GATE 102 (Digidave) - 1 Port VoIP Phone Adapter with Bu...

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GATE 102 - 1 Port VoIP Phone Adapter with Built in Router

This device allows you to use a standard analogue telephone for VOIP (Voice over IP) and connect a computer, hub or switch to share your internet connection. This device also allows you to connect an analogue telephone (PSTN) line to use as a backup phone line. You can select on a per call basis whether to use the VOIP line or the analogue (PSTN) line for making a call.
The adaptor has the following connectors.
LINE - To connect your adaptor to a normal analogue phone (PSTN) line.
PHONE - To connect a regular telephone.
LAN - To connect to your PC or other peripherals.
WAN - To connect to your ISP's modem.
POWER - To power the device.

An example setup for this device

Please refer to the user manual for full details on connecting your phone and configuring the router.

[Link to manual (external PDF)]

IMPORTANT: Always confirm you are connected to the Internet before trying to setup your phone. An easy way to do this is visit a public web page from a PC connected to LAN connector on your Adaptor (like http://www.freespeech.co.uk).

Logging on

To log on to the Adaptor you will need to connect a PC to the LAN interface. Connect to the web configuration page by typing http://192.168.2.1 (default url) into your browser navigation bar.

Default password:voip

Configuring Freespeech account

IMPORTANT: The Gate 102 has a built in router, it is recommended you do not cascade it behind another router (ie. do not connect the WAN port into a router). This will ensure you have minimal call connection problems and get the best quality for your calls.
After logging in to your adaptor select SUPER OPTIONS > SIP Settings from the menu. Fill in the following details where 0844XXXXXX is your freespeech username/phone number. Make sure you click "SaveSet" at the bottom of the page after changes.

In SUPER OPTIONS > Audio Settings - set your preferred codec to G729, this is the best compromise between speech quality and low bandwidth.

Now REBOOT

Check you are online

Problem solving
If you have problems you may be behind a NAT (router) without knowing it. As we said above it's best not to connect this device behind a router. If you have problems getting the device online, incoming calls or silent calls you can try activating STUN in SUPER OPTIONS > SIP Settings as follows.