

Application Note 502-20-0121-001

Adding a GAI-Tronics SIP phone to Cisco Call Manager 5.0 or 6.0

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 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 502-20-0115-001 & 502-20-0119-001).

Configuring the phone

Make the following changes to the phone's parameters using its web pages (or serial port or a configuration file if more convenient).

The default IP address of the phone is 192.168.1.2 – consult the manual for more information.

N.B. To avoid unnecessary network activity you should clear the SYSLOG & SYSLOG2 addresses, & turn off the Email messages unless these are required.

Audio settings

CODEC 4, 2, 1
Frames G.711 20
Frames G.729 2
DTMF RFC2833

Clock

SNTP the address of the time server

Unit

Hostname Directory Number (DN) to be used for the phone
ANI 'from' display name of phone

SIP

LOCALID Directory Number for phone
DOMAIN IP address of CCM
PROXY IP address of CCM
REGISTRAR IP address of CCM
USERNAME Directory Number for VoIP phone (CCM End User)
PASSWORD password for the CCM end user

IP

DHCP ON/OFF
ADDRESS IP Address of the VoIP phone (Not required if DHCP is being used)
MASK Subnet Mask of the VoIP phone (Not required if DHCP is being used)



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.

