CONFIGURING THE GRANDSTREAM GXV3000 IP VIDEO PHONE
WITH ASTERISK™

How do I configure Asterisk and the GXV3000?

PLEASE NOTE: for the GXV3000 to work in an Asterisk™ environment, video support must be on and the related video codec must be turned on in Asterisk™. Please be sure your sip.conf file contains the following line in the [general] section:

```plaintext
videosupport = yes
```

CONFIGURING THE GXV3000

First, configure the GXV3000 as you would for a traditional SIP-based application using either the DHCP server or a static IP. To access the Web Configuration Menu type `http://Phone-IP-Address` on your web browser - where the Phone-IP-Address is the IP address of the phone displayed on the LCD screen. The GXV-3000 operates in two modes:

- **DHCP mode**: all the field values for the Static IP mode are not used (even though they are still saved in the Flash memory.) The GXV-3000 acquires its IP address from the first DHCP server it discovers on its LAN. The DHCP option is reserved for NAT router mode.
  - To use the PPPoE feature, set the PPPoE account settings. The GXV-3000 establishes a PPPoE session if any of the PPPoE fields is set.

- **Static IP mode**: configure all of the following fields: IP address, Subnet Mask, Default Router IP address, DNS Server 1 (primary), DNS Server 2 (secondary). These fields are set to zero by default.

After you’ve set-up your GXV3000, you are now ready to configure it for Asterisk. Using the GUI interface, pull up the Grandstream Device Configuration GUI and enter your password “admin”. Click on the Account 1 tab where you configure the SIP server.
**CONFIGURING THE GXV3000 IN ASTERISK™**

The following parameters need to be updated:

- **Account Active** Yes
- **Account Name** name of the account
- **SIP Server** Asterisk IP address
- **Outbound Proxy** Asterisk IP address
- **SIP User ID** SIP user ID, commonly the extension number
- **Authenticate ID** re-enter SIP user ID
- **Authentication Password** password for the SIP account, configured in Asterisk as the account’s ‘secret’
- **Name** name on the display (not the asterisk configuration name)
- **Preferred VoCoder** this parameter determines the voice codecs. By default, PCMU (ulaw) is “Choice 1”
- **Preferred Video Codec** this parameter determines which video codec is used. If you are using video codec H.264, make sure you have Asterisk 1.4.x or later; if you are using H.263, make sure you have Asterisk 1.2.x or later and a GXV-3000 with f/w 1.0.1.7 or later.
- **Enable Video** Yes

**Asterisk sip.conf configuration:**

```
[709]
type = friend
username = 709
secret = password
context = default
dtmfmode = rfc2833
host = dynamic
qualify = yes
allow = h263 (and/or ‘allow = h264’ for Asterisk 1.4.x)
```