KVB-311 Broadband VoIP Router

User's Manual

Version 101a.

Preface

➢ About this User's Manual

This user's guide includes specifications, installation guide, web management and command line configuration interface for the KVB-311 VoIP Gateway.

➢ *Revision History:*

Version	***Date***	***Author***	***Modified Contents***
1.0	Sep., 17th, 2004	Sabrina	*1st Revision for KVB-311 SIP Gateway.*
1.1	Dec., 4th, 2004	Eason	*1. Add "Prefix" function in CLI and Web Management Interface.* *2. Modify the PSTN line application.* *3. Modify "call id" function.* *4. Add the description of Installing from Installation Wizard.* *5. Modify the description of Forward function*
1.2	Feb, 2005	Eason	*Revision for KVB-311 SIP Gateway for Rom Application version 101a.*

Part I: VoIP Gateway Overview

This part introduces the software/hardware specifications and default settings of the Gateway.

1.1 Overview

The KVB-311 is a one-port telephone extension and three ports SOHO Router to IP network gateway. It provides Data transfer by 10/100Mbps, telephone services and T.38 fax over IP network with easily operation and configuration. It is most suitable for SOHO and small-to-medium enterprise in Internet communication environment.

The KVB-311 provides IP telephone number for end users with FreeTalk voice service. User can make phone call via Internet now. No more long distance and international telephony fee! It also connects three computers without another IP sharing as showed as following diagram.

The KVB-311 provides two telephone numbers that one is IP telephone number and the other is PSTN telephone number in one device for end users. You can make phone call via Internet or PSTN in one telephone set now. No more long distance and international telephony fee! Especially, User still can make phone call when external power is failure.

The KVB-311 also can connect three computers with embedded IP sharing and DHCP server function.

1.2 Software Specifications

➢ **KVB-311 Gateway Features**

- Provide Voice over IP and Fax over IP features.
- SIP RFC 3261 compliance
- Built-in NAT/IP sharing function
- Provided call features: Hold, forward and transfer
- Automatic FAX detection (Support T.38 protocol)
- Codec: G.711 a/μ law, G.723.1, G.729A
- PPPoE connection
- VAD (Voice Activity Detection), CNG (Comfort Noise Generate)
- G.168/165 echo cancellation
- FSK and DTMF Caller ID
- Provide both IP telephone number and PSTN telephone number in one device for end users
- PSTN backup: user still can make phone call when external power is failure

➢ **Audio feature**

- Codec: G.711 a/μ law, G.723.1 (6.3kbps), G.729A
- VAD (Voice Activity Detection)
- CNG (Comfort Noise Generation)
- G.168/165-compliant adaptive echo cancellation
- Dynamic Jitter Buffer
- Bad Frame Interpolation
- Voice/DTMF Gain Settings

➢ **System Monitoring**

- System status (WAN, LAN, TEL, Status, Power)

➢ **Remote Firmware Upgrade**

You can use FTP/TFTP to perform firmware upgrade for the VOIP Gateway from a remote location.

➢ **Security**

- Password protection for system management
- Built-in NAT function.

➢ **Certification**

- CE, FCC

1.3 Hardware Specifications

➢ **Chassis**

- 190mm(W) x 110mm(D) x 51.5mm(H)
- Weight (unit): 0.3 kg

➢ **Interface**

- Four 10/100 Base-T Ethernet RJ-45 ports (Auto LAN MDI/MDIX).
- Input AC 100V-240V, Output DC 12V.
- Two RJ11 Telephone Port.

➢ **Special Housing**

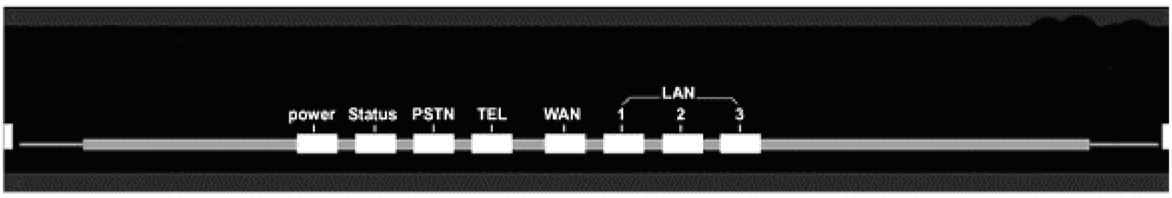
- The plastic housing can be adjustable by manual (Vertical type or Horizontal type)

➢ **Environment**

- Operational Humidity: 10 to 90 % (Non-condensing)
- Operational Temperature: 0 to +40 °C
- Storage Humidity: 10 to 90 % (Non-condensing)
- Storage Temperature: -10 to +50 °C

➢ **Front Panel**

The LEDs on the front panel indicate the operational status of the Gateway.



- **Power (Green):**
  (1) Light on: The gateway is connected with power adapter correctly and power on.
  (2) Light off: The gateway is not connected with power adapter correctly or not power on.
- **Status (Green):**
  (1) Light on: The gateway is under Proxy mode and successfully register to Proxy.

(2) Light off: Under Peer-to-Peer mode.

(3) Light Blanking: The gateway is under Proxy mode and not successfully registers to the Proxy.

- **TEL (Orange):**

(1) Light Blinking: The IP side has incoming call.

(2) Light On: The IP side is in communication.

(3) Light Off: IP Line of gateway is in standby mode.

- **WAN/LAN (Green):**

(1) Light on: Ethernet port successfully connected with network.

(2) Light Blanking: Ethernet port is transmitting or receiving data.

- **PSTN (Orange):**

(1) Light Blinking: The IP side has incoming call.

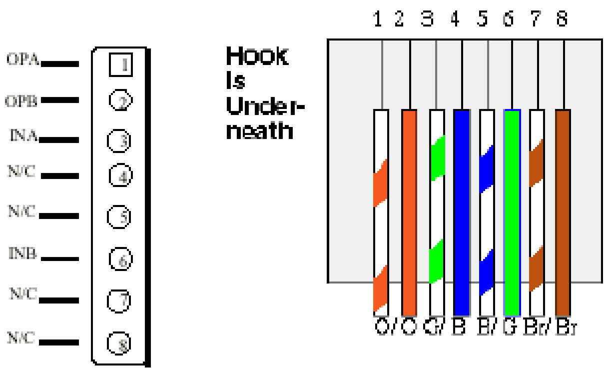
(2) Light On: The PSTN side is in communication.

(3) Light Off: PSTN Line of the gateway is in standby mode.

➢ Back Panel

- **Ethernet Port:**

Ethernet port is for connecting VoIP Gateway to network, transmit rate supports 10/100 Base-T.



Ethernet connector（LAN/WAN）

- **TEL Port:**
  RJ-11 connector, gateway interface to connect analog phone sets or trunk port of PABX.
- **PSTN Port:**
  RJ-11 connector, gateway interface to CO PSTN line or extension port of PABX.
- **DC 12V Port:**
  DC 12V Power supply.

:

Part II: Start-UP

This part explains how to configure essential and basic items before user can run the gateway.

2.1 Software Installation Guide

This guide covers all essential configurations under different application, user can follow steps below to configure basic items to run the gateway.

2.1.1 Default Settings of VoIP gateway

➢ **WAN IP Parameters**

(1) WAN

- IP Address = 10.1.1.3
- Subnet mask = 255.0.0.0
- Default gateway = 10.1.1.254

(2) LAN

- IP Address = 192.168.123.123
- Subnet mask = 255.255.255.0

➢ **Telnet and Web Login Password**

- Login User Name= root
  Password = “Null” (default)

2.1.2 Additional Installation Requirements

In addition to the contents of your package, there are other hardware and software requirements you need before you can install and use your VoIP Gateway. These requirements include:

1. A computer with an Ethernet NIC (Network Interface Card) installed.
2. Use Internet Explorer 5.5 or later / Netscape Navigator 6 or later versions.
3. Analog telephone set.
4. Software tools: SIP Proxy Server (optional)
5. Installation Wizard (optional): This is a configuration tool for users can easy access products and configuring IP address. Please contact with your retailer for more information.

Please follow steps below to access gateway configuration interface:

Step 1.Connect WAN Port of the Gateway to public network

Connect the WAN port (silver) on the Gateway to the Ethernet port of your network (e.g. Cable Modem, ADSL Modem) using the standard

CAT-5 straight Ethernet cable.

Step 2. Connect your PC to the LAN port of the gateway

Connect your PC to the LAN port of the gateway with standard CAT-5 straight Ethernet cable.

Step 3. Set your PC as DHCP mode

Please go to the network setting of your PC and set it as DHCP mode, let your PC can automatically search for DHCP server and get one valid IP address. The gateway has embedded DHCP server (default is enabled) so that your PC will get one IP address from gateway DHCP server.

Step 4.Open your browser and input IP address *192.168.123.123*

Once your PC has got an IP from the gateway, you may connect to the gateway via WEB browser to do more configurations. The default LAN IP address is "**192.168.123.123**"; type this IP address on web browser address bar and click "enter" to connect with web management interface. Refer to Part III Web management for more information.

Step 5.Advanced Setting via Telnet (Optional)

If user wants to do more advanced and complete settings that cannot be found via web management interface, please Telnet to the gateway to do more detail configurations.

Step 6.Connect other PC with LAN Ports (Optional)

If you have more than one PC, you can connect them with LAN Ports (black) on the Gateway. Configure these PCs as DHCP mode so that they can automatically get IP from gateway DHCP server. DHCP server can assign 253 IP addresses at most.

> **Caution:**
> **To prevent damage to the Gateway, please make sure you have connected with the supplied power adapter.**

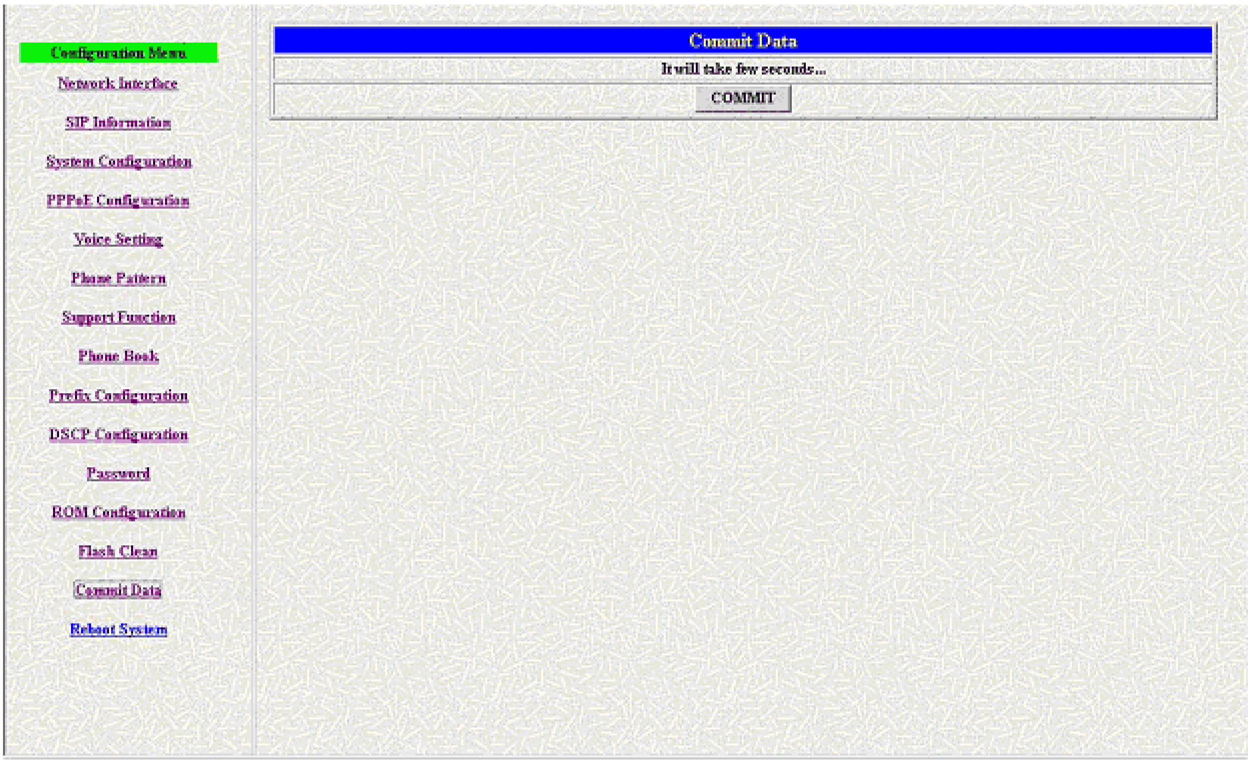
2.1.3 Essential Configuration via Web Management interface

This section describes how to setup the KVT-311 gateway via Web management interface. Follow procedures below to configure essential parameters before you use the gateway.

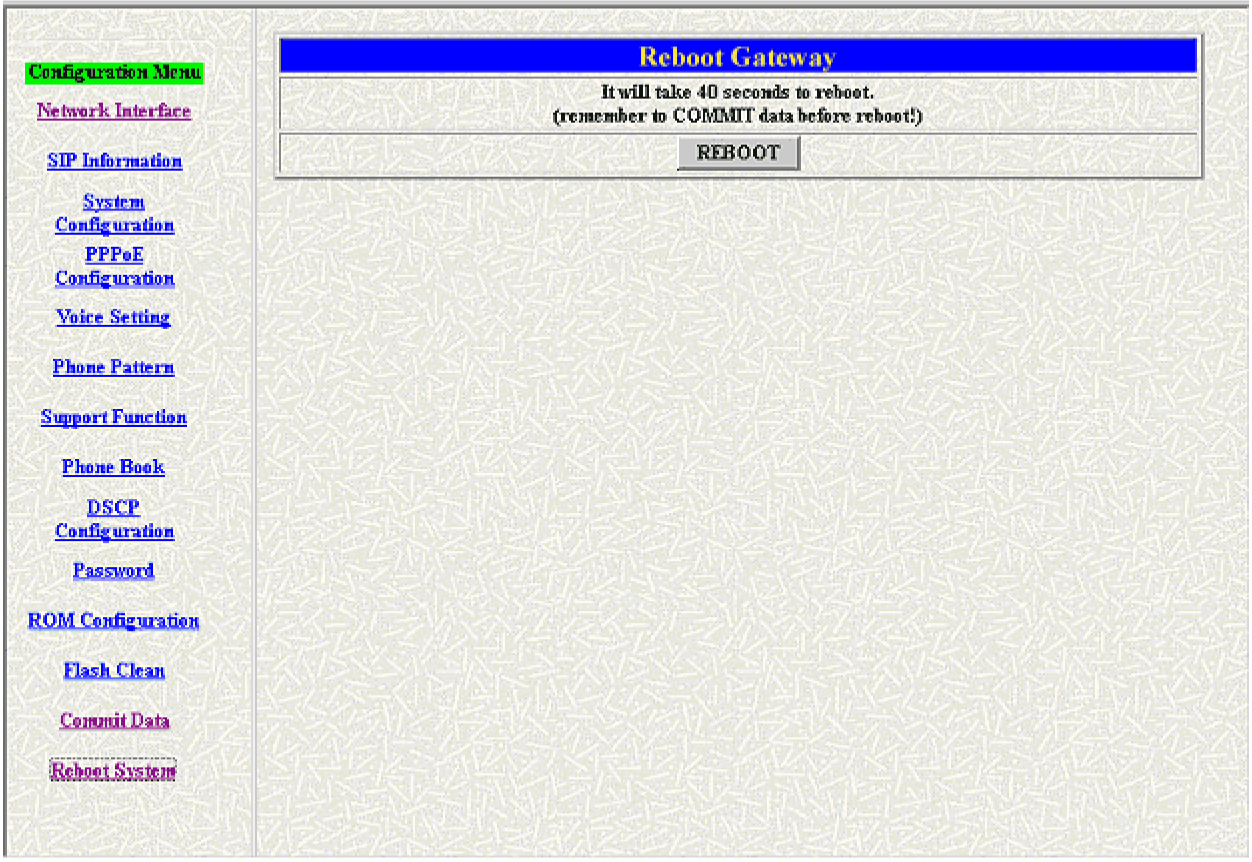
2.1.3.1 Save Data and Reboot

After any configuration has been made, user has to save all data and reboot system to make configurations take effect.

Step 1. Click [Commit Data] on the navigation panel. In the Commit Configuration Data screen, click the [Commit] button. In the Commit Configuration Data screen will Display [Commit to Flash OK!], when the gateway has finished committing data.



Step 2. Click [Reboot System] on the navigation panel. In the Gateway screen, click the [Reboot] button. It will take around 40 seconds to reboot.

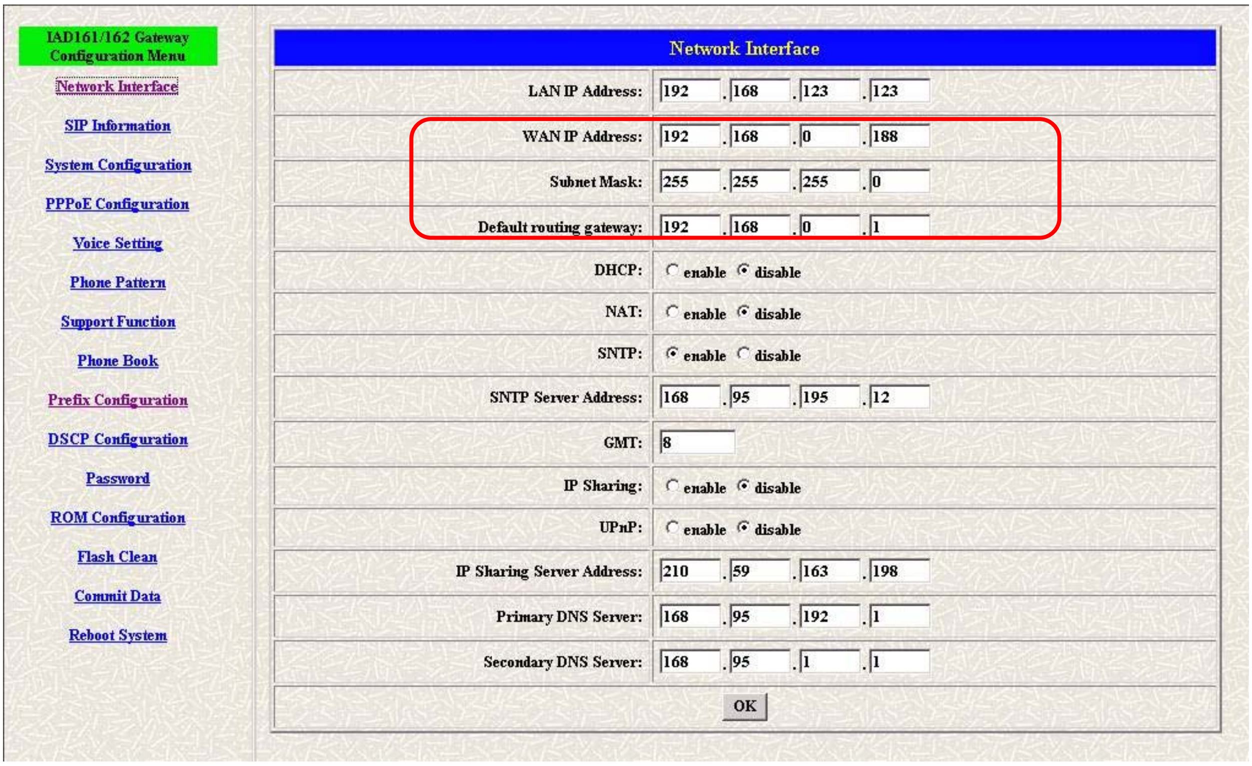


Step 3. Close the current browser windows and launch your web browser again.

2.1.3.2 Setup Network

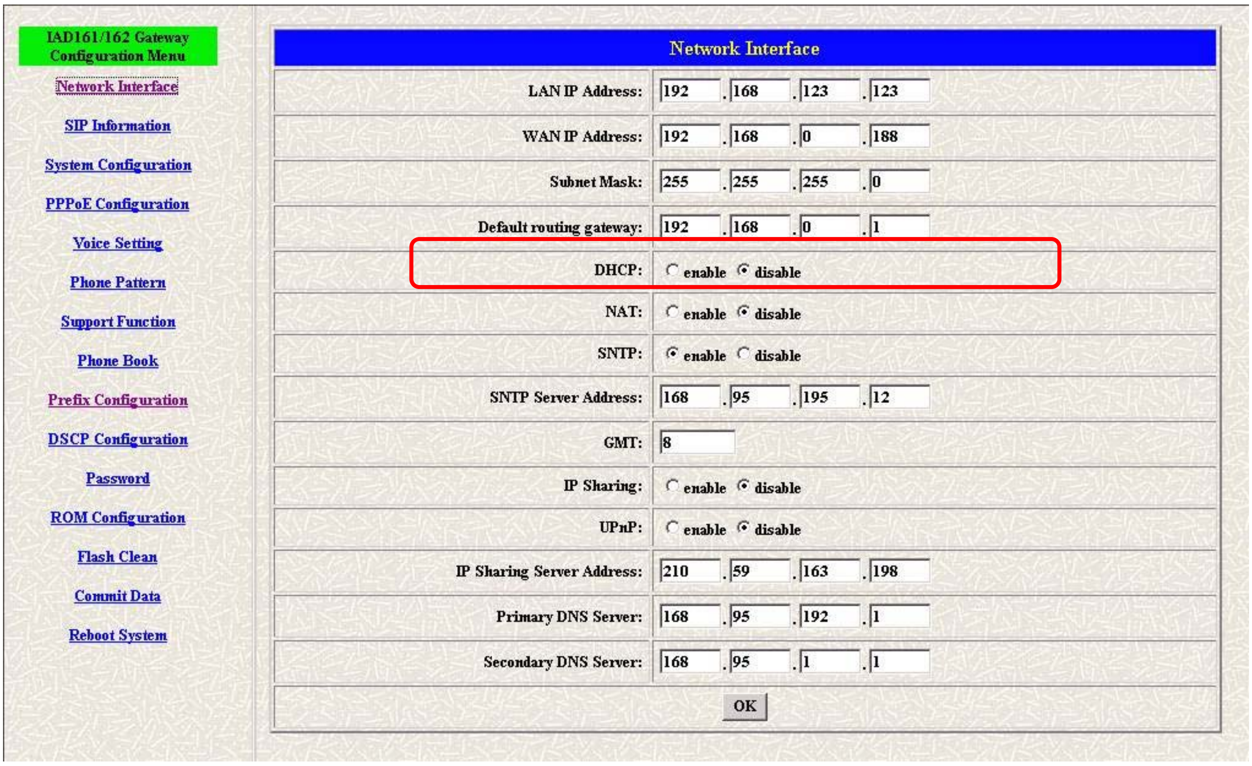
(1) Fixed IP

To configure the VoIP Gateway IP address, please click [Network Interface] on the navigation panel. In the Network Interface screen, type a new IP address, subnet mask and the default routing gateway (e.g. IP Address: 192.168.0.188, Subnet mask: 255.255.255.0, Default routing gateway: 192.168.0.1) and click the OK button.



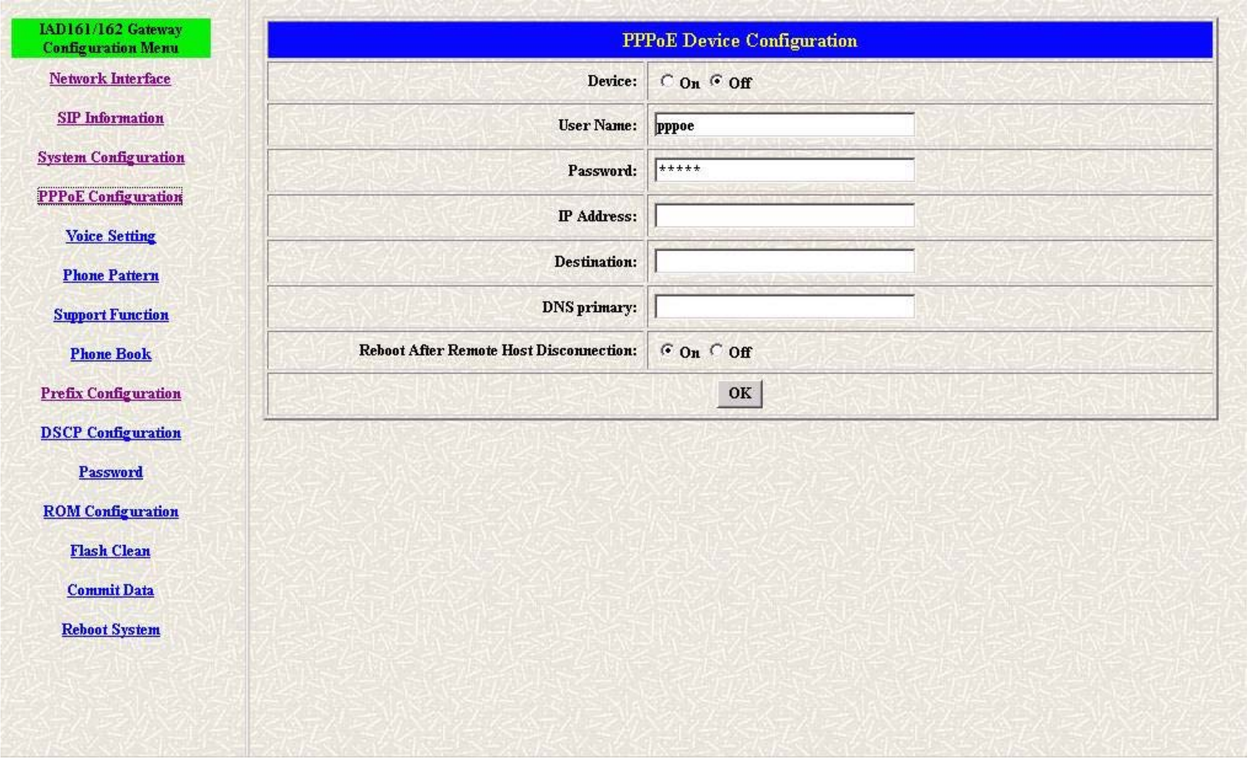
(2) DHCP

Click [Network Interface] on the navigation panel. In the Network Interface screen, enable the DHCP function if you are using the cable modem or DHCP server and click the [OK] button.



(3) PPPoE

Click [PPPoE Configuration] in the navigation panel and open the [PPPoE Configuration] Screen.



- Device: Set PPPoe function to be On or Off.
- User Name: Set PPPoE authentication User Name.
- Password: Set PPPoE authentication password.
- Reboot After Remote Host Disconnection: Enable/Disable auto reboot after PPPoE disconnection

  If user enables this function, after PPPoE being disconnected, the gateway will automatically reboot to re-connect, and after rebooting, if the gateway still can't get contact with server, it will keep trying to connect. After re-connected, it will also restart system. On the other hand, if user disables this function, it won't reboot and keep trying to connect.
- Other items: for reference only, cannot allow to be configured.

2.1.3.3 Application mode-Proxy/Peer-to-Peer Mode

After setting IP address, user must assign the gateway to work under Proxy mode or Peer-to-Peer mode. If there is no Gatekeeper, configure your gateway as Peer-to-Peer Mode.

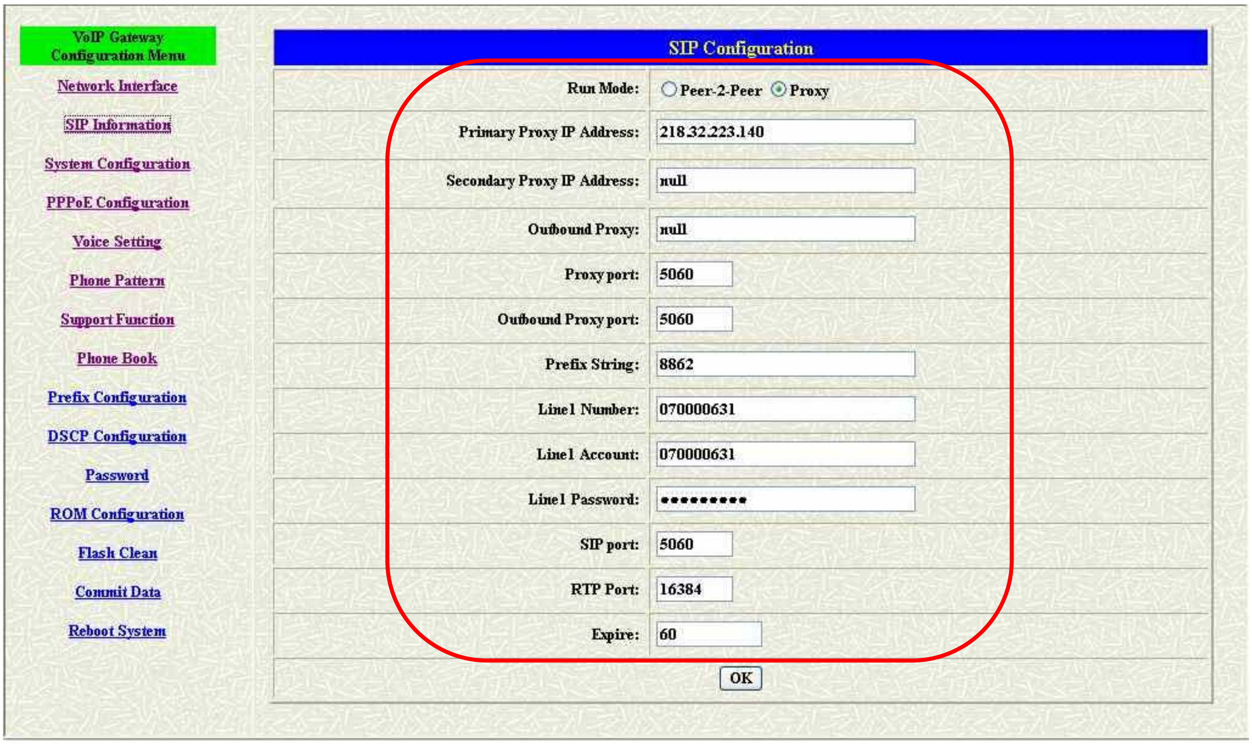
2.1.3.3.1 Proxy mode

Proxy mode means that there will be an intermediate Proxy Server between the VoIP Gateway and the remote entity. While operating at this mode, the Gateway will first register to the Proxy Server located at the ISP side. For the following operation, it sends the INVITE message to the Proxy Server once you initiate a session. Then the Proxy server will forward the INVITE message to the right place. And the Response message from the remote entity will be forwarded back to you via Proxy server.

Step 1. Configure the VoIP Gateway SIP Configuration. Click SIP Information on the navigation panel. In the SIP Information screen, select Proxy routed Mode function.

Step 2. Set the SIP information from your service provider: Proxy IP Address, Line1 Number, Line1 Account, Line1 Password, and click the OK button.

> Note:
> 1. Please contact with your Proxy vendor to obtain user account information.
> 2. If no need to enter password, user also has to set security information, please set "name" the same with line number.
> 3. The Gateway uses "line number" to register to proxy server, the "name" is only for authentication.

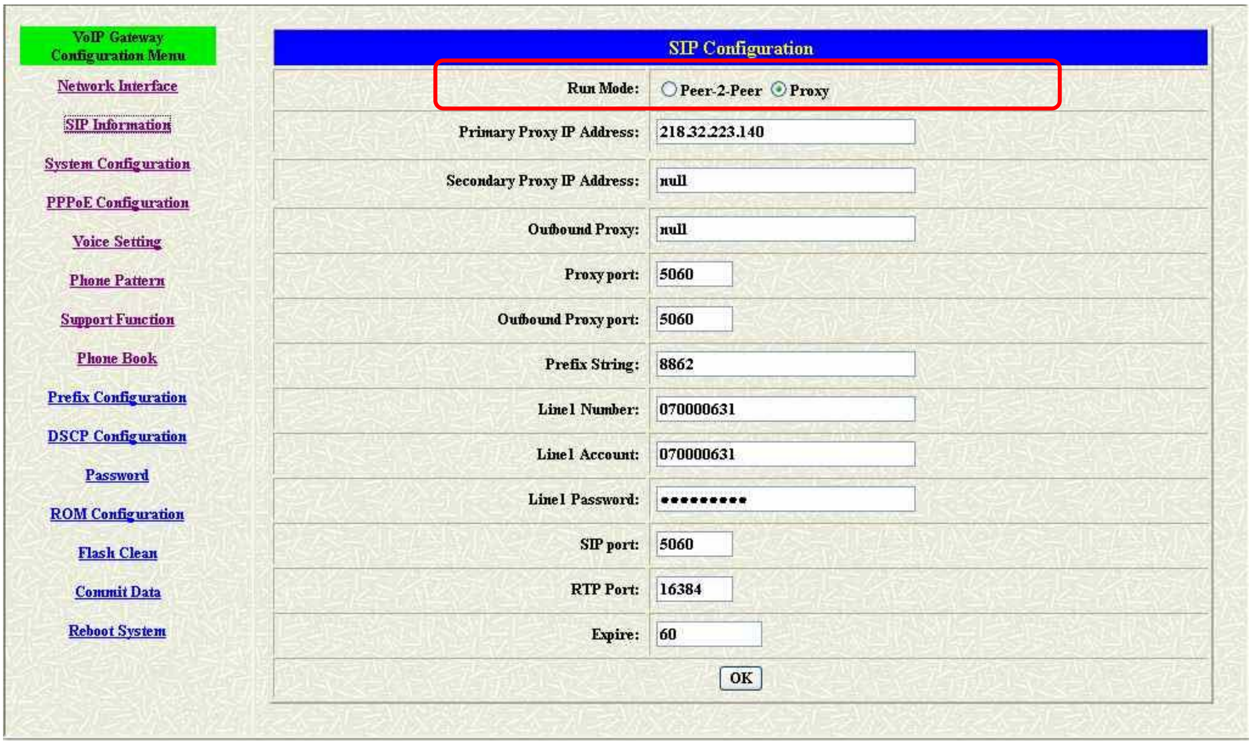


2.1.3.3.2 Peer-to-Peer Mode

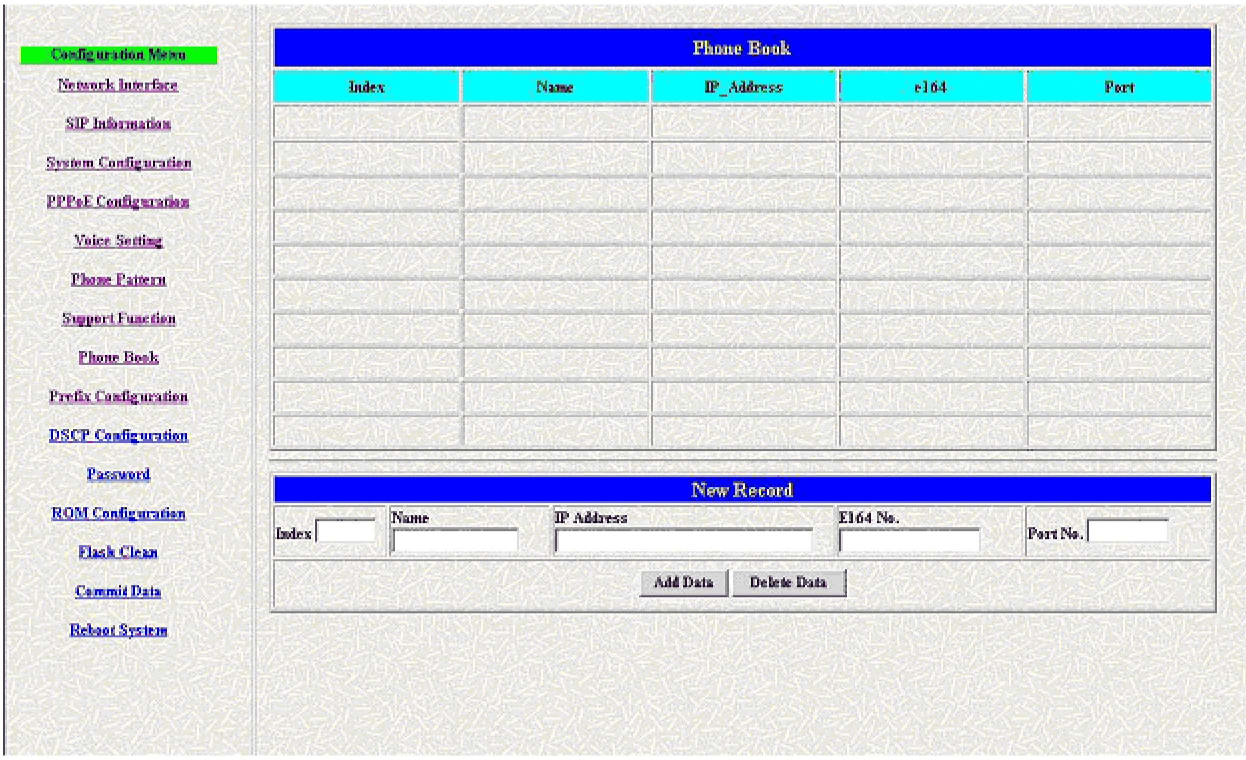
Peer-to-Peer Mode allows users to call other VoIP devices without the proxy server. When in Peer-To-Peer mode, the Gateway use Phone Book, which will dial predefined phone number, and press "#" (optional, to accelerate the dial) as end of dial.

To configure Peer-To-Peer Mode in the Gateway, follow the steps below:

Step 1. Configure the Gateway SIP information. Click [SIP information] on the navigation panel. In the SIP information screen, select Peer-to-Peer Mode function, set line number, line account and click the [OK] button. Line account must be the same with Line number.



Step 2. Configure Phone Book in the Gateway. Click [Phone Book] on the navigation panel. In the Phone Book screen, enter the Index, Name, IP address and e164 (phone number) of the destination and click the Add Data button.



2.1.4 Essential Configuration via Telnet Command Line interface

This section describes how to setup the gateway via Telnet command line interface. Please follow procedures below to configure essential items.

2.1.4.1 Save Data and Reboot

After any configuration has been made, user has to save all data and reboot system to make configurations take effect.

Step 1. Confirm the changed configurations, input [commit] and press [enter] key to save it.

Step 2. Input [reboot] then press [enter] key to restart Gateway.

Step 3. After around 40 seconds, Gateway will take effect in new configurations.

Do not turn off your Gateway or remove the Gateway while saving your configuration.

2.1.4.2 Setup Network

Use command [ifaddr] to configure Gateway IP Address and related information.

(1) Fixed IP

```
usr/config$ ifaddr –ip 192.168.1.11 –mask 255.2555.255.0 –gate 192.168.1.254
```

In this case is to configure Gateway IP Address as [192.168.1.11], subnet mask as [255.255.255.0], default router gateway as [192.168.1.254].

(2) DHCP

```
usr/config$ ifaddr –dhcp 1
```

In this case is to enable DHCP mode of the gateway, once the gateway reboot the system, it will automatically capture IP from DHCP server.

(3) PPPoE

Step 1. To Set PPPoE mode, please use [pppoe] command:

```
usr/config$ pppoe –dve 1 (PPPoE used)
usr/config$ pppoe –open (PPPoE open)
```

Step 2. Input the user id & password provided by your ISP:

```
usr/config$ pppoe –id 123@hinet.net (PPPoE login account)
usr/config$ pppoe –pwd 123 (PPPoE login Passowd)
```

For example:

```
usr/config$ pppoe -print

PPPoE adapter information
      Device            : Enabled
      Status            : Not initialized
      User name         : 84460791@hinet.net
      Password          : ********
      Reboot            : Yes

usr/config$
```

Step 4. Commit and reboot the gateway.

```
usr/config$ commit
usr/config$ reboot
```

Step 5. When it successfully establish PPPoE connection, use command [pppoe -print] to see detail information.

For example:

```
usr/config$ pppoe -print

PPPoE adapter information
      Device            : Enabled
      Status            : Ready
      User name         : 84460791@hinet.net
      Password          : ********
      Reboot            : Yes
      IP address        : 218.160.239.35
      Destination       : 61.223.128.254
      DNS primary       : 168.95.1.1
      Subnet Mask       : 255.255.255.255
      Authenticate      : PAP
      Protocol          : TCP/IP
      Device            : PPP/PPPoE

usr/config$
```

2.1.4.3 Application mode-Proxy/Peer-to-Peer Mode

After setting IP address, user must assign the gateway to work under Proxy mode or Peer-to-Peer mode. If there is no Proxy, please set your gateway as Peer-to-Peer Mode.

2.1.4.3.1 Proxy mode

Proxy mode means that there will be an intermediate Proxy Server between the gateway and the remote entity. While operating in this mode, the Gateway will first register to the Proxy Server located at the ISP side. For the following operation, it sends the INVITE message to the Proxy Server once you initiate a session. Then the Proxy server will forward the INVITE message to the right place. And the Response message from the remote entity will be forwarded back to you via Proxy server.

Step 1. Set Proxy Mode, using "sip" command

```
usr/config$ sip –mode 1
```

Mode 0 is for Peer-To-Peer mode, while mode 1 is for Proxy mode.

Step 2. You must specify Proxy address obtained from your service provider. And the Proxy address can be IPv4 address as well as DNS name.

Several important SIP parameters are listed below when setting proxy mode:"–px", "–line1".

For example:

```
usr/config$ sip –px 210.68.222.33 -line1 12345
```

In this case is to set proxy IP address as "210.68.222.23", line number as "12345".

Step 3. You must configure the accounts using "security" command.

An example is demonstrated below:

```
usr/config$ security –line 1 –name 12345 –password 12345
```

This is to set username (userid) as "12345", password as "12345" into line1, which means line1 can accept incoming calls after successfully registered to Proxy server.

Note:

1. Please contact with your Proxy vendor to obtain user account information. 2. If no need to enter password, user also has to set security information, please set "name" the same with line number.

2.1.4.3.2 Peer-to-Peer Mode

Peer-to-Peer Mode allows users to call other VoIP devices without the proxy server. When in Peer-To-Peer mode, the Gateway use Phone Book, which will dial predefined phone number, and press "#" (optional, to accelerate the dial) as end of dial.

To configure Peer-To-Peer Mode in the Gateway, follow the steps below:

Step 1. Set Peer-To-Peer Mode, using "sip" command

```
usr/config$ sip –mode 0
```

Mode 0 is for Peer-To-Peer mode, while mode 1 is for Proxy mode.

Step 2. Configure Phone Book, using "pbook" command.

```
usr/config$ pbook –add name TEST1 ip 10.1.1.1 e164 10
```

In this case user add one callee record named as TEST1, IP address as 10.1.1.1, and mapping e.164 number as 10. After phone book data has been set, user can dial 10 to make a call for IP 10.1.1.1.

After the command completed, you can type "pbook –print" to see if the input record is correct.

When adding a record to Phone Book, user does not have to reboot the machine, and the record will be effective immediately.

2.1.5 Essential Configuration via Installation Wizard

Installation Wizard is a friendly software tool that can provide you an easy way to configure your WellTech VoIP devices. You only need to Input the MAC address of your product and Click [Search Device]; you can configure your VoIP device without changing your PC's setting.

Additionally, when you forget IP address of the VoIP device, Installation Wizard gives you a solution to solve this problem.

For more information, please refer to the Installation Wizard user manual.

2.1 Special Housing Installation Guide

The VoIP gateway router has special adjustable housing for vertical or horizontal type. Please follow procedures as below to change type you like.

2.1.1 Horizontal Type

2.1.1.1

Insert stand board on one side.

2.1.1.2

Insert the other stand board on the other side.

2.1.1.3

The gateway can stand as horizontal type.

2.1.2 Vertical Type

2.1.2.1

Insert stand board on one side.

2.1.2.2

Insert the other stand board on the same side.

2.1.2.3

The gateway can stand as vertical type.

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Part III: Special Applications and Features

This part explains how to configure VoIP Gateway under special application mode, such as behind NAT, and how to upgrade firmware.

3.1 Behind IP-Sharing

3.1.1 IP Sharing Configuration

3.1.1.1 One VoIP Gateway behind VoIP Gateway

This application is only for the user who is using the IP Sharing device, and the VoIP gateway is connected behind the IP sharing device. The IP Sharing Device must support the DMZ or Virtual server functions such as ADSL network.

Step 1. The WAN IP Address obtained from ADSL has two kinds of methods. One is fixed IP Address, while user applies for one or more fixed IP Addresses. Another is dynamic IP Address while user applies for dial-up connection way. Only when the IP address is fixed, user can put the gateway behind NAT device.

Step 2. The LAN IP Address of User's PC can be set as DHCP client in order to gain a valid one.

Step 3. One can also assign a fixed IP address, which belongs to the same network segment as the LAN interface of IP Sharing device.

Step 4. VoIP Gateway must enable the IP Sharing function for the fixed / dynamic WAN IP Address.

> ***Note:***
> With Dynamic WAN IP Address, a valid Proxy for the Gateway to get register on is a must. In other word, it is not workable in Peer-to-Peer mode while dynamic WAN IP Address.

Step 5. IP Sharing device must have a function to do IP/Port mapping. Some is named as DMZ, some is named as virtual server. The VoIP messages from WAN have to completely pass forward to the LAN. It means that if the Gateway is assigned a virtual fixed IP Address such as 192.168.1.5, IP Sharing device must forward the VoIP message to 192.168.1.5.

Please see following for example:

>Advanced setting > NAT setting > DMZ Host setting

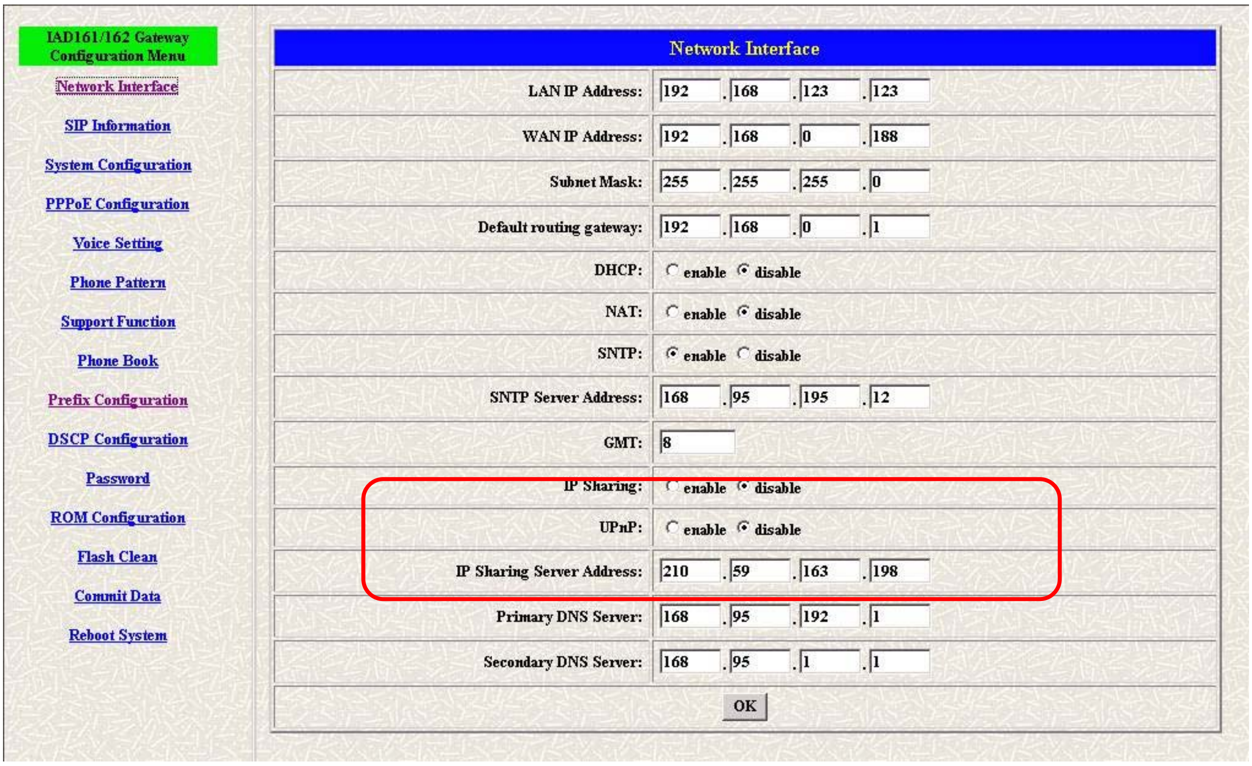
DMZ Host setting

☐ **Activate DMZ**

DMZ Host IP: 192.168.1.5

Step 6. Configure the Gateway IP address for IP Sharing Mode. Click [Network Interface] on the navigation panel. In the Network Interface screen, enter the IP address, Subnet mask and the default gateway in the network table. Please follow up your IP Sharing device

Step 7. Enable the IP sharing function and input the static IP address in the IP Sharing server address (e.g. 210.59.163.198) and click the OK button.



Step 8. Click [Commit Data] on the navigation panel. In the Commit Configuration Data screen, click the Commit button. In the Commit Configuration Data screen will Display [Commit to Flash OK!], when the gateway finished committing data.

Step 9. Click [Reboot System] on the navigation panel. In the VoIP Gateway screen, click the [Reboot] button. It will take around 40 seconds to reboot.

Step 10. Close the current browser windows and launch your web browser again. Enter the new IP address in the Location or Address field.

3.1.1.2 More than one VoIP gateway behind the same IP Sharing Device

1. Assign an IP address to each gateway using fixed address.
2. Enable the IP Sharing function for each set using following command.
   Fixed IP Address – *usr/config$ ifaddr –ipsharing 1 “public IP of IP Sharing”*
3. Configure separate SIP port and RTP port for each set to prevent from port conflict. For example, if set A uses the default settings (SIP port: 5060, RTP port: 16384), you must change set B’s setting to SIP port equal to 5061 and RTP port equal to 26384 for instance.
   Change SIP port – *usr/config$sip –port 5061*
   Change RTP port– *usr/config$sip –rtp 26384*
4. Use the Port Forwarding or Port Redirection function provided by IP Sharing device (Router). See following for example.

>Advanced setting > NAT setting > Port Redirection

Active Configuration

Items	**Service name**	**Protocol**	**Actual Port**	**Virtual IP**	**Virtual Port**	**Enable**
1	1	UDP	5060	192.168.1.10	5060	**V**
2	2	UDP	16384	192.168.1.10	16384	**V**
3	3	UDP	16394	192.168.1.10	16394	**V**
4	4	UDP	5061	192.168.1.11	5061	**V**
5	5	UDP	26384	192.168.1.11	26384	**V**
6	6	UDP	26394	192.168.1.11	26394	**V**
7		---	0		0	X
8		---	0		0	X
9		---	0		0	X
10		---	0		0	X

Note:

With Dynamic WAN IP Address, when the WAN IP is changed, we need to change the external IP of VoIP Gateway using above command.

1. Different Vendor's Router will have different appearance of setting.
2. Once you set the DMZ Host, you don't need to configure the Port Forwarding and vice versa.
3. If there is only one Gateway attached to the IP Sharing device, it is recommended to use DMZ Host setting to enable the NAT traverse and disable the Port Forwarding.
4. If there are two or more sets of Gateways attached to the IP-Sharing device, please configure the Port Redirection (Forwarding) to enable the NAT traverse and disable the DMZ Host.
5. After the IP Sharing configuration of the Gateway and IP Sharing device is complete, you must reboot the Gateway to activate the new settings.

3.2 NAT mode (PPPoE)

Step 1. Set PPPoE mode, using [pppoe –dve 1], input the user id & password provided by your ISP, using [pppoe –id –pwd], reboot the device after disconnection, using [pppoe –reboot 1]

```
usr/config$ pppoe –dve 1 (PPPoE used)
usr/config$ pppoe –open (PPPoE open)
usr/config$ pppoe –id 123@hinet.net (PPPoE login account)
usr/config$ pppoe –pwd 123 (PPPoE login Passowd)
usr/config$ pppoe –reboot 1 (Enable)
```

Step 2. Set NAT function (Default NAT function is enable)

```
usr/config$ ifaddr -nat 1
```

For example:

```
usr/config$ ifaddr -print

Internet address information
      LAN IP address          : 192.168.123.123
      WAN IP address          : 192.168.13.71
      Subnet mask             : 255.255.248.0
      Default gateway         : 192.168.8.254
      NAT enabled             : ON
      DHCP startup            : OFF
      SNTP                    : mode=1
                                server 168.95.195.12
                                time zone : GMT+8
                                cycle=1024 mins

      IPSharing               : no IPSharing device.

      Primary DNS Server      : 168.95.192.1
      Secondary DNS Server : 168.95.1.1
usr/config$
```

Step 3. When Gateway connection succeed. **Setup PC use LAN IP connection Network**

Select [Specify an IP Address] and enter [192.168.123.xxx] in the [IP Address] location (where xxx is a number between 2 and 254 used by the VoIP Gateway to identify each computer), and the default [Subnet Mask 255.255.255.0]. Please notice that two computers on the same LAN cannot have the same IP address. Set Default Gateway value as 192.168.123.123 in the [new gateway] field. Then save your change. PC can also use DHCP mode when DHCP server of VoIP gateway is enabled.

3.3 Call Hold, Transfer and Forward

Gateway provides call features including call hold, transfer and forward. Please be noted that both calling and called site have to support this feature. For call forward function, it only works under Proxy mode. Of course, Proxy must support these call features, too.

It is better for user to prepare a telephone set supported [FLASH] function on keypad. If telephone set does not support [FLASH] function on keypad, user can click the Hook quickly by sending FLASH message.

Note:
The default FLASH length for Gateway is between 400ms to 800 ms.. This value must be compliant with your phone set, if user press flash but not work, please check the flash time value of your phone set and adjust it on the VoIP Gateway.

3.3.1 Call Hold – press [FLASH]

By pressing the FLASH after making a call, both sites shall hear the 2nd dial tone generated by Gateway. To retrieve the call back, just press the FLASH again.

3.3.2 Call Transfer – press [FLASH], then [transferring number]

A makes a call to B, B press FLASH, A and B hear 2nd dial tone, B presses C's number, C will ring, A will hear Ring Back tone, B Hangs up this call, and A and C can communicate.

3.3.3 Call Forward:

User has to activate/deactivate call forward function via pressing keypad of phone set. This function is only available under Proxy mode, and the Proxy must support Call Forward function. There are three conditions for user to set forward function:

3.3.3.1 No response/ Answer:

While no one answers the call, incoming call will be forwarded to the assigned number.

(1) Activate: *75 [Forward No.] #

(2) Deactivate: #75#

3.3.3.2 Busy Forward:

While line is engaged or phone set is been off-hook, incoming call will be forwarded to the assigned number.

(1) Activate: *76 [Forward No.] #

(2) Deactivate: #76#

3.3.3.3 Unconditional:

Incoming call will be forwarded to the assigned number unconditionally.

(1) Activate: *77 [Forward No.] #

(2) Deactivate: #77#

3.4 Upgrade Your VoIP Gateway

.

3.4.1 Upgrade via Web management interface

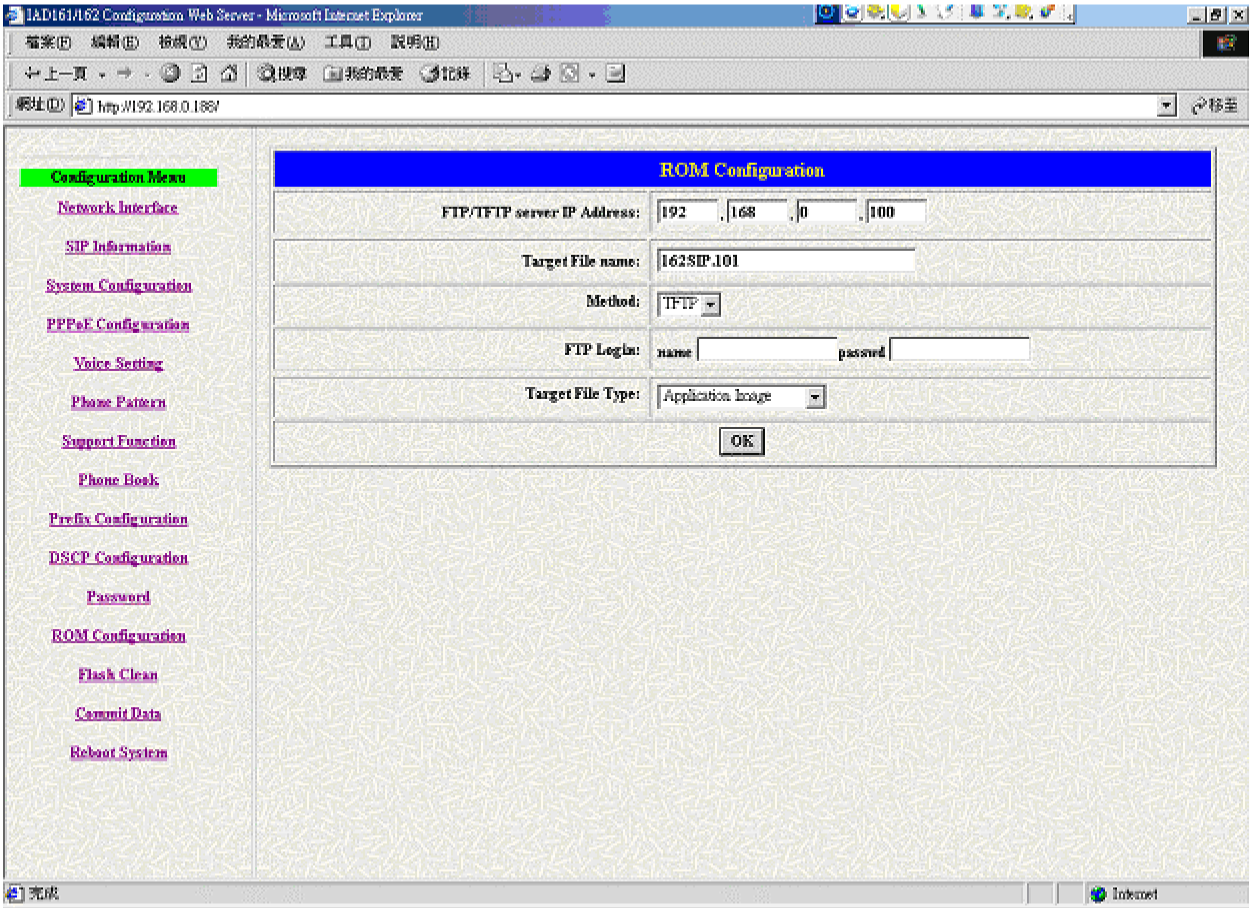
3.4.1.1 Before start

Step 1. Please confirm Host PC, which is installed as TFTP / FTP server and is in available network.

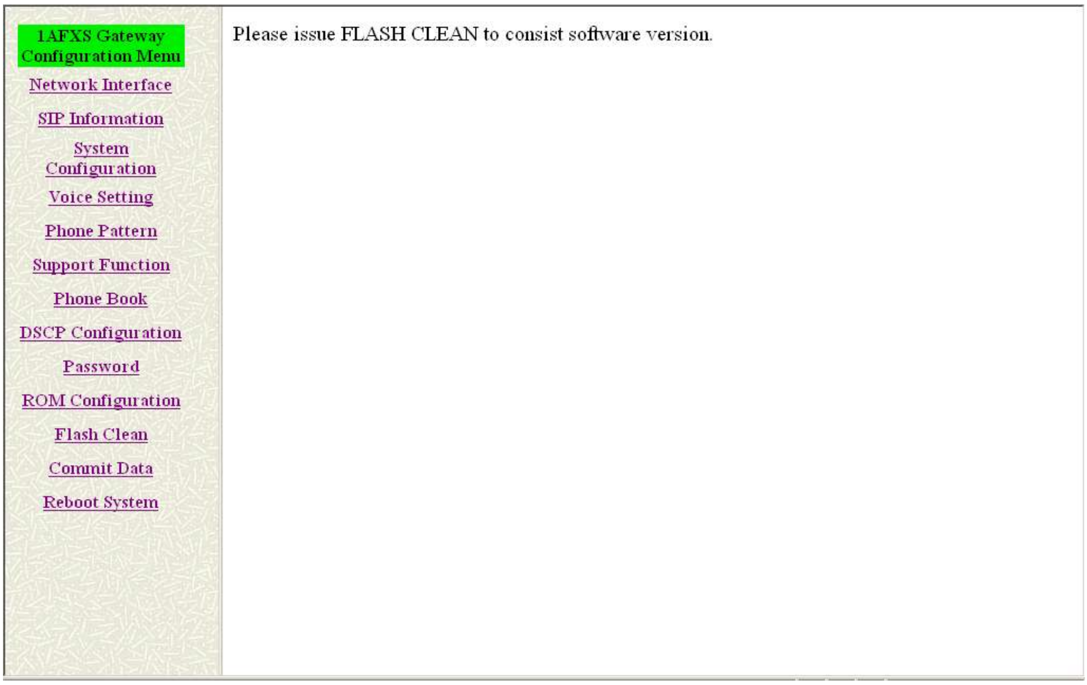
Step 2. Note down your current configurations, such as [SIP Information], [Phone Book].

3.4.1.2 Upgrade Version

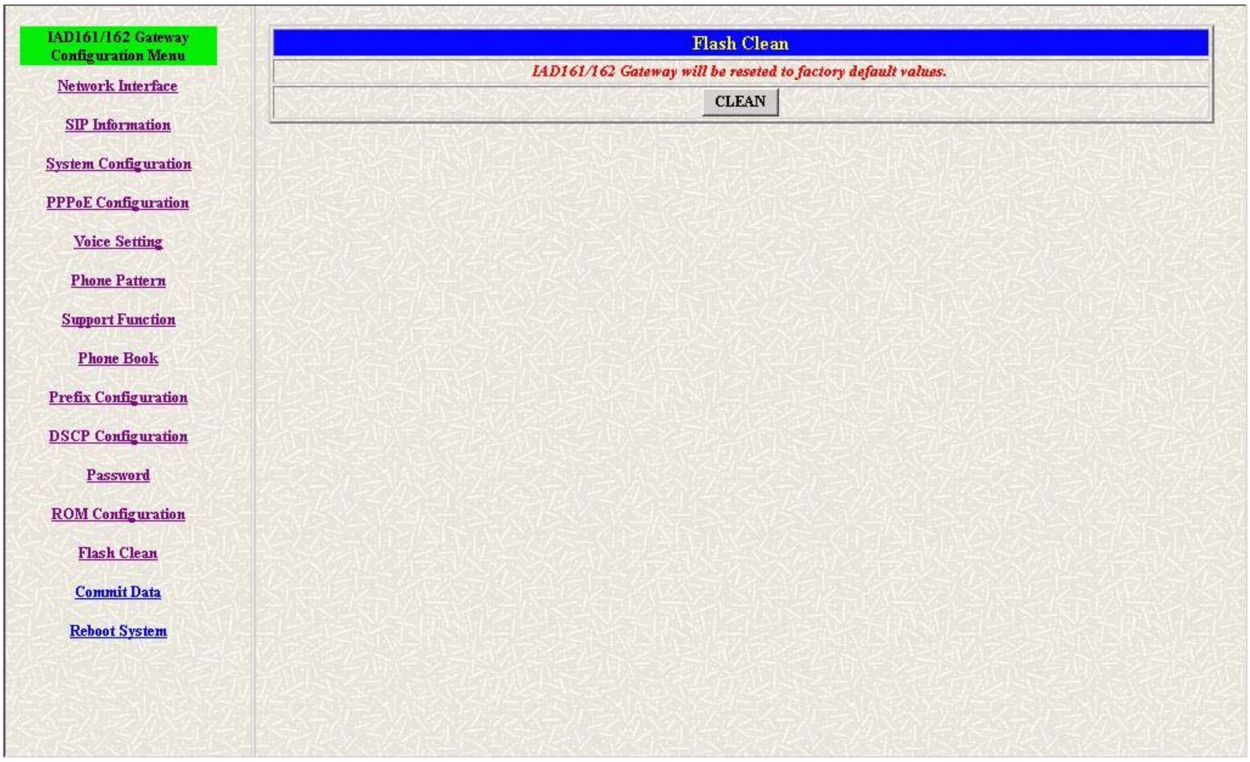
Step 1. To update the Gateway ROM Version, please click [ROM Upgrade] on the navigation panel. In the [ROM Configuration] screen, type TFTP/FTP Server IP address, Target File Name, Method, Target File Type (e.g. Server IP Address: 192.168.0.100, Target File Name: 162SIP.101, Method: TFTP, Target File Type: Application image) and click the [OK] button.



Step 2. After upgrade finished, on screen will display [Please issue FLASH CLEAN to consist software version.] information.



Step 3. Click [Flash Clean] on the navigation panel. In the Flash Clean screen, click the [CLEAN] button.



Step 4. In the Flash Clean screen to Display [Flash cleaned!! Please reboot your

system!!], when Flash Clean Ok.

Step 5. Click [Reboot System] on the navigation panel. In the Reboot Gateway screen, click the [Reboot] button. It will take 40 seconds to reboot.

Step 6. Close the current browser windows and launch your web browser again. Enter the IP address in the Location or Address field.

3.4.2 Upgrade via Telnet Command interface

Use [rom] command to upgrade software.

```
usr/config$ rom

ROM files updating commands
Usage:
rom [-print] [-app] [-boot] [-dsptest] [-dspcore] [-dspapp]
      [-ht] [-method used] [-boot2m]
      -s TFTP/FTP server ip -f filename
rom -print
      -print      show versions of rom files. (optional)
      -app        update main application code(optional)
      -boot       update main boot code(optional)
      -boot2m     update 2M code(optional)
      -ht         updata Hold Tone PCM file(optional)
      -dsptest    update DSP testing code(optional)
      -dspcore    update DSP kernel code(optional)
      -dspapp     update DSP application code(optional)
      -s          IP address of TFTP/FTP server (mandatory)
      -f          file name(mandatory)
      -method     download via TFTP/FTP (TFTP: mode=0, FTP: mode=1)
      -ftp        specify username and password for FTP
      Note:
      This command can run select one option in 'app', 'boot',
      , 'dsptest', 'dspcore', and 'dspapp'.
Example:
      rom -method 1
      rom -ftp vwusr vwusr
      rom -app -s 192.168.4.101 -f app.bin

usr/config
```

Parameter Usages:

-print: show versions of all rom files.

-app, boot, boot2m, dsptest, dspcore, dspapp, ht: To update main Application program code, Boot code, DSP testing code, DSP kernel code, or DSP application code, and Hold Tone file.

Note: Most of all, the Rom file needed to get upgrade is App or Boot2m. Please check the exact Rom file before doing download procedure.

-s: To specify TFTP server's IP address when upgrading ROM files.

-f: To specify the target file name, which will replace the old one.

-method: To decide using TFTP or FTP as file transfer server. [0] stands for TFTP, while [1] stands for FTP.

-ftp: If users choose FTP in above item, it is necessary to specify pre-defined username and password when upgrading files.

For example:

```
usr/config$ rom -print


Download Method  :  TFTP
       Boot Rom  :  sdboot.200
Application Rom  :  1asipIAD.107
        DSP App  :  48302ce3.140
     DSP Kernel  :  48302ck.140
  DSP Test Code  :  483cbit.bin

      Hold Tone  :  holdtone.101

usr/config$
```

After software like application has been upgraded, please execute [flash -clean] to clear old configurations and make upgrade complete. This will keep all configurations under [ifaddr].

```
usr/config$ flash –clean
```

3.5 KVT-311 PSTN Line Application

3.5.1 PSTN Outgoing Call

3.5.1.1 Make PSTN call from TEL Phone set

Default TEL line is intended to make IP call, if user wants to make a call via PSTN line, please dial "***#**", then user will hear dial tone from PSTN side. The dial tone between IP side and PSTN are different. If the PSTN port doesn't connect a PSTN line, press "*#", the signal will be back to IP side then play busy tone.

3.5.1.2 PSTN Backup

If the gateway is working under Proxy mode, and fail to register to Proxy server, TEL line will automatically switch relay to PSTN line, which means when VoIP system is failed, user can still make a call from PSTN line without pressing "***#**".

3.5.2 PSTN Incoming Call

3.5.2.1 VoIP TEL is busy

Caller from PSTN side can still make a call to VoIP router, PSTN LED will be blanking, when user hangs up IP call, TEL phone set will ring, and user can pick up the call from PSTN side. The Gateway has "call waiting tone" function, when IP side is under calling, if PSTN side has an incoming call, the user in IP side will hear a call waiting tone.

3.5.2.2 VoIP TEL is available

When the gateway has incoming call from PSTN side, TEL phone set will ring, user can pick the call from PSTN side.

3.5.3 VoIP Outgoing Call

Default TEL line is intended to make IP call, user can just pick up the phone set connected with TEL port and make VoIP call while VoIP network system is available.

3.5.4 VoIP Incoming Call

When user is communicating with PSTN side with TEL Phone set, the gateway can't have VoIP incoming call.

3.6 Hotline mode

The Hotline Mode is applied in limited two peers. User just picks up the phone set and then hears ring back tone or dial tone depended on configurations of destination device.

Step 1. Specify gateway service type as Hotline service.

Step 2. Create a Hotline table with [line] command.

```
usr/config$ sysconf –service 1
usr/config$ bureau –hotline 1 10.2.2.2 201
```

This example means that if user picks up phone set of FXS Line1, gateway will automatically dial out IP address of [201].

Note:

If this gateway is under P2P mode, please set the phone book firstly.

The IP address of “bureau” command indicates the IP address of called party in P2P mode, or the proxy server IP address in proxy mode.

Step 4. After the configuration, [commit] and [reboot] the device.

```
usr/config$ commit
usr/config$ reboot
```

:

Part IV: Web Management Interface

This part explains how to configure the the Gateway via WEB management interface.

4.1 Login and welcome screen

Step 1. Start your web browser.

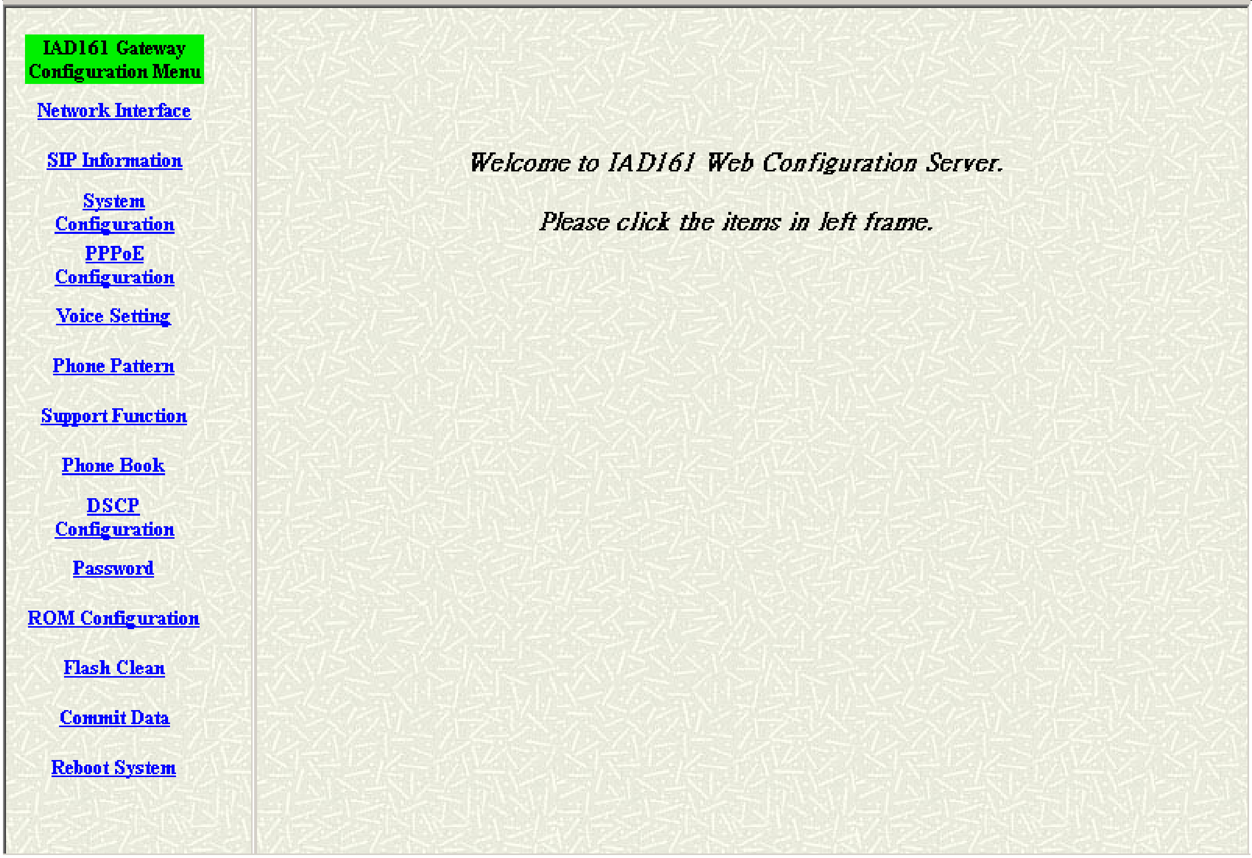
Step 2. Launch your web browser and enter [192.168.123.123] (the default IP address of the LAN port) in the **Location** or **Address** field. Press **Enter.**

Step 3. Password request screen will appear as below. Please input "**root"** in the user name field and **no password** in the password field.

Step 4. Click **OK.**

Step 5. After a successful login, you will see the welcome screen described next. User can click links on the navigation panel at left to go to corresponding configuration screen

.



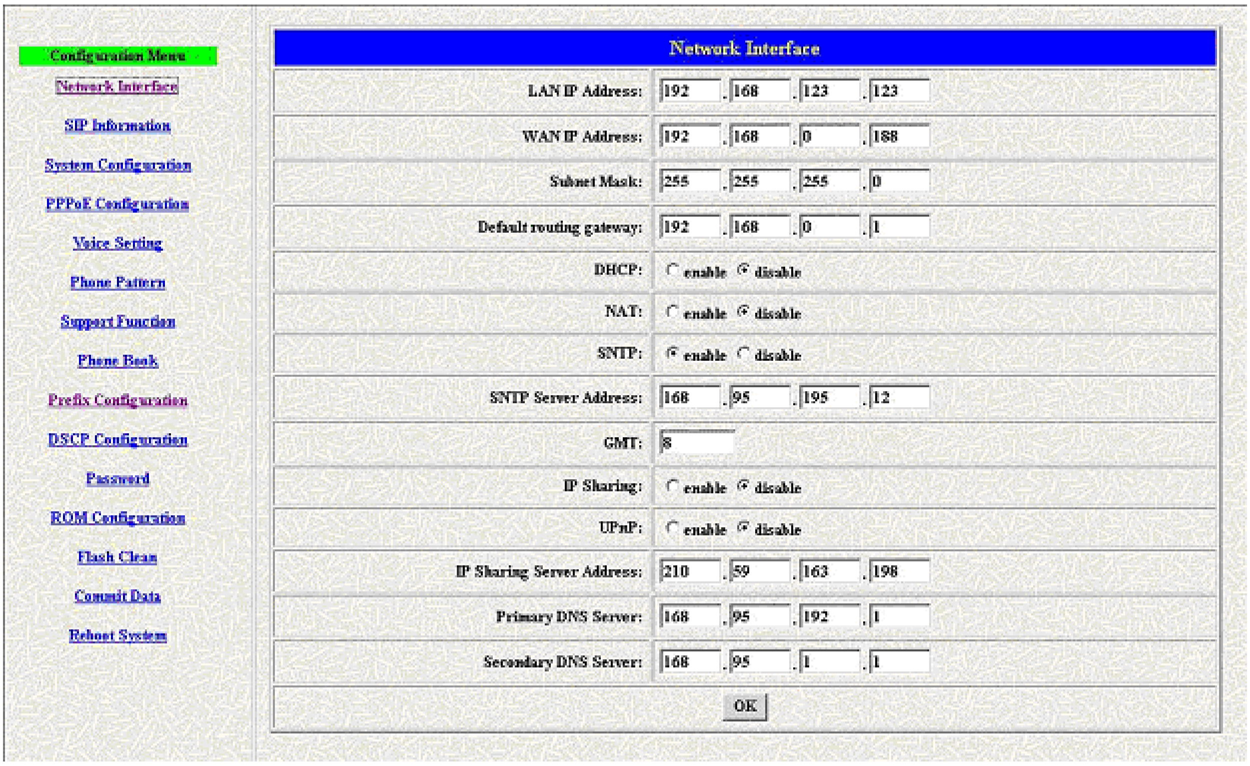
4.2 Save and Reboot

Click **OK** at the end of every configuration page to confirm your changes. All configurations will not take effect before reboot system. Please remember to do [**Commit Data**] to save all configuration then **[Reboot System]** to reboot the gateway.

4.3 Web Management Configuration

4.3.1 Network Interface

Click [Network Interface] in the navigation panel and open the Network Interface Screen.



- LAN IP Address: Set LAN IP Address of gateway (range: 192.168.1.1-192.168.254.254)
- WAN IP Address: Set WAN IP Address of the gateway
- Subnet Mask: Set the Subnet Mask of the gateway
- Default routing gateway: Set Default routing gateway of the VoIP router
- Get IP Mode: User has to set the gateway to use which network mode.
- DHCP: When DHCP function enables, the gateway will automatically search DHCP server after reboot.
- NAT: Enable / Disable the Network Address Translation function
- SNTP: Enable / Disable the Simple Network Time Protocol function
- SNTP Server Address: Set SNTP Server Address

  When SNTP server is available, enable gateway SNTP function to point to SNTP server IP address so that the gateway can get correct current time.
- GMT: Set time zone for SNTP Server time

  User can set different time zone according to the location of the gateway. For example, in Taiwan the time zone should be set as 8,which means GMT+8.
- IP Sharing: Enable it if the gateway is behind IP Sharing router.
- UPnP: Enable it if IP sharing or NAT device supports UPnP function so that no

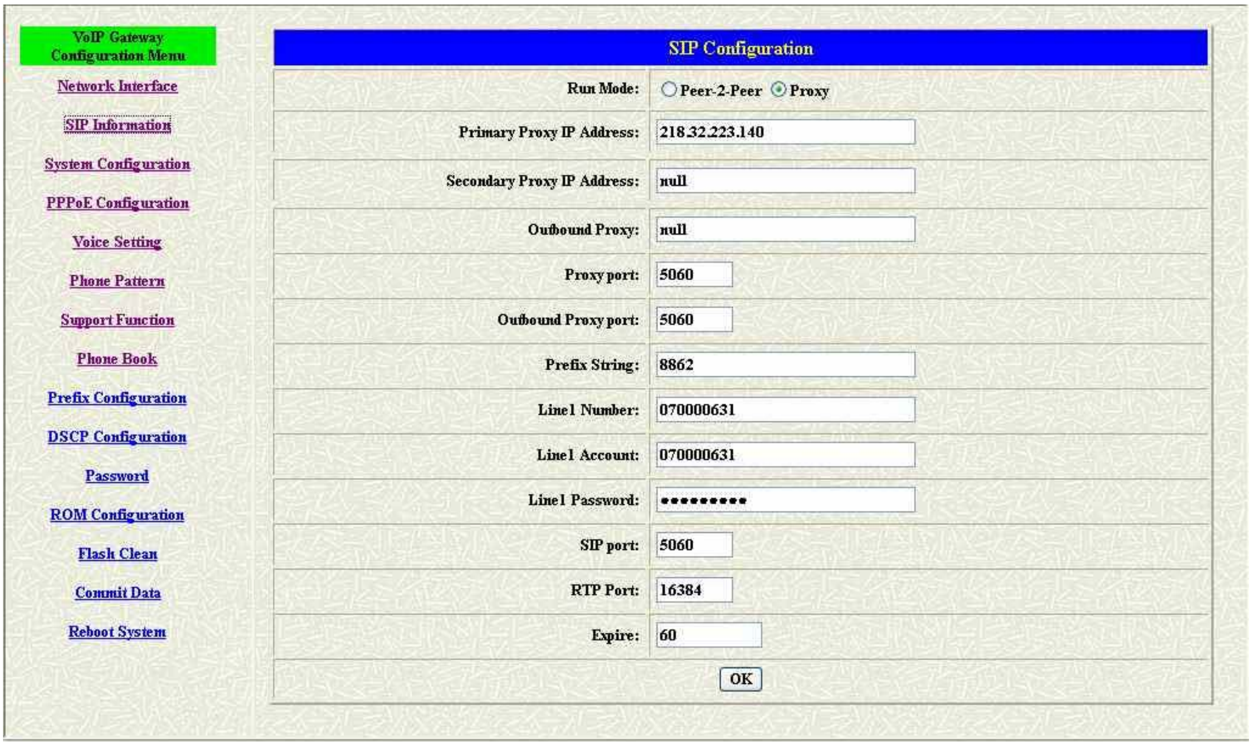
need to configure IP sharing or the gateway when it is behind NAT device.

- IP Sharing Server Address: Set Public IP Address of IP Sharing router for the gateway to work behind IP sharing.
- Primary DNS Server: Set Primary Domain Name Server IP address.
  User can set Domain Name Server IP address. Once the gateway can connect with DNS server, user can specify URL address instead of IP address for Proxy and phone book IP address.
- Secondary DNS Server: Set Secondary Domain Name Server IP address.

Note:
When the gateway is behind IP sharing device, if Proxy support behind NAT function, both gateway and IP sharing don't need to do any configuration. Please contact with your proxy vendor more correct information before configuring the VoIP router.

4.3.2 SIP Information Screen

Click [SIP Configuration] in the navigation panel and open the SIP Information Screen.



- Run Mode: Select the gateway to work under Peer-to-Peer mode or Proxy mode.
- Primary Proxy IP Address: Set primary Proxy IP Address or URL address (Domain Name Server must be configured. Please refer to **Network Interface**).
- Secondary Proxy IP Address: Set secondary Proxy IP Address or URL address (Domain Name Server must be configured. Please refer to **Network Interface**). When gateway fail to register to primary Proxy, it will try to register to secondary Proxy, when it fails again, it will retry to register to Primary Proxy.
- Outbound Proxy: Set IP Address or URL address (Domain Name Server must be configured. Please refer to **Network Configure**) of outbound Proxy server.
- Proxy port: Set Proxy port for the gateway to send message, default value is 5060, if there is no special request of Proxy server, please don't change this value.
- Outbound proxy port: Set outbound Proxy port for the gateway to send message, default value is 5060, if there is no special request of Proxy server, please don't change this value.
- Prefix String: set prefix string. If user ID contains alphabets, user can set it as prefix string here. For example, if Account Name is 123, the gateway will sent out messages as Account Name @"IP address of Proxy", if user set prefix as

abc, the gateway will set out as abc123@"IP address of Proxy". This function is for special proxy server.

- Line Number: identify one number for the the gateway register to the Proxy.
- Line Account: set user name of the gateway for registering. User can set user name and password for registering. If password is no need, please set user name the same as line number or the gateway won't register successfully.
- Line Password: set password for registering.
- SIP Port: set SIP UDP port.
- RTP Port: set RTP port for sending voice data.
- Expire: set expire time of registration. The gateway will keep re-registering to proxy server before expire timed out

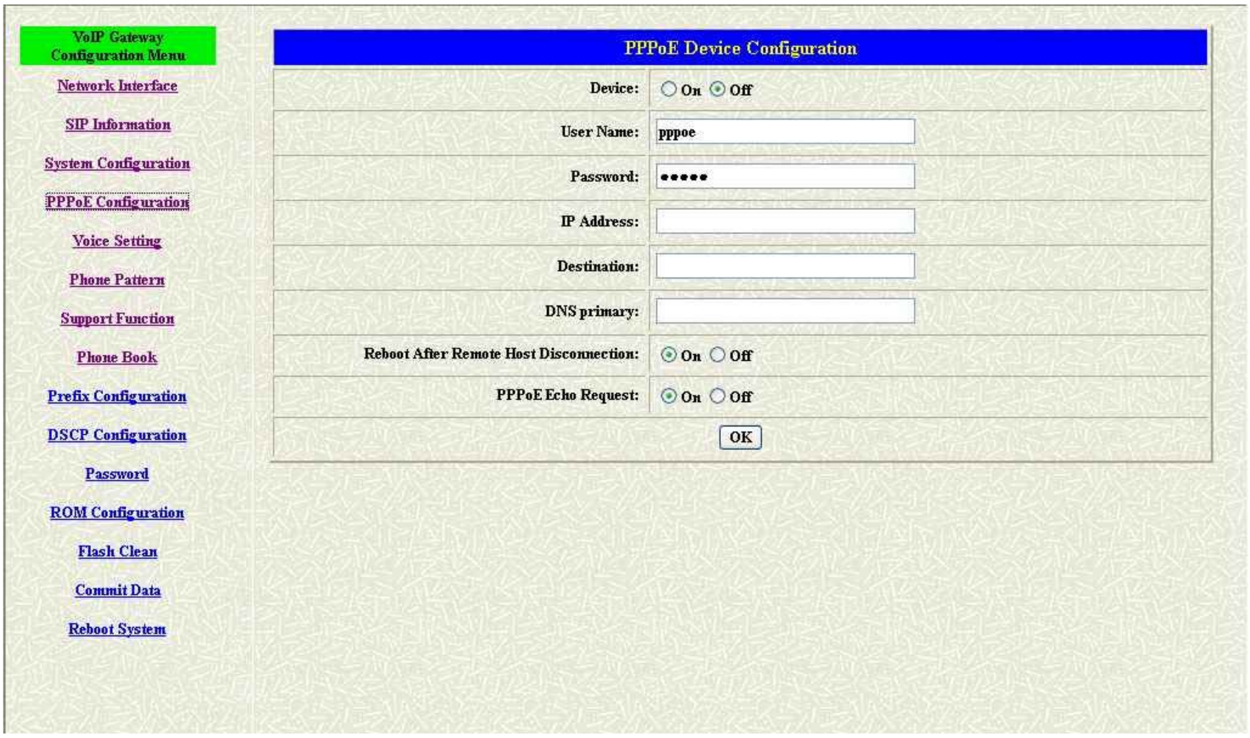
4.3.3 System Configuration

Click [System Configuration] in the navigation panel and open the [System Configuration] Screen.

- Keypad DTMF Type: set DTMF type. User can select DTMF type the gateway transmits.
- RFC2833 Payload Type: change RFC2833 Payload type. This is for special request from the other site, if RFC2833 payload types of 2 sites are different, it may cause some problem of connection.
- FAX Payload Type: Change FAX payload type of the gateway.
- Inter Digit Time: Set the DTMF inter digit time (second)
  To set the duration (in second) of two pressed digits in dial mode as timed out. If after the duration user hasn't pressed next number, the gateway will dial out all number pressed (The inter digit time range is 1~10 secends).
- Caller ID Type: Set Caller ID function. If user set disable, the gateway won't display caller ID on Phone set when received caller ID information from remote site. User can also select caller ID type to be FSK, DTMF or NTT according on which type your phone set supports.
- End of Dial Digit: select end of dialing key, e.g. set end of dial key as ***** button, after finished pressing dialing number then press ***** will dial out.

4.3.4 PPPoE Configuration Screen

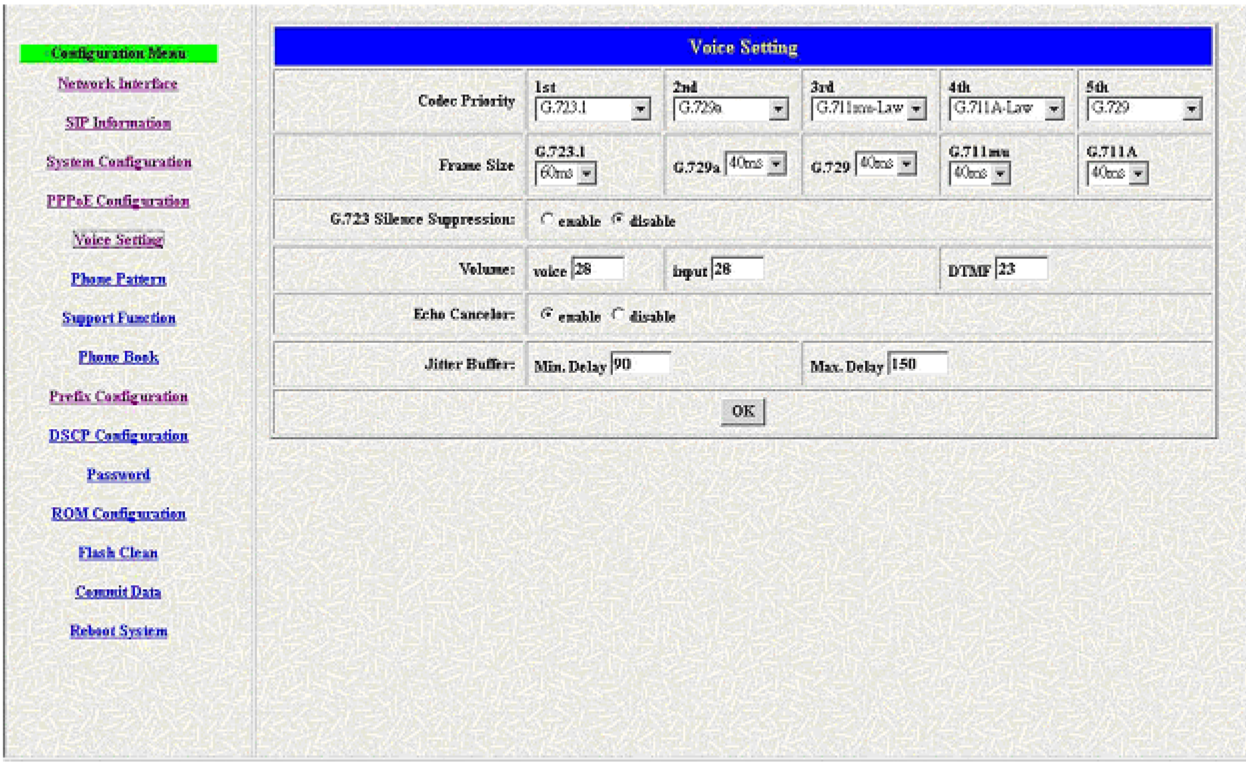
Click [PPPoE Configuration] in the navigation panel and open the [PPPoE Configuration] Screen.



- Device: Set PPPoe function to be On or Off.
- User Name: Set PPPoE authentication User Name.
- Password: Set PPPoE authentication password.
- Reboot After Remote Host Disconnection: Enable/Disable auto reboot after PPPoE disconnection
  If user enables this function, after PPPoE being disconnected, the gateway will automatically reboot to re-connect, and after reboot, if it still can't get contact with server, it will keep trying to connect. After re-connected, it will also restart system. On the other hand, if user disables this function, it won't reboot and keep trying to connect.
- PPPoE Echo Request: Enable or Disable PPPoE echo request function.
- Other items: for reference only, cannot allow to be configured.

4.3.5 Voice Configuration Screen

Click [Voice Configuration] in the navigation panel and open the [Voice Configuration] Screen.



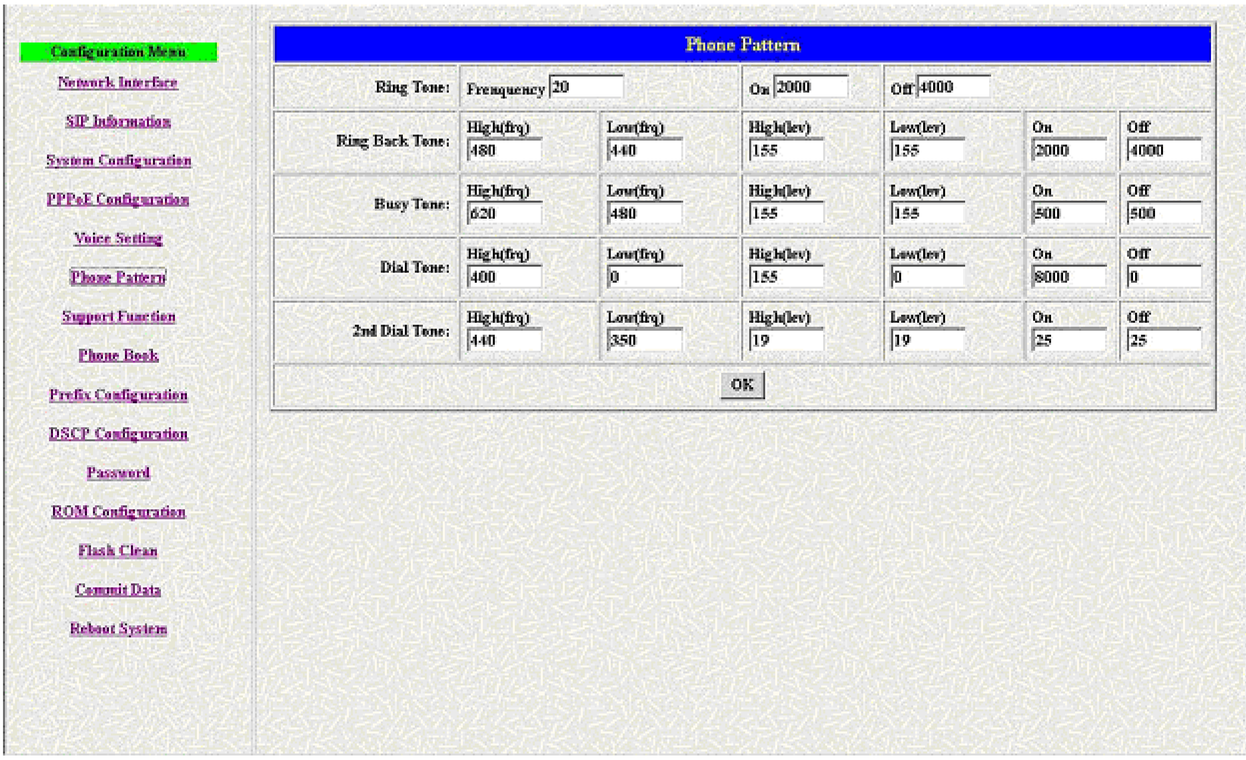
- Codec Priority: set codecs priority in order. Please notice that user can set from 1 to 5 codecs as their need. For example, user can only set first priority as G.723.1, and set the others as x, that means only G.723.1 is available.
- Frame Size: User can set different packet size for each codec.
- G.723 Silence Suppression: Enable / Disable sound compression and comfort noise generation. It is only for codec G.723.1
- Volume: Adjust the volume in "Voice" (sending out); "Input" (receiving); " DTMF" (DTMF sending out).
- Echo Cancelor: Enable / Disable (suggested always Enable this function).
- Jitter Buffer: Set Min. Delay and Max. Delay of Jitter Buffer for voice packets.

Note:

Well the application before you change voice parameters, because this might cause incompatibility.

4.3.6 Phone Configuration Screen

Click [Phone Configuration] in the navigation panel and open the [Phone Configuration] Screen. For tone simulation, the Gateway adopts dual frequencies as traditional telephone does. If users want to have their own call progress tone, they can change the value of tones.



- Ring Tone: Set Ring frequency, on time, off time, voltage level. The gateway will give ring to phone set to trigger the ring. If user found that phone set cannot ring when having incoming call, please try to increase ring frequency here.
  - ➢ ringing frequency: 15 ~ 100 (Unit: Hz)
  - ➢ ringing ring ON/OFF: 0 ~ 8000 (Unit: ms)
  - ➢ ringing level: 0 ~ 94 (Unit: V)
  - ➢ tone frequency: 0 ~ 65535 (Unit: Hz)
  - ➢ tone freqLevel: 0 ~ 65535 (Unit: mVrms)
  - ➢ tone Tone ON/OFF: 0 ~ 8000 (Unit: ms)

- Ring Back Tone: Set ring back tone parameters.
- Busy Tone: Set busy tone parameters.
- Dial tone: Set Dial tone parameters.
  - ➢ Low(frq) : Frequency value of Low frequency
  - ➢ High(frq) : Frequency value of High frequency

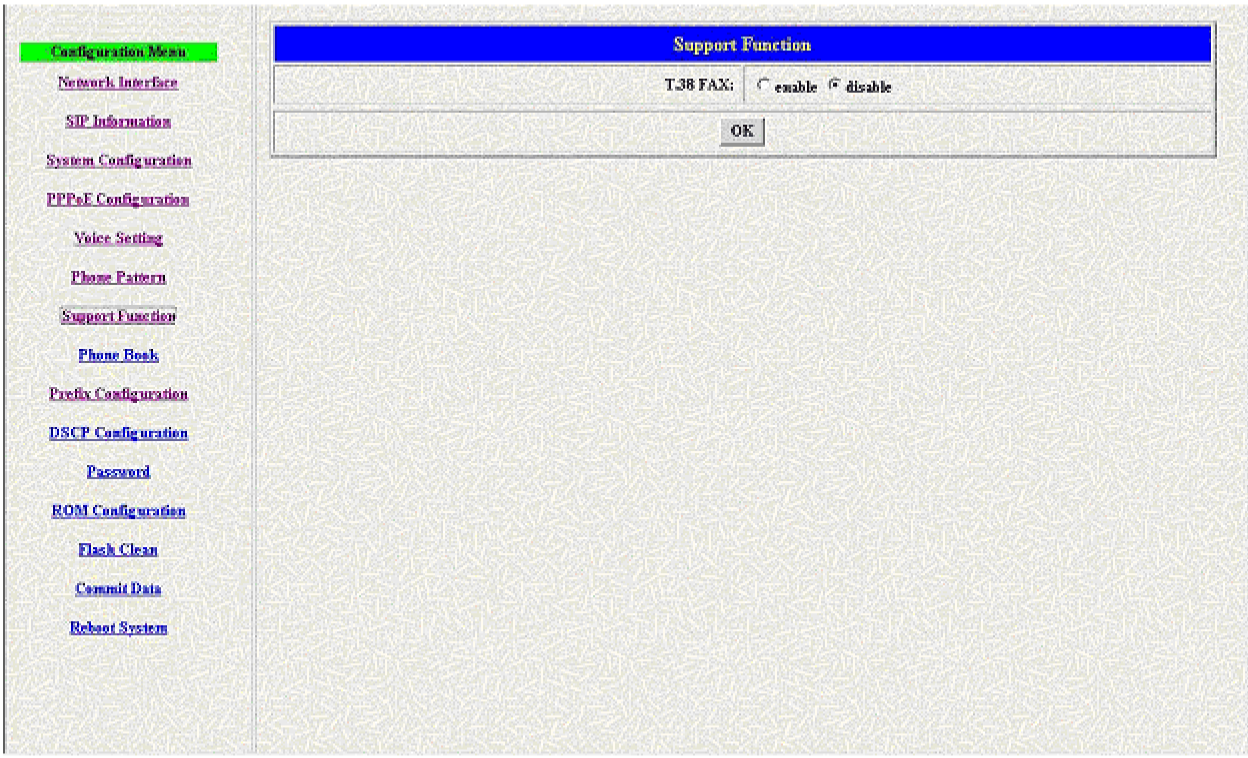
- Low(lev) : Level (volume) of Low frequency
- High(lev) : Level (volume) of High frequency
- On1 : On cadence of first cycle
- Off1 : Off cadence of first cycle
- On2 : On cadence of second cycle
- Off2 : Off cadence of second cycle

Note:

1. If disconnect tone is single-frequency, user has to configure the same frequency value of “Low frequency” and “High frequency”; the same level of “Low frequency” and “High frequency”
2. For On/Off cadence, user must set “1023” instead of “0”, if there is only one set of cycle, please as in second set columns

4.3.7 Support Configuration Screen

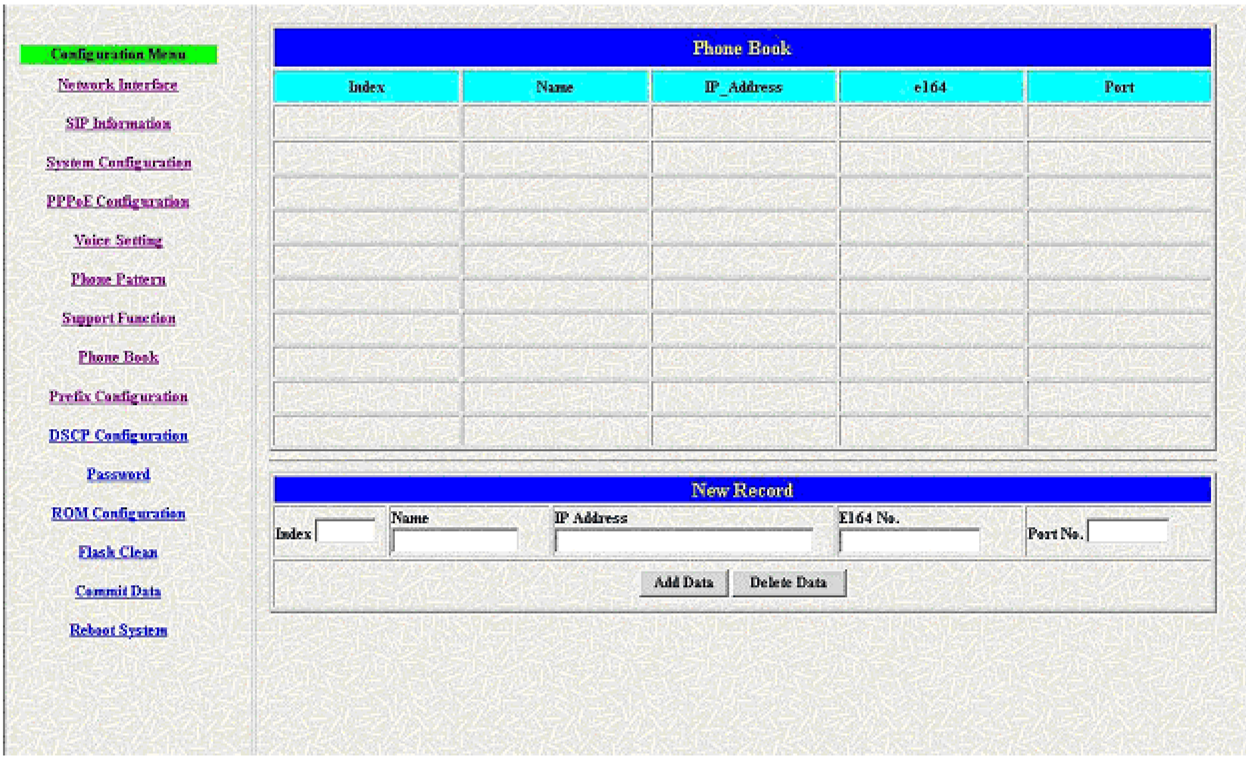
Click [Support configuration] in the navigation panel and open the [Support Configuration] Screen.



- T.38 FAX: enable/disable FAX function. If user wants to fax with the gateway, this function must be enabled.

4.3.8 Phone Book Configuration

Click [Phone Book Configuration] in the navigation panel and open the [Phone Book] Screen.



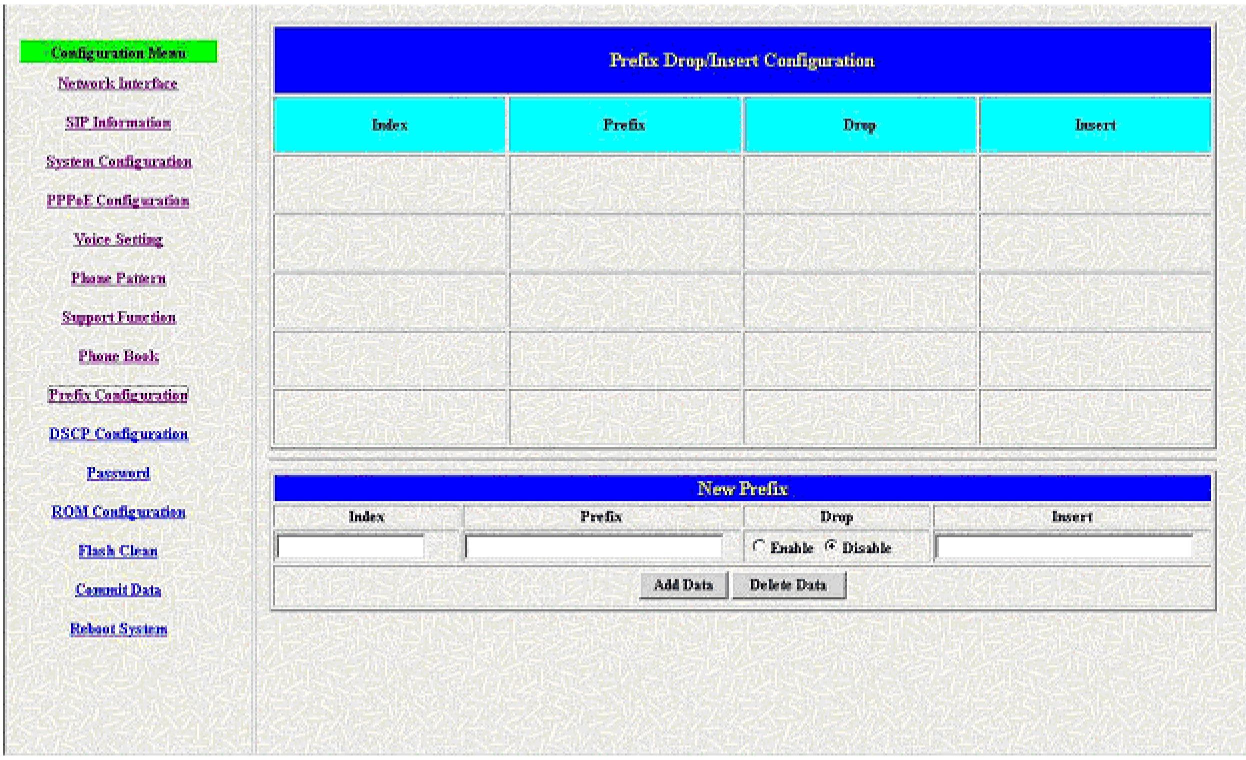
- Add Data: User can specify 10 sets of phone book via Web Management Interface. Please input index, Name, IP Address and E.164 number of the destination device.
- Delete Date: User can delete any configured phone book data by index.

Note:

The e164 number defined in phone book will be fully sent to destination. It is not just a representative number for destination's IP Address. In other words, user dial this e164 number to reach destination, destination will receive the number and find out if it is matched to its line number.

4.3.9 Prefix Configuration Screen

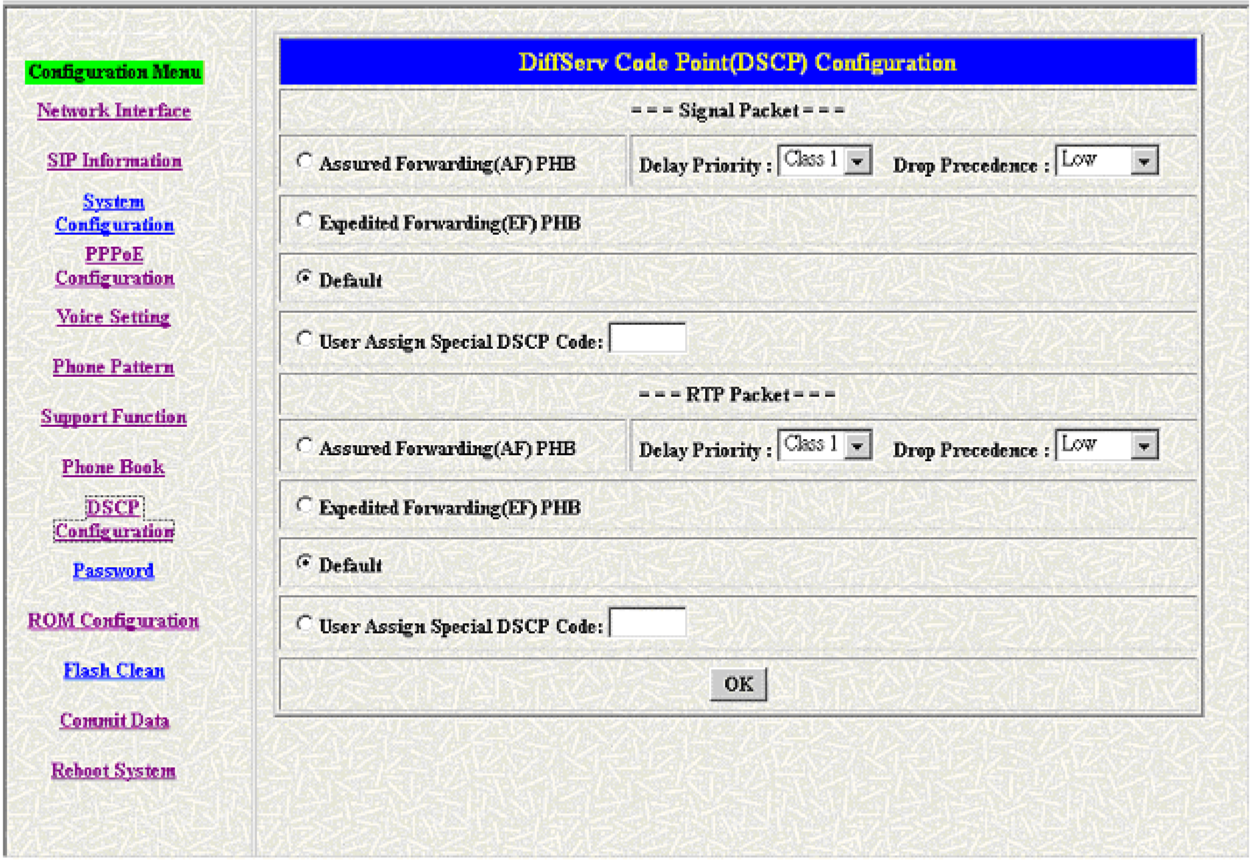
Click [Prefix configuration] in the navigation panel and open the [Prefix Drop/Insert Configuration] Screen.



- Index: Setting the index number for prefix record (max 30 record).
- Prefix: Setting the prefix number of the whole numbers that could be into this VoIP gateway (1~20 digits).
- Drop: Select enable or disable drop prefix function. The function is enabled means to drop prefix number when dialing out. The function is disabled means to keep prefix number.
- Insert: Setting the digits that you want to insert in this number (1~30 digits).

4.3.10 DSCP Configuration Screen

Click **DSCP Configuration** in the navigation panel and open the **DSCP Configuration** Screen.



Set Signal or RTP Packet DSCP value:

- Assured Forwarding (AF) PHB: Select Delay priority and Drop Precedence
- Expedited Forwarding (EF) PHB: Select TOS value as EF
- Default: Select TOS value as 0
- User Assign Special DSCP Code: User can set other unspecified value here.

TOS/DiffServ (DS) priority function can discriminate the Differentiated Service Code Point (DSCP) of the DS field in the IP packet header, and map each Code Point to a corresponding egress traffic priority. As per the definition in RFC2474, the DS field is Type-of-Service (TOS) octet in IPv4. The recommended DiffServ Code Point is defined in RFC2597 to classify the traffic into different service classes. The mapping of Code Point value of DS-field to egress traffic priorities is shown as follows.

DROP Precedence	Class #1	Class #2	Class #3	Class #4

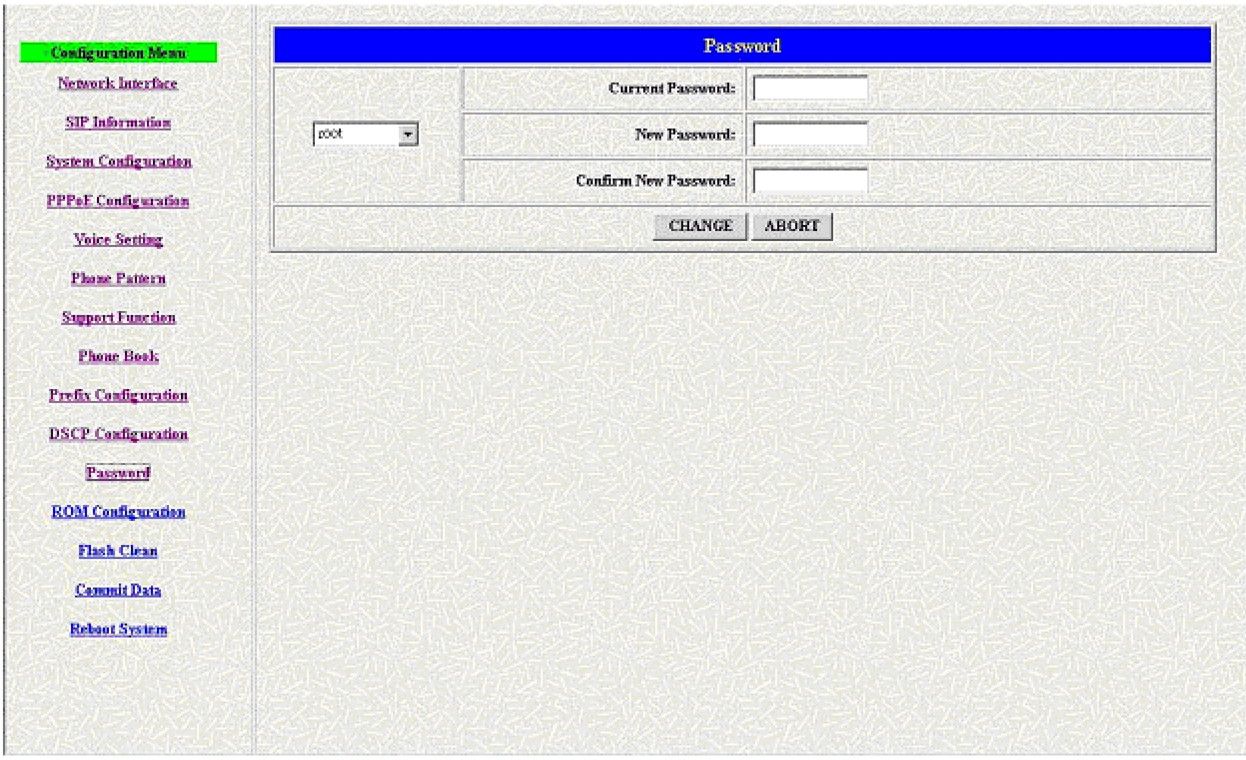
Low Drop Precedence	(AF11) 001010	(AF21) 010010	(AF31) 011010	(AF41) 100010
Medium Drop Precedence	(AF12) 001100	(AF22) 010100	(AF32) 011100	(AF42) 100100
High Drop Precedence	(AF13) 001110	(AF23) 010110	(AF33) 011110	(AF43) 100110

Please refer to RFC standard documents for more information about what is DSCP.

4.3.10 Password Configuration Screen

Click [Password configuration] in the navigation panel to open the [Password Configuration] screen.

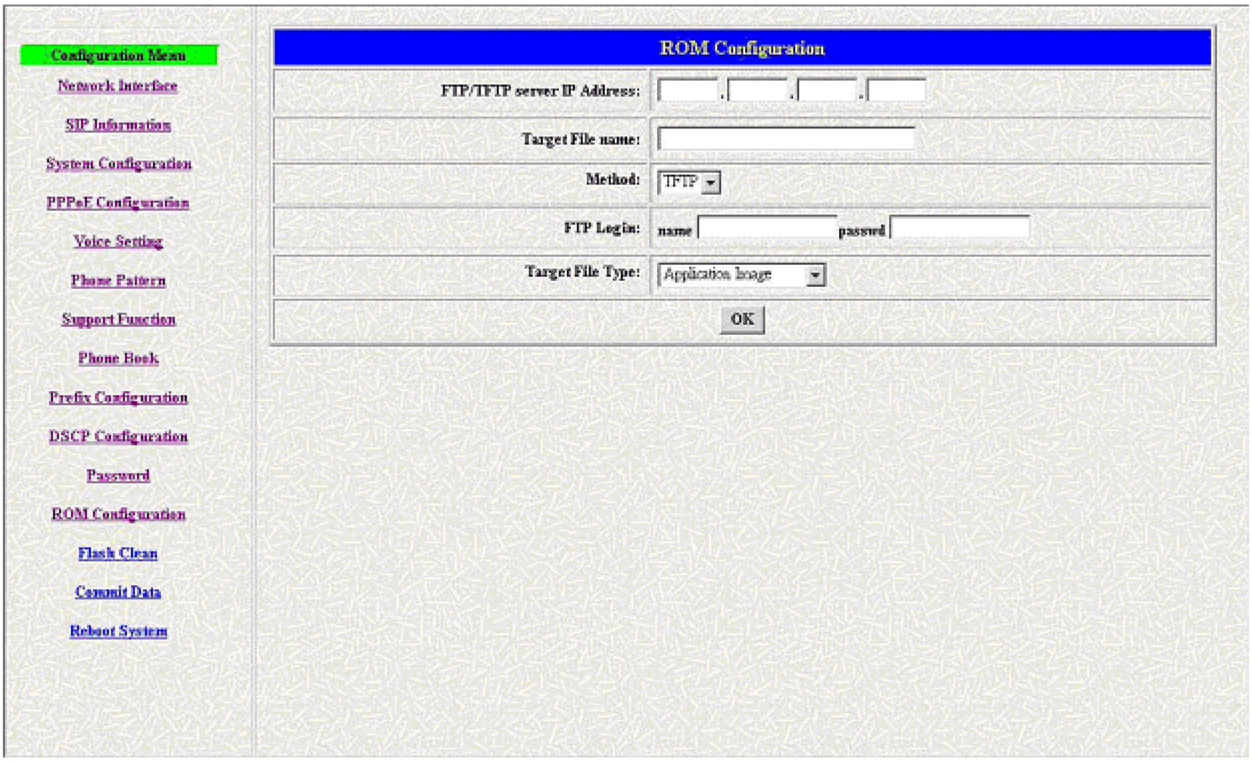
It is highly recommended that you change the default password ([Null]).



- Change: First select login name as root or administrator, then enter current password, new password and confirm new password again to set new password.
- Abort: Press abort will clean all inputs.

4.3.11 ROM Configuration Screen

Click [ROM Upgrade] in the navigation panel and open the [ROM Configuration] Screen.

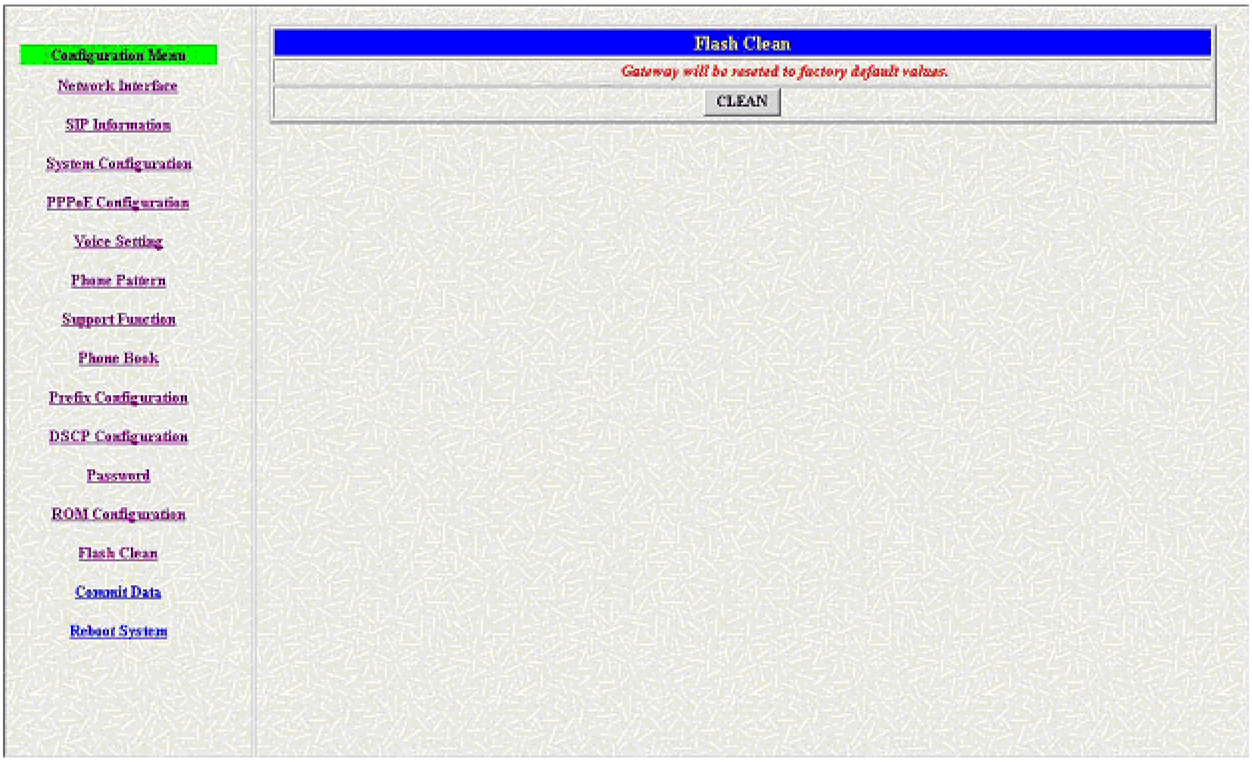


- FTP/TFTP Server IP Address: Set TFTP server IP address
- Target File name: Set file name prepared to upgrade
- Method: Select download method as TFTP or FTP
- FTP Login: Set FTP login name and password
- Target File Type: Select which sector of the gateway to upgrade

Note: After upgrading 2mb file or Application, please remember to execute Flash Clean, which will clean all configurations become factory values except IP address.

4.3.12 Flash Clean Screen

Click [Flash Clean] in the navigation panel and open the [Flash Clean] Screen.

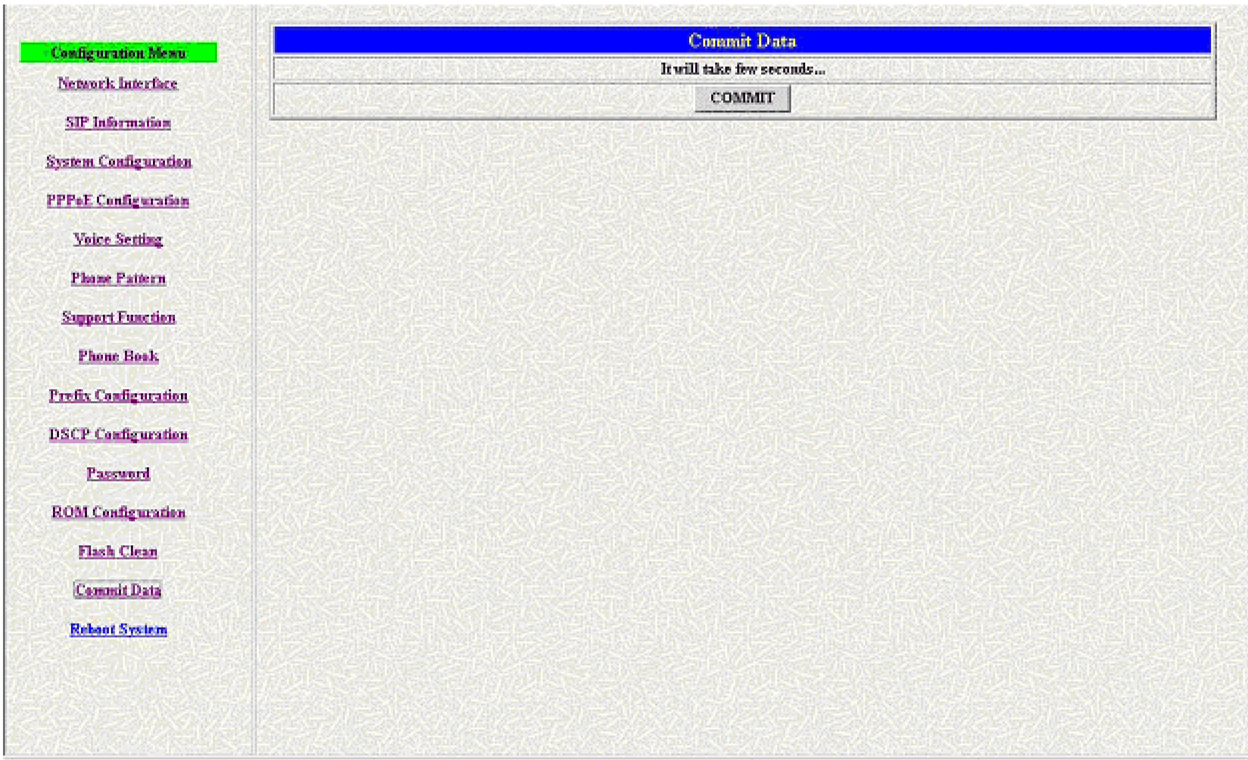


- Press CLEAN will clean all configurations of the gateway and reset to factory default value.

Note: User must re-configure all commands all over again (except Network Configure) once execute this function,

4.3.13 Commit Configuration Data Screen

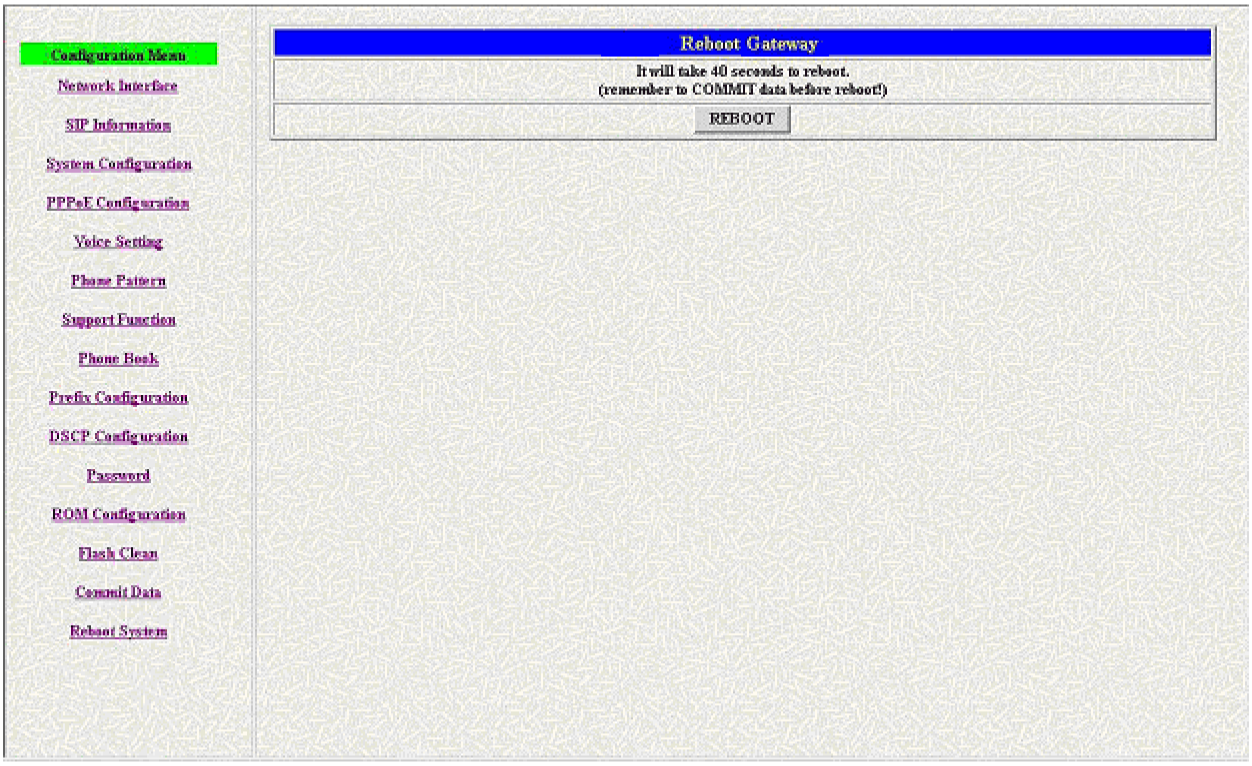
Click [Commit Data] in the navigation panel and open the [Commit Configuration Data] Screen.



- Commit Date to save all configurations. Please remember to commit data before reboot your gateway.

4.3.14 Reboot VoIP gateway System screen

Click [Reboot System] in the navigation panel and open the [Reboot Gateway] Screen.



- Press reboot to reset the gateway.

Note: To execute reboot, please remember to do **Commit Data** before **Reboot System**.

Part V: Telnet Command Interface

This part gives information on how to configure gateway via Telnet command line interface.

5.1 Login

For you first login, enter the login: [root] and default no password.

```
login: root
password:
Welcome to Terminal Configuration Mode
Please enter your configuration item

usr/config$
```

Note:
Login account [root] or [administrator] is the default login account and there is no password needed.

5.2 Save and Reboot

After any configuration has been made, user has to save all data and reboot system to make configurations take effect.

Step 1. Confirm the changed configurations, input [commit] and press [enter] key to save it.

Step 2. Input [reboot] then press [enter] key to restart Gateway.

Step 3. After around 40 seconds, Gateway will take effect in new configurations.

Do not turn off your Gateway or remove the Gateway while saving your configuration.

```
usr/config$ passwd -set root voip

Setting
Login: root
Password: voip
OK

usr/config$
```

5.3 System Commands Overview

5.3.1 [help]

Press help/man/ ? will display all command list of the gateway. The following table lists all of the commands that you can use with the Gateway. Refer to the following chapters for descriptions of commonly used commands.

This user's guide describes commands that are helpful for configuring the Gateway. Using commands not documented in the user's guide can damage the unit and possibly render it unusable.

VoIP Gateway Commands

Command	DESCRIPTION
help	Input help/man/? to list all command list.
quit	Input quit/exit/close to exit telnet connection.
debug	Add debug flag and display debug messages.
reboot	Reboot local machine.
commit	Save all data.
ifaddr	Internet address manipulation.
time	Show current time.
ping	Test if an IP address is reachable.
pbook	Phone book information and configuration.
pppoe	PPPoE parameters manipulation.
flash	Clean all configuration from flash rom.
sysconf	System information manipulation.
sip	Configure SIP related parameters.
security	This command is used to configure the account information included username and password obtained from the service provider.
voice	Voice information manipulation.
support	Special functions support manipulation.
tos	Set DSCP values for QOS.
phone	Setup of call progress tones and ring (SLIC control).
bureau	To set Hotline function which must be under Peer-to-Peer mode and switch to hotline service.
rom	ROM file update.
auth	Set configuration items for "administrator" user.

Command	DESCRIPTION
passwd	Password setting information and configuration.
prefix	Prefix drop/insert information manipulation

5.3.2 [quit]

Type [quit] will quit the Gateway configuration mode. And turn back to login prompt.

```
usr/config$ quit

Disconnecting..
login: root
Welcome to Terminal Configuration Mode
Please enter your configuration item

usr/config$
```

Note:

It is recommended that type the [quit] command before you leave the console. If so, Gateway will ask password again when next user connects to console port.

5.3.3 [debug]

Open debug message will show up specific information while Gateway is in operation. After executing the debug command, it should execute command [debug –open] as well.

```
usr/config$ debug

Debug message information and configuration
Usage:
debug [-add type1 [[type2]...]] | -open | -close | -status

     -status     Display the enabled debug flags.
     -add        Add debug flag.
     -delete     Remove specified debug flag.
     -open       Start to show debug messages.
     -close      Stop showing debug messages.
Example:
     debug -add sip msg
```

```
    debug -open

usr/config$
```

Parameter Usages:

-status: Display the enabled debug flags.

-add: Add debug flag.

-- sip: sip related information

-- msg: voice related information

-delete: Remove specified debug flag.

-open: Start to show debug messages.

-close: Stop showing debug messages.

In this example, user open debug flags including sip, vp, msg.

```
usr/config$ debug -add sip msg
usr/config$ debug –open
```

For example:

```
usr/config$ debug -status

Current debug type enabled :
Debug Mode is open
DEBUG-> SIP    MSG
usr/config$
```

5.3.4 [reboot]

After [commit], type [reboot] to reload Gateway in new configuration. The procedure is as below:

```
usr/config$ reboot

 Start to Unregister ...
 Unregister complete...
. Rebooting...It will take 40 seconds....Attached TCP/IP interface to cpm unit 0
Attaching interface lo0...done

HTTPD initialized...
Flash Check
 WorkMode : PROXY_MODE
 Start registering to Proxy server
```

```
 AC4804[0] is ok
 AC4804[1] is ok
 successful 2 2
 Initialize OSS libraries...OK!
VP v1.42 stack open sucessfully.

login:
```

5.3.5 [commit]

Save changes after configuring Gateway.

```
usr/config$ commit

This may take a few seconds, please wait..
Commit to flash memory ok!

usr/config$
```

Note:
Users shall use [commit] to save modified value, or they will not be activated after system reboot.

5.3.6 [ifaddr]

Configure and display Gateway network information.

```
usr/config$ ifaddr

LAN information and configuration
Usage:
ifaddr [-print]|[-dhcp used]|[-sntp mode [server]]
ifaddr [-ip ipaddress] [-mask subnetmask] [-gate defaultgateway]
ifaddr [-dns index [dns server address]] [-ipsharing used[ip address]]
ifaddr [-autodns used]

     -print       Display LAN information and configuration.
     -ip           Specify WAN ip address.
     -lanip       Specify LAN ip address.
```

```
      -mask        Set Internet subnet mask.
      -gate        Specify default gateway ip address
      -nat         Set NAT service flag (On/Off).
      -dhcp        Set DHCP client service flag (On/Off).
      -sntp        Set SNTP server mode and specify IP address.
      -autodns     Specify the way to obtain DNS Server (0:Manual/1:Auto).
      -dns         Specify IP address of DNS Server.
      -timezone    Set local timezone.
      -ipsharing Specify usage of an IP sharing device and specify IP address.
      -server      Specify EMS Server IP address
      -id          Specify EMS Server ID
      -pwd         Specify EMS Server password
      -emstime     Specify EMS cycle time
Note:
      Range of ip address setting (0.0.0.0 ~ 255.255.255.255).
      DHCP client setting value (On=1, Off=0). If DHCP set to 'On',
      Obtain a set of Internet configuration from DHCP server assgined.
      SNTP mode (0=no update, 1=specify server IP, 2=broadcast mode).
Example:
      ifaddr -ip 210.59.163.202 -mask 255.255.255.0 -gate 210.59.163.254
      ifaddr -nat 1
      ifaddr -dhcp 1
      ifaddr -sntp 1 210.59.163.254
      ifaddr -ipsharing 1 210.59.163.254
      ifaddr -autodns 1
      ifaddr -dns 1 168.95.1.1

usr/config$
```

Parameter Usages:

-print: Print current IP setting and status

-ip: Assigned IP address for Gateway

-lanip: Specify LAN port IP address (For NAT function), use this command setup LAN IP address assigned to PC or other machine.

```
usr/config$ ifaddr -lanip 192.168.XXX.YYY
(The range of LAN IP is XXX: 1-254, YYY: 1-254)
```

-mask: Assigned internet subnet mask

-gate: Assigned IP default gateway

-nat: Provide DHCP Server and NAT function.

-dhcp: Dynamic Host Configuration (1 = ON; 0 = OFF)

-dns: Setup DNS Server IP Address.

-sntp: Simple Network Time Protocol (0=No update, 1=Specify server IP, 2=broadcast mode). When SNTP function is activated, users have to specify a SNTP server as network time source. An example is demonstrated below while 10.1.1.1 stands for SNTP server's IP address:

```
usr/config$ ifaddr -sntp 1 10.1.1.1
```

-autodns: Auto or manual configures the DNS IP address when gateway device is under DHCP and PPPoE mode.

-timezone: set local time zone according to GMT

-ipsharing: To enable or disable IP sharing function. When this function is enabled, user must specify a public fixed IP address.

```
usr/config$ ifaddr -ipsharing 1 210.11.22.33
```

Note:
If the public IP address is not a fixed one, the gateway cannot work behind NAT with peer-to-peer mode.

-server: set EMS server IP address. EMS is software to help user can easily configure products. Please contact with your reseller for more information.

-id: specify EMS ID to login EMS Server.

-pwd: specify EMS password to login EMS Server.

-emstime: specify EMS cycle time.

For example:

```
usr/config$ ifaddr -print

Internet address information
      WAN IP address        : 192.168.13.71
      Subnet mask           : 255.255.248.0
      Default gateway       : 192.168.8.254
      NAT enabled           : OFF
      DHCP startup          : OFF
```

```
    SNTP                        : mode=1
                                  server 168.95.195.12
                                  time zone : GMT+8
                                  cycle=1024 mins

    IPSharing                  : no IPSharing device.

    Primary DNS Server     : 168.95.1.1
    Secondary DNS Server : 168.95.1.1

   EMS IP Address:   null
   EMS User ID     :   vwusr
   EMS Password   :   vwusr
   EMS cycle time:   0
usr/config$
```

5.3.7 [time]

When SNTP function of Gateway is enabled and SNTP server can be found as well, type [time] command to show current network time.

```
usr/config$ time
Current time is WED SEP 17 12:36:49 2003

usr/config$
```

5.3.8 [ping]

Use [ping] to test whether a specific IP is reachable or not.

For example: if 192.168.1.2 is not existing while 210.63.15.32 exists. Users will have the following results:

```
usr/config$ ping 192.168.1.2
no answer from 192.168.1.2
usr/config$ ping192.168.1.254
PING 192.168.1.254: 56 data bytes
64 bytes from 192.168.1.254: icmp_seq=0. time=5. ms
64 bytes from 192.168.1.254: icmp_seq=1. time=0. ms
64 bytes from 192.168.1.254: icmp_seq=2. time=0. ms
64 bytes from 192.168.1.254: icmp_seq=3. time=0. ms
----192.168.1.254 PING Statistics----
```

```
4 packets transmitted, 4 packets received, 0% packet loss
round-trIP (ms)    min/avg/max = 0/1/5
210.63.15.32 is alive
usr/config$
```

5.3.9 [pbook]

Phone Book function allows users to define their own numbers, which mapping to real IP address. It is effective only in peer-to-peer mode. When adding a record to Phone Book, users do not have to reboot the machine, and the record will be effective immediately.

```
usr/config$ pbook

Phonebook information and configuration
Usage:
pbook [-print [start_record] [end_record]]
pbook [-add [ip ipaddress] [name Alias] [e164 phonenumber]]
pbook [-search [ip ipaddress] [name Alias] [e164 phonenumber]]
pbook [-insert [index] [ip ipaddress] [name Alias] [e164 phonenumber] [port numb
er]]
pbook [-delete index]
pbook [-modify [index] [ip ipaddress] [name Alias] [e164 phonenumber] [port numb
er]]

      -print        Display phonebook data.
      -add          Add an record to phonebook.
      -search       Search an record in phonebook.
      -delete       Delete an record from phonebook.
      -insert       Insert an record to phonebook in specified position.
      -modify       Modify an exist record.
Note:
      If parameter 'end_record' is omited, only record 'start_record' will be disp
lay.
      If  both parameters 'end_record' and 'start_record' are omited, all records
will be display.
      Range of ip address setting (0.0.0.0 ~ 255.255.255.255).
      Range of index setting value (1~100),
Example:
```

```
    pbook -print 1 10
    pbook -print 1
    pbook -print
    pbook -add name Test ip 210.59.163.202 e164 1001
    pbook -insert 3 name Test ip 210.59.163.202 e164 1001
    pbook -delete 3
    pbook -search ip 192.168.4.99
    pbook -modify 3 name Test ip 210.59.163.202 e164 1001

usr/config$
```

Parameter Usages:

-print: Print out current contents of Phone Book. Users can also add index number, from 1 to 100, to the parameter to show specific phone number.

Note:
Index number: means the sequence number in phone book. If users do request a specific index number in phone book, Gateway will give each record a automatic sequence number as index.

-add: add a new record to phone book. When adding a record, users have to specify name, IP, and e164 number to complete the command.

--name: Name to represent caller.

--e164: e.164 number for mapping with IP address of caller

--ip: IP address of caller

--port: Call signal port number of caller

--drop : Drop e.164 number when dial out. 0 means to keep e.164 number, 1 means to drop e.164 number when dialing out.

--inert: Insert digits.(1~10 digits)

```
usr/config$ pbook –add name test e164 100 ip 192.168.13.78
```

-modify: modify an existing record. When using this command, users have to specify the record's index number, and then make the change.

```
usr/config$ pbook –modify 1 name test e164 5678 ip 192.168.1.10 port 1730 drop 0
```

-delete: delete a specific record. [pbook –delete 3] means delete index 3 record.

```
usr/config$ pbook –delete 3
```

PhoneBook Rules:

The e164 number defined in phone book will fully carry to destination. It is not just a representative number for destination's IP Address. In other words, user dial this e164 number to reach destination, destination will receive the number and find out if it is matched to its e164, including Line number in some particular device.

For example:

```
usr/config$ pbook -print

index     Name              IP                    E164              Port
==========================================================
1          74                192.168.13.74          74
-----------------------------------------------------
usr/config$
```

5.3.10 [pppoe]

Display PPPoE related information.

```
usr/config$ pppoe

PPPoE device information and configuration
Usage:
pppoe [-print]|[-open]|[-close]
pppoe [-dev on/off][-id username][-pwd password]

     -print       Display PPPoE device information.
     -dev          Enable(=1) or Disable(=0) device.
     -open         Open PPPoE connection.
     -close        Disconnect PPPoE connection.
     -id           Connection user name.
     -pwd          Connection password.
     -reboot       Reboot after remote host disconnection.
     -echo         PPPoE Echo Request (0=disable, 1=enable).
usr/config$
```

Parameter Usages:

-print: print PPPoE status.

-dev: Enable PPPoE Dial-up function

-open: Open the connection

-close: Close the connection

-id: Input the User name ID provided by ISP

-pwd: Input the User name password provided by ISP

-reboot: Reboot the PPPoE connection.

-echo: Enable or Disable PPPoE echo request function.

For example:

```
usr/config$ pppoe -print

PPPoE adapter information
        Device          : Enabled
        Status          : Not initialized
        User name       : pppoe
        Password        : *****
        Reboot          : No
        Echo            : Enable
usr/config$
```

5.3.11 [flash]

Clean the configuration stored in flash.

```
usr/config$ flash

Flash memory information and configuration
Usage:
flash [-clean]
flash -clean    Clean the configuration stored.

Note:
          This command will clean the configuration stored in
          the flash and reboot it.

usr/config$
```

Parameter Usages:

-clean: clean all the user defined value, and reboot Gateway in factory default mode.

Note:

1. It is recommended to execute [flash –clean] after application firmware been

upgraded.

2. Only User who login with root can execute it. After flash clean, all configurations in command [ifaddr] and [pppoe] will still be kept.

For example:

```
usr/config$ flash -clean

Flash clean start
Flash clean success!!


!! rebooting ...
Attached TCP/IP interface to cpm unit 0
Attaching interface lo0...done

HTTPD initialized...
Flash Check
 WorkMode : PROXY_MODE
 Start registering to Proxy server

 AC4804[0] is ok
 AC4804[1] is ok
 successful 2 2
 Initialize OSS libraries...OK!
VP v1.42 stack open successfully.

login:
```

5.3.12 [sysconf]

This command displays system information and configurations.

```
usr/config$ sysconf

System information and configuration
Usage:
 sysconf [-print] [-idtime digit] [-bf digit] [-keypad dtmf]
         [-faxtype type][-2833type type][-lcdrop ON/OFF]
```

```
            [-droptime digit][-eod digit] [-callerid type]
            [-service used][-dtmfstart digits] [-dtmfend digits]
 sysconf -print

 -print          Display system overall information and configuration.
 -idtime          Inter-Digits time.(1~10 sec)
 -service         Specify gateway service type. (0: Dial in service,
                    1: HotLine service.)
 -keypad           Select DTMF type: 0=In-band,
                                   1=RFC2833.
                                   2=INFO.
 -faxtype                  FAX    Payload    Type             (range:96~128
inter-used:100,102~105)
 -2833type                  RFC2833     Payload     Type     (range:96~128
inter-used:100,102~105)
 -lcdrop          Disconnect Supervision(Loop Current Drop) (ON:1 / OFF:0)
 -droptime        Period of Loop Current Drop (ms)
 -eod              End of Dial Digit setting(0: none, 1: *, 2: #)
 -callerid       Caller ID Type setting, 0: Disable,
                                         1: FSK(BELLCORE),
                                         2: DTMF,
                                         3: NTT.
 -dtmfstart      DTMF CallerID Start Symbol.
 -dtmfend         DTMF CallerID End Symbol.
Example:
   sysconf -keypad 0 -eod 2 -callerid 1

usr/config$
```

Parameter Usages:

-print: Print current sysconf settings.

-idtime: Set the duration (in second) of two pressed digits in dial mode as timed out. If after the duration user hasn't pressed next number, it will dial out all number pressed (1-10 seconds).

-service: set SIP Phone to be normal mode or under hotline mode. (sysconf –service 0/1, 0 for normal service, 1 for hotline service.)

-keypad: DTMF replay type. When value is "1", the Gateway will transfer DTMF signal via RTP payload as defined in RFC2833. When the value is set to "0", the DTMF type is set as In-band, and 2 for SIP info DTMF.

-faxtype: FAX Payload Type. range: 96~128 inter-used:100,102~105.

-2833type: RFC2833 Payload Type. Range: 96~128 inter-used: 100, 102~105.

-lcdrop: Disconnect Supervision (Loop Current Drop) (ON:1 / OFF:0).

-droptime: Period of Loop Current Drop (ms).

-eod: It will transfer the DTMF in [#] if users disable the end of dial function. Users have to press the keypad in [#] if the end of dial function is enable.

Note:

User can also define IP address here in P2P mode, once user press "#", Gateway will call out this IP address.

-callerid: Support Bell Core, DTMF caller ID and NTT caller ID function. After the first ring at destination site, device will send line number as caller ID to called site.

-dtmfstart: DTMF Caller ID Start Symbol.

-dtmfstart: DTMF Caller ID End Symbol.

For example:

```
usr/config$ sysconf -print

System information
        Gateway Service                  : 0
        Inter-Digits time                : 3
        BusyForward                      : OFF
        Keypad DTMF type                 : In-band
        End of Dial Digit                : #
        Caller ID Type                   : x
        DTMF Caller ID Start Symbol : D
        DTMF Caller ID End Symbol     : C
        RFC2833 Payload Type             : 96
        FAX Payload Type                 : 101
        Disconnect Supervision           : OFF
        Loop Current Drop Time(ms)    : 500
usr/config$
```

5.3.13 [sip]

This command is to configure SIP related parameters.

```
usr/config$ sip

SIP stack information and configuration
Usage:
sip [-print] [-mode   pxmode] [-outpx IPaddmress] [-transport type]
```

```
sip [-px address] [-px2 address] [-pxport number] [-outpxport number]
      [-line1 number]
      [-prefix prefixstring] [-expire t1] [-port udpPort] [-rtp rtpPort]
sip -print

      -print       Display SIP stack information and configuration.
      -mode        Configure as Peer-to-Peer mode:0/Proxy mode:1.
      -px          Primary Proxy server address. (IPv4 address or dns name)
      -px2         Secondary Proxy server address. (IPv4 address or dns name)
      -pxport      Proxy server port.     (the port of proxy)
      -outpx       OutBound Proxy server address. (IPv4 address or dns name)
      -outpxport OutBound Proxy server port. (the port of OutBound proxy)
      -prefix      Specify prefix string, use it when UserID contains alphabets
                     (if UserID uses numerals, specify as null)
      -line1       TEL1 Phone number.
      -pbsearch    Search phone book     0:off/1:on.
      -expire      The relative time after which the message expires(0 ~ (2^31-1))
      -port        SIP local UDP port number (5060~5070), Default: 5060
      -rtp         RTP port number (2326~65534), Default: 16384
Example:
       sip -mode 1
       sip -px 210.59.163.171 -line1 70

usr/config$
```

Parameter Usages:

-mode: Configure as Proxy mode or Peer-to-Peer mode (0: Peer-to-Peer mode, 1: Proxy mode).

-px: to specify Proxy address when the Gateway is in proxy mode. Proxy address can be IPv4 address or DNS name.

-px2: to setting Secondary Proxy server address. Proxy address can be IPv4 address or DNS name.

-pxport: Set Proxy port for the gateway to send message, default value is 5060, if there is no special request of Proxy server, please don’t change this value.

-outpx: Set IP Address or URL address (Domain Name Server must be configured. Please refer to Network Configure) of outbound Proxy server.

-outpxport: Set outbound Proxy port for SIP-Phone to send message, default value is 5060, if there is no special request of Proxy server, please don’t change this value.

-prefix: when your username contains alphabets, for example sip1123, then specify the prefix string as “sip”.

-line1: assign line 1 number.

–pbsearch: enable/disable phone book search function under Proxy Mode. If user enabled this function, the gateway will search dialed number in phone book to see if there is any matched table before send to Proxy server, and if there is a matched data in phone book, it will make call to related IP address.

-expire: this parameter is used to inform proxy server the valid duration of the registration information.

-port: SIP local UDP port which uses to listen incoming SIP Messages.

-rtp: Specify the RTP received port number.

Note: *One will need to configure port and rtp parameters only when you deploy two or more sets behind the IP sharing device (Router).*

For example:

```
usr/config$ sip –print

    Run Mode                 : PEER-2-PEER MODE
    Prefix string            : null
    Line1                    : 1001
    pbook search             : OFF
    SIP listen port          : 5060
    RTP receive port         : 16384
usr/config$
```

```
usr/config$ sip –print

    Run Mode                 : PROXY MODE
    Primary Proxy address    : 10.1.1.2
    Secondary Proxy address  : null
    Proxy port               : 5060
    OutBound Proxy address   : null
    Transport Type (TCP/UDP) : UDP
    Prefix string            : null
    Line1                    : 1001
    Line2                    : 1002
    Line3                    : 1003
    Line4                    : 1004
    pbook search             : OFF
```

```
      SIP listen port              : 5060
      RTP receive port               : 16384
      Expire                          : 3600
usr/config$
```

5.3.14 [security]

This command is used to configure the account information included username and password obtained from the service provider

```
usr/config$ security

Secuirty information and configuration
Usage:
 security [-line number][-name username] [-pwd password]
 security [-print]

 -print          Display system account information and configuration.
 -line           Specify which line number you want to set the account.
 -name              Specify user name.
 -pwd              Specify password.
Example:
   security -line 1 -name 1001 -pwd 1001

usr/config$
```

Parameter Usages:

-print: print current setting in security command.

-line: Specify which line number you want to set into the account

Note: *If you have only one account, you can set into line1 or line2 using this parameter. For example, if you set the account into line1, line1 can accept incoming calls.*

-name: Specify the username of your account information.

-pwd: Specify the password of your account information.

For example:

```
usr/config$ security -print

 Line1 account information
      Username      : 1001
```

```
    Password      : ***
usr/config$
```

5.3.15 [voice]

The voice command is associated with the audio setting information. There are four voice codecs supported by Gateway.

```
usr/config$ voice

Voice codec setting information and configuration
Usage:
voice [-send [G723 ms] [G711U ms] [G711A ms] [G729 ms] ]
        [-volume [voice level] [input level] [dtmf   level]]
        [-nscng [G711U used1] [G711A used2] [G723 used3]]
        [-echo used] [-mindelay t1] [-maxdelay t2]
voice -print
voice -priority [G723] [G711U] [G711A] [G729]

    -print       Display voice codec information and configuration.
    -send          Specify sending packet size.
                   G.723   (30/60 ms)
                   G.711U (20/40/60 ms)
                   G.711A (20/40/60 ms)
                   G.729   (20/40/60/80 ms)
    -priority   Priority preference of installed codecs.
                   G.723
                   G.711U
                   G.711A
                   G.729
    -volume        Specify the following levels:
                   voice volume (0~63, default: 25),
                   input gain (0~38, default: 25),
                   dtmf   volume (0~31, default: 23),
    -nscng         No sound compression and CNG. (G.723.1 only, On=1, Off=0).
    -echo          Setting of echo canceller. (On=1, Off=0, per port basis).
    -mindelay    Setting of jitter buffer min delay. (0~150, default: 90).
    -maxdelay    Setting of jitter buffer max delay. (0~150, default: 150).
Example:
    voice -send g723 60 g711u 60 g711a 60 g729 60
    voice -volume voice 20 input 32 dtmf 27
```

```
    voice -echo 1 1 1 1
usr/config$
```

Parameter Usages:

-print: Print current voice information and configurations.

-send: To define packet size for each codec. 20/40/60/80 ms mean to send a voice packet per 20/40/60/80 milliseconds. The smaller the packet size, the shorter the delay time. If network is in good condition, smaller sending packet size is recommended. In this parameter, 20/40/60ms is applicable to G.711u/a law, 20/40/60/80 ms is applicable to G.729 codec, while 30/60ms is applicable to G.723.1 codec.

-priority: Codec priority while negotiating with other SIP device. The codec listed in left side has the highest priority when both parties determining final codec.

```
usr/config$ voice –priority g723    (only select this codec)
usr/config$ voice –priority g723 g729 g711u g711a (select four codecs, and
g723 is the first choice)
```

-volume: There are three adjustable value.

- --voice volume stands for volume, which can be heard from Gateway side(range 0~63, default: 28).
- --input gain stands for volume, which the opposite party hears (range 0~38, default: 28).
- --dtmf volume stands for DTMF volume/level, which sends to its own Line (range 0~31, default: 23).

-nscng: Silence suppression and comfort noise generation setting (1 = ON; 0 = OFF). It is applicable to G.723 codec only.

```
usr/config$ voice -nscng g723 1
```

-echo: On or Off the activate each canceler.

-mindelay: The minimum jitter buffer size (Default value= 90 ms).

-maxdelay: The minimum jitter buffer size (Default value= 150 ms).

```
usr/config$ voice –mindelay 90 –maxdelay 150
```

> ***Note:***
> Be sure to know well the application before you change voice parameters because this might cause incompatibility.

For example:

```
usr/config$ voice -print
```

```
Voice codec setting relate information
     Sending packet size   :
               G.729A          : 40 ms
               G.723.1         : 60 ms
               G.711U          : 40 ms
               G.711A          : 40 ms
     Priority order codec :
               g729a g7231 g711u g711a
     Volume levels            :
               voice volume : 25
               input gain     : 25
               dtmf volume   : 23
 No sound compress & CNG :
               G.729A          : There is no setting
               G.723.1         : Off
               G.711(U-Law) : Off
               G.711(A-Law) : Off
     Echo canceller            : On On On On
     Jitter buffer             :
               Min Delay       : 90
               Max Delay       : 150
usr/config$
```

5.3.16 [support]

This command provides some extra functions that might be needed by users.

```
usr/config$ support

Special Voice function support manipulation
Usage:
support [-t38 enable]
          [-busy number] [-noanswer number] [-uncon number]
support -print
     -t38         T.38(FAX) enabled/disabled.
     -busy        Busy Forward number.(if empty, please fill "null")
     -noanswer No Anser Forward number.(if empty, please fill "null")
     -uncon       Unconditional Forward number.(if empty, please fill "null")
Example:
     support   -t38 1
     support   -busy 1001
```

```
      support   -uncon null

usr/config$
```

Parameter Usages:

-print: print current settings in support command.

-t38: Enable or disable T.38 fax ability. The function is will automatically defer codec (G.723 or G.729a) to T.38 when FAX signal is detected.

-busy: Provide setting busy forwrd to other number, when you setting this function. Then this channel busy, auto forward to setting phone number.

-noanswer: Provide setting noanser forwrd to other number, when you setting this function. Then this channel not answer, auto forward to setting phone number.

-uncon: Provide setting noanser forwrd to other number, when you setting this function. Then all call this channel number, will all auto forward to setting phone number.

> ***Note:***
> It is not recommended to change the value in this command, only if users do know well the application. This might cause incompatibility with other devices.

For example:

```
usr/config$ support -print

Special Voice function support manipulation
      T.38(FAX) support : Disabled
Forward Numbers
      Busy Forward number: 0123456789
      NoAnswer Forward number: 0212345678
      Uncondition Forward number:
usr/config$
```

5.3.17 [tos]

This command is for setting Differentiated Service Code Point configuration.

```
usr/config$ tos

IP Packet ToS(type of Service)/Differentiated Service configuration
Usage:
tos [-rtptype dscp]
tos [-sigtype dscp]
```

```
tos -print
      [-rtpreliab mode]
tos -print

Example:
     tos -rtptype 7 -sigtype 0
usr/config$
```

Parameter Usages:

-rtptype: the packages of voice (0~63).

-sigtype: the package of call signal (0~63).

Note:
The value of rtptype and sigtype is from 0 to 63. It's working if it supported by your network.

For example:

```
usr/config$ tos –print

IP Packet ToS information:
     Signalling Packet:
              DSCP Code : 0
     Media Packet        :
              DSCP Code : 0

usr/config$
```

5.3.18 [phone]

Gateway progress tone is configurable. Default tone value is set according to U.S. tone specification. Users may adjust the values according to their own country's tone specification or users-defined tone specification.

```
usr/config$ phone

Phone ringing , ringback tone , busy tone , dial tone setting and notes
Usage:

phone [-ring [freq     ] [ringON    ] [ringOFF ] [ringLevel]]
        [-rbt    [freqHi ] [freqLo    ] [freqHiLev] [freqLoLev]
```

```
                [Tone1ON] [Tone1OFF] [Tone2ON ] [Tone2OFF ]]
        [-bt    [freqHi ] [freqLo ] [freqHiLev] [freqLoLev]
                [Tone1ON] [Tone1OFF] [Tone2ON ] [Tone2OFF ]]
        [-dt    [freqHi ] [freqLo ] [freqHiLev] [freqLoLev]
                [Tone1ON] [Tone1OFF] [Tone2ON ] [Tone2OFF ]]
        [-flash [freqLo ] [freqHi ]]
        [-level [loopCurrentLevel] [onhookLineVoltageLevel ]]
phone [-print [ring]|[rbt]|[bt]|[dt]|[flash]]

        -print  Display phone ringing/tone configuration.
                 ring :   ringing
                 rbt  :   ringback tone
                 bt   :   busy tone
                 dt   :   dial tone
                 flash:   flash tone
        -ring   ringing configuration set .
        -rbt    ringback tone configuration set .
        -bt     busy tone configuration set .
        -dt     dial tone configuration set .
        -flash  flash configuration set .
        -level  Loop Current and On-Hook Line Voltage level set .
Note:
        ringing frequency    : 15 ~ 100   (Unit : Hz)
        ringing ring ON/OFF : 0  ~ 8000   (Unit : ms)
        ringing level        : 0  ~ 94    (Unit : V)
        tone    frequency    : 0  ~ 65535 (Unit : Hz)
        tone    freqLevel    : 0  ~ 65535 (Unit : mVrms)
        tone    Tone ON/OFF : 0  ~ 8000   (Unit : ms)
        level   loopCurrent : 0  ~ 7      (20mA ~ 41mA, Step : 3mA)
        level   OnHookVol    : 0  ~ 63    ( 0V ~ 94.5V, Step : 1.5V)
Example:
        phone -print rbt
        phone -ring 20 2000 4000 94
        phone -rbt 480 440 125 105 2000 4000 2000 4000
        phone -bt 620 480 125 105 500 500 500 500
        phone -dt 440 350 96 96 8000 0 8000 0
        phone -flash 400 800
        phone -level 1 32
```

```
usr/config$
```

Parameter Usages:

-print: Print current call progress tone configurations (ring: ring tone, rbt: ring back tone, bt: busy tone, dt: dial tone). This parameter should be accompanied with tone type.

-ring: To set RING tone value. The played tone type, when Gateway is receiving a call.

-rbt: To set RingBackTone value. The played tone type, when Gateway receives a Q.931 Alerting message. In condition that Gateway is the originate side.

-bt: To set BusyTone value. The played tone type, when destination is busy.

-dt: To set DialTone value. The played tone type, when hook off a phone set of workable Gateway.

-flash: Set the detective flash range in ms, for example, 300-500 ms.

> ***Note:***
> For tone simulation, Gateway adopts dual frequencies as traditional telephone does. If users want to have their own call progress tone, they can change the value of tones. High and Low frequency/level/cadence can be configured respectively.

For example:

```
usr/config$ phone -print rbt
Phone ringback tone paramter
     Ringback Tone frequency high            : 480
     Ringback Tone frequency low             : 440
     Ringback Tone frequency high level    : 155
     Ringback Tone frequency low level     : 155
     Ringback Tone tone1 on                  : 2000
     Ringback Tone tone1 off                 : 4000
     Ringback Tone tone2 on                  : 2000
     Ringback Tone tone2 off                 : 4000

usr/config$
```

```
usr/config$ phone -print rbt

Phone ring back tone paramter
     Ringback Tone frequency high            : 440
```

```
     Ringback Tone frequency low              : 480
     Ringback Tone frequency high level    : 13
     Ringback Tone frequency low level     : 13
     Ringback Tone tone1 on                     : 100
     Ringback Tone tone1 off                    : 200
     Ringback Tone tone2 on                     : 100
     Ringback Tone tone2 off                    : 200

usr/config$
```

```
usr/config$ phone -print bt

Phone busy tone paramter
     Busy Tone frequency high              : 620
     Busy Tone frequency low               : 480
     Busy Tone frequency high level    : 155
     Busy Tone frequency low level     : 155
     Busy Tone tone1 on                      : 500
     Busy Tone tone1 off                     : 500
     Busy Tone tone2 on                      : 500
     Busy Tone tone2 off                     : 500

usr/config$
```

```
usr/config$ phone -print dt

Phone dial tone paramter
     Dial Tone frequency high              : 440
     Dial Tone frequency low               : 350
     Dial Tone frequency high level    : 155
     Dial Tone frequency low level     : 155
     Dial Tone tone1 on                      : 8000
     Dial Tone tone1 off                     : 0
     Dial Tone tone2 on                      : 8000
     Dial Tone tone2 off                     : 0
usr/config$
usr/config$
```

```
usr/config$ phone -print flash

Phone flash paramter
      Flash frequency high : 800
      Flash frequency low    : 400

usr/config$
```

5.3.19 [bureau]

To set Hotline function.

```
usr/config$ bureau

Bureau line setting information and configuration
Usage:
bureau    [-hotline [Port DestIP TELnum]]
bureau -print

      -print         Display Bureau line information and configuration.
      -hotline       Set Hot line information. (Port range: 1~6)
Note:
      Hotline feature should be used together with:
            $sysconf -service 1 (HotLine service)
Example:
      bureau -hotline 1 192.168.4.69 628 2 192.168.4.200 999

usr/config$
```

Parameter Usages:

-print: Display current Hotline table.

-hotline: Define Line1 Hotline table respectively. The table is included [Line number], [destination IP Address] and [destination Port or Number].

For example

1. Destination is a FXS device, 628 is its Line1 number

usr/config$ bureau -hotline 1 200.168.4.69 628

User picks up the Line1, and then hears the ring back tone generated from destination. Of course, 628 are ringing simultaneously.

For example:

```
usr/config$ bureau -print

Bureau line setting relate information
      Hot line table
      ==========================================
      Port      Destination Address       Remote TEL
      ------------------------------------------
       1           192.168.13.78            629
      ==========================================

usr/config$
```

5.3.20 [rom]

ROM file information and firmware upgrade function.

```
usr/config$ rom

ROM files updating commands
Usage:
rom [-print] [-app] [-boot] [-dsptest] [-dspcore] [-dspapp]
      [-ht] [-method used] [-boot2m]
      -s TFTP/FTP server ip -f filename
rom -print
      -print       show versions of rom files. (optional)
      -app          update main application code(optional)
      -boot         update main boot code(optional)
      -boot2m       update 2M code(optional)
      -ht           updata Hold Tone PCM file(optional)
      -dsptest     update DSP testing code(optional)
      -dspcore      update DSP kernel code(optional)
      -dspapp       update DSP application code(optional)
      -s            IP address of TFTP/FTP server (mandatory)
      -f            file name(mandatory)
      -method       download via TFTP/FTP (TFTP: mode=0, FTP: mode=1)
      -ftp          specify username and password for FTP
      -server       specify EMS Server IP address
Note:
      This command can run select one option in 'app', 'boot',
      , 'dsptest', 'dspcore', and 'dspapp'.
```

```
Example:
    rom -method 1
    rom -ftp vwusr vwusr
    rom -app -s 192.168.4.101 -f app.bin

usr/config
```

Parameter Usages:

-print: show versions of all rom files.

-app, boot, boot2m, dsptest, dspcore, dspapp, ht: To update main Application program code, Boot code, DSP testing code, DSP kernel code, or DSP application code, and Hold Tone file.

Note:

Most of all, the Rom file needed to get upgrade is App or Boot2m. Please check the exactly Rom file before doing download procedure.

-s: To specify TFTP server's IP address when upgrading ROM files.

-f: To specify the target file name, which will replace the old one.

-method: To decide using TFTP or FTP as file transfer server. [0] stands for TFTP, while [1] stands for FTP.

-ftp: If users choose FTP in above item, it is necessary to specify pre-defined username and password when upgrading files.

- server: specify EMS Server IP address. Provide auto upgrade rom application verion, but you must use EMS Server it work.

For example:

```
usr/config$ rom -print


Download Method   :   TFTP
        Boot Rom   :   sdboot.200
Application Rom   :   1asipIAD.107
         DSP App   :   48302ce3.140
      DSP Kernel   :   48302ck.140
   DSP Test Code   :   483cbit.bin

       Hold Tone   :   holdtone.101

usr/config$
```

5.3.21 [auth]

For security concern, the "root" user can customize some configurable items for "administrator" user.

```
usr/config$ auth

Root control what command administrator can use.
Usage:
auth -print Display auth switch configuration.
                    Use item name to do config name (0=Disable, 1=Enabled).
Example: auth -ifaddr 1

usr/config$
```

Parameter Usages:

-"item name": Assign the configurable item for "administrator" user.

```
usr/config$ auth -ifaddr 1
usr/config$ auth -h323 1
usr/config$ auth -voice 1
```

Now the Administrator user can use the command which Root user assigned.

-print: Display the configurable items for "administrator" user.

For example:

```
usr/config$ auth -print

Root can control what command administrator can use.
      ifaddr     : Enable
      sip        : Disable
      line       : Disable
      pbook      : Enable
      support    : Disable
      sysconf    : Disable
      voice      : Disable
      phone      : Disable
      rtp        : Disable
      tos        : Disable
      prefix     : Disable
      passwd     : Enable
      rom        : Disable
      flash      : Disable
```

```
      bureau     : Enable
      pppoe      : Enable

usr/config$
```

5.3.22 [passwd]

For security concern, users have to input the password before entering configuration mode. [passwd] command is for password setting purpose.

```
usr/config$ passwd

Password setting information and configuration
Usage:
passwd [-set [Login name] [Password]][-clean]

passwd -set      Loginname Password.
          -clean    Clear all password stored in flash.
Note:
    1. Loginname can be only 'root' or 'administrator'
    2. passwd -clean will clear all passwd stored in flash,
        please use it with care.
Example:
    passwd -set root Your_Passwd_Setting
    passwd -clean

usr/config$
```

Parameter Usages:

–set: Set login name and password, input login name then input new password.

-clean: Will clear all password setup, and change null.

Note: Gateway Login name only use [root] or [administrator]. [root] and [administrator] have the same authorization, except commands that can be excuted by [Login name: root] only [passwd –set root], [rom –boot], [room-boot2m] and [flash –clean].

For example:

```
usr/config$ passwd -set root root1234
```

```
Setting
login: root
Password: root1234
OK
usr/config$
```

```
sr/config$ passwd -clean

Please wait a moment!!

Clean password OK.
usr/config$
```

5.3.23 [prefix]

This command is for make rules for drop or insert prefix digits.

```
usr/config$ prefix

Prefix drop/insert information and configuration
Usage:
prefix -add [prefix number][drop number][insert digits]
prefix -delete index
prefix -modify index [prefix number][drop number][insert number]
prefix -print     Prefix drop/insert information.
         prefix     The prefix of dialed number.
         drop         Drop prefix(Enable:1/Disable:0).
         insert     Insert digits.
Example:
       prefix -add prefix 100 drop 1 insert 2000
       prefix -add prefix 100 drop 1
       prefix -add prefix 100 drop 0 insert 200
       prefix -delete 1
       prefix -modify 1 prefix 100 drop 0 insert 300

usr/config$
```

Parameter Usages:

-add: Add a rule to drop or insert prefix digits of incoming call.

--prefix: Set which prefix number to implement prefix rule.

--drop: Enable or disable drop function. If this function is enabled, Gateway will drop prefix number on incoming call.

--insert: Set which digit to insert on incoming call.

```
usr/config$ prefix -add prefix 100 drop 1 insert 2000
```

-modify: Modify a rule to drop or insert prefix digits of incoming call.

```
usr/config$ prefix –modify 100 drop 0 insert 200
```

-delete: Delete a rule to drop or insert prefix digits of incoming call.

```
usr/config$ prefix –delte modify 100 drop 0 insert 200
```

For example:

```
usr/config$ prefix -print

Prefix drop/insert information and configuration
Index   Prefix                    Drop      Insert
==========================================================
1       100                        Enable    2000

usr/config$
```