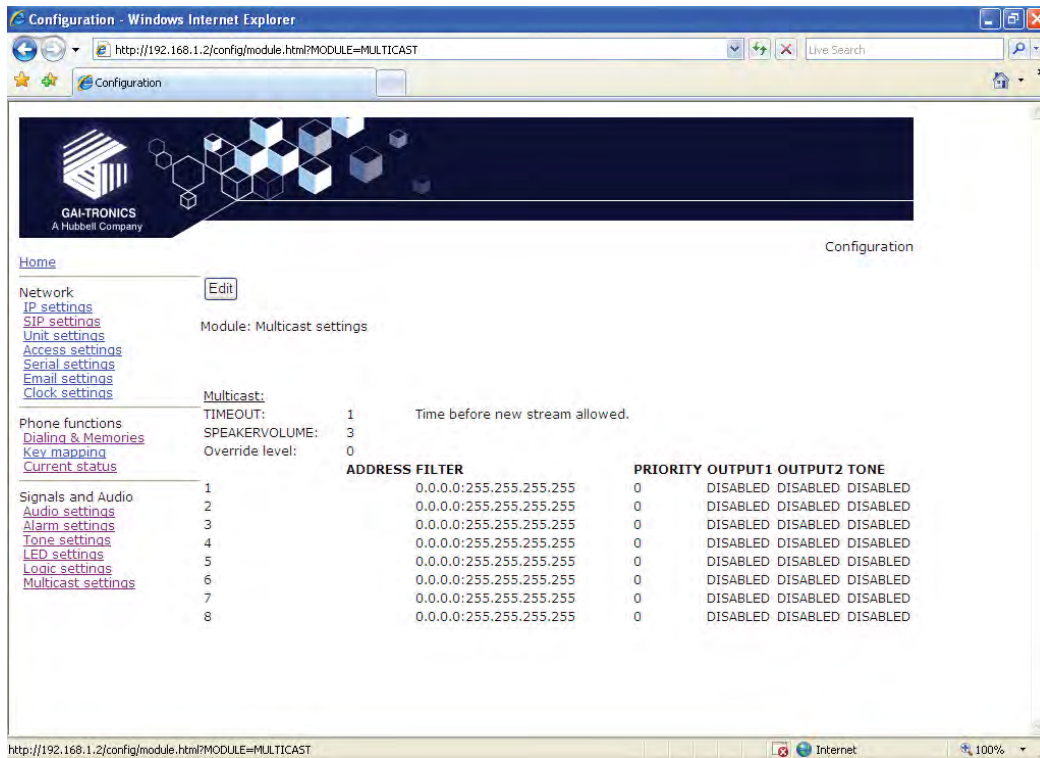
Multicast requires the use of a SIP server that specifically supports it, and each endpoint (ie phone) must be individually configured to receive multicast packets.

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 phone can be programmed with up to 8 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.

A phone with multicast enabled 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.



TIMEOUT: sets an enforced delay (in seconds) between one Multicast session ending and another beginning. Range 1-120. Default 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. Range is 1 to 10, default value 3.

Override level: sets the override level (between 0 and 8) for normal phone 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 phone whilst a higher priority multicast is in progress, the caller may hear the multicast audio but a speech call will not be connected to the phone until the multicast has ended.

Note: Emergency calls started from the phone (ie using a button designated as an Emergency button) will always override any normal or multicast call, regardless of priority or override level.

The following fields can be set for 8 individual multicast sources:

ADDRESS: the phone will accept multicast calls sent to this address by the SIP server. Must be an IP address complete with port, eg 242.0.1.75:5000. Reserved addresses for multicast channels are normally in the range 224.0.0.0 to 224.0.0.255.

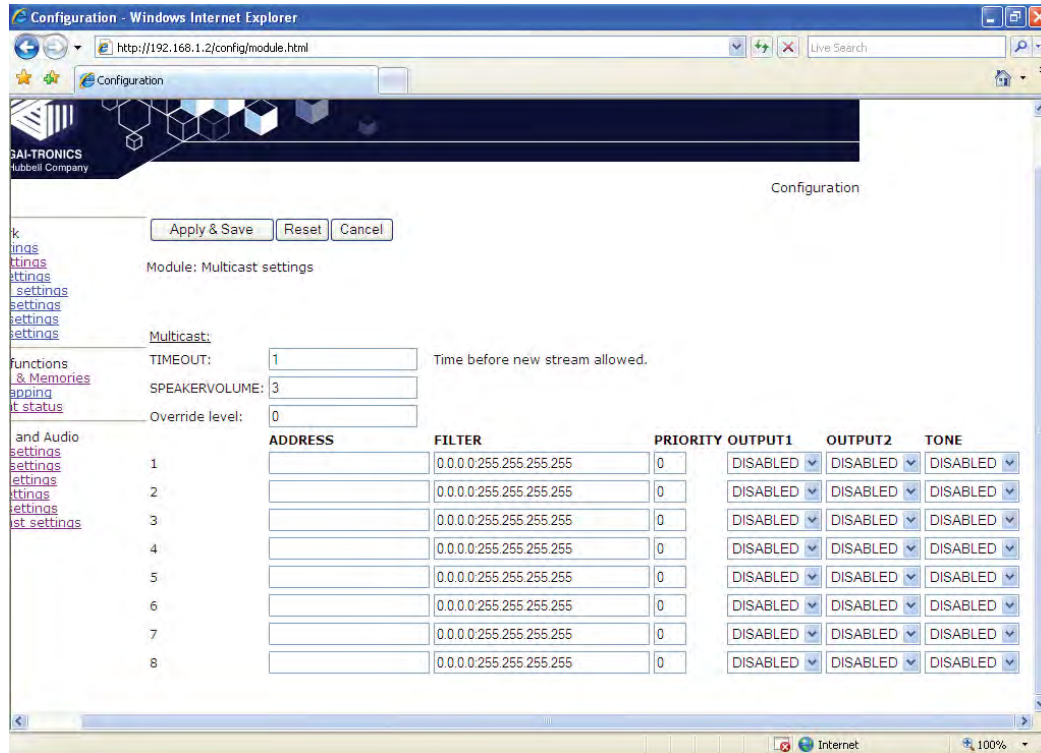
FILTER: Sets a range of acceptable multicast source IP addresses. The phone will only accept a multicast if the source is within this IP address range. The format is 2 IP addresses separated by a colon. The default is 0.0.0.0:255.255.255.255

PRIORITY: sets a priority level for this multicast with respect to other multicasts and normal phone 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 phone call) occurs during an existing multicast, it will be interrupted and resume after the higher priority event has finished.

OUTPUT1 : sets if OUTPUT1 is ENABLED or DISABLED during this multicast. If ENABLED it will be in a permanently energised 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 if OUTPUT2 is ENABLED or DISABLED during this multicast. If ENABLED it will be in a permanently energised 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 if tone 10 (TONES page) is ENABLED or DISABLED during this multicast.



6. Configuration File update

GAI-Tronics VoIP telephones can be updated by downloading files from a server. This provides a powerful method of updating multiple units across a network. Security features are built in to reduce the possibility of accidental or malicious damage. All 3 components of the unit's software can be upgraded:

- The configuration (equivalent to the settings on the web pages)
- The firmware
- The kernel (effectively the operating system on which the firmware runs)

These 3 elements can be downloaded independently, with the exception that if the kernel is updated, the firmware must be updated at the same time, since it references the kernel version.

The update process is as follows:

1. The telephone has 4 parameters that control updates (on the UNIT page):
 - SERVER: the IP address of the host running the TFTP server
 - FILE: the name of the update control file on the server
 - INTERVAL: a period in hours between download attempts
 - CONFIGID: used to identify the current configuration
2. Note that the update is initiated by the telephone. It can also be done on demand by clicking the "Update now" button on the UNIT page. After the update occurs the telephone will reset itself to activate the update - this means that web pages will be unavailable for a few seconds.
3. The update control file is a small text file containing up to 13 lines, each of which starts with a keyword. Normal routine updates would be configuration changes only, when the update control file would contain only 4 lines. An example file would be:


```

      USERNAME=user
      PASSWORD=password
      CONFIGVERSION=18but7
      CONFIGFILE=VoIP3.cfg
      
```

Where:

USERNAME & PASSWORD are used by the unit to decide if this is a valid update.
 CONFIGVERSION is the string that will be checked to see if the configuration needs upgrading. (If this matches CONFIGID in the telephone, no update will be carried out)
 CONFIGFILE is the configuration file to upgrade to.

4. The configuration file itself is again a text file, composed with a fixed syntax (see section 6.1 below). An example of a simple configuration file to change 4 memories and put them into 2 memory lists (the same example as shown in the FAQs at the start of this guide) is shown below:

```

[DIALPLAN]
MEMORY=1 888
MEMORY=2 sip:info@mydomain.com
MEMORY=3 sip:security@mydomain.com
MEMORY=4 223344@mydomain.com
COMFORT=1 888
COMFORT=2 223344
COMFORT=3 223344
COMFORT=4 223344
LIST=1 2, 3, 4
LIST=2 1
WAKEANDDIAL=1 ON
WAKEANDDIAL=2 ON
[UNIT]
CONFIGID=18but7
  
```

Note that the configuration file is divided into sections, with each section header in square brackets. Note that the CONFIGID in the UNIT section is used to change the CONFIGID in the phone to the CONFIGVERSION in the update control file. This provides an indication for the current configuration in the telephone itself, and prevents repeated downloading of the same file triggered by the UPDATEINTERVAL.

5. In summary, configuration file updates are achieved using 2 files: an update control file and a configuration file.
6. When the firmware (and kernel if required) need to be updated, the update control file is expanded to its full form, for example:

```

USERNAME=user
PASSWORD=password
CONFIGVERSION=18but7
CONFIGFILE=VoIP3.cfg
SERVER=192.168.1.6
VERSION=1.1.11
ROOTFS=incaip.root.jffs2
USERFS=incaip.usrlocal.jffs2
KERNEL=1
KERNELFILE=ulmage_quiet
KERNELMD5=ad785ffb47ccd95224f8844addc7ec05
ROOTFSMD5=f5d417c3b94a8b34e2c6afecfc985128
USERFSMD5=d5c978f26d351a9428d9c390fbb5e1ed

```

Where:

SERVER is the address from which the firmware and kernel files are downloaded.

VERSION is the version of the firmware code available.

ROOTFS & USERFS are filenames of the 2 files required to upgrade the firmware.

KERNEL is a flag that decides whether the kernel needs updating for this version of code. (1 – needed, 0 – not needed)

KERNELFILE is the kernel file to upgrade to.

The xxxMD5 lines are the MD5 sums of the padded files to be upgraded to. This is to ensure the integrity of the files.

7. If firmware or kernel upgrades are necessary, the files will be supplied by Gai-Tronics together with the appropriate xxxMD5 codes.
8. All filenames used in this process, ie the FILE field on the UNIT page, and the CONFIGFILE, ROOTFS, USERFS and KERNELFILE names in the update control file, can contain the predefined macros %h (hostname), %i (IP address) or %m (MAC address) or any combination of them in the filename string. Eg: 'update.cfg%m' would expand to 'update.cfg0001df123456' (for a MAC address of 0001df123456).

6.1 Configuration File Syntax

Configuration files are text files divided into sections. Each section comprises a header followed by a series of configuration lines for parameters within that section.

Sections roughly correspond to the web pages, and every web page parameter can be set using a configuration file.

The 15 possible sections are:

Section Name	Section Description
ACCESS	This section allows the configuration of the user access name and password
ALARMS	This section allows the configuration of the alarm control parameters
AUDIO	This section facilitates the configuration of the analogue front-end parameters and the treatment of the audio streams.
CLOCK	This section allows the configuration of parameters associated with the time keeping facilities within the unit.
DIALPLAN	This section allows the configuration of parameters associated with the dialling of destination telephone numbers, and associated dialling parameters.
IP	This section permits the configuration of the IP parameters specific to this unit.
KEY	This section allows the configuration of the mapping between the keypad and the associated key function.
LED	This section allows the configuration of the function of the two LED indicators
LOCAL	This section controls the configuration and settings of the Local Asynchronous configuration port
LOGIC	This section accesses the configuration and settings of the relay outputs and sensor inputs.
MULTICAST	This section allows the configuration of multicast parameters. Only available if the phone is factory configured for multicast.
SIP	This section allows the configuration of parameters specific to the SIP signalling protocol.
SMTP	This section permits the configuration and settings for the SMTP client.
TONES	This section allows the configuration of the units call progress tones, and ring cadence information.
UNIT	This section looks after various other miscellaneous functions and configuration options for the unit

Each configuration file need only contain the sections required to be changed. Within each section, only the parameters to be changed need to be included. Sections (and parameters within each section) can be in any order. A configuration file will incrementally patch the existing configuration

All section headers are enclosed within square brackets and followed by comment character or a Carriage return / Line feed character combination.

[Section Name]cr/lf

Comments can be placed within the file by preceding the comment with the // symbol combination and ending the line with a carriage return / line feed combination

// this line is a comment

Individual module configuration lines are made up of a configuration item identifier followed by the = sign, and then any configuration values or parameters.

Item Identifier = value1 value2 value3

The Item Identifier can consist of one or more words before the equals sign, the configuration values or parameters follow the = character and are separated by spaces or tab characters. The line is ended by a comment character combination (//) or a cr/lf combination.

6.2 Configuration File Commands

The table below lists the valid section names, the valid Item Identifiers within that section and the allowable values that can be assigned to each item identifier. In some cases a fuller description of the various options is contained in the section on the relevant web page above.

Configuration lines that are not understood will be ignored and processing will continue at the next line. Configuration lines that are in the wrong section will be ignored, and processing of the rest of the file will continue at the next line.

If an invalid or incorrectly positioned configuration line is encountered then an "Invalid Configuration file" error will be generated and signalled as an alarm.

Commands will be read and actioned as the parser proceeds down the file. If a subsequent command contradicts an earlier command, then the later command will be acted upon and the earlier command overridden.

For Example:-

```
[ALARMS]
ALARM=1 ON           //initial command is actioned
ALARM=1 ON+OFF      // subsequent command is actioned overriding previous command
```

Multiple values for the same item identifier are permitted on the same line, and are separated by space, comma or tab characters.

Example:-

```
[AUDIO]
CODEC=1,4,6         // Selects preferences as G.711 A-law, G.729 &
                   // G.723.1 ACELP in that order
```

An example configuration file is included in Section 8, showing entries in each section. This example (or parts of it) can be used as a basis to construct files as required.

Section	Item Identifier	Value(s)	Comments
ACCESS	USERNAME	UserName	Sets the user name used in local or remote access to be "UserName". UserName can be up to 30 characters long, and can contain only the alphanumeric characters a-z, A-Z , 0-9. The default value is "user". The Username cannot be blank. Example USERNAME=franklin IMPORTANT: The word 'root' is a reserved username and must not be used or assigned a password. Setting a user name of "root" will make it impossible to access the phone, and will require a reset to factory defaults.
	PASSWORD	Password	Sets the password used in local or remote access to be "Password". "Password" can be up to 30 characters long, and contains only the alphanumeric characters a-z, A-Z , 0-9. The default value is password. Example PASSWORD=sugar1
<p>Note: please make sure to record the user name and password securely. They will be required to access the phone every time whether by web page, command line or configuration file.</p> <p>In the event that the username and password are lost, the unit will need to be reset to factory defaults. This can be done by holding down a button on the main circuit board. See section 10.</p>			
ALARMS	ALARMx (where x is 1 to 20) See ALARMS web page section (5.14) for alarm descriptions and appropriate settings	REPORT ON	This command specifies if the alarm will be generated on assertion of an alarm condition only. Example ALARM4=ON
		REPORT OFF	This command specifies if the alarm will be generated on removal of an alarm condition only.
		REPORT ON+OFF	This command specifies if the alarm will be generated on assertion and removal of an alarm condition.
		REPORT NONE	This command disables, the generation of all alarm reports for the alarm. The generation of alarms in the status pages, and the generation of SYSLOG and MAIL reporting are all disabled. The generation of alarms must be reapplied for status reporting, SYSLOG reporting and MAIL reporting if required. The default value for all alarm conditions is NONE.
		SYSLOG ON	This command enables SYSLOG reporting for the selected alarm number.
		SYSLOG OFF	This command disables SYSLOG reporting for the selected alarm number.
		MAIL ON	This command enables email reporting for the selected alarm number XX. Example ALARM6=MAIL ON ALARM4=MAIL OFF
		MAIL OFF	This command disables email reporting for the selected alarm number.
		MSG "text"	This command replaces the default text message ALARM Δ<alarm_number> with the text entered as the value after the = text. The text is delimited by quote marks and is a maximum of 40 characters. The status <on/off> is appended to the end of the text.
		ONTIME xxx	This command assigns an activation de-bounce period to a specific alarm number. See ALARMS web page section (5.14) for appropriate values for each alarm number. XXX has a value between 0 and 30000 (seconds). Example ALARM1=ONTIME 5
		OFFTIME xxx	This command assigns an alarm removal de-bounce period to a specific alarm number. See ALARMS web page section (5.14) for appropriate values for each alarm number. XXX has a value between 0 and 30000 (seconds).

Section	Item Identifier	Value(s)	Comments
AUDIO	CODEC	A<,B><<,C><<,D>< <,E><<,F>	This command sets the order of preference that will be used in the SDP by the phone. Where A B C D E and F can have the values 1 -6 that corresponds to the following settings 1 = G.711 A-Law 4 = G.729 a/b 2 = G.711 u-Law 5 = G.723.1 MP-MLQ 3 = G.722 6 = G.723.1 ACELP The values should be separated by commas. EXAMPLE: CODEC=3,4,5,1,2 NOTE: If codecs 5 & 6 are both used, they must be next to one another in the list.
	SAMPLE	G711 [10 20]	Sets the audio sample period for G.711 A-law or u-law to be either 10 ms or 20ms. See 5.13
	SAMPLE	G722 [10 20]	Sets the audio sample period for G.722 to be either 10 ms or 20ms See 5.13
	SAMPLE	G729 [10 20]	Sets the audio sample period for G.729 to be either 10 ms or 20ms See 5.13
	VAD	[ON OFF]	This command enables or disables the use of Voice Activity Detection. This command is only valid for G723 and G729 Codec settings. VAD must be set to ON if using G.729 codec. The default value is ON.
	DTMF	[INBAND RFC2833]	This command sets the transmission method for DTMF as either in-band or out of band.
	DTMFPT	[96-127]	Sets the payload type parameter in the RTP packets when sending DTMF tones 'out-of-band' according to RFC2833. The default value is 96, but should be changed to 101 for compatibility when using Cisco CallManager™.
	DTMFPLAYBACK	[ON OFF]	sets whether DTMF tones are heard in the earpiece when digit buttons 0-9, * or # are pressed. Default value OFF.
	FRAMES	[G723 G722 G711 G729] X	This command sets the number of audio sample periods or "frames" per IP packet to be X. See 5.13
	HANDSETGAIN HANDSETVOLUME LINEGAIN LINEVOLUME RINGERVOLUME HANDSFREEGAIN HANDSFREEVOLUME	XX	These commands specify the various gains and volumes. Refer to Audio settings web page section (5.13) for definitions and acceptable value ranges.
	SIDETONELEVEL	[0-255]	If sidetone is set to ON, this parameter sets its level. Range is 0-255, default value 127.
	SIDETONE	[ON/OFF]	Turns sidetone on or off. Default is off.
	JTTERMIN	XX	Minimum dynamic jitter buffer size
	JITTERMAX	XX	Maximum dynamic jitter buffer size
CLOCK	SNTP	xxx.xxx.xxx.xxx or FQDN	Sets the IP address of the SNTP server to be xxx.xxx.xxx.xxx or resolves the FQDN using DNS to locate the host from which the phone obtains time data.
	DST ADJUST	[ON OFF]	This command determines if the unit's clock will automatically adjust to daylight saving time. The default value is OFF
	FORMAT	[DD/MM MM/DD]	Sets the format of the date to be either US style MM/DD or UK style DD/MM.
	DST ADJUST	[+/-]HH:MM	Sets the adjustment offset when DST is active. Default is DST=ADJUST +01:00
	TIMEZONE	XXXX	This entry sets the current time zone for local time where XXXX is an entry selected from the time zone table in Section 7
	DST STARTDAY	[0 – 31]	If DST is active, these commands define when it starts and ends. Refer to Clock Settings web page section (5.9) for detailed explanation. The example file in Section 8 lists the commands to set DST to be active between 2am on the last Sunday in March until 2am on the last Sunday in October.
	DST STARTDOW	[1-7]	
	DST STARTMONTH	[1-12]	
	DST STARTWOM	[1-6 8]	
	DST ENDDAY	[1-31]	
	DST ENDDOW	[1-7]	
	DST ENDMONTH	[1-12]	
	DST ENDWOM	[1-6 8]	
	DST STARTTIME	HH:MM	
	DST ENDTIME	HH:MM	

Section	Item Identifier	Value(s)	Comments
DIALPLAN	MEMORY	[1-20] <phone_num or SIP URI>	This command sets the dial plan memory storage position to be the telephone number or SIP URI specified in the value field. The entry can be cleared by omitting the value. Examples: MEMORY=3 0015551234 MEMORY=13 r.copper@bigweld.com
	COMFORT	[1-20] <phone_num>	This command is used to set a dial string to be used to generate DTMF digits as a memory dial feedback to the user. 30 Characters maximum length. Example COMFORT=8 1234567890
	LIST	[1-10] A B C D up to 20 entries	This command sets the dial plan list associated with a memory or emergency key function to contain the memory locations specified by A B C D and so on. The order of call attempts will be as specified.
	OFFHOOK	XX	This command sets the memory list that is associated with the off hook state. Omitting the value parameter sets the list to be non applicable and entering the off hook state will not cause a memory /emergency list to be applied. The default value is blank (non applicable)
	TERMINATOR	[* #]	This command sets the dial string terminator character to be either #, * or not used (if omitted). The default value is blank (not used). If the user dials the selected character the call setup will be initiated.
	MAXLEN	XX	This command sets the maximum dial string length to XX digits. Once the user has dialled XX digits the call will be initiated. The default value is 25 digits
	DIALTIME	X	This command sets the inter-digit timeout value to be X seconds. Once the user has entered the off hook state, then failure to receive another digit within the timeout period will result in the call being initiated with the dialled digits received so far. A value of 0 seconds disables the use of the inter-digit timeout. The default value is 5 seconds.
	CALLLIMIT	X	This command sets the maximum time allowed for a call in minutes. The value of X can 0 – 240. The value 0 disables the timer. The default value is 0.
	PRECALL	X	This command sets length of time in seconds that a phone will remain in the initial off hook state generating dial tone without a dialling key being pressed. After this delay the phone will cease dial tone and enter the on hook state even if the hook switch is off hook. The value 0 disables this timeout. The default value is 30.
	CALLFAIL	X	This command set the length of time that the phone will remain in the off hook state after the call has ended. The default value is 30. A value of 0 disables this timeout.
	WAKEANDDIAL	[0-10] [ON/OFF]	Turns ON or OFF the wake and dial functionality for the associated memory LIST.
	FAILOVERCAUSES	<CC,...>	Lists the Q.931 alike cause codes that cause the phone to fail over to the next entry in a dial LIST. The default list is FAILOVERCAUSES=1,1718,,21,27,38,41,50,88
	REMOTEAALERTINGTIMEOUT	[0-600]	sets the maximum length of time in seconds that the phone will ring on an outgoing call before timing out and returning "Number Unobtainable" (Tone 6) to the user. A value of 0 disables the timer, meaning it will ring until the caller hangs up or the remote end refuses the call. Default value 0.
	LOCALALERTINGTIMEOUT	[0-600]	sets the maximum length of time in seconds that the phone will ring on an incoming call before timing out and returning a "no answer" result to the caller. A value of 0 disables the timer, meaning it will ring until the caller stops ringing. Default value 0.
REMOTEAALERTINGTIMEOUTCAUSECODE	XX	sets the cause code (from the table above) that will be entered in the call description record (CDR) if an outgoing call times out without being answered. Default value 18.	

Section	Item Identifier	Value(s)	Comments
	LOCALALERTINGTIME OUT	XX	sets the cause code (from the table above) that will be entered in the call description record (CDR) if an incoming call times out without being answered. It also sets the failover code that is returned to the calling party (from the table in section 5.10.3). Default value 18
IP	DHCP	[ON OFF]	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.(Default value is ON)
	ADDRESS	xxx.xxx.xxx.xxx	Sets the static IP Address of this unit to be that given in parameter 2. (default value is 192.168.1.2)
	MASK	xxx.xxx.xxx.xxx	Sets the static sub-net mask for this unit to be that given in parameter 2. (default value is 255.255.0.0)
	GATEWAY	xxx.xxx.xxx.xxx	Sets the static default gateway address for this unit to be that given in parameter 2 (default value is 0.0.0.0)
	DNS1	xxx.xxx.xxx.xxx	Sets the IP address of the primary static DNS server. If DHCP is enabled then this DNS server will be used in addition to the DNS servers supplied by DHCP. (Default Value is 0.0.0.0)
	DNS2	xxx.xxx.xxx.xxx	Sets the IP address of the secondary DNS server. If DHCP is enabled then this DNS server will be used in addition to the DNS servers supplied by DHCP. (Default Value is 0.0.0.0)
	LOCALDOMAIN		
	WEBPORT	XX	The TCP port through which the Telephone Web server can be accessed (Default Value is 80)
	WEB	[ON/OFF]	This command enables or disables access to the web server (Default value is ON)
	TELNETPORT	XX	The TCP port through which the Telephones telnet server can be accessed (Default Value is 23)
	TELNET	[ON/OFF]	This command enables or disables access to the telnet server (Default value is ON)
	SYSLOGPORT	XX	This command sets the port number to be used for syslog messages. The default value is 514
	SYSLOG	xx.xxx.xxx.xxx or FQDN	This command sets the destination address for syslog server message
	STUN	xx.xxx.xxx.xxx or FQDN	This command sets the IP address or the URL of the STUN server used by the unit. The absence of the value disables the use of STUN.
	SYSLOGFACILITY	xx	The SYSLOG message facility level, as per RFC3164.
	SYSLOGSEVERITY	xx	The SYSLOG message severity level, as per RFC3164.
KEY	RECALL	xxx.xxx.xxx.xxx	This command sets the IP address of the remote telephone that will be signalled when the RECALL Key is pressed on the local phone.
	INHIBIT	<DIGIT> <+> <MEMORY> <+> <CLEAR> or <NONE>	Functions to inhibit in emergency mode. See Key mapping web page section (5.11) for detailed explanation.
	RELAY1ONCODE	XXXX[XX]	These codes set relays OUTPUT1 and OUTPUT2 to activate on receipt of a matching string of DTMF characters from the remote party during a call. Outputs can be set to turn ON, OFF or to PULSE for a fixed duration below. Strings must be min 4 max 6 characters. Valid characters are 0-9, A-D, *,#. Blank disables. Default value blank.
	RELAY2ONCODE		
	RELAY1OFFCODE		
	RELAY2OFFCODE		
	RELAY1PULSECODE		
	RELAY2PULSECODE	1-60	Duration in seconds that an output will remain ON following receipt of a RELAYxPULSECODE. Range 1-60.
RELAY1PULSELEN			
RELAY2PULSELEN			
LED	MODE	HELPPPOINT	These commands preset the LED functions to mimic existing analogue telephone models: HELPPPOINT: Sets LED1 to RING+HOOK, LED2 to OFF DDA: Sets LED1 to HOOK+RINGOUT, and LED2 to CONNECT+RING. OFF: Sets both LEDs OFF
		DDA	

Section	Item Identifier	Value(s)	Comments																										
		OFF																											
	[LED1 LED2 LED3]	GENERATE (Items marked "+" can be combined in any order separated by a "+" symbol)	<table border="1"> <tr> <td>OFF</td> <td>This command sets the function of LED 1 or 2 to be off. (Default State)</td> </tr> <tr> <td>PULSE+</td> <td>This command will cause the LED to illuminate for the period defined by the TIMER command, when an activation message is received from a remote unit</td> </tr> <tr> <td>MUTE+</td> <td>This command sets the function of LED 1 or 2 to indicate if the audio input is muted.</td> </tr> <tr> <td>RING+</td> <td>This command sets the function of LED 1 or 2 to be enabled when an incoming call is present and ringing, and the on /off period will be as defined by the SET CADENCE commands</td> </tr> <tr> <td>CALL+</td> <td>This command sets the function of LED 1 or 2 to be flashing when an outgoing call is present with an on /off period which is as defined by the LOGIC SET CADENCE commands</td> </tr> <tr> <td>CONNECT+</td> <td>This command set the function of the LED output 1 OR 2 to be enabled when a call is connected. If the ANI feature is enabled, the LED output is enabled only when the call is connected and DTMF # has been received.</td> </tr> <tr> <td>HOOK+</td> <td>This command sets the function of the LED output 1 OR 2 to show the status of the hook switch or ON/OFF/TOGGLE button states. When the unit is OFF_HOOK or ON the LED will be enabled. When the unit is ON_HOOK or OFF the LED will be disabled</td> </tr> <tr> <td>INUSE+</td> <td>This command sets the function of the LED output 1 or 2 to be enabled when an incoming call arrives or when the user goes off hook for an outgoing call, and disabled when the call ends.</td> </tr> <tr> <td>RINGCADENCE+</td> <td>Causes the LED to flash in time with the incoming ring signal cadence. This cadence is set by the parameters for Tone 7 (TONES page).</td> </tr> <tr> <td>PAGE +</td> <td>Sets the LED to flash when a call is present that has been signalled as a PAGEMODE call (see UNIT page, section 5.5) The flashing on /off periods are set by the CADENCE field.</td> </tr> <tr> <td>REGISTERED +</td> <td>Sets the LED to flash when the phone is registered with at least one SIP server. Can be used as a "phone available" indicator. The flashing on /off periods are set by the CADENCE field.</td> </tr> <tr> <td>EMERGENCY +</td> <td>Sets the LED to flash whenever there is an outgoing call present that has been initiated by an EMERGENCY button. The flashing on /off periods are set by the CADENCE field.</td> </tr> <tr> <td>ON</td> <td>This command sets the function of the LED 1 or 2 outputs to be enabled when an incoming call arrives or when the user goes off hook for an outgoing call, and disabled when the call ends.</td> </tr> </table>	OFF	This command sets the function of LED 1 or 2 to be off. (Default State)	PULSE+	This command will cause the LED to illuminate for the period defined by the TIMER command, when an activation message is received from a remote unit	MUTE+	This command sets the function of LED 1 or 2 to indicate if the audio input is muted.	RING+	This command sets the function of LED 1 or 2 to be enabled when an incoming call is present and ringing, and the on /off period will be as defined by the SET CADENCE commands	CALL+	This command sets the function of LED 1 or 2 to be flashing when an outgoing call is present with an on /off period which is as defined by the LOGIC SET CADENCE commands	CONNECT+	This command set the function of the LED output 1 OR 2 to be enabled when a call is connected. If the ANI feature is enabled, the LED output is enabled only when the call is connected and DTMF # has been received.	HOOK+	This command sets the function of the LED output 1 OR 2 to show the status of the hook switch or ON/OFF/TOGGLE button states. When the unit is OFF_HOOK or ON the LED will be enabled. When the unit is ON_HOOK or OFF the LED will be disabled	INUSE+	This command sets the function of the LED output 1 or 2 to be enabled when an incoming call arrives or when the user goes off hook for an outgoing call, and disabled when the call ends.	RINGCADENCE+	Causes the LED to flash in time with the incoming ring signal cadence. This cadence is set by the parameters for Tone 7 (TONES page).	PAGE +	Sets the LED to flash when a call is present that has been signalled as a PAGEMODE call (see UNIT page, section 5.5) The flashing on /off periods are set by the CADENCE field.	REGISTERED +	Sets the LED to flash when the phone is registered with at least one SIP server. Can be used as a "phone available" indicator. The flashing on /off periods are set by the CADENCE field.	EMERGENCY +	Sets the LED to flash whenever there is an outgoing call present that has been initiated by an EMERGENCY button. The flashing on /off periods are set by the CADENCE field.	ON	This command sets the function of the LED 1 or 2 outputs to be enabled when an incoming call arrives or when the user goes off hook for an outgoing call, and disabled when the call ends.
OFF	This command sets the function of LED 1 or 2 to be off. (Default State)																												
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ON	This command sets the function of the LED 1 or 2 outputs to be enabled when an incoming call arrives or when the user goes off hook for an outgoing call, and disabled when the call ends.																												
	[LED1 LED2 LED3]	TIMER	XX	This command sets the timer value for the LED, used in the PULSE command where XX is the time in seconds. Default value is 3																									
	[LED1 LED2 LED3]	CADENCE	ON:OFF	This command sets the cadence for an LED. To be on for ON/10 seconds where ON is in tenths of seconds, and then OFF for OFF/10 seconds. The default value for ON is 10, and OFF is 0 implying the contact does not "flash".																									
LOCAL	SPEED	9600		Sets the Port speed to be 9.6 Kbps																									
		19200		Sets the Port speed to be 19.2 Kbps																									
		38400		Sets the Port speed to be 38.4 Kbps																									
		57600		Sets the Port speed to be 9.6 Kbps																									
		115200		Sets the Port speed to be 115.2 Kbps (Default)																									
LOGIC	[INPUT1 INPUT2 INPUT 3 INPUT 4]	DETECT ON		This command enables the generation of messages for the assertion of an auxiliary input. Default setting is NONE for all inputs.																									
		DETECT OFF		This command enables the generation of messages for the de-assertion of an auxiliary input. Default setting is NONE for all inputs.																									
		DETECT NONE		This command disables the generation of messages for the auxiliary input.																									

Section	Item Identifier	Value(s)	Comments	
		DETECT ON+OFF	This command enables the generation of messages for both assertion and de-assertion of the auxiliary input.	
		SENSE [NORMAL INVERT]	This command inverts the sense of the auxiliary input. If set to NORMAL a logic 0 on the input is regarded as being the OFF state. If Set to INVERT, a Logic 1 on the input is regarded as being in the OFF state. Default is NORMAL.	
		SYSLOG ON	This command enables SYSLOG reporting for the selected auxiliary input number.	
		SYSLOG OFF	This command disables SYSLOG reporting for the selected auxiliary input number.	
		MAIL ON	This command enables email reporting for the selected auxiliary input.	
		MAIL OFF	This command enables email reporting for the selected auxiliary input	
		[INPUT1 INPUT2 INPUT 3 INPUT 4] MSG	"text"	This command replaces the default text message Aux_in Δ<input number> with the text entered as the value after the = text. The text is delimited by quote marks and is a maximum of 40 characters. The status <on/off> is appended to the end of the text. If "text" is blank , the message defaults back to the Aux_IN
	[OUTPUT1 OUTPUT2]	GENERATE (Items marked "+" can be combined in any order separated by a "+" symbol)	OFF	This command sets the function of output 1 or 2 to be off. (Default State)
			MUTE+	This command sets the function of output 1 or 2 to indicate if the audio input is muted.
			PULSE+	This command will cause the output to operate for the period defined by the TIMER command, When an activation message is received from a remote unit
			RING+	This command sets the function of output 1 or 2 to be enabled when an incoming call is present and ringing.
			CALL+	This command sets the function of Volt Free Contact 1 or 2 to be enabled when an outgoing call is present with an on /off period which is as defined by the LOGIC SET CADENCE commands
			CONNECT+	This command set the function of the output 1 OR 2 to be enabled when a call is connected.
			HOOK+	This command sets the function of output 1 OR 2 to show the status of the hook switch or ON/OFF/TOGGLE button states. When the unit is OFF_HOOK or ON the LED will be enabled. When the unit is ON_HOOK or OFF the output will be disabled
			INUSE+	This command sets the function of output 1 or 2 to be enabled when an incoming call arrives or when the user goes off hook for an outgoing call, and disabled when the call ends.
RINGCADENCE+			Causes the output to pulse in time with the incoming ring signal cadence. This cadence is set by the parameters for Tone 7 (TONES page).	
PAGE +			Sets the output to pulse when a call is present that has been signalled as a PAGEMODE call (see UNIT page, section 5.5) The pulsing on /off periods are set by the CADENCE field.	
REGISTERED +			Sets the output to pulse when the phone 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.			
		ON	This command sets the function of output 1 or 2 outputs to be enabled when an incoming call arrives or when the user goes off hook for an outgoing call, and disabled when the call ends.	
	[OUTPUT1 OUTPUT2]	TIMER	XX	This command sets the value of the timer used by the PULSE action when the appropriate command is received from a remote station

Section	Item Identifier	Value(s)		Comments	
	[OUTPUT1 OUTPUT2]	CADENCE	ON:OFF	This command sets the cadence for a contact output. To be on for ON/10 seconds where ON is in tenths of seconds, and then OFF for OFF/10 seconds. The default value for ON is 10, and OFF is 0 implying the contact does not "flash"	
MULTICAST	TIMEOUT	[1-120]		sets an enforced delay (in seconds) between one Multicast session ending and another beginning. Range 1-120. Default 120	
	SPEAKERVOLUME	1-10		sets the speaker volume during a multicast. Volume will revert to the setting on the AUDIO page when the multicast session has ended. Range is 1 to 10, default value 3.	
	OVERRIDE	0-8		Priority level. See section 5.18.	
	ADDRESS	[1-8]	xxx.xxx.xxx.xxx:ppp p	Multicast address, must include multicast port.	
	FILTER		xxx.xxx.xxx.xxx:xxx .xxx.xxx.xxx	Sets a range of acceptable multicast source IP addresses. The phone will only accept a multicast if the source is within this IP address range. The format is 2 IP addresses separated by a colon. The default is 0.0.0.0:255.255.255.255	
	PRIORITY		[0-8]	Priority level. See section 5.18.	
	OUTPUT1		[ENABLED DISABLED]	sets if OUTPUT1 is ENABLED or DISABLED during this multicast	
	OUTPUT2			sets if OUTPUT2 is ENABLED or DISABLED during this multicast	
	OUTPUT2			sets if tone 10 (TONES page) is ENABLED or DISABLED during this multicast.	
OUTPUT2	sets if tone 10 (TONES page) is ENABLED or DISABLED during this multicast.				
TONE					
SIP	LOCALPORT	XXXX		Configures the port number used for the local SIP signalling socket to be XXXX. The default value is 5060.	
	PROXYFAILOVERSTAT USES	XXX, XXX ...		Contains a list of SIP status codes that will trigger a fail over from one proxy to the next.	
	DONTSTARTMEDIAAT RING	[ON OFF]		Delay the sending of media packets to end points until the call has been answered.	
	SENDDTMFLAST	[ON OFF]		Reorder the codec sequence to end points, so that the DTMF codec is sent last.	
	RTPTOS	XX		This command sets the value of the TOS/Diffserv field in the UDP packets carrying RTP data. (Default value = 46)	
	SINGLEPTIME	[0-63]		Forces a single packet time to the value set in ms. See 5.4.	
	SENDMULTIPARTMIME	[ON OFF]		This option is for future enhancement and should always be set to 'OFF'. Default value OFF.	
	NEWBRANCHONAUTH BYE	[ON OFF]		This is a legacy option that is no longer used, and must always be set to 'ON'. Default value 'ON'.	
	MODE	[SERIAL MULTIPLE]		Sets whether multiple proxies and registrars are used serially or concurrently.	
	REGTIMEOUT	xxxx		This command sets the Registration timeout value that will be suggested by the telephone to a Registrar to be xxxx seconds. The default value is 3600 seconds. Valid values are 0 to $2^{32} - 1$	
	REREGTIMEOUT			Sets a period in seconds after which the phone will force a re-registration period and the server cannot override it.	
	LOCALID	[1-4]	"identity"		Sets the identity of the user/phone that will be used in the registration process with a registrar. The Identity parameter can consist of any alpha numeric string, and will be used within the To:, From: and Contact: headers. EXAMPLE LOCALID= john.smith would give: To: <sip: john.smith @ gai-tronics.co.uk >
	DOMAIN		FQDN minus host name		Sets the domain name to be a FQDN without the leading host part. Example DOMAIN= gai-tronics.co.uk
	PROXY		xxx.xxx.xxx.xxx or FQDN		Sets the IP address or the FQDN of the SIP proxy server to be used for incoming/outgoing calls.
PROXYPORT	XXXX		The port number on the proxy used for SIP protocol signalling can be changed to XXXX using this command. The default value is 5060		
PRIORITY	[1 - 4]		Sets the failover sequence between the 4 proxies		

Section	Item Identifier	Value(s)	Comments		
	REGISTRAR	xxx.xxx.xxx.xxx or FQDN	This command sets the address of the Registrar to be xxx.xxx.xxx.xxx or the FQDN specified. The registrar address and the proxy may or may not be the same address, but the address for registration must be set here.		
	REGISTRARPORT	XXXX	The port to send the register to.		
	USERNAME	"username"	This command sets the username for the authorisation realm to be username (Default value is blank)		
	PASSWORD	"password"	This command sets the password for the authorisation realm to be password (Default value is blank)		
	ENDPOINT	[ENABLED DISABLED]	Sets whether this set of proxy parameters is active.		
SMTP	SERVER1	xxx.xxx.xxx.xxx or FQDN	Sets the IP address of the primary SMTP server to be xxx.xxx.xxx.xxx or uses the FQDN to resolve the IP address through DNS. Email will be sent on assertion of an alarm condition via the primary server if configured.		
	SERVER2	xxx.xxx.xxx.xxx or FQDN	Sets the IP address of the secondary SMTP server to be xxx.xxx.xxx.xxx or uses the FQDN to resolve the IP address through DNS. Email will be sent on assertion of an alarm condition via the secondary server if configured.		
	TOADDRESS	abc@hij.com	Sets the To: Address		
	CCADDRESS	abc@hij.com	Sets the CC: Address		
	FROMADDRESS	abc@hij.com	Sets the FROM: Address		
	SUBJECT	"SubjectText"	Set the contents of the subject field to be "SubjectText". The Subject Text field can be up to 50 characters in length, and can contain any printable character except double quotes.		
	SMTP	[ON OFF]	This setting enables or disables the use of SMTP within the unit.		
TONES	[TONE1 –TONE10]	FREQ	1-13	These commands set the various call tones produced by the telephone. See Tone Settings web page section (5.15) for detailed explanation. For example to set a standard UK and US ring tones the commands would be:	
		ON1	0-600 (*25ms)		
		OFF1			
		ON2		UK	US
		OFF2		TONE3=FREQ 5 TONE3=ON1 0 TONE3=OFF1 0 TONE3=ON2 16 TONE3=OFF2 8 TONE3=ON3 16 TONE3=OFF3 80	TONE3=FREQ 3 TONE3=ON1 0 TONE3=OFF1 0 TONE3=ON2 80 TONE3=OFF2 160 TONE3=ON3 0 TONE3=OFF3 0
		ON3			
		OFF3			
UNIT	HOSTNAME	xxxxxxx	Sets the unit host name for this unit to be XXXXXXXXXXXX where X is any alphanumeric character. Maximum of 15 characters. Default Value is "UNNAMED".		
	UPDATE	SERVER	xxx.xxx.xxx.xx x or FQDN	Sets the IP address of the host running the TFTP server to be xxx.xxx.xxx.xxx or resolves the FQDN using DNS to access the host containing the update file(s).	
	UPDATE	FILE	Filename	The name of the file on the TFTP server. This name may contain the macro symbols %m, %h and %i. These symbols are expanded to the MAC address, host name and IP address respectively.	
	UPDATE	INTERVAL	X	This command forces the unit to attempt a configuration file download every X hours where X can be an integer value between 0 and 1000. A value of 0 disables the periodic update request. The default value is 1. Any non-zero value will cause the unit to attempt a configuration file download at boot time.	
	HELPSERVER	xx.xxx.xxx.xxx or FQDN	Sets the address for the Web Support Page link on the Home page.		
	LAN	SPEED	AUTO 10 100	Ethernet speed	

Section	Item Identifier	Value(s)		Comments
	LAN	DUPLEX	FULL HALF	Selects the duplex of the ethernet connection. If the speed is automatic, the duplex is too.
	CONFIGID	FreeFormString		String used to compare to see if configuration needs updating.
	ANI	"Identity"		Used as an identifying token to GAI-Tronics CMA Call Management Application.
	DEFAULT_ANS_MODE:	[RING PICK-UP PAGE STEALTH]		Default behaviour determining how an incoming call is answered. See section 5.5. Default value RING.
	ANSMODE1	"password"		Set "passwords" that can be used to activate 3 special auto-answer modes, usually for hands-free telephone types. See web page section for details.
	ANSMODE2	"password"		
	PAGEMODE	"password"		
	APTENABLE	[ON OFF]		APT parameters. See section 5.5.1
	APPTIME	Xx:xx,xx		
	APTCOUNT	1-10		
	APTOKCOUNT	1-10		
	APTREPORT	[ON OFF]		

7. Time Zone Table

Abbreviation	UTC/GMT Offset	Time Zone Name	Cities
IDL	GMT-12:00	IDL (International Date Line), IDLW (International Date Line west)	Eniwetok
NT	GMT-11:00	BT (Bering Time), NT (NomeTime)	Midway
AHST	GMT-10:00	AHST (Alaska-Hawaii Standard Time), HST (Hawaiian Standard Time), CAT (Central Alaska Time)	Hawaii
IMT	GMT-09:30	Isle Marquises	Isle Marquises
YST	GMT-09:00	YST (Yukon Standard Time)	Yukon
PST	GMT-08:00	PST (Pacific Standard Time)	Los Angeles
MST	GMT-07:00	MST (Mountain Standard Time), PDT (Pacific Daylight Time)	Phoenix
CST	GMT-06:00	CST (Central Standard Time), MDT (Mountain Daylight Time)	Dallas, Mexico City, Chicago
EST	GMT-05:00	EST (Eastern Standard Time), CDT (Central Daylight Time), NYC	New York
AST	GMT-04:00	AST (Atlantic Standard Time), EDT (Eastern Daylight Time)	La Paz
NST	GMT-03:30	NST (Newfoundland Standard Time)	Newfoundla nd
BST	GMT-03:00	BST (Brazil Standard Time), ADT (Atlantic Daylight Time), GST (Greenland Standard Time)	Buenos Aires
AT	GMT-02:00	AT (Azores Time)	Mid-Atlantic
WAT	GMT-01:00	WAT (West Africa Time)	Azores
GMT	GMT+00:00	GMT (Greenwich Mean Time), WET (Western European Time), UT (Universal Time)	London
CET	GMT+01:00	BST (British Summer Time), CET (Central European Time), FWT (French Winter Time), MET (Middle European Time), MEWT (Middle European Winter Time), SWT (Swedish Winter Time)	Paris
EET	GMT+02:00	EET (Eastern European Time), USSR-zone1, MEST (Middle European Summer Time), FST (French Summer Time)	Athens, Rome
BT	GMT+03:00	BT (Baghdad Time), USSR-zone2	Baghdad, Moscow
IT	GMT+03:30	IT (Iran Time)	Tehran
ZP4	GMT+04:00	USSR-zone3, ZP4 (GMT Plus 4 Hours)	Abu Dhabi
AFG	GMT+04:30	Afghanistan	Kabul
ZP5	GMT+05:00	USSR-zone4, ZP5 (GMT Plus 5 Hours)	Islamabad
IST	GMT+05:30	IST (Indian Standard Time)	Bombay, Delhi
ZP6	GMT+06:00	USSR-zone5, ZP6 (GMT Plus 6 Hours)	Colombo
SUM	GMT+06:30	NST (North Sumatra Time)	North Sumatra
WAST	GMT+07:00	SST (South Sumatra Time), USSR-zone6, WAST (West Australian Standard Time)	Bangkok, Hanoi

Abbreviation	UTC/GMT Offset	Time Zone Name	Cities
HST	GMT+08:00	CCT (China Coast Time), HST (Hong Kong Standard Time), USSR-zone7, WADT (West Australian Daylight Time)	Beijing, Hong Kong
JST	GMT+09:00	JST (Japan Standard Time/Tokyo), KST (Korean Standard Time), SSR-zone8	Tokyo, Seoul
CAST	GMT +09:30	SAST (South Australian Standard Time), CAST (Central Australian Standard Time)	Darwin
EAST	GMT+10:00	GST (Guam Standard Time), USSR-zone9, EAST (East Australian Standard Time)	Brisbane, Guam
EADT	GMT+11:00	USSR-zone10, EADT (East Australian Daylight Time)	Solomon Islands
NZST	GMT+12:00	NZT (New Zealand Time/Auckland), NZST (New Zealand Standard Time), IDLE (International Date Line East)	Auckland

8. Example Configuration File

The file below is an example with at least one entry in every possible section.

The key features of the phone are:

- 2 button handsfree with an emergency button (dials 888) and an information button (dials three alternative end points in cascade).
- Output 1 activates a beacon when the unit is making or receiving a call.
- Output 2 activates a door release on command from another telephone
- Call tones are set to UK patterns
- Alarms 2, 3, 4 & 8 report via SYSLOG
- Alarm 6 (stuck button) reports via email
- Input 1 is configured as a vandal alarm and also reports via email

```
[ACCESS]
USERNAME=newuser          //User name changed
PASSWORD=newpassword     //Password changed
[ALARMS]
ALARM1=REPORT NONE
ALARM2=REPORT ON
ALARM2=ONTIME 0
ALARM2=OFFTIME 0
ALARM2=SYSLOG ON
ALARM2=MAIL OFF
ALARM2=MSG "Config Error"
ALARM3=REPORT ON
ALARM3=ONTIME 0
ALARM3=OFFTIME 0
ALARM3=SYSLOG ON
ALARM3=MAIL OFF
ALARM3=MSG "Cold Reset"
ALARM4=REPORT ON
ALARM4=ONTIME 0
ALARM4=OFFTIME 0
ALARM4=SYSLOG ON
ALARM4=MAIL OFF
ALARM4=MSG "Warm Reset"
ALARM6=REPORT ON
ALARM6=ONTIME 10
ALARM6=OFFTIME 2
ALARM6=SYSLOG OFF
ALARM6=MAIL ON
ALARM6=MSG "Stuck button"
ALARM8=REPORT ON
ALARM8=ONTIME 360
ALARM8=OFFTIME 0
ALARM8=SYSLOG ON
ALARM8=MAIL OFF
ALARM8=MSG "Registration Fail"
[AUDIO]
CODEC=5,6,4,3,1,2
DTMF=RFC2833
FRAMES=G711 20
FRAMES=G722 20
FRAMES=G729 2
FRAMES=G7231 1
HANDSFREEGAIN=3
HANDSFREEVOLUME=3
JITTERMAX=60
JITTERMIN=30
```

```
RINGERVOLUME=10
SAMPLE=G711 20
SAMPLE=G722 20
SAMPLE=G729 20
SIDETONE=OFF
VAD=OFF
[CLOCK]
DST=ADJUST ON
DST=OFFSET +01:00
DST=STARTDAY 0
DST=STARTDOW 1
DST=STARTMONTH 3
DST=STARTWOM 8
DST=STARTTIME 02:00
DST=ENDDAY 0
DST=ENDDOW 1
DST=ENDMONTH 10
DST=ENDWOM 8
DST=ENDTIME 02:00
FORMAT=DD/MM
SNTP=ntp2b.mcc.ac.uk
SNTPINTERVAL=60
TIMEZONE=GMT
[DIALPLAN]
MEMORY=1 888
MEMORY=2 sip:info@mydomain.com
MEMORY=3 sip:security@mydomain.com
MEMORY=4 223344@mydomain.com
COMFORT=1 888
COMFORT=2 223344
COMFORT=3 223344
COMFORT=4 223344
LIST=1 2, 3, 4
LIST=0 1
WAKEANDDIAL=1 ON
WAKEANDDIAL=2 ON
LIST=2
LIST=3
LIST=4
LIST=5
LIST=6
LIST=7
LIST=8
LIST=9
LIST=10
CALLFAIL=30
CALLLIMIT=7
DIALTIME=4
FAILOVERCAUSES=1,17,18,21,27,38,41,50,88
MAXLEN=25
OFFHOOK=
PRECALL=30
TERMINATOR=
[IP]
ADDRESS=192.168.1.2
DHCP=OFF
DNS1=0.0.0.0
DNS2=0.0.0.0
GATEWAY=0.0.0.0
LOCALDOMAIN=mydomain.com
MASK=255.255.0.0
```

```
STUN=
SYSLOG=192.168.1.25
SYSLOG2=192.168.1.26
SYSLOGFACILITY=14
SYSLOGPORT=514
SYSLOGPORT2=514
SYSLOGSEVERITY=5
TELNET=ON
TELNETPORT=23
WEB=ON
WEBPORT=80
[KEY]
INHIBIT=MEMORY
RECALL=
[LED]
MODE=DDA
LED1=CADENCE 10: 10
LED2=CADENCE 10: 10
[LOCAL]
SPEED=115200
[LOGIC]
INPUT1=DETECT ON
INPUT1=SENSE NORMAL
INPUT1=SYSLOG OFF
INPUT1=MAIL ON
INPUT1=MSG "Tamper Alarm"
OUTPUT1=TIMER 3
OUTPUT1=CADENCE 10:0
OUTPUT1=GENERATE RING + INUSE
OUTPUT2=TIMER 3
OUTPUT2=CADENCE 10:0
OUTPUT2=GENERATE PULSE
[SIP]
LOCALPORT=5060
RTPTOS=46
PROXYFAILOVERSTATUSES=5xx, 6xx, 49x, 403, 406, 9xx
DONTSTARTMEDIAATRING=OFF
SENDDTMFLAST=OFF
MODE=SERIAL
REGTIMEOUT=3600
REREGTIMEOUT=0
DOMAIN=1 192.168.1.25
LOCALID=1 12345
PROXY=1 192.168.1.25
PROXYPORT=1 5060
PRIORITY=1 1
REGISTRAR=1 192.168.1.25
REGISTRARPORT=1 5060
ENDPOINT=1 ENABLED
ENDPOINT=2 DISABLED
ENDPOINT=3 DISABLED
ENDPOINT=4 DISABLED
[SMTP]
CCADDRESS=
FROMADDRESS=Help12345@mydomain.com
SERVER1=192.168.1.25
SERVER2=192.168.1.26
SMTP=ON
SUBJECT="Help Point Alarm"
TOADDRESS=maintenance@mydomain.com
[TONES]
```

TONE1=FREQ 4
TONE1=ON1 0
TONE1=OFF1 0
TONE1=ON2 80
TONE1=OFF2 0
TONE1=ON3 0
TONE1=OFF3 0
TONE2=FREQ 4
TONE2=ON1 0
TONE2=OFF1 0
TONE2=ON2 16
TONE2=OFF2 4
TONE2=ON3 0
TONE2=OFF3 0
TONE3=FREQ 5
TONE3=ON1 0
TONE3=OFF1 0
TONE3=ON2 16
TONE3=OFF2 8
TONE3=ON3 16
TONE3=OFF3 80
TONE4=FREQ 1
TONE4=ON1 0
TONE4=OFF1 0
TONE4=ON2 15
TONE4=OFF2 15
TONE4=ON3 0
TONE4=OFF3 0
TONE5=FREQ 1
TONE5=ON1 0
TONE5=OFF1 0
TONE5=ON2 16
TONE5=OFF2 14
TONE5=ON3 9
TONE5=OFF3 21
TONE6=FREQ 1
TONE6=ON1 0
TONE6=OFF1 0
TONE6=ON2 80
TONE6=OFF2 0
TONE6=ON3 0
TONE6=OFF3 0
TONE7=FREQ 10
TONE7=ON1 0
TONE7=OFF1 0
TONE7=ON2 16
TONE7=OFF2 8
TONE7=ON3 16
TONE7=OFF3 80
[UNIT]
CONFIGID=HelpPoint1

9. Command Line Interface

Configuration settings within GAI-Tronics VoIP telephones can be changed by typing commands into a Command Line Interface (CLI).

This section describes the syntax required for CLI commands. Generally, the CLI commands match those used in configuration files. Therefore the feature descriptions listed below may be abbreviated. For fuller descriptions refer to the sections on the relevant web pages or configuration file syntax.

The CLI can be accessed in 2 ways:

- Over a direct serial link via the integral serial port
- Via a Telnet session over the network

Generally speaking, the CLI is not the preferred access method, but it may offer advantages in certain circumstances. In particular it offers a method of accessing the telephone and discovering the IP address if it has been lost. (Password security is maintained).

To start a CLI session via serial link, connect a standard RS232 serial cable between the telephone and a PC serial port, and connect using a terminal program such as HyperTerminal. The default port settings are 115200 baud, 8 data bits, 1 stop bit, no parity. Note that the speed can be changed (See the Serial Settings web page section).

A CLI session can also be started by entering "Telnet 192.168.1.2" from a command prompt on a computer that can see the telephone on the network (substitute the IP address if it has been changed).

The behaviour of the CLI is the same regardless of the access method, and its first response is to request the USERNAME and PASSWORD.

At log in, the following information is displayed (the values presented here are examples):

```
Welcome <username>
Welcome to the GAI Tronics SIP Phone CLI.
Board type: a
Board serial: b
Daughter type: c
Daughter serial: d
Unit type: e
Unit serial: f
MAC address: 00:01:df:65:43:21
2.1.7
2.1.7 GAICLI
2.1.7 GAIUISERVER
2.1.7 GAIPHONE
2.1.7 GAIGW
2.1.7 CONFIGACTIVATOR
```

To view it again, log out and log back in again.

9.1 CLI Syntax

The Command Line interface will provide a Command Line Prompt as shown below: -

[UNIT IDENTIFIER]>>

The unit identifier is a configuration option that provides a user configurable name up to a maximum of 32 characters that can be used to identify the unit. By default the Unit identifier is set to "UNNAMED".

The command line interface syntax consists of three parts, a module name, an action verb and a variable set of action parameters. Each command is terminated by a carriage return.

[Module Name] < Action Verb > < Parameter List > [CR]

Although each module name or action verb may consist of several letters, only sufficient letters to uniquely identify the module name or action verb are required. For example to enter the module name **LOCAL**, only three characters **LOC** are required to differentiate it from the module name **LOGIC**.

The module names are the same as the section names listed in section 6.1.

If a module name is entered without an action verb to follow, the command line focus enters the module name given, for example the command: -

UNIT [CR]

Will cause the command line interface focus to enter the UNIT module, and the Command line prompt will change to:

[UNIT IDENTIFIER]>>UNIT>>

When the Command line focus is within a specific module, then only the action verbs specific to that module will be effective. To return focus to the highest level, use:

EXIT [CR]

A list of all the commands applicable to the current module can be obtained by:

HELP [CR]

Some commands allow multiple parameters. For example to set both **KEY SET INHIBIT DIGIT** and **KEY SET INHIBIT MEMORY**, enter them together in one command by placing a + sign between the parameters, as **KEY SET INHIBIT DIGIT + MEMORY**. Entering **KEY SET INHIBIT DIGIT** and **KEY SET INHIBIT MEMORY** separately would cause the last entered command to overwrite the earlier one.

Command parameters that can be combined in this way are indicated by a + sign following their definition.

A history of the last 100 commands issued can be obtained by using:

HISTORY[CR]

The history list is accessed using the up and down arrows on your keyboard.

In all of the following actions where the action is **SET**, the **SET** can be replaced with **SHOW** along with the first parameter to display the individual configuration information.

9.2 ACCESS Module Command Line Syntax

Action Verb	Parameters		Comment	
	1	2		
SET	USERNAME	UserName	Sets the user name used in local or remote access to be "UserName". UserName can be up to 30 characters long, and can contain only the alphanumeric characters a-z, A-Z , 0-9 . The default value is "user". The Username cannot be blank. IMPORTANT: The word 'root' is a reserved username and must not be used or assigned a password. Setting a user name of "root" will make it impossible to access the phone, and will require a reset to factory defaults.	
	PASSWORD	<Password>	Sets the password used in local or remote access to be "Password". <Password> can be up to 30 characters long, and can contain only the alphanumeric characters a-z, A-Z , 0-9 . The default value is password.	
QUIT			This command returns the user to the login prompt, effectively logging the user out and returning the CLI to the secured state.	
EXIT			This command returns the user to the top level of the Command line Interface	
SHOW			ALL	Shows the current access Name and Password and also the access error counters
RESET			COUNTERS	Resets the access error counters

Note: please make sure to record the user name and password securely. They will be required to access the phone every time whether by web page, command line or configuration file.

In the event that the username and password are lost, the unit will need to be reset to factory defaults. This can be done by holding down a button on the main circuit board. See section 10.

9.3 ALARMS Module Command Line Syntax

Action Verb	Parameters			Comment
	1	2	3	
SET	ALARM[1-20]	REPORT	[ON OFF NONE]	This command specifies if an alarm will be generated on assertion of an alarm condition only (ON) , on removal of the alarm condition only (OFF) or on either event (ON+OFF) or not at all (NONE)
		ONTIME	[0- 30000]	This command assigns alarm activation De-bounce Period to a specific alarm number. The alarm event must be present at the start and at the end of the de-bounce Period before the alarm status will be signalled using e-mail or syslog messaging (If enabled). The period is specified in seconds and can take a value of 0 – 30,000. A value of 0 indicates that there is no de-bounce period for this alarm type and a message will be generated immediately the alarm event is detected.
		OFFTIME	[0-30000]	This command assigns an alarm removal De-bounce Period to a specific alarm number. The alarm event must be absent at the start and at the end of the de-bounce period before the alarm clearance will be signalled using e-mail or syslog messaging (if enabled). The period is specified in seconds and can take a value of 0 – 30,000. A value of 0 indicates that there is no de-bounce period for this alarm type and a message will be generated immediately the alarm event is detected.
		MSG	"text"	This command replaces the default text message ALARM Δ<alarm_number> with the text entered as Parameter 3. The text is delineated by Quote marks and is a maximum of 40 characters. The status <on/off> is appended to the end of the text. If the "text" value is blank , the default message is reinstated.
		SYSLOG	[ON OFF]	This command enables or disables SYSLOG reporting for the selected alarm number. This command will also set STATUS reporting if not already applied.
		MAIL	[ON OFF]	This command enables or disables SMTP reporting for the selected alarm number. This command will also set STATUS reporting if not already applied.
SHOW	ALL			This command displays the current settings of the alarm module. It also shows status information for the alarms. An alarm can be :- <ul style="list-style-type: none"> a) ON - the alarm is in the up condition b) OFF – the alarm is in the down condition c) PENDING ON – the alarm has occurred but not reached the end its de-bounce period d) PENDING OFF- the alarm has been cleared but not yet been signalled

9.4 KEY Module Command Line Syntax

Action Verb	Parameters		Comment
	1	2	
SET	RECALL	xxx.xxx.xxx.xxx	This command sets the destination unit IP address and output number for a relay contact that will be operated when the RECALL key is pressed. The command will be sent using an extension to the proprietary protocol defined for passing logic states. (No implemented in first release)
	INHIBIT	DIGIT +	If this parameter is set then all digit keys will be disabled during an emergency call.
		MEMORY +	If this parameter is set then all memory keys will be disabled during an emergency call.
		CLEAR +	If this parameter is set then any key will capable of initiating a call clear be disabled during an emergency call.
		NONE	This command clears all key inhibit settings (DEFAULT VALUE)
	RELAY1ONCODE	XXXX[XX]	These codes set relays OUTPUT1 and OUTPUT2 to activate on receipt of a matching string of DTMF characters from the remote party during a call. Outputs can be set to turn ON, OFF or to PULSE for a fixed duration below. Strings must be min 4 max 6 characters. Valid characters are 0-9, A-D, *,#. Blank disables. Default value blank.
	RELAY2ONCODE		
	RELAY1OFFCODE		
	RELAY2OFFCODE		
	RELAY1PULSECODE		
RELAY2PULSECODE			
RELAY1PULSELEN	1-60		
RELAY2PULSELEN			
SHOW	ALL		This command shows the current settings of the RECALL and INHIBIT settings and lists the key map settings.

9.5 LED Module Command Line Syntax

Action Verb	Parameters			Comment
	1	2	3	
MODE	HELPPPOINT			Refer to web page section for function descriptions
	DDA			
	OFF			
SET	LED [1 2 3]	GENERATE	OFF	
			PULSE +	
			MUTE +	
			RING +	
			CALL +	
			CONNECT +	
			HOOK +	
			INUSE +	
			RINGCADENCE +	
			PAGE +	
REGISTERED +				
EMERGENCY +				
		ON		
	TIMER	XX		
	CADENCE	ON:OFF		
SHOW	ALL			Shows the current settings of all the LED parameters

9.6 DIALPLAN Module Command Line Syntax

Action Verb	Parameters			Comment
	1	2	3	
SET	MEMORY	1-20	<phone_num or SIP URI>	This command sets the dial plan memory storage position specified in parameter 2 to be the telephone number or SIP URI specified in Parameter 3. The entry can be cleared by omitting parameter 3.
	COMFORT	1-20	<phone_num>	This command is used to set a dial string to be used to generate DTMF digits as a memory dial feedback to the user if the memory store contains a URI rather than a telephone number. If the call reaches the alerting or connected state prior to the completion of the play out of the comfort string or the real stored number, the play out will cease. Max 30 characters.
	LIST	0 – 10	A,B,C,D up to 20 entries	This command sets the dial plan list associated with a memory or emergency key function to contain the memory locations specified by A, B, C, D and so on. The order of call attempt will be as specified.
	OFFHOOK	0-10		This command sets the memory list that is associated with the off hook state. Omitting parameter 2 sets the list to be non applicable and entering the off hook state will not cause a memory /emergency list to be applied. The default value is blank (non applicable)
	TERMINATOR	[#]*		This command sets the dial string terminator character to be either #, * or if omitted (not used). The default value is blank (not used). If the user dials the selected character the call setup will be initiated.
	MAXLEN	XX		This command sets the maximum dial string length to XX digits. Once the user has dialled XX digits the call will be initiated. The default value is 25 digits. Range is 1-99.
	DIALTIME	X		This command sets the inter-digit timeout value to be X seconds. Once the user has entered the off hook state, then failure to receive another digit within the timeout period will result in the call being initiated with the dialled digits received so far. A value of 0 seconds disables the use of the inter-digit timeout. The default value is 5 seconds. The maximum is 20 seconds.
	CALLLIMIT	X		This command sets the maximum time allowed for a call in minutes. The value of X can 0 – 240 in minutes. The value 0 disables the timer. The default value is 0.
	PRECALL	X		This command sets length of time in seconds that a phone will remain in the initial off hook state generating dial tone without a dialling key being pressed. After this delay the phone will cease dial tone and enter the on hook state even if the hook switch is off hook. The value 0 disables this timeout. The default value is 30. Maximum is 60
	CALLFAIL	X		This command set the length of time that the phone will remain in the off hook state after the call has ended. The default value is 30 seconds a value of 0 disables this timeout. Range is 0-30
	FAILOVERCAUSES	1-127		Comma separated list of cause codes that would allow the phone to try the next entry in a list of memories. It is in no particular order. The default list is: 1,17,18,21,27,38,41,50,88
	REMOTEALERTINGTIMEOUT	[0-600]		sets the maximum length of time in seconds that the phone will ring on an outgoing call before timing out and returning "Number Unobtainable" (Tone 6) to the user. A value of 0 disables the timer, meaning it will ring until the caller hangs up or the remote end refuses the call. Default value 0.
	LOCALALERTINGTIMEOUT	[0-600]		sets the maximum length of time in seconds that the phone will ring on an incoming call before timing out and returning a "no answer" result to the caller. A value of 0 disables the timer, meaning it will ring until the caller stops ringing. Default value 0.
	REMOTEALERTINGTIMEOUTCAUSECODE	XX		sets the cause code (from the table above) that will be entered in the call description record (CDR) if an outgoing call times out without being answered. Default value 18.
	LOCALALERTINGTIMEOUT	XX		sets the cause code (from the table above) that will be entered in the call description record (CDR) if an incoming call times out without being answered. It also sets the failover code that is returned to the calling party (from the table in section 5.10.3). Default value 18

Action Verb	Parameters			Comment
	1	2	3	
SHOW	ALL			This command lists the current settings of the memory dial locations 1 – 20 and also lists the order of use of lists 1-10 and the emergency list It also displays the values for the other variables that can be configured in this module.

9.8 CLOCK Module Command Line Syntax

Action Verb	Parameters			Comment	
	1	2	3		
SET	SNTP	xxx.xxx.xxx.xxx or FQDN		Sets the IP address of the SNTP server to be xxx.xxx.xxx.xxx or resolves the FQDN using DNS to locate the host from which the phone obtains time data.	
	TIMEZONE	XXXX		This command sets the current time zone for local time where XXXX is an abbreviation selected from the time zone abbreviations in section 7.	
	FORMAT	[US UK]		Sets the format of the date to be either US style MM/DD or UK style DD/MM. Default value is UK.	
	ADJUST	ADJUST	ON/OFF		This command determines if the unit's clock will automatically adjust to daylight saving time. The default value is OFF
		OFFSET	+/-HH:MM		This command sets the value of the offset from the current clock time applied when the DST period starts. Valid values for HH are 00 – 12 and for MM they are 00 – 59. The default value is +01:00 (1 hour)
	DST	STARTDAY	[0 – 31]		If DST is active, these commands define when it starts and ends. Refer to Clock Settings web page section for detailed explanation. The example file in Section 8 lists the commands to set DST to be active between 2am on the last Sunday in March until 2am on the last Sunday in October.
		STARTDOW	[1-7]		
		STARTMONTH	[1-12]		
		STARTWOM	[1-6 8]		
		ENDDAY	[1-31]		
		ENDDOW	[1-7]		
		ENDMONTH	[1-12]		
		ENDWOM	[1-6 8]		
STARTTIME	HH:MM				
ENDTIME	HH:MM				

9.9 AUDIO Module Command Line Syntax

Action Verb	Parameters			Comment	
	1	2			
SET	CODEC	A,B,C,D,E,F		This command sets the order of preference that will be used in the SDP by the phone, where A B C D E and F can have the values 1 -6 that corresponds to the CODECS listed in the Audio web page section above. Note VAD must be set to ON when using G.729 Codec.	
	SAMPLE	G711	10	Set the sample period in ms. See 5.13	
			20		
		G729	10		
	20				
	FRAMES	G7231 G722 G729 G711	X		This command sets the number of audio sample periods or "frames" per IP packet . See 5.13
	VAD	[ON OFF]			This command enables or disables the use of Voice Activity Detection. This command is only valid for G723 and G729 Codec settings. Note VAD must be set to ON when using G.729 Codec. The default value is ON.
	DTMF	[INBAND RFC2833]			This command sets the transmission of DTMF digits to be either in band or out of band. The default setting is OUTBAND, DTMF is transmitted using RFC 2833
	DTMFPT	[96-127]			Sets the payload type parameter in the RTP packets when sending DTMF tones 'out-of-band' according to RFC2833. The default value is 96, but should be changed to 101 for compatibility when using Cisco CallManager™.
DTMFPLAYBACK	[ON OFF]		sets whether DTMF tones are heard in the earpiece when digit buttons 0-9, * or # are pressed. Default value OFF.		
HANDSETGAIN HANDSETVOLUME LINEGAIN LINEVOLUME RINGERVOLUME HANDSFREEGAIN HANDSFREEVOLUME	XX		These commands specify the various gains and volumes. Refer to Audio settings web page section for definitions and acceptable value ranges.		
SET	SIDETONE	ON/OFF		Enables or disables sidetone. OFF by default.	
	SIDETONELEVEL	[0-255]		If sidetone is set to ON, this parameter sets its level. Range is 0-255, default value 127.	
	JITTERMIN	30 -> 120		Minimum size of dynamic jitter buffer.	
	JITTERMAX	30 -> 120		Maximum size of dynamic jitter buffer.	
SHOW	ALL			Lists all the configuration settings for the Audio module	

9.10 TONES Module Command Line

Action Verb	Parameters			Comment
	1	2	3	
SET	TONE[1-10]	FREQ	[1-13]	This command sets the frequency (or frequency pair) [1-10] referenced in parameter 3 for the Tone referenced in parameter 1. See Tone Settings web page section (5.15) for detailed explanation.
		[ON1 ON2 ON3 OFF1 OFF2 OFF3]	X	This command sets up the timing for the cadence period referenced in parameter 2 with the value X where X is an integer value between 0 and 600 that is equivalent to the period's time in 25ms increments. See Tone Settings web page section (5.15) for detailed explanation.
SHOW	ALL			This command displays the current settings of all the tone parameters.

IP Module Command Line Syntax

Action Verb	Parameters			Comment
	1	2	3	
SET	DHCP	[ON/OFF]		Enables or disables the use of DHCP for the assignment of IP parameters.
	ADDRESS	xxx.xxx.xxx.xxx		Sets the static IP Address of this unit
	MASK	xxx.xxx.xxx.xxx		Sets the static sub-net mask for this unit
	GATEWAY	xxx.xxx.xxx.xxx		Sets the static default gateway address for this unit
	DNS1	xxx.xxx.xxx.xxx		Sets the IP address of the primary static DNS server.
	DNS2	xxx.xxx.xxx.xxx		Sets the IP address of the secondary static DNS server.
	WEBPORT		XXXX	The TCP port through which the Telephone Web server can be accessed (Default Value is 80)
	WEB		[ON/OFF]	This command enables or disables access to the web server (Default value is ON)
	TELNETPORT		XX	The TCP port through which the Telephones telnet server can be accessed (Default Value is 23)
	TELNET		[ON/OFF]	This command enables or disables access to the telnet server (Default value is ON)
	SYSLOGPORT		xx	This command sets the port number to be used for syslog messages. The default value is 514
	SYSLOG	xxx.xxx.xxx.xxx or FQDN		This command sets the destination address for syslog server messages.
	SYSLOGPORT2		xx	This command sets the port number to be used for syslog messages (second syslog server). The default value is 514
	SYSLOG2	xxx.xxx.xxx.xxx or FQDN		This command sets the destination address for syslog server messages (second syslog server).
SYSLOGFACILITY	xx		The SYSLOG message facility level, as per RFC3164.	
SYSLOGSEVERITY			The SYSLOG message severity level, as per RFC3164.	
STUN	xxx.xxx.xxx.xxx or FQDN		This command sets the IP address or URL for the STUN server that will be used to resolve STUN requests. The absence of parameter 2 will disable the STUN facility.	
SHOW	ALL		Lists all the configuration settings for the IP Module. The current static settings will be displayed along with the current settings if the unit is using a Dynamic IP address	
PING	xxx.xxx.xxx.xxx or FQDN		Sends an ICMP ping to the IP address or the FQDN given in parameter 1	
TRACEROUTE	xxx.xxx.xxx.xxx or FQDN		Executes a series of PING messages with varying HOP numbers in order to determine the routing used to reach the destination address.	

9.12 LOCAL Module Command Line Syntax

Action Verb	Parameters		Comment
	1	2	
SET	SPEED	9600	Sets the Port speed to be 9.6 Kbps (Default)
		19200	Sets the Port speed to be 19.2 Kbps
		38400	Sets the Port speed to be 38.4 Kbps
		57600	Sets the Port speed to be 57.6 Kbps
		115200	Sets the Port speed to be 115.2 Kbps
SHOW	ALL		

9.13 LOGIC Module Command Line Syntax

Action Verb	Parameters			Comment
	1	2	3	
SET	INPUT[1-4]	DETECT	ON +	This command enables the generation of messages for the auxiliary input given by the 2 nd parameter when it goes to the ON state .Default setting is NONE for all inputs.
			OFF +	This command enables the generation of messages for the auxiliary input given by the 2 nd parameter when it goes to the OFF State. Default setting is NONE for all inputs.
			NONE	This command disables the generation of messages for the auxiliary input given by the 2 nd parameter. Default setting is NONE for all inputs.
		SENSE	[NORMAL INVERT]	This command inverts the sense of the logic input. If set to NORMAL a logic 0 on the input is regarded as being the OFF state. If Set to INVERT, a Logic 1 on the input is regarded as being in the OFF state. Default is NORMAL
		SYSLOG	[ON OFF]	This command enables or disables the sending of a logic input change over SYSLOG. The default value is SYSLOGOFF
		MAIL	[ON OFF]	This command enables the sending of a logic input change over SMTP. The default value is MAILOFF
		MSG	"text"	This command replaces the default text message Aux_in Δ<input_number> with the text entered as Parameter 3. The text is delineated by Quote marks and is a maximum of 40 characters. The status <on/off> is appended to the end of the text. If the "text" value is blank, the default message is reinstated.
	OUTPUT[1-2]	GENERATE	OFF	This command sets the function of Volt Free Contact output 1 or 2 to be off. (Default State)
			MUTE +	This command sets the function of Volt Free Contact Output 1 or 2 to indicate the output should be enabled when the audio is muted by the user.
			PULSE +	This command sets the function of Volt Free Contact output 1 or 2 to indicate if the output should be pulsed for the period set by the associated timer.
			RING +	This command sets the function of Volt Free Contact 1 or 2 to be enabled when an incoming call is present and ringing , and the on /off period will be as defined by the LOGIC SET CADENCE commands
			CALL +	This command sets the function of Volt Free Contact 1 or 2 to be enabled when an outgoing call is present with an on /off period which is as defined by the LOGIC SET CADENCE commands
			CONNECT +	This command set the function of the Volt Free Contact outputs 1 or 2 to be enabled when a call is connected. If the ANI feature is enabled, the volt free contact output is enabled only when the call is connected and DTMF # has been received.
			HOOK +	This command sets the function of the Volt free contact 1 or 2 to show the status of the hook switch or ON/OFF/TOGGLE button states. When the unit is in the off hook state the output will be enabled. When the unit is in the on hook state the output will be disabled.
			INUSE +	This command sets the function of the volt free contact output 1 or 2 to be enabled when an incoming call arrives or when the user goes off hook for an outgoing call, and disabled when the call ends.
			RINGCADENCE +	Causes the relay to operate at the same time as the ring tone cadence.
			PAGE +	Sets the output on when a call is present that has been signalled as a PAGEMODE call (see UNIT page, section 5.5)
			REGISTERED +	Sets the output to pulse when the phone 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.
			ON	This command sets the function of the volt free contact output 1 or 2 to be enabled permanently.
TIMER	XX	This command sets the pulse timer value for Volt free contact output 1 or 2 to be XX. XX is specified in seconds and can have a value in the range 0 -3600. Default value is 3 seconds.		

		CADENCE	ON:OFF	This command sets the cadence for a contact output. To be on for ON/10 seconds where ON is in tenths of seconds, and then OFF for OFF/10 seconds. The default value for ON is 10, and OFF is 0 implying the contact does not "flash". Values in the range 0-6000.
SHOW	ALL			This command outputs the current settings of the Logic Module

9.14 SIP Module Command Line Syntax

Action Verb	Parameters			Comment
	1	2	3	
SET	LOCALPORT		XXXX	Configures the port number used for the local SIP signalling socket to be XXXX. The default value is 5060.
	RTPTOS		XX	This command sets the value of the TOS/Diffserv field in the UDP packets carrying RTP data. Valid values are 1->63 (Default value = 46)
	REGTIMEOUT		xxxxx	This command sets the Registration timeout value that will be suggested by the telephone to a Registrar to be XXXXX Seconds .
	REREGTIMEOUT		XXXXX	Sets a period in seconds after which the phone will force a re-registration period and the server cannot override it.
	PROXYFAILOVERSTATUS		XXX, XXX ...	Contains a list of SIP status codes that will trigger a fail over from one proxy to the next.
	DONTSTARTMEDIAATTRING		[ON OFF]	Delay the sending of media packets to end points until the call has been answered.
	SENDDTMFLAST		[ON OFF]	Reorder the codec sequence to end points, so that the DTMF codec is sent last.
	SINGLEPTIME		[0-63]	Forces a single packet time to the value set in ms. See 5.4.
	SENDMULTIPARTMIME		[ON OFF]	This option is for future enhancement and should always be set to 'OFF'. Default value OFF.
	NEWBRANCHONAUTHBYE		[ON OFF]	This is a legacy option that is no longer used, and must always be set to 'ON'. Default value 'ON'.
	MODE		[SERIAL MULTIPLE]	Sets whether multiple proxies and registrars are used serially or concurrently.
	LOCALID	[1-4]	Identity	Sets the identity of the user/phone that will be used in the registration process with a registrar.
	DOMAIN	[1-4]	FQDN minus host name	Sets the domain name to be a FQDN without the leading host part.
	PROXY	[1-4]	xxx.xxx.xxx.xx x or FQDN	Sets the IP address or the FQDN of the SIP proxy server to be used for incoming/outgoing calls.
	PROXYPORT	[1-4]	XXXX	The port number on the proxy used for SIP protocol signalling can be changed to XXXX using this command. The default value is 5060
	PRIORITY	[1-4]	[1 – 4]	Sets the failover sequence between the 4 proxies
	REGISTRAR	[1-4]	xxx.xxx.xxx.xx x or FQDN	This command sets the address of the Registrar to be xxx.xxx.xxx.xxx or the FQDN specified. The registrar address and the proxy may or may not be the same address , but the address for registration must be set here.
	REGISTRARPORT	[1-4]	XXXX	Port number to send the requests to. Is 5060 by default or if unspecified.
USERNAME	[1-4]	"username"	This command sets the username for the authorisation realm to be username (Default value is blank)	
PASSWORD	[1-4]	"password"	This command sets the password for the authorisation realm to be password (Default value is blank)	
ENDPOINT	[1-4]	[ENABLED DISABLED]	Sets whether this set of proxy parameters is active.	
SHOW	ALL			

9.15 SMTP Module Command Line Syntax

Action Verb	Parameters			Comment
	1	2	3	
SET	SERVER1 or SERVER2		xxx.xxx.xxx.xxx or FQDN	Sets the IP address of the primary SMTP server to be xxx.xxx.xxx.xxx or uses the FQDN to resolve the IP address through DNS. E-mail will be sent on assertion of an alarm condition via the primary server if configured.
			xxx.xxx.xxx.xxx or FQDN	Sets the IP address of the secondary SMTP server to be xxx.xxx.xxx.xxx or uses the FQDN to resolve the IP address through DNS. E-mail will be sent on assertion of an alarm condition via the secondary server if configured.
	TOADDRESS			Sets the To: Address
	CCADDRESS		abc@hij.com	Sets the CC: Address
	FROMADDRESS			Sets the FROM: Address
	SUBJECT		"SubjectText"	Set the contents of the subject field to be "SubjectText". The Subject Text field can be up to 50 characters in length, and can contain any printable character except double quotes.
	OFF			This command disables the sending of SMTP alerts
ON			This command enables the sending of SMTP alerts if the above server settings and addresses are configured	
SHOW	ALL			

9.16 Status Module Command Line Syntax

Action Verb	Parameters			Comment
	1	2	3	
SHOW	ALL			This command shows the current call status, the state of all auxiliary inputs and logic outputs, and the registration status of the 4 SIP proxies and registrars

9.17 UNIT Module Command Line Syntax

Action Verb	Parameters			Comment	
	1	2	3		
SET	NAME	XXXXXXXXXX		Sets the unit host name for this unit to be XXXXXXXXXXXX where X is any alphanumeric character. Maximum of 15 characters. Default Value is "UNNAMED"	
	UPDATE	SERVER	xxx.xxx.xxx.xxx or FQDN		Sets the IP address of the host running the TFTP server to be xxx.xxx.xxx.xxx or resolves the FQDN using DNS to access the host containing the update file(s).
		FILE	Filename		The name of the file on the update server.
		INTERVAL	X		This command forces the unit to attempt a configuration file download every X hours.
	HELPSERVER		xxx.xxx.xxx.xxx or FQDN		This command is used to set the default address for the Help web page server.
	LAN	SPEED	[AUTO 10 100]		This command is used to set the speed or auto negotiation status for the WAN Ethernet port. The default value is AUTO for auto negotiation. If the speed is auto negotiated the duplex setting has no effect.
		DUPLEX	[FULL HALF]		This command sets the duplex value for the WAN Ethernet port. The default value is Full duplex
	CONFIGID	"IDSTRING"		This is used by the configuration upgrade script to determine if the local configuration is the same as the one it wants to upgrade to.	
	ANI	"Identity"		Used as an identifying token to GAI-Tronics CMA Call Management Application.	
	DEFAULT_ANS_MODE:	[RING PICK-UP PAGE STEALTH]		Default behaviour determining how an incoming call is answered. See section 5.5. Default value RING.	
	ANSMODE1	"password"		Set "passwords" that can be used to activate 3 special auto-answer modes, usually for hands-free telephone types. See web page section for details.	
	ANSMODE2	"password"			
	PAGEMODE	"password"			
	APTENABLE	[ON OFF]		APT parameters. See section 5.5.1	
	APPTIME	Xx:xx,xx			
	APTCOUNT	1-10			
	APTOKCOUNT	1-10			
APTREPORT	[ON OFF]				
REBOOT	NOW		Resets the telephone.		
UPDATE	NOW		Fetches updates immediately.		
SHOW	ALL		Lists all the current settings and information for the Unit module. (This includes unit name, Mac Address and Firmware version.		

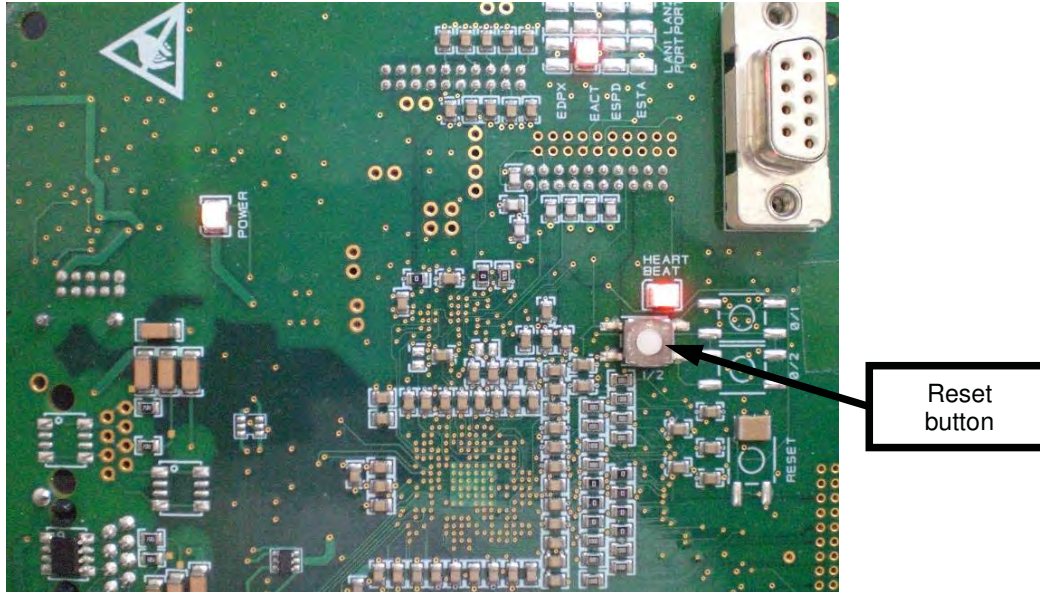
9.18 Multicast Module Command Line Syntax

Action Verb	Parameters			Comment
	1	2	3	
SET	TIMEOUT		[1-120]	sets an enforced delay (in seconds) between one Multicast session ending and another beginning. Range 1-120. Default 120
	SPEAKERVOLUME		1-10	sets the speaker volume during a multicast. Volume will revert to the setting on the AUDIO page when the multicast session has ended. Range is 1 to 10, default value 3.
	OVERRIDE		0-8	Priority level. See section 5.18.
	ADDRESS		xxx.xxx.xxx.xxx:pppp	Multicast address, must include multicast port.
	FILTER		xxx.xxx.xxx.xxx:xxx.xxx.xxx.xxx	Sets a range of acceptable multicast source IP addresses. The phone will only accept a multicast if the source is within this IP address range. The format is 2 IP addresses separated by a colon. The default is 0.0.0.0:255.255.255.255
	PRIORITY	[1-8]	[0-8]	Priority level. See section 5.18.
	OUTPUT1		[ENABLED DISABLED]	sets if OUTPUT1 is ENABLED or DISABLED during this multicast
	OUTPUT2			sets if OUTPUT2 is ENABLED or DISABLED during this multicast
TONE		sets if tone 10 (TONES page) is ENABLED or DISABLED during this multicast.		
SHOW	ALL			

10. Reset to Defaults

If the user name and / or password have been lost, the unit can be reset to allow access to its configuration settings.

To reset to defaults, locate the reset button on the main PCB, next to the HEARTBEAT LED:



NOTE: on Titan and handsfree telephones the main PCB is covered by a plastic backbox secured by 4 screws. Carefully remove the screws and backbox to expose the circuit boards underneath.

With the unit powered up (heartbeat flashing), hold the reset button in for 5 seconds, then release.

The unit will reboot and all configuration settings will revert to factory defaults.

Note all configuration settings will be lost.

The unit will have:

User name : "user"

Password : "password"

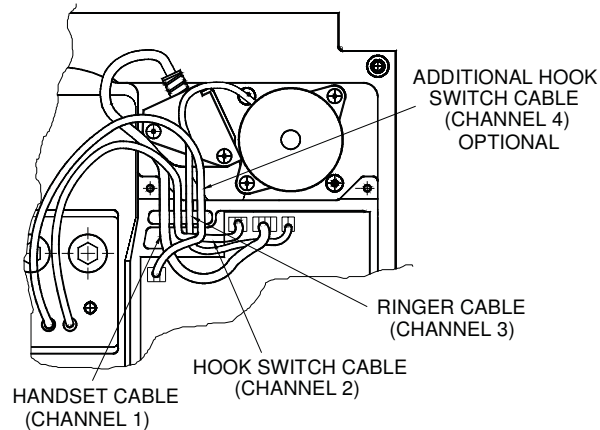
IP address ; 192.168.1.2 (static)

Net mask : 255.255.255.0

NOTE: THIS OPTION IS ONLY AVAILABLE ON UNITS FACTORY CONFIGURED WITH FIRMWARE VERSION 3 AND ABOVE.

Units upgraded to v3 in the field will normally not have this function enabled. Units originally shipped with v1 or v2 must be returned to factory to have this function enabled.

For Titan and handsfree telephones, carefully replace the backbox replacing the 4 screws. On Titan models, when replacing the backbox, take care to seat the cables correctly in their channels as shown.



11. Troubleshooting

This 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.

11.1 Is the unit powered up?

Look for 2 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 40s after power is applied.

Note on some models (for example Titan and VR / Help Point) the circuit board is covered by a plastic cover. It is still possible to see whether the LEDs are operating through the aperture for the serial port.

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.

NOTE: Auteldac 4 VoIP telephones require PoE alternative B (ie spare pair only). PoE LAN switches do not commonly use this alternative (using data pair method instead) whereas PoE injectors commonly do.

11.2 I can't access the web pages

If the unit is correctly powered up, but you cannot browse to its webpages over the network, you will usually need to make a serial connection to the unit (see section 9) and check the following using the Command Line Interface:

- Are the IP and UNIT settings correct?
- Is the web server enabled?
- Can the phone 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.

11.3 I can't make calls

If the unit can ping (and be pinged by) its intended call destination, call connection problems are usually due to proxy or registration issues.

- Check that the SIP server is listed on the GAI-Tronics website as having been tested with GAI-Tronics phones.
- Check that the proxy settings are correct and that both end points are properly registered.
- If the SIP server requires authentication, check that the user names and passwords match between the phone and the server.
- 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.

Note that GAI-Tronics VoIP units are SIP only - calls will not connect using H.323, SCCP or other VoIP call connection protocols.

11.4 Calls connect but there is no speech (or sound is garbled)

Audio problems are usually due to codec issues.

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.

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.

Ensure that the FRAMES and SAMPLE values are set correctly according to instructions in section 5.13. Setting these values wrongly can cause speech to jitter.

12. Licensing Notices

The firmware in GAI-Tronics VoIP products contains modules subject to licensing and copyright as follows:

Module	License
u-boot	GPL V2
Linux kernel	GPL V2
Busybox	GPL V2
Opal/PWLib	Mozilla Public License V1.1
Modutils	GPL V2
MTD	GPL V2
NTP	David L. Mills Copyright Notice

These licence and copyright notices are available in full from our website at www.gai-tronics.co.uk/voipsupport.htm