GAI-TRONICS
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Application Note GAI-ENG-583

Adding a GAI-Tronics SIP phone (1193 model) to Cisco Call Manager 5.0 and above

These instructions show settings for both a GAI-Tronics VoIP telephone and a Cisco Call Manager to enable them to work together.
Note that these instructions are not intended as an exhaustive guide to setting up Cisco Call Manager, merely as an aid to adding a GAI-Tronics SIP phone. This is only one example of the parameter settings required; many of them can be modified if desired. For more information refer to the manuals for Cisco Call Manager and the GAI-Tronics SIP phone (GAI-Tronics manuals documents reference GAI-ENG-568).

Configuring the phone

Make the following changes to the phone's parameters using its web pages (or a configuration file or via SNMP if more convenient).
The phone is shipped with DHCP enabled. If no DHCP server is found within 60 seconds the phone will revert to a default IP address of 192.168.1.2 – consult the manual for more information.

SYSTEM – Date/Time

phone timeserver enable	enable
phone timeserver domain name	The address of the time server

SYSTEM – Network Configuration

net static IP address	IP address for the VoIP phone
net static netmask	Subnet mask for the VoIP phone
net isp dhcp enable	enable/disable

VoIP ACCOUNTS – VoIP Account <*n*> *Information*

provider enable	enable
user name	Directory Number for phone
domain name	IP address of CUCM
auth user password	Password for the CUCM end user
proxy domain name	IP address of CUCM
registrar enable	enable
registrar domain name	IP address of CUCM

Using the 'Advanced' web page, you may also want to set

VoIP PARAMETERS – Audio Settings

voip preferred audio codecs
voip DTMF transmit method

Configuring Call Manager

Add the directory number

Check that the Directory number you want to use for the phone exists &, if necessary, add it by going to Call Routing -> Directory Number->Add New.

Add the End User

Add an end user to the system with the same name as the directory number to be used
User Management -> End User->Add New
Set :-

USER I.D.	Directory Number (SIP/Registrar USERNAME on the phone)
Password	the SIP/Registrar password set on the phone

You will also need to set (& confirm) a 5 or more digit PIN, and set a 'Last Name' for the user

Add the phone to the system

Add the phone to the system by going to Device -> phone-> add new
Set the phone type to 'Third-party SIP Device (Basic)', then click on 'next'.

Set:

MAC address	MAC address of the phone you are adding
Description	could be the DN, or the phone location, or a combination
Device Pool	Default
Phone button Template	Third party SIP device (Basic)
Owner User I.D.	Select the D.N. (end user added above)
SIP phone security profile	Standard SIP profile for auto registration
SIP profile	Standard SIP Profile
Digest User	Select the D.N.

Note: In addition to "Digest User", there may be another drop down list containing the same values on the page. Double check that you have set the "Digest User" before you click save.

Click on the 'Media Termination Point required' check box to enable the M.T.P. if required.
Note: In some installations, MTP may be required to enable some features (such as call transfer) on certain system configurations.

Values also need to be set for 'Common Phone Profile', 'Location', and 'Presence Group'; normally the default values are correct.
Click on 'save'

When the screen has refreshed, click on the link 'Line[1] – Add a new DN'

Set: -

Directory Number	Enter the D.N.
Max number of calls	1
Busy Trigger	1

The 'Alerting Name' is displayed on another phone when it makes a call to this phone.
The 'Display (internal caller ID)' string is displayed on another phone when it receives a call from this phone.
The other default values are normally correct.

Click on 'Save'
When the screen has refreshed, click on 'Reset' and reset or restart the phone.

It may take a few minutes for the new settings to take effect.