Gate 300 User Manual

Key Features

1.Built-in Modem function:

- The built-in V.90 modem function let Gate 300 dial-up to any ISP and connect to Internet easily.
- The WAN Port can also be used to connect to Internet by using DHCP, static IP or PPPoE configuration flexibly.

2.Following H.323 version 4 standard or RFC-3261 SIP standard :

- Support Fast start, Tunneling and H.245 DTMF relay.
- Support password authentication using MD5 digest and RFC-2833 for DTMF relay.

3.Dynamic IP support(DHCP and PPPoE) :

- Getting IP from DHCP server using DHCP protocol or through ADSL modem using PPPoE protocol, automatically reconnect when PPPoE lost connection.

4.Passing through NAT devices :

- Can make outgoing and incoming calls under any NAT devices(even under two layer NAT devices) when working with the specific gatekeeper/proxy devices.

5.Remote software upgrade capability (via ftp) :

- FTP protocol provides reliable remote upgrade through Internet.

6.Advanced Digital Signal Processing technology to ensure superior audio quality :

7.Support G.723.1, G.729A/B, G.711(A-law/U-law) voice codecs :

- Following ITU-T standard to support best compatibility.

8.Support supplementary services, including immediate (unconditional) call forwarding, busy call forwarding , no answer call forwarding and call hold/transferring.

9.Support Silence Suppression, VAD (Voice Activity Detection), CNG (Comfort Noise Generation) :

- Silence suppression can save about half of the network bandwidth needed during normal VoIP conversation.

10. Ping function supported :

- Ping other device in the Internet from EG-202/EMG-202 to make sure the Internet connection is ok.

11.System status display on the LCD panel :

- User can easily know if the EG-202/EMG-202 is working normally and monitor the system status(network status, registering status) from the LCD panel display.

12.Call with or without gatekeeper/proxy(direct IP dialing) :

- Following standard SIP/H.323 protocol and is compatible with most of existing SIP proxy/H.323 gatekeeper.

13.Provide easy configuration methods :

- Very easy setting by keypads on the connected phone set.
- Setting by web browser.

14.Support RFC-3261, H.323, TCP/UDP/IP, RTP/RTCP, HTTP, ICMP, ARP, DNS, DHCP, NTP/SNTP, FTP, PPP, PPPoE protocols.

15.Interoperable with most of the existing SIP/H.323 VoIP devices(IP-phone, gateway, gatekeeper, proxy, softswitch, IP-PBX), including Microsoft Netmeeting, Cisco gateways/gatekeepers :

- Please refer to the section 6.2/6.3 Interoperability List for the complete listing.

16.The WAN Port automatically works for paralleled Ethernet cable and crossed Ethernet cable.

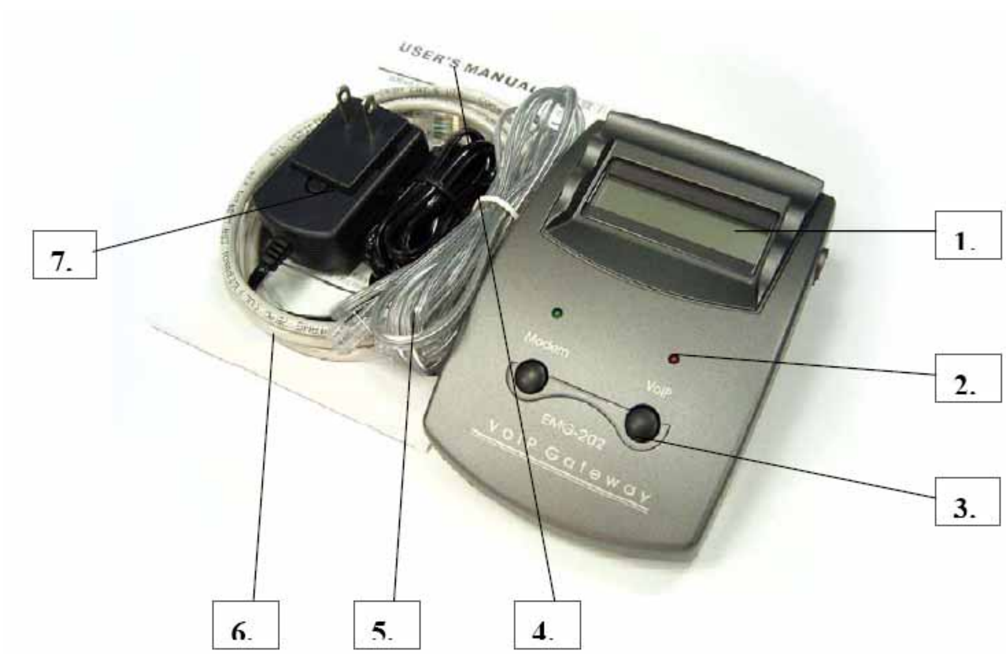
* **Hardware Specification**

HARDWARE SPEC	
Spec\Model	Gate 300
Phone Port	1xRJ11 connect to a telephone set. This telephone set can make PSTN call through the line port or VoIP call through Ethernet or modem connection
Line Port	1xRJ11 connect to the PSTN line. Built-in V.90 modem function. Work between PSTN call and modem function.
WAN Port	1xRJ45 10/100 Base-T Ethernet, line auto-sensing/switching.
LCD display	2x16 characters
Case	2-button keypad
Universal Switching Power Adaptor	Input: 100-240V AC
	Output: +12V DC, 350mA
Dimension	146 mm(W)x80mm(D)x32mm(H)
Weight	230 g
Operating Temperature	32 - 104°F (0 – 40°C)
Humidity	10%-95% (non-condensing)
EMI Compliance	UL/EN/FCC Class B

Basic Installation

Figure 1. Key parts of Gate 300 VoIP Gateway

1. 2x16 LCD
2. Led
3. Button
4. User Manual
5. Phone line for Gate 300
6. Ethernet Cable
7. Power Adaptor



1. RJ-45 Ethernet Port
2. RJ-11 Phone Port, connect to a phone set.
3. RJ-11 Line Port, connect to PSTN line

1. Power Adaptor

Basic Installation:

- Plug in the power supply.
- Install the Gate 300:

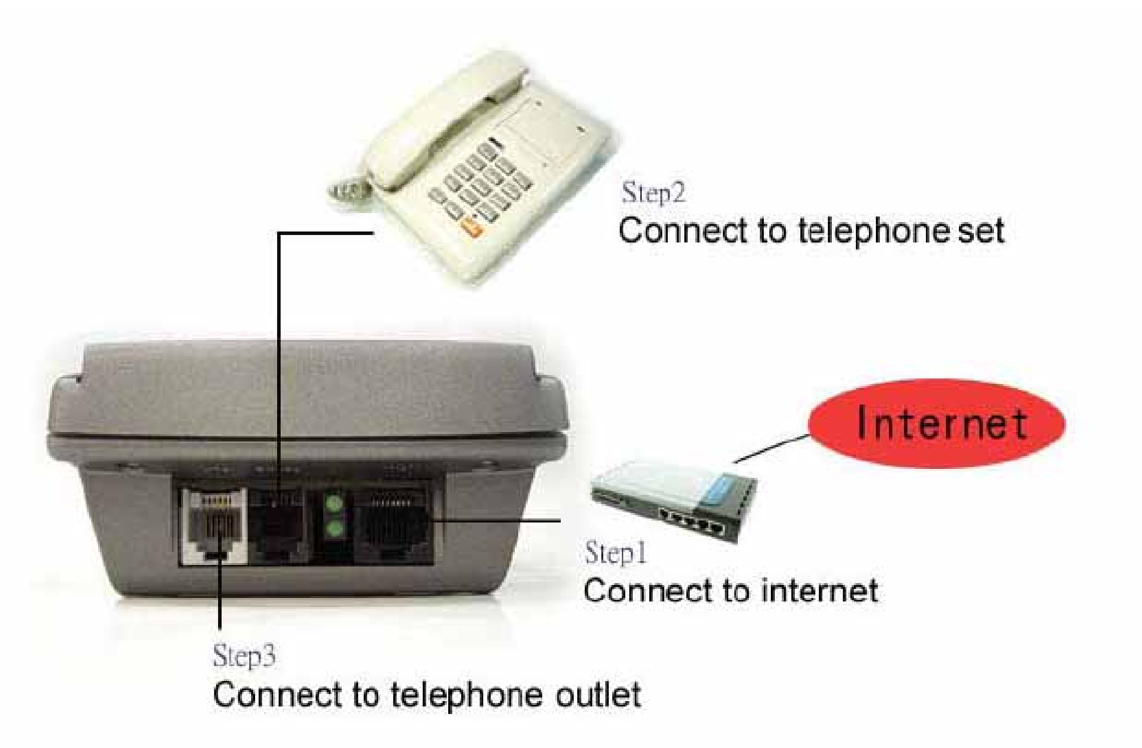
Step 1：Please take out your VoIP Gateway and its accesories. If you use RJ-45 broadband connection, please connect your network cable from your broadband device(such as NAT、HUB、ADSL modem、CABLE modem) to WAN port of your VoIP Gateway.

Step 2：

Then connect a phone set to the “phone” port of the VoIP Gateway.

Step 3：

Please connect the “line” port of your VoIP Gateway to the telephone line.



***On-Device Button and Led definition**

There are two buttons on the upper of the gateway, the right side one is “VoIP” button, the left side one is “Modem/Info” button, explained below :

A. “VoIP” button and led:

The button is pressed to switch the device from making a VoIP call to a PSTN call alternatively. When the device is on VoIP call mode, the led is on, all the call made from the attached phone set will be VoIP call going through the Ethernet port. When the device is on PSTN call mode or when the device is powered off, the led is off, all the call made from the attached phone set will be PSTN call going through the line port.

When there is an incoming call either from PSTN line or VoIP side, whether the device is on “VoIP” or PSTN mode, the device will automatically switch to the correct mode to ring the attached phone set, so the user will not miss any call. After call disconnected, the device will automatically switch back to the previous mode.

B. “Modem/Info” button and led:

This button is pressed to make the built-in modem dial-up to PSTN line or hang-up the built-in modem connection. When the modem connection is connected, the upper led is on. Pressing this button again will hang-up the modem connection.

C. Switch between VoIP and PSTN :

The “VoIP” button on the right side of the gateway case can be pressed to toggle between VoIP and PSTN mode. When in VoIP mode(led is on), all the outgoing call will go to the VoIP network. When in PSTN mode(led is on), all the outgoing call will go to the PSTN network.

D. Incoming call :

There are two kinds of incoming call, one is from PSTN network, another is from VoIP network. For any kind of incoming call, unless the phone set is already busy, otherwise, the gateway will automatically switch to the correct PSTN or VoIP mode to ring the phone set. After the call is disconnected, the mode will go back to the original mode.

E. Outgoing call :

When a user want to make a call, it can choose to make a VoIP call or PSTN call by pressing the “VoIP” button to toggle between them. There is another method to toggle from VoIP mode to PSTN mode -- when in VoIP mode, and after the phone is off-hooked to hear the dial tone, the user can now press ‘*’ key on the phone set to switch to PSTN mode.

3 Configuration with keypad

The Gate 300 can be directly configured through the keypad on the attached phone set and on-device LCD panel. Please follow the steps below to do the configuration :

A. Make sure the device is on "VoIP" mode("VoIP" led is on).

B. Off-hook the phone set(or press the "spk"), you should hear the dial tone.

C. Press '#' then '8' on the phone set, the device is now entered into the configuration mode by displaying the menu items on the LCD display.

D. You can then use the combination key

1. '#'+'7' for left/up :
   When the VoIP Gateway is entered into the menu selection, this key is used to scroll up the menu items.
   And when the VoIP Gateway is editing some menu item's contents, this key is used as " left delete" to delete a digit per each key press.
2. '#'+'8' for ok/menu
   When inside the menu selection/setting on the LCD display, this key is used as "OK" key to enter into a lower layer of menu selection or to accept the edited item's contents.
3. '#'+'9' for down :
   When the VoIP Gateway is entered into the menu selection, this key is used to scroll down the menu items.
4. '#'+'0' for Cancel :
   When the VoIP Gateway is entered into the menu selection, this key is used to escape to an upper layer of the menu selection.
   And when the VoIP Gateway is editing some menu item's contents, this key is used to cancel current edit and escape to an upper layer of the menu selection.

E. For normal keypad operation of the phone set connected to EG-202/EMG-202 gateway, it's described below:

- "#" Key
  After phone number is completed during phone dial, pressing the '#" key will force the device to call the dialed number immediately.
- "*" Key
  Work as symbol "." (dot) when input IP address. For EMG-202, Work as "stop for one second" when input modem dial number.
- Digits "0~9" Keys
  These keys are used to dial the call out number when the handset is off-hooked. But when in the menu item editing mode, these keys can be used to toggle between digits and all English characters by quickly re-pressing the same key.

Notice :
When need to input an English character in any menu item, please press that key button quickly to switch between different characters to set the correct one needed.

F. Network Configuration

Configure ► Password：135 ► Network ► (Yes/No) Modem On

Please press ‘#’+’8’(“ok”) button to select Yes on (Yes/No)
Modem On when using dialup method.

The followings are some other parameters in the menu items needed to be set when ModemOn is enabled:

(a). Dial Number – please input the telephone number of the account given by internet service provider(ISP) to dial up.

If want to wait one second in the dial number, please key in ‘.’(‘*’) inside the number. This happened in the case when the modem line is inside a PABX , need to put “.” (dot) in order to receive a second dial tone. For example, set dial number 0.xxxxxxxx, 0 is the outline access digit, then ‘.’ will wait one second to get the second dial tone to dial out.

(b). Dial Username – please input the user name of the account given by ISP.

(c). Dial Password – please input the password of the account given by ISP.

(d). Disc time – unit in minutes.

This parameter determines the minutes when the modem connection is connected but is idle. i.e. if this value is 5, then after 5 minutes, the modem will disconnect if no call is made . If set as 0, modem will not disconnect for any idle duration.

G. Registration to a proxy server (SIP system)

When the VoIP vendor/operator is running the SIP system, use Gate 300 to register to the proxy server. Configure the following parameters to do the registration.

Configure ► Password：135 ► SIP

Please press ‘#’+’8’(“ok”) button to enter into.

(a). Number – please input the phone number to register to the proxy server.

(b) Password – please input the password to register to the proxy server.
This password is carried in the SIP Proxy-Authorization field using MD5 digest method for authentication purpose. Not every proxy server needs this field, if not needed, keep it empty.
Gate 300 follows the standard RFC-2617 to do the authentication.

(c). (Yes/No) Proxy On – please select Yes and register to the proxy server.
Proxy server address – Input Proxy sever address
proxy server port - set the port of the proxy server, usually 5060
If the VoIP Gateway does not register to any proxy server, it still can call to other VoIP Gateway by calling the IP address directly.

(d). (Yes/No) Outbound Proxy – please set this item to Yes if the registration needs to pass through the Outbound Proxy server.

(e) SIP Domain Name – Please input SIP server address.

(f) If you need to input STUN server information, go to “Advanced”

Advanced ► Password：1230 ► SIP Advanced ► Protocal ► (Yes/No)STUN server

(Yes/No) STUN server – please select Yes and register to the STUN server.
STUN server address – Input Proxy sever address
STUN server port - set the port of the STUN server

Notice : Most of the proxy server now have the built-in ability to let VoIP Gateway pass through the NAT/router devices, and this pass-through function does not need the VoIP Gateway to change anything. This method is also more reliable and more easier and successful than the other NAT pass-through method like STUN or others. The Outbound Proxy method is just very like the Proxy server built-in NAT pass-through solution, except that the packets need to pass through the Outbound proxy server. And of course, this pass-through method does not need the VoIP Gateway to change anything.

H. After all the above parameters are set, users can press “Modem” key to dial up, the built-in modem will attempt the dialing 3 times automatically.
When ModemOn is enabled, if users try to make an outgoing call and the modem is not connected, the built-in modem will attempt to connect itself onto internet automatically.

I. After these, please plug in your power adaptor to your VoIP Gateway and power source. LCD of your VoIP Gateway will display “Starting…….” and then “SIP(xxxxx)” menu within approximately 4 seconds.

4 Configuration on Web

A. Make sure the device is on “VoIP” mode(“VoIP” led is on).

B. Off-hook the phone set(or press the “spk”), you should hear the dial tone.

C. Press ‘#’ then ‘8’ on the phone set, the device is now entered into the configuration mode by displaying the menu items on the LCD display.

Configure ► Password：135 ► Network ► (Yes/No) DynamicIP

Please press ‘#’+’8’(“ok”) button to select Yes on (Yes/No)

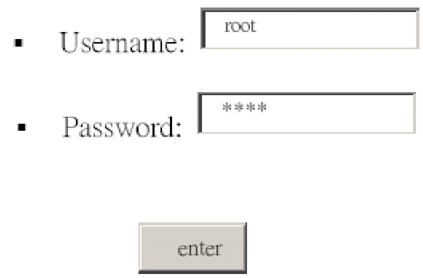
D. Press ‘#’ then ‘0”, Go back to “View” to restart the Gate 300 and get IP address:

View ► Restart

Please press '#'+'8'("ok") button

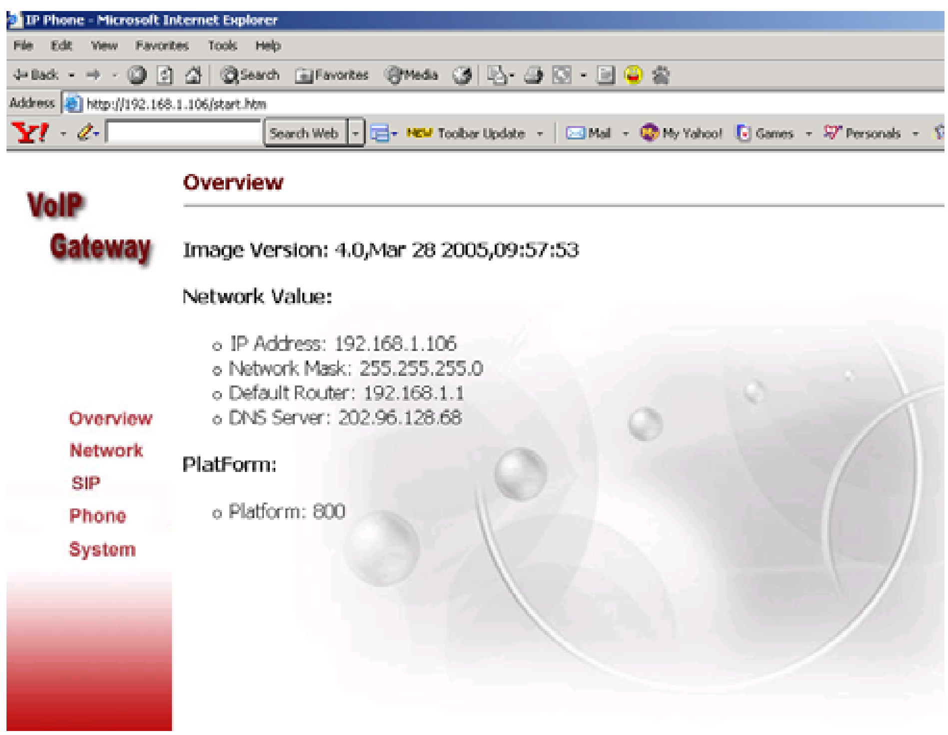
E. Push modem butten to get IP address:

F. Input the IP address into IE, and open the page.

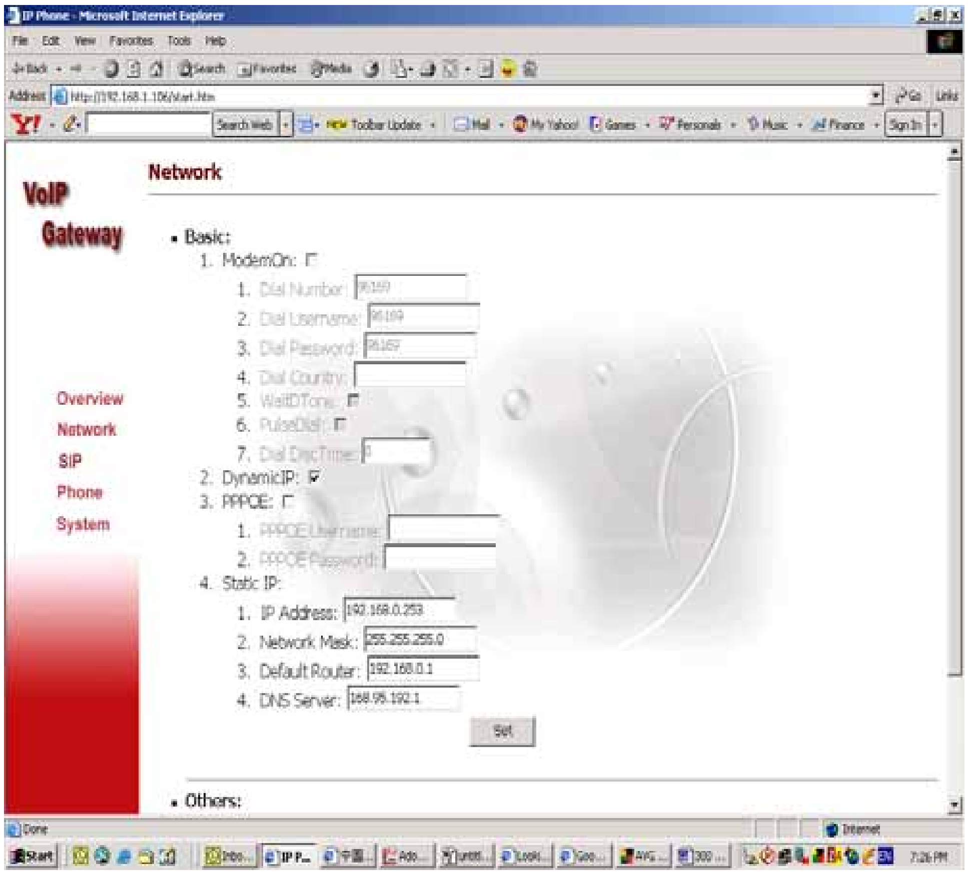


Username: root
Password: 1234

G After Enter you can see the following page. From this first page, you will get the basic information of the VoIP Gateway, also can set the configurations by selecting the specific pages



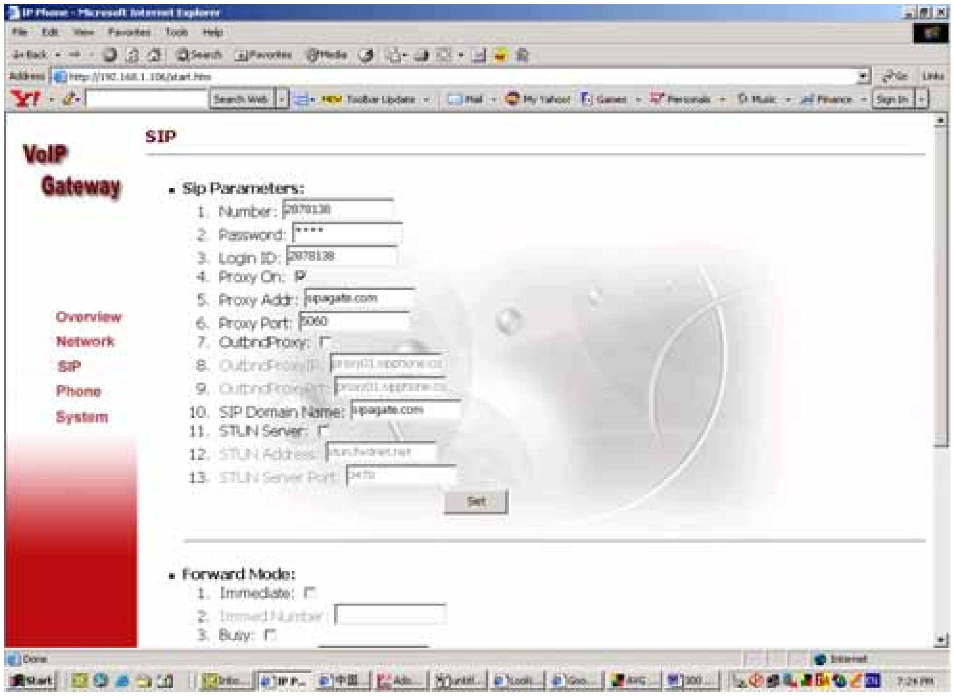
H. Select “ Network” to configure Network:



- Basic
  Please see Net configuration on Page 9 for the explanation of each fields.
- MAC Address –
  View the MAC Address of the VoIP Gateway. Not changeable here.
- NTP Server –
  Set the NTP(Network Time Protocol) server’s IP Address for the VoIP Gateway to get current date/time and display it on the LCD screen.
- Time Zone –
  Specify the time zone of your area, you could click the nearby “View” icon to see the time zone of your area. E.g. set the value to 8 for Taiwan area

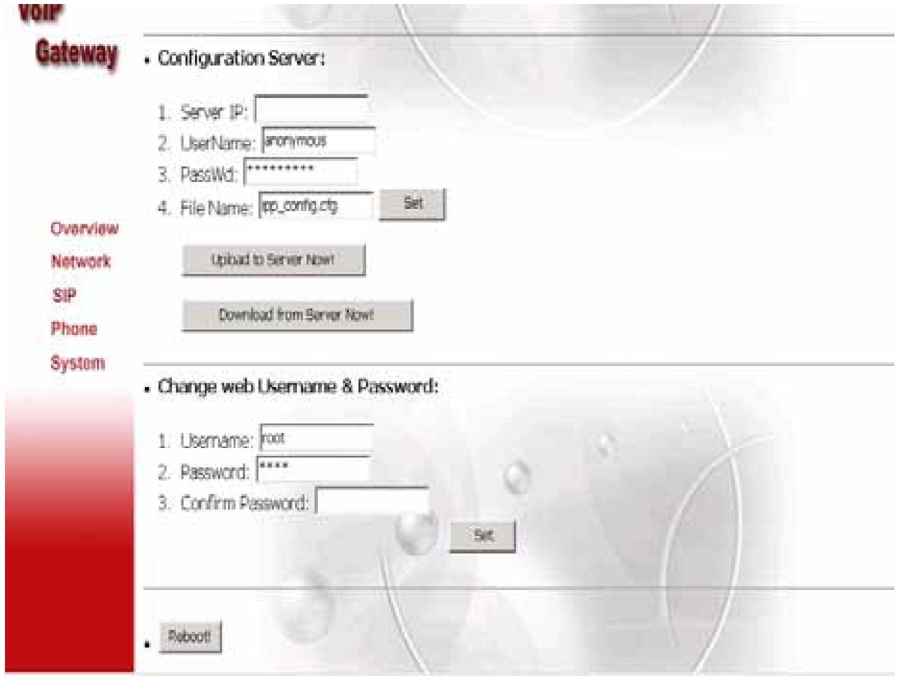
I. SIP Configuration

Input all the SIP information into the necessary fields.



J. Reboot

Select “system” page, go the bottom to “ reboot”, waiting for “ starting” on Gateway 300.



5 LCD Instructions

When the VoIP Gateway is registering to a SIP proxy server successfully, the LCD screen will display the following message :

(a)

SIP(number)
Date Time

Means that the Gate 300 is working OK and ready for outgoing/incoming calls. The number inside braces () is the VoIP Gateway's number.

(b)

SIP(Proxy Off)
Date Time

Means that the Gate 300 is working OK and ready for outgoing/incoming calls. But the "Proxy On" flag is "No", i.e. the VoIP Gateway not need to register. In this case, can call the IP address of other VoIP Gateway directly.

(c).

Registering(number)
Date Time

Means that the Cate300 is configured to use a gatekeeper or proxy server, but not yet registered successfully or failed. The VoIP Gateway can not make any calls now. But the menu selection and the onhook/offhook function can work OK.

(d).

RegFail(failed message) Date Time

Means that the VoIP Gateway is failed in registering to SIP proxy server, the LCD screen will display the following message :

The failed message could be one of the followings :

(1). Duplicate : means that the registering number is duplicated with others, or the VoIP Gateway's previous registration information is still kept in the gatekeeper/proxy server if not unregistered last time(this could happened when the VoIP Gateway is powered off instead of restarted from the menu item). If the previous registration information is not cleared, may need to wait about 4 minutes to let VoIP Gateway register successfully again.

(2). Security : means that the account (username/password/SIP ID) is not correct, please check your account again.

When the RegFail message displayed, the VoIP Gateway can not make any calls now. But the menu selection and the onhook/offhook function can work OK.

Users can disconnect the modem connection at any time by pressing “Modem” key again when the “Modem” led is on.