ViP 3001
ViP 3002

IP Telephony Adapters

User's Manual

aethra®
TELECOMMUNICATIONS

Contents

1. Overview

The ViP 3001 and 3002 are external standalone devices, that can provide cost effective voice communication over an IP Network. The ViP 3001 and 3002 voice gateways are available in one and two channel models (respectively). Both models connect directly to analog phones, fax machines, PBX, and the IP Networks without extra equipment and setup. With the Ethernet interface of the voice gateways connected to another device with a WAN interface (e.g. xDSL, cable modem...), the ViP 3001 and 3002 voice gateways can provide toll quality voice communication in terms of voice quality and reliability for the user.

1.1 Package Contents

Carefully unpack the shipping package containing the voice gateway and make sure that you have the following items. If you find anything missing, mismatched or damaged, promptly contact your dealer who you purchased your product from for assistance.

- One VoIP Residential Gateway
- One RJ-11 telephone line for first telephone
- One RJ-45 Ethernet cable
- One power adapter
- One user's manual

1.2 System Requirements

- One RJ-45 Broadband Internet connection via cable modem or ADSL modem
- One PC with 10Mbps, 100Mbps, or 10/100 Mbps Ethernet card installed
- TCP/IP protocol for each PC
- Microsoft Internet Explorer 4.0 or later (5.0 is strongly recommended for web configuration)
- One or two standard touch-tone telephone(s)
- Subscribe to a VoIP service provider for VoIP services

1.3 Features

- The (ViP 3002) supports simultaneous voice and data communications, two simultaneous G.711 μ, G.729a, G.723 voice conversations via the FXS ports, as well as IP packet transmission through the Ethernet port
- A 10/100 Base-T Ethernet port allows connecting to broadband access media, such as cable modem or ADSL modem
- Manual/automatic selection between PSTN and IP network for VoIP phone
- Two RJ-11 Foreign Exchange Station (FXS) ports for IP telephony (ViP 3002)

- One RJ-11 port for PSTN backup use
- Supports DHCP client in WAN port
- Supports PPPoE function
- Supports SNMP network management
- Remote software upgrade
- Web-based configuration
- Supports Plug & Play
- High transmission throughput

2. Hardware Installation

2.1 Front View (LEDs)

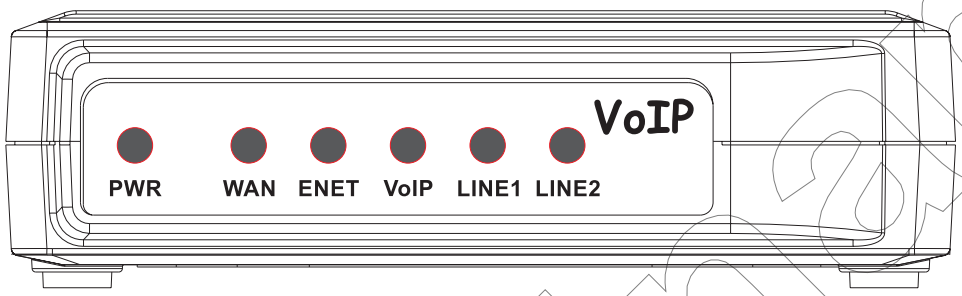


Figure 1: The front panel of the *ViP 3001/2* Voice Gateways

LED	Color	Status	Description
PWR	Green	On	When the ViP 3001 and 3002 Voice Gateways is powered on
		Off	No power supply
WAN	Green	Blinking	When data is being transmitted or received
		On	When Ethernet connection is established
		Off	When there is no ethernet connection
LAN	Green	Blinking	When data is being transmitted or received
		On	When Ethernet connection is established
		Off	When there is no Ethernet connection
VoIP	Green	On	When VoIP telephone service is ready
		Off	When VoIP telephone service is not ready
LINE1 & LINE2	Green	Blinking	When there is an incoming call (the telephone is ringing)
		On	When the telephone is in use
		Off	Switches to PSTN back-up line

2.2 Rear View (Ports)

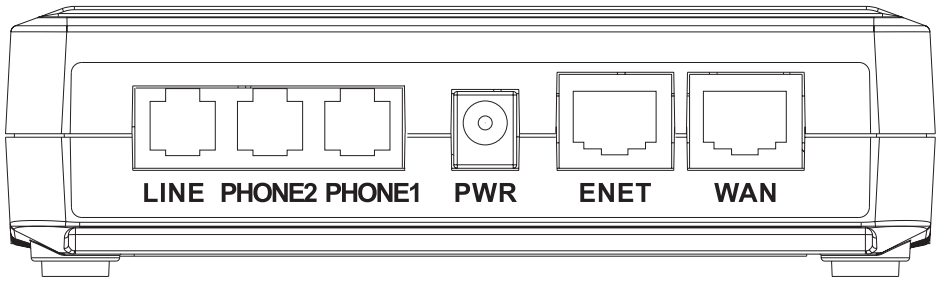


Figure 2: The rear panel of the ViP 30002 Voice Gateways

- **LINE:** RJ-11 connector, connected to PSTN back-up line
- **PHONE1 & PHONE2:** RJ-11 connectors, connected to telephones or fax machine
- **PWR:** Power connector, connected to the power adapter packaged with the VoIP Gateway
- **ENET:** Ethernet RJ-45 connector, connected to PC using a RJ-45 Ethernet Cable
- **WAN:** Ethernet RJ-45 connector, connected to WAN access device, such as the cable modem or ADSL modem

2.3 Installation Procedure

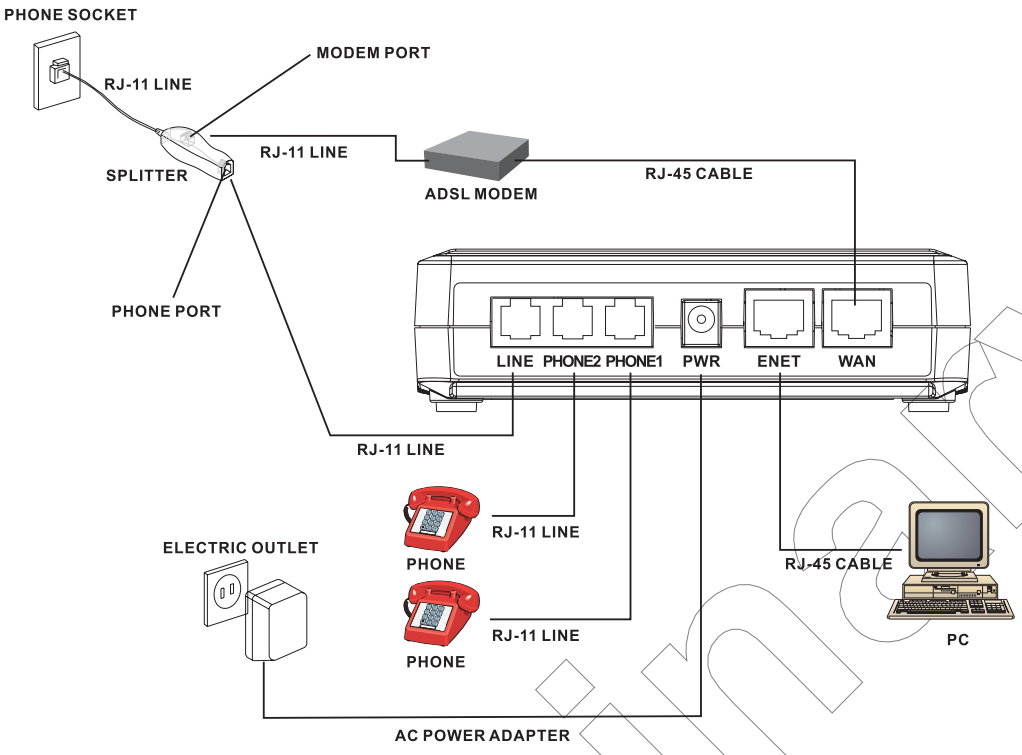


Figure 3: Typical Application

1. **LINE:**
   Plug one end of the RJ-11 telephone line into the LINE port and plug the other end into the phone port of the splitter. Then connect the splitter to the phone socket in the wall using a RJ-11 telephone line.

   The LINE port is for back-up use. The telephone is using VoIP service by default. However, if the ViP 3001 and 3002 Voice Gateways loses WAN connection or the VoIP function is not available, the ViP 3001 and 3002 Voice Gateways will make the telephone to use PSTN (Public Switched Telephone Network) service.

2. **PHONE1 & PHONE2:**
   Plug one end of the RJ-11 telephone line into the PHONE1 or PHONE2 port and plug the other end into the phone socket on a telephone set.

3. **PWR:**
   Plug one end of the power adapter into the PWR port and plug the other end into an electric outlet in the wall.

4. **ENET:**
   Plug one end of the RJ-45 Ethernet cable into the ENET port and plug the other end into the Ethernet socket of NIC on your PC.

5. **WAN:**
   Plug one end of the RJ-45 Ethernet cable into the WAN port and plug the other end into the Ethernet port of the Internet service device, such as the cable modem or ADSL modem. Then connect the cable modem or ADSL modem to the modem port of the splitter using a RJ-11 telephone line.

3. Configuring TCP/IP Protocol for Your PC

To communicate with and configure this device, each PC on your LAN must install TCP/IP protocol. If you enable static IP addressing, make sure your PC resides in the same subnet with this device's LAN port (default IP Address: 192.168.100.1, default subnet mask: 255.255.255.0).

3.1 Using Windows 98/ME

1. From the **Start** menu, click **Settings,** and then click **Control Panel**.
2. Double-click **Network**.
3. On the **Configuration** tab, check if TCP/IP protocol is installed on the components list.
4. If yes, go to Step 8. If no, then click **Add**.
5. Highlight **Protocol** and click **Add**.
6. Select **Microsoft** from the Manufactures list and select **TCP/IP** from the Network Protocols list.
7. Click **OK**. You will see **TCP/IP** displayed on the network components list.
8. Highlight **TCP/IP** and click **Properties**.
9. Select the **IP Address** tab and check **Specify an IP address**.
10. Set **IP address** as 192.168.100.100, **Subnet mask** as 255.255.255.0 and press **OK**.

3.2 Using Windows 2000/XP

1. From the **Start** menu, click **Settings,** and then click **Network and Dial-up Connections**.
2. Double-click **Local Area Connection**.
3. Click **Properties**.
4. Click **Internet Protocol (TCP/IP)** and then click **Properties**.
5. Check **Use the following IP address.**
6. Set **IP address** as 192.168.100.100, **Subnet mask** as 255.255.255.0 and press **OK**.

3.3 Using Windows NT

1. From the **Start** menu, click **Settings,** and then click **Control Panel**.
2. Double-click **Network**.
3. On the **Protocol** tab, check if TCP/IP protocol is installed on the components list.
4. If yes, go to Step 7. If no, then click **Add**.
5. Highlight **TCP/IP Protocol** and click **OK**.
6. When asked to use DHCP, click **No**.
7. Select **TCP/IP Protocol** and click **Properties**.
8. When Information Message appears, click **OK**.
9. On the **IP Address** tab, check **Specify an IP address.**

10. Set **IP address** as 192.168.100.100, **Subnet mask** as 255.255.255.0 and press **OK**.
11. When asked to restart your computer, click **Yes**.

3.4 Using Linux

1. In the command line interface, enter **netconf**.
2. Highlight and click **Host name and IP network devices**.
3. Set **IP address** as 192.168.100.100, **Subnet mask** as 255.255.255.0.
4. Highlight and click **Accept** to save the configuration.

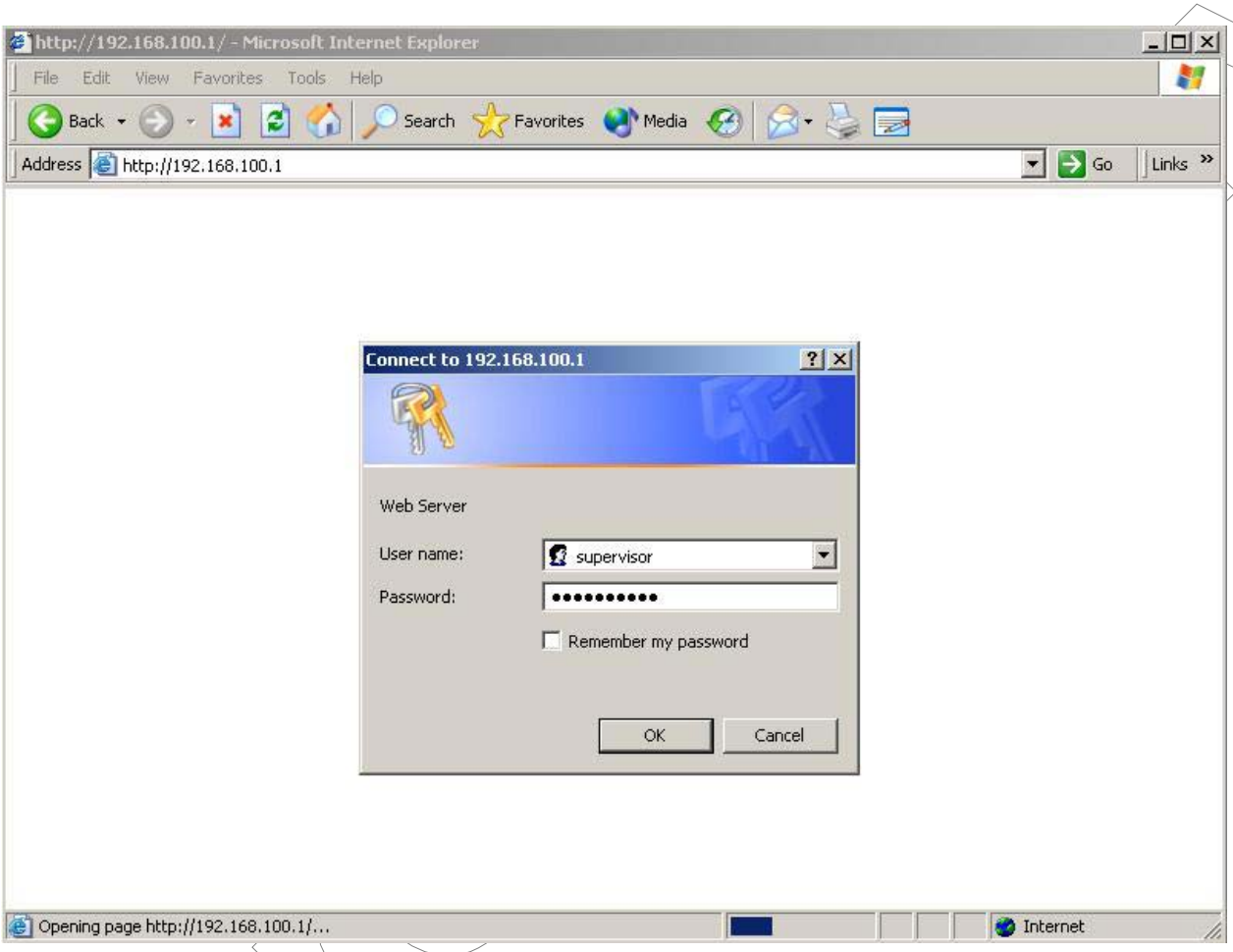
4. Configuring the ViP 3001 and 3002 Voice Gateways

The ViP 3001 and 3002 Voice Gateways implement an embedded Web server allowing you to check the connection status and configure this device via the Web interface.

3.5 Configuring via Web Browser

Once your PC is properly configured, please proceed the following steps:

1. Start the web browser.
2. Enter the default IP address **192.168.100.1** of this device in the Address box to access the web configuration menu.



3. The web configuration menu provides two operation modes: the user mode and the supervisor mode. The web configuration menu varies according to the different operation mode.

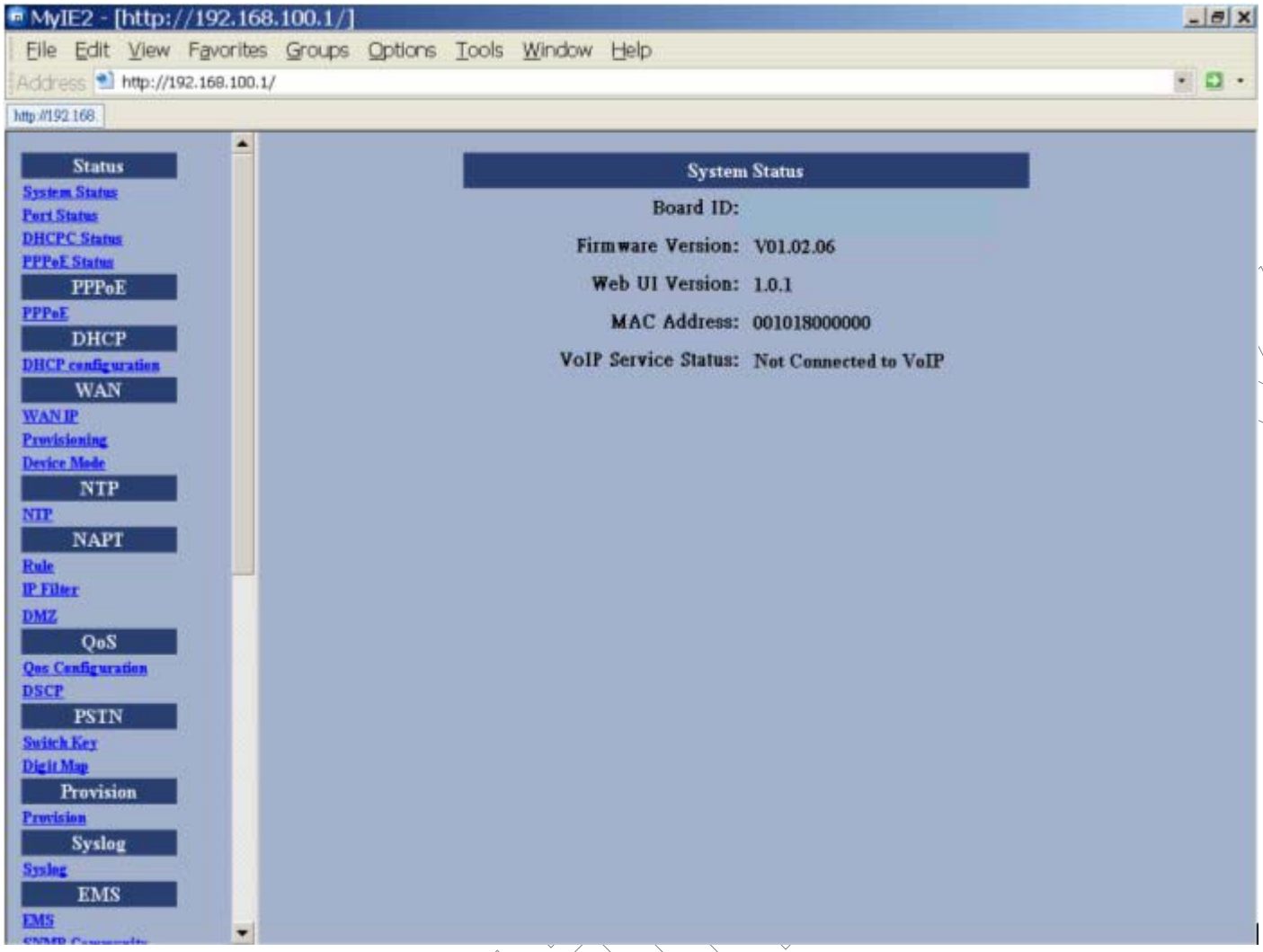
   When the following window pops up, enter the predefined user name as **Supervisor** and password as **aethra (12345 after sw reset)** and then press **OK** key.

4. On the following pages you will find a brief generic description and a more detailed description on additional functions for each web page. To apply any settings you've altered on any page, click **OK**. To clear any values you've entered on any page, click **Reset**. Changing to another screen without clicking **OK** does not save any settings you have made. Remember to click **OK** before browsing screens or your configuration will be ignored.

   After making all necessary settings, you need to save the configurations and then restart the VoIP Gateway to make the new settings take effect.

3.5.1 System Status (Supervisor):

When you first connect. a setup screen, will be displayed as in the example below:

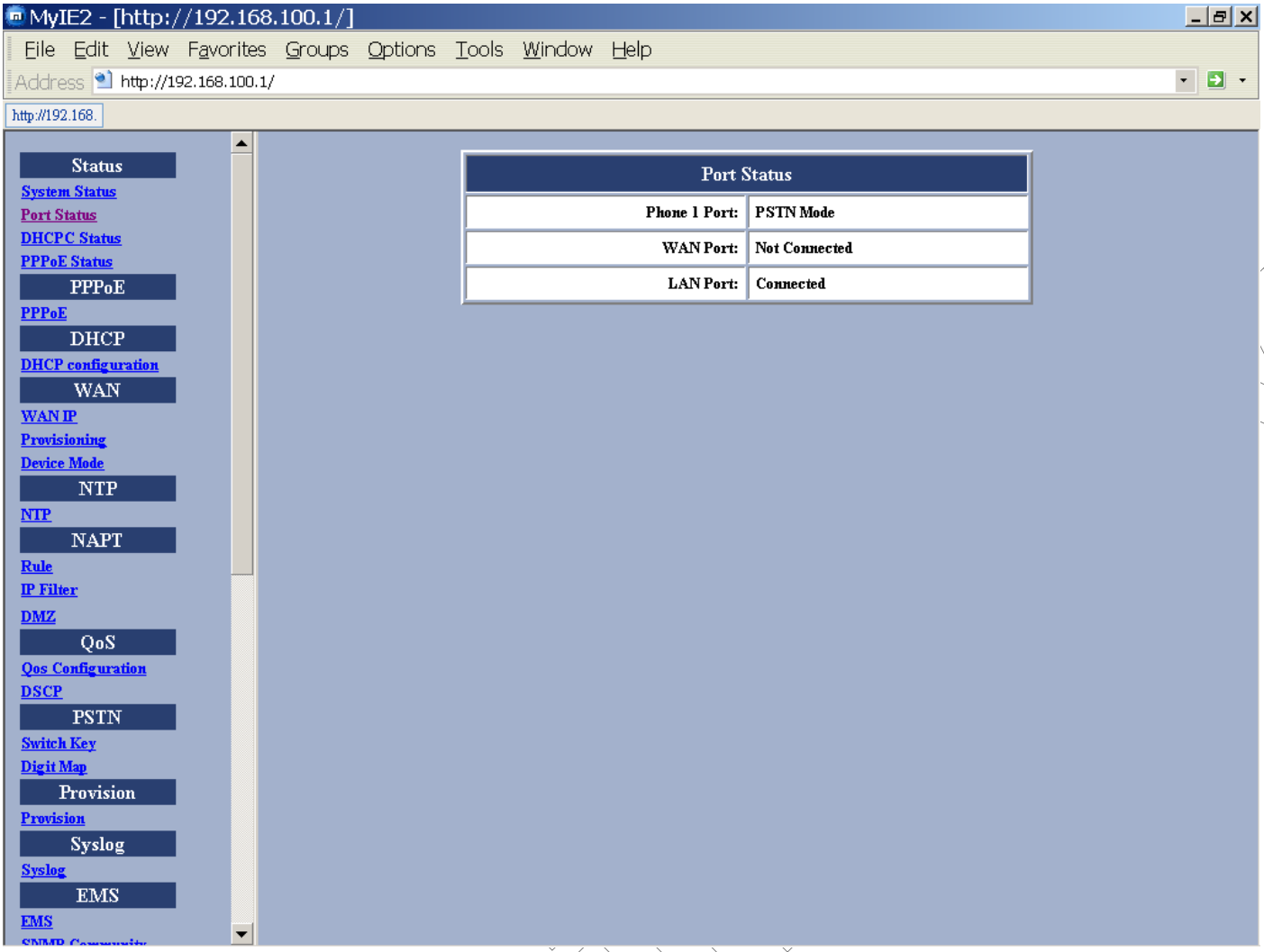


This screen contains Board ID, Firmware Version, Web UI Version, Build Date and MAC Address.

Item	Description
Board ID	This number is used to identify different customers for their respective HW/PCB design.
Firmware Version	Specifies the installed firmware version.
Web UI Version	Specifies the current Web UI version.
Build Date	Specifies the date to establish the Web UI.
MAC Address	Specifies the unique hardware number of the ViP 3001 and 3002 VoIP Gateways.

3.5.2 Port Status:

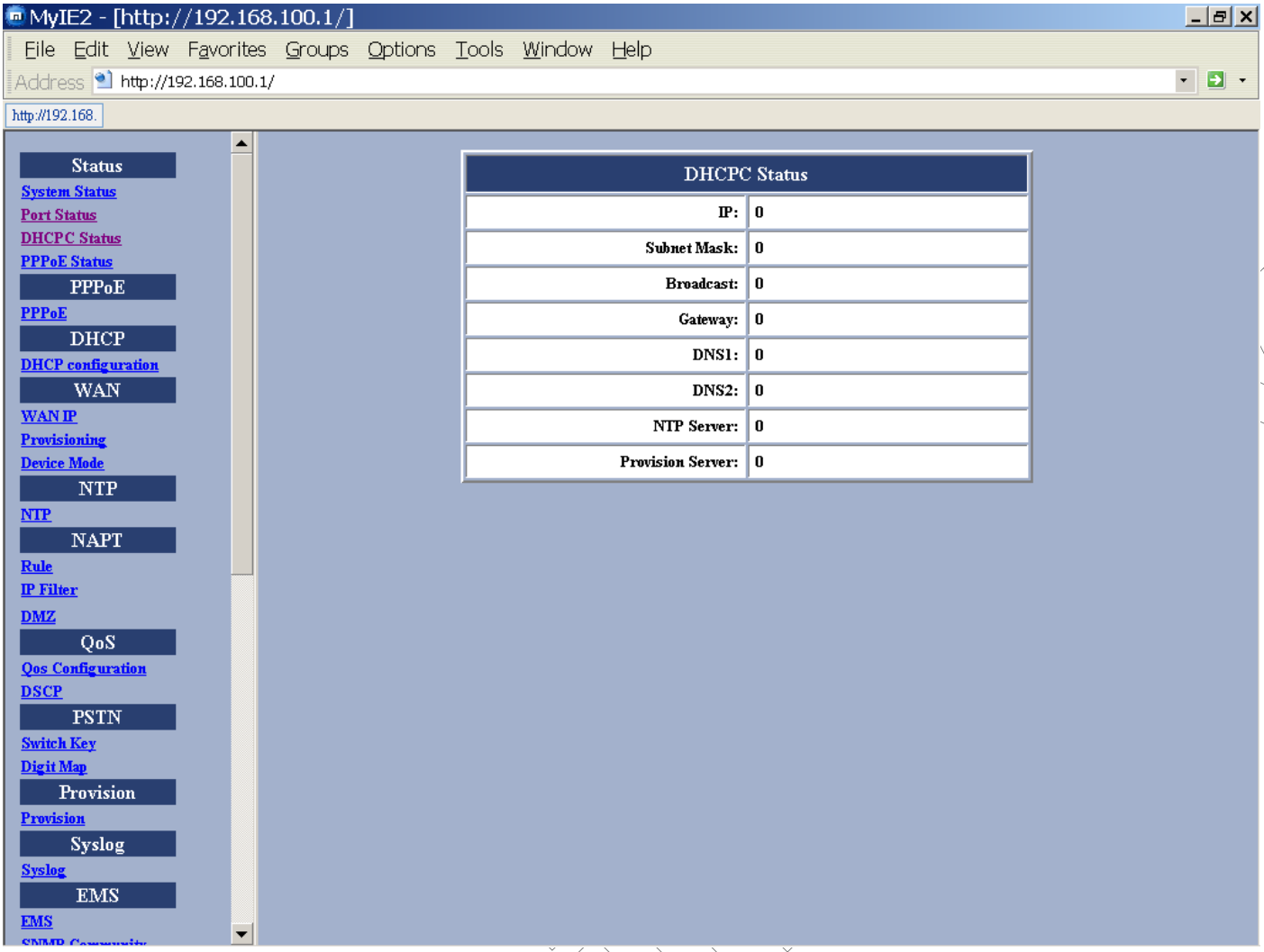
You can check the status of all I/O ports by clicking **Port Status.**



Item	Description
WAN IP Address	The IP Address of the ViP 3001 and 3002 VoIP Gateways provided by your ISP.
WAN Subnet Mask	The Subnet Mask of the ViP 3001 and 3002 VoIP Gateways provided by your ISP.
Default Gateway	The Gateway Address of the ViP 3001 and 3002 VoIP Gateways provided by your ISP.

3.5.3 DHCPC Status:

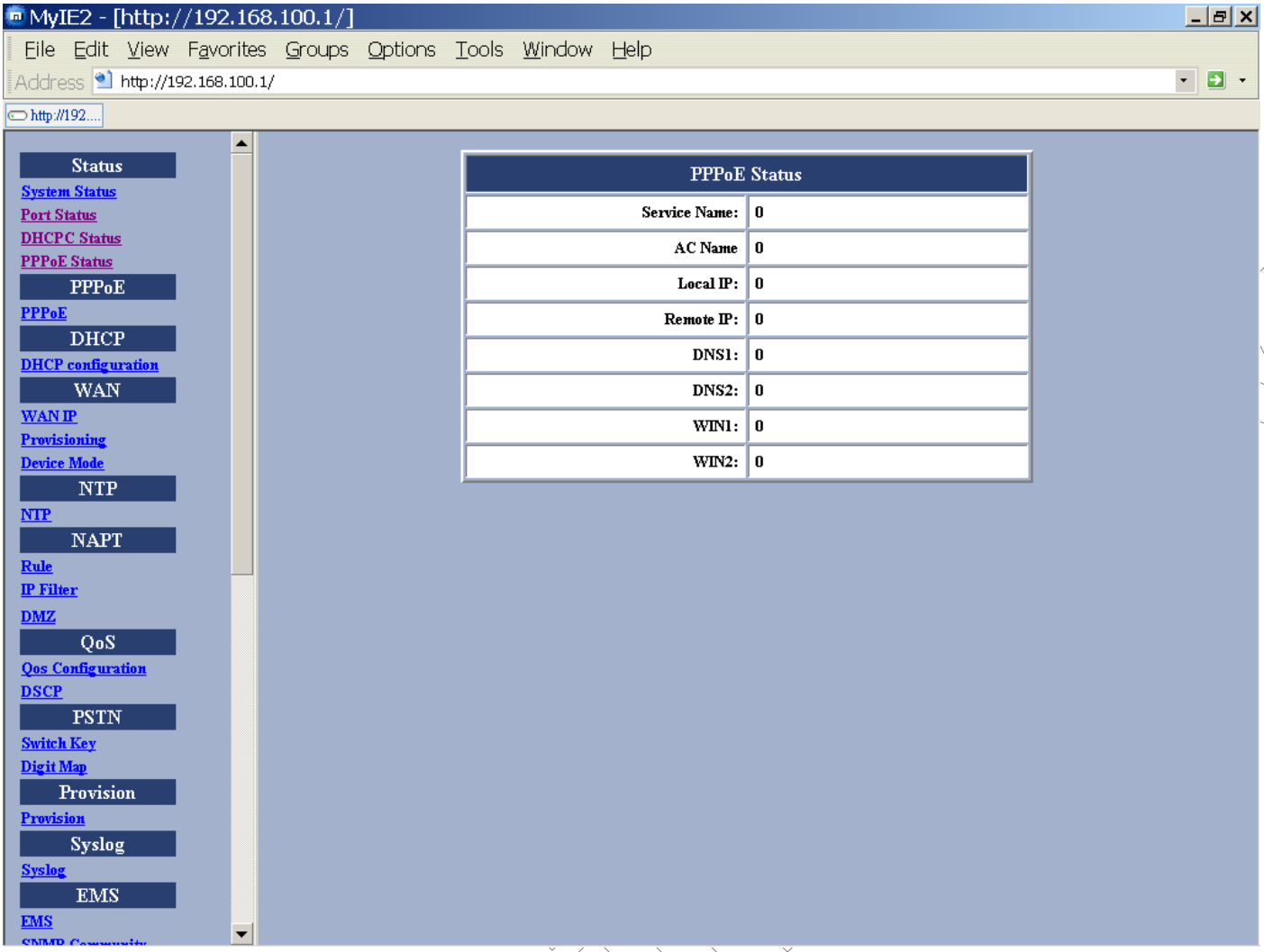
If you enable **DHCP** mode, you can check the status by clicking **DHCPC Status**.



Item	Description
IP	The IP Address of the ViP 3001 and 3002 Voice Gateways.
Subnet Mask	The Subnet Mask of the ViP 3001 and 3002 Voice Gateways.
Broadcast	The broadcast IP of the ViP 3001 and 3002 Voice Gateways.
Gateway	The Gateway Address of the ViP 3001 and 3002 Voice Gateways.
DNS1	The IP Address of Domain Name Server.
DNS2	The IP Address of Domain Name Server.
NTP Server	The IP Address of NTP Server.
Prevision Server	The IP Address of Prevision Server.

3.5.4 PPPoE Status:

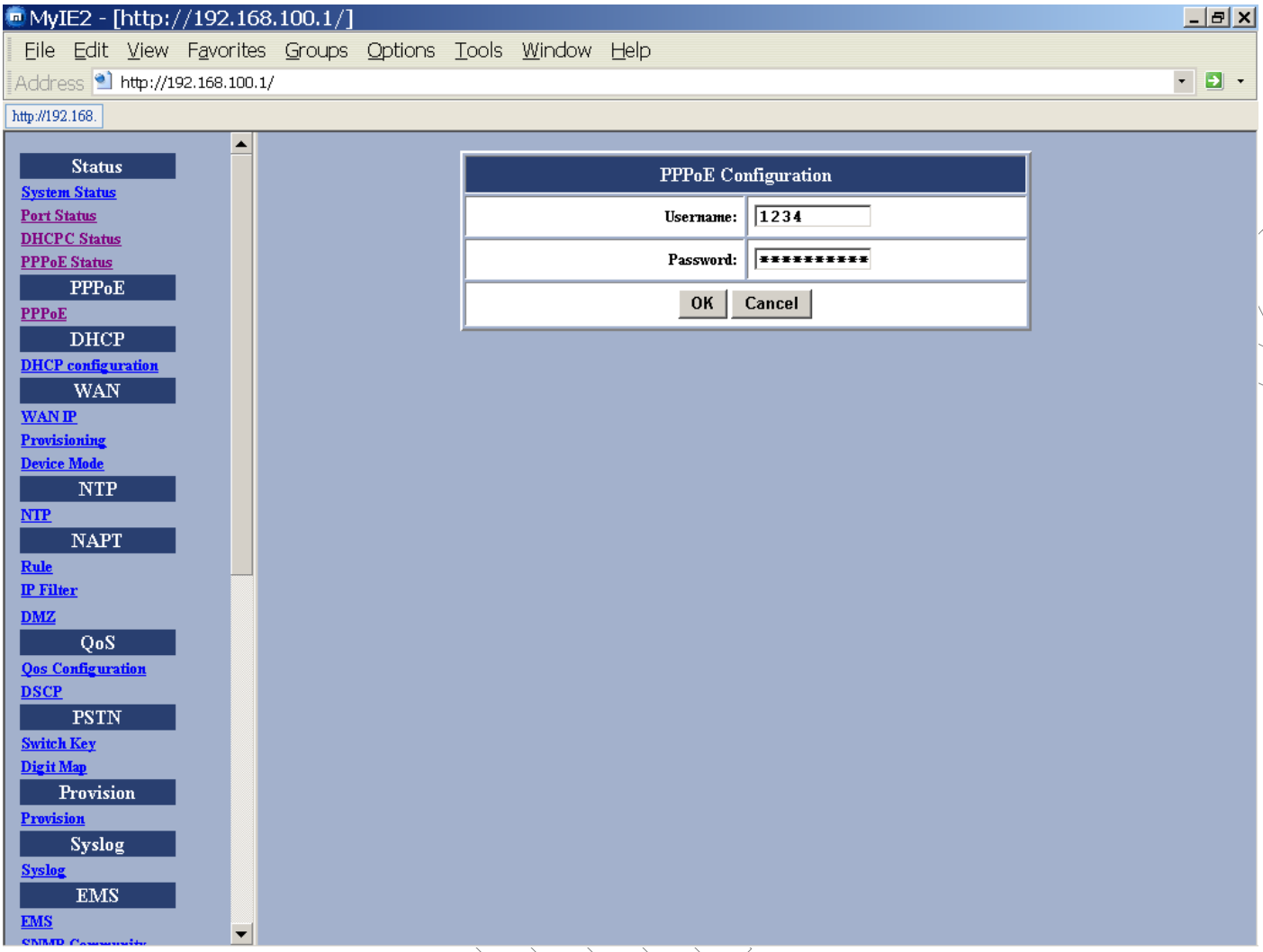
If you enable **PPPoE** mode, you can check the status by clicking **PPPoE Status**.



Item	Description
Service Name	Specifies different service group name.
AC Name	Indicates to use specific server.
Local IP	The Client IP Address.
Remote IP	The Server IP Address.
DNS1	The IP Address of Domain Name Server.
DNS2	The IP Address of Domain Name Server.
WIN1	The IP Address of WIN Server.
WIN2	The IP Address of WIN Server.

3.5.5 PPPoE Configuration:

If you select **PPPoE** to get WAN IP Address of the ViP 3001 and 3002 Voice Gateways, you need to enter the **User name** and **Password** provided by your ISP.



DHCP

3.5.6 DHCP Configuration

If the device mode is gateway mode, we support DHCP Server on LAN side. You can set the Status, Last IP and Mode etc. on this page. After you make the settings, click **OK** for the settings to immediately take effect.

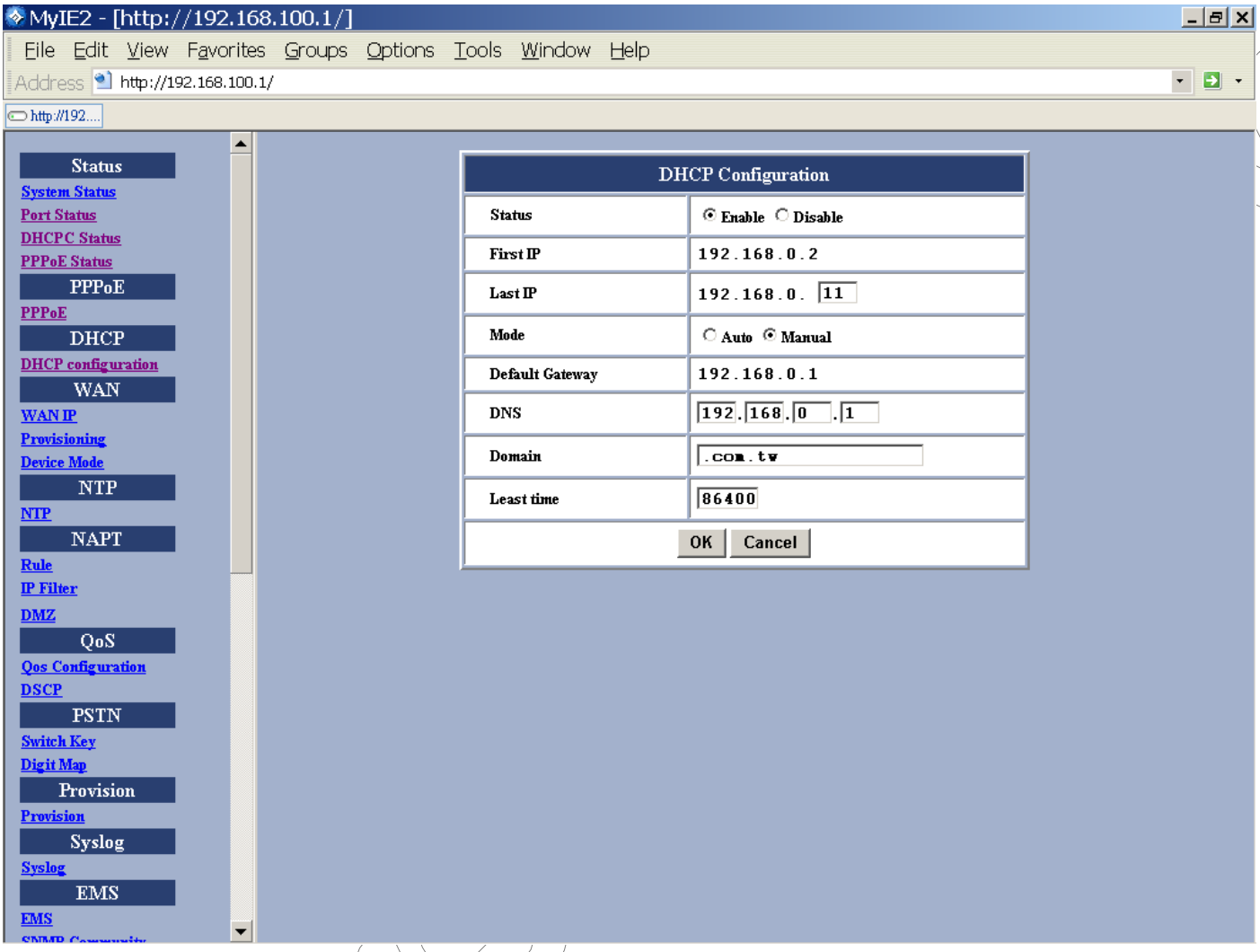


Figure 12 – DHCP Configuration Window

Item	Description
Status	The DHCP Server is enabled or disabled.
Last IP	The highest IP address that can be assigned by the DHCP server.

Mode	The network settings assigned to the DHCP Client is Auto mode or Manual Mode. In Auto mode, the network settings are from the WAN side. In Manual mode, the network settings are from the user's input in this page.
DNS	You can manually set the value in Manual mode or it takes the value from WAN side in Auto mode.
Domain	You can manually set the value in Manual mode or it takes the value from WAN side in Auto mode.
Least time	The least time of the DHCP client to holding the network settings. The value is useful in Auto mode.

WAN Configuration

4.1.7 WAN IP

You can decide the method to obtain the IP address of the VoIP Gateway by selecting one of the following modes. After the Gateway is restarted, the settings you have made will take effect.

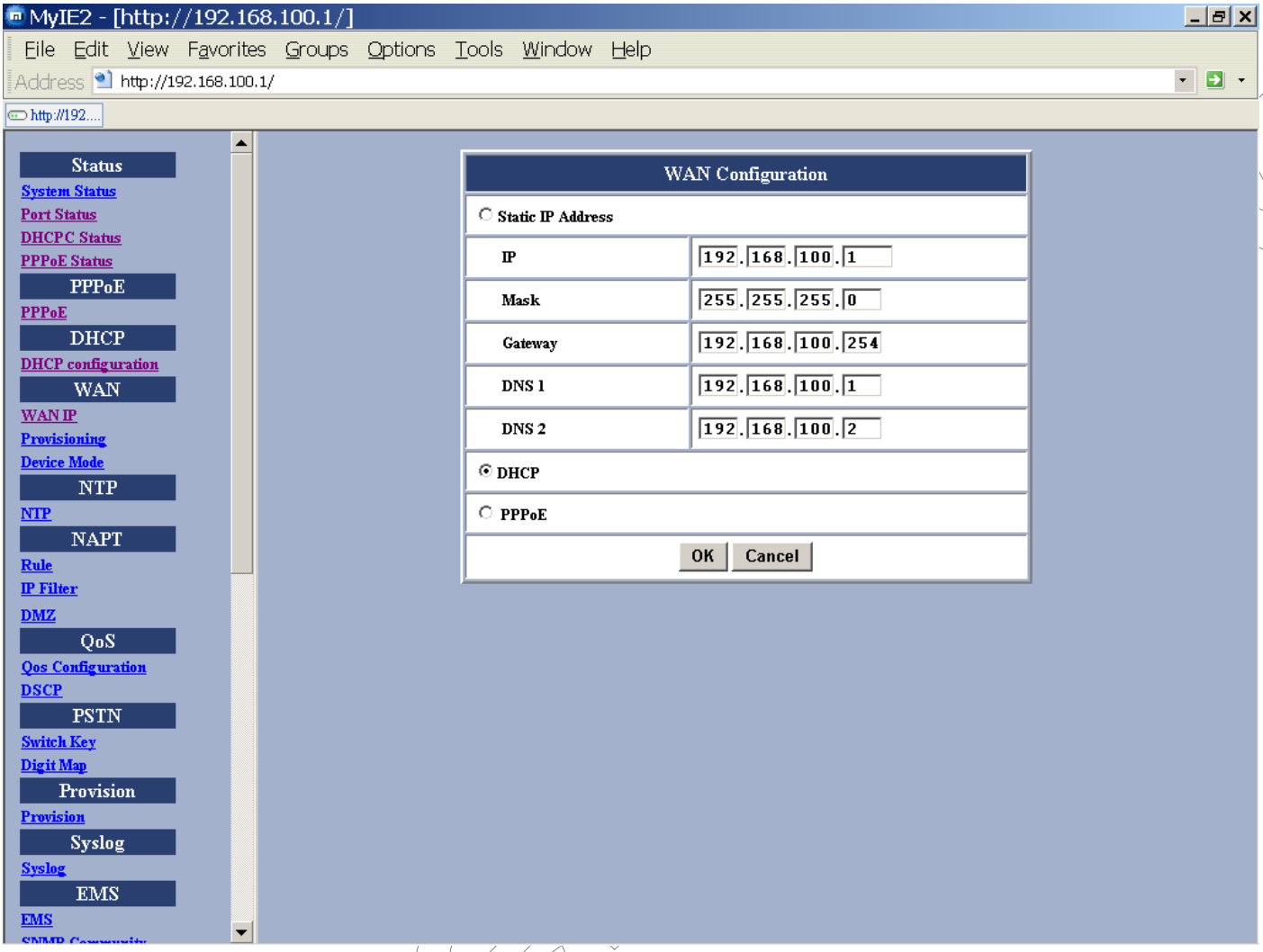


Figure 13 – WAN Configuration Window

Item	Description
Static IP Address	The IP address of the WAN side is assigned by the user.
IP	The IP address of the WAN side.
Mask	The subnet mask of the WAN side.

Gateway	The IP address of the Gateway in WAN side.
DNS1	The IP Address of the primary Domain Name Server in WAN side
DNS2	The IP Address of the secondary Domain Name Server in WAN side
DHCP	The IP Address of the WAN side is assigned by the DHCP server.

4.1.9 Provisioning

If the **Provisioning** is **off**, the TA will block telnet and http access from WAN port, so it would be active in gateway mode.

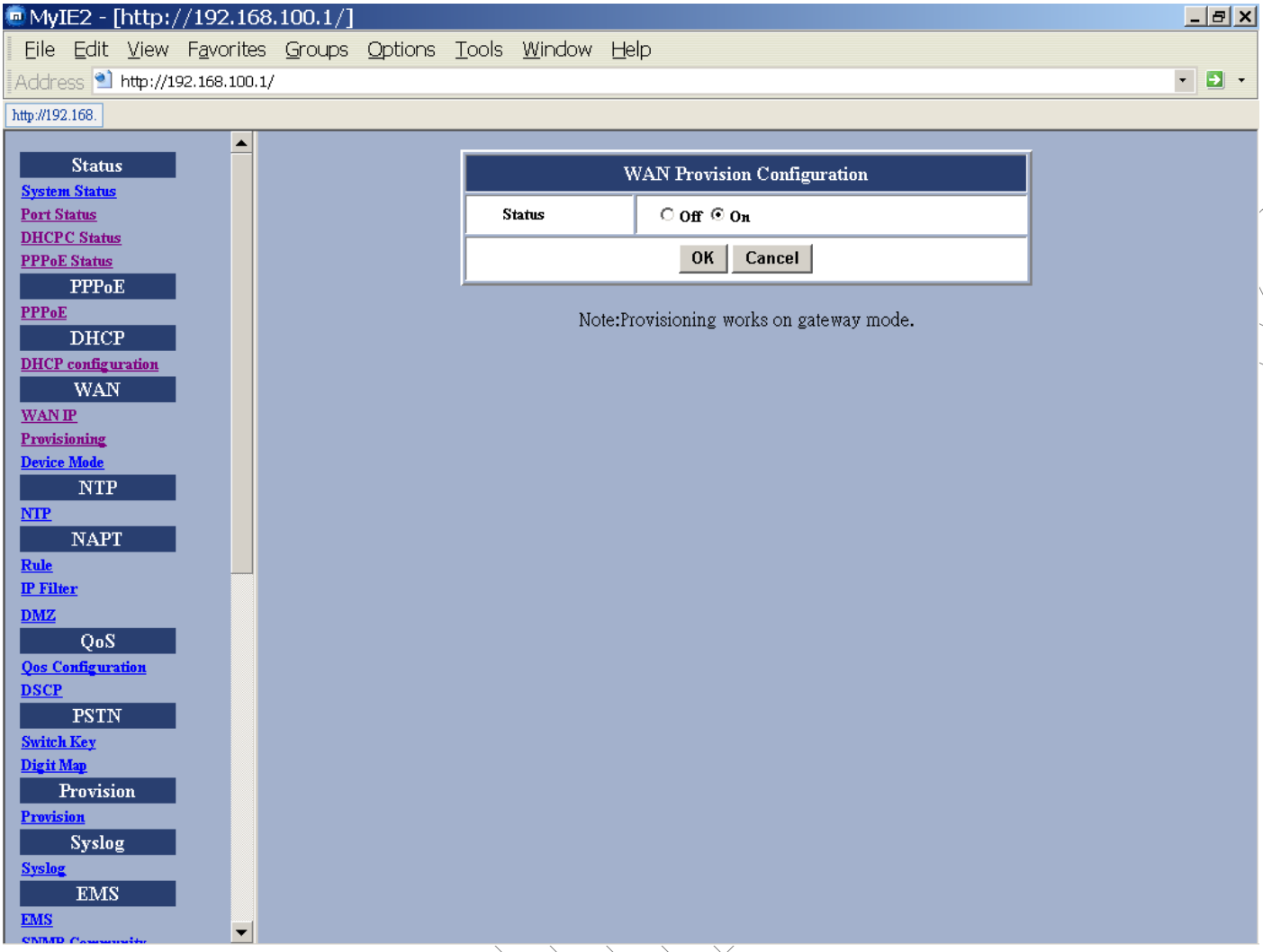


Figure 15 – provisioning Configuration Window

4.1.9 Device Mode

If the **Device Mode** is **Gateway**, NAPT is enabled. On the contrary, it's **Bridge**, NAPT is disabled.

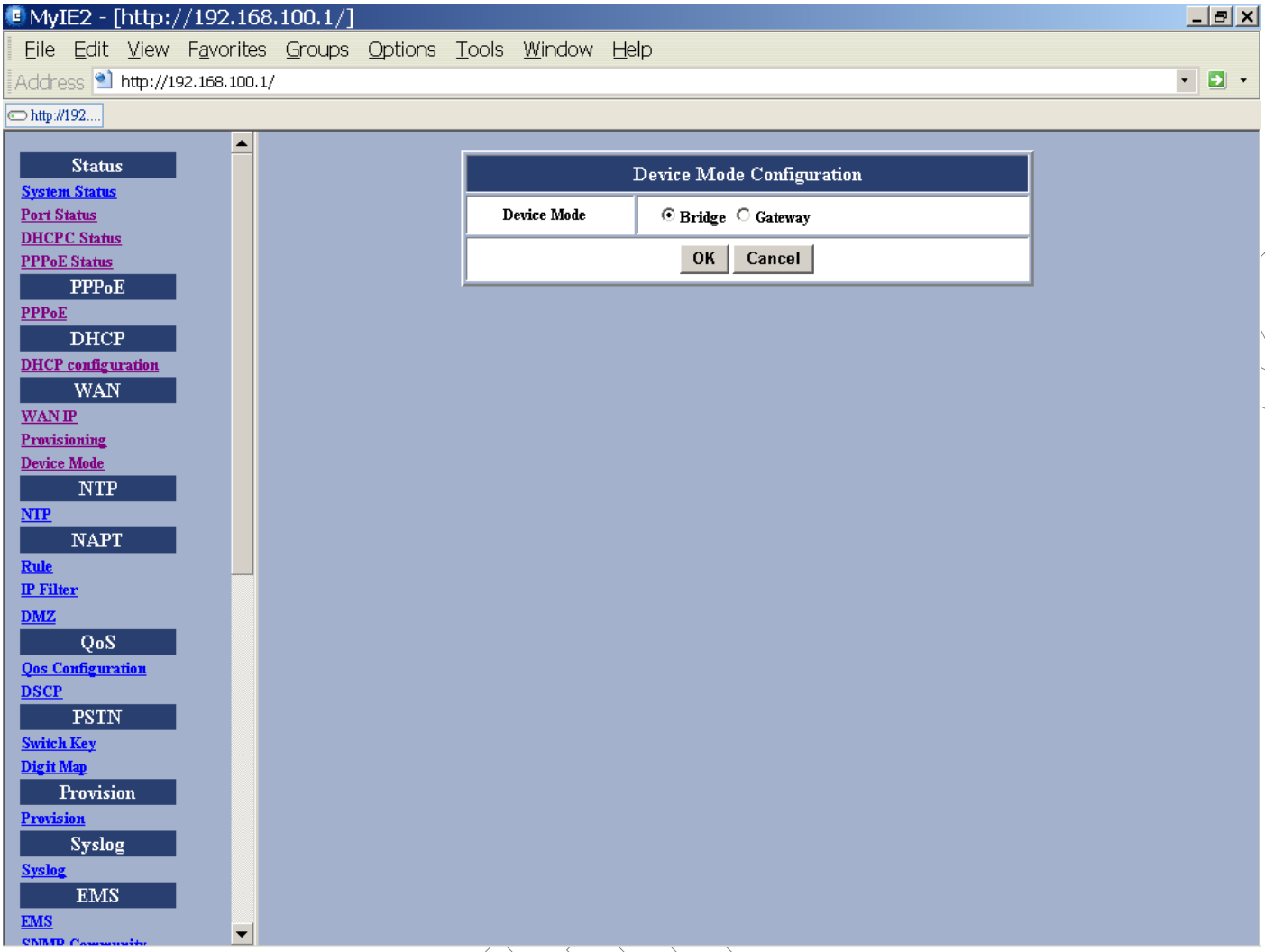


Figure 16 – Device Mode Configuration Window

NAPT Configuration

Port Forwarding

You can add or delete **Port Forwarding Rule** to the device in **Gateway mode**. When the packet goes into the VoIP Gateway, if the port of the packet matches the port of port-forwarding rule, the packet will be forwarded to the private IP address configured of the matched rule.

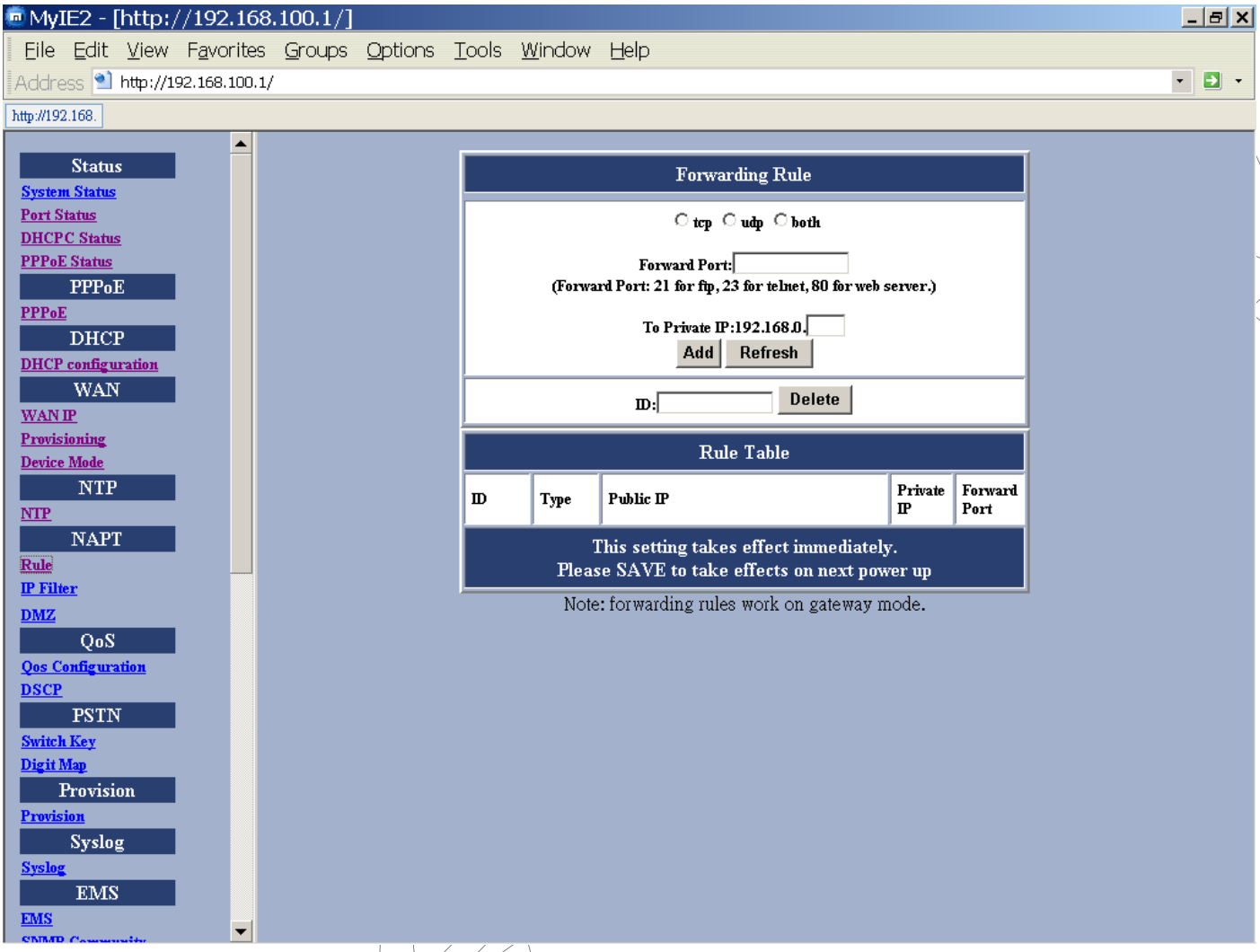


Figure 17 – Port Forwarding Rule/Rule Table Window

Item	Description
Tcp/udp/both	Select if you want to forward the packet based on tcp, udp or both.

Forward Port	The tcp or udp port number for which you want to check against.
To Private IP	The IP Address of the pc in the LAN side is forwarding to.
ID	The ID of the port forwarding rule is to be deleted.

IP Filter

You can add or delete **IP Filter Rule** to the device in **Gateway mode**. When the packet goes into the VoIP Gateway, the packet will be blocked if the source IP of the packet matches the rule of IP Filter.

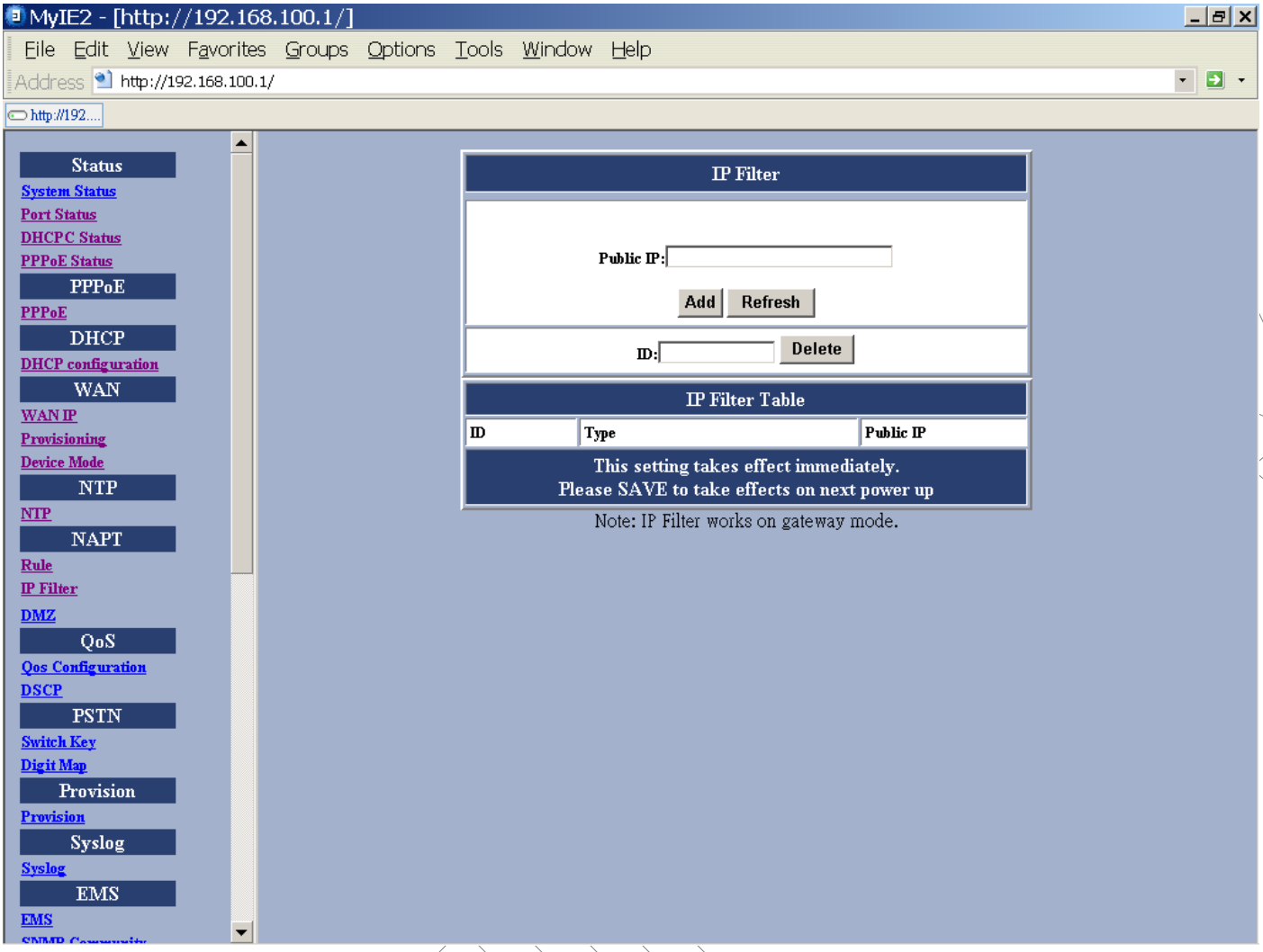


Figure 18 – IP Filter Configuration Window

Item	Description
Public IP	The Public IP Address is to be filtered.
ID	The ID of the IP Filter rule is to be deleted.

DMZ

You can **enable or disable** DMZ and specify the **IP address** of DMZ in **Gateway mode**. When the packet goes into the VoIP Gateway, the packet will be transferred to the DMZ if packet is not filtered, not port-forwarded, and not matched for the NAPT binding.

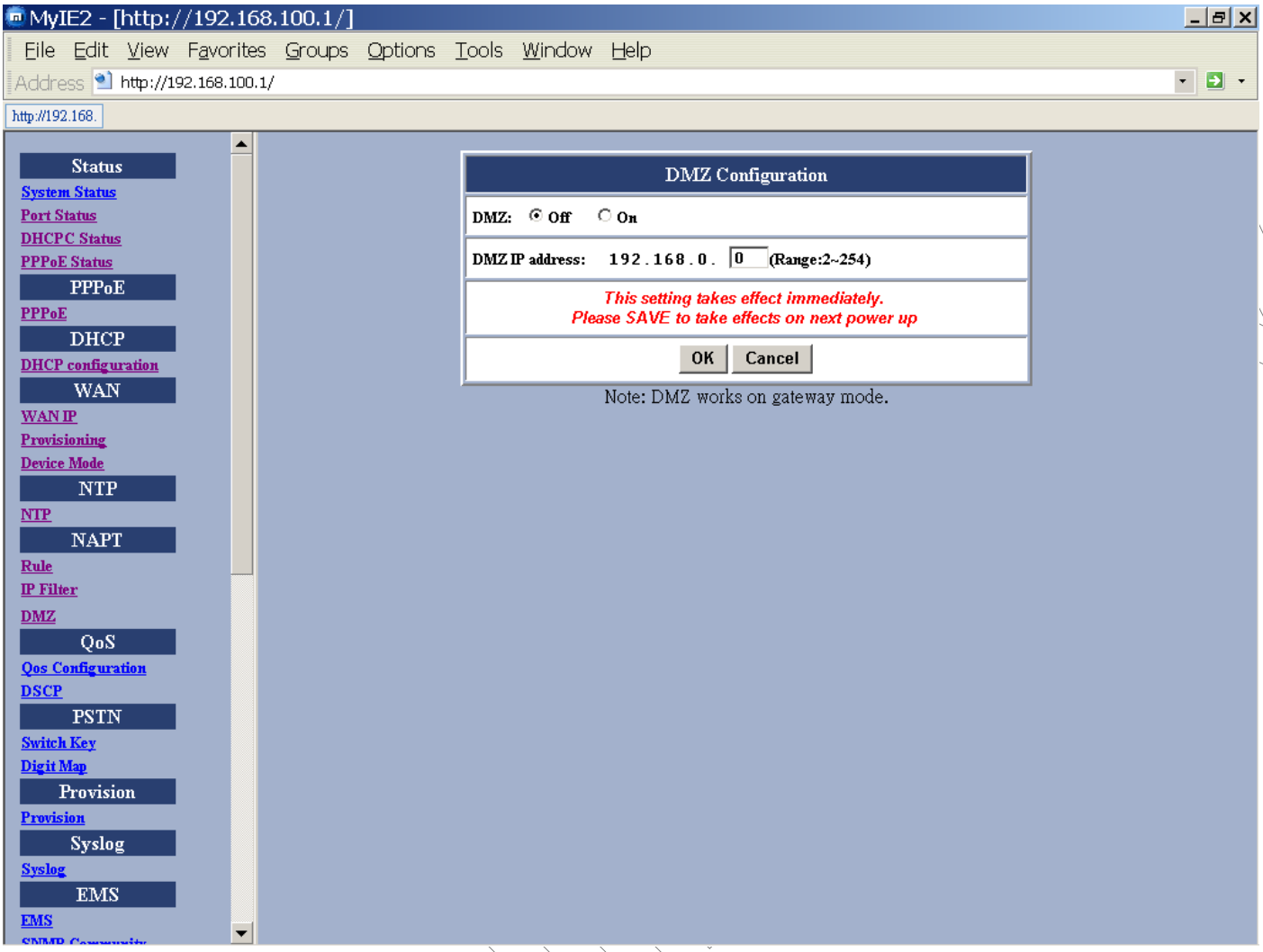


Figure 19 – DMZ Configuration Window

Item	Description
DMZ	The DMZ is disabled or enabled.
DMZ IP address	The IP address of the DMZ.

QoS

QoS Configuration

You can decide the **QoS type** of the packets coming out from the VoIP Gateway. If the type of **QoS** is DiffServ, you can also specify the different values for **Signal DSCP** and **Media DSCP**. Both ToS and DSCP QoS are supported for the VoIP packets sending out from the VoIP Gateway.

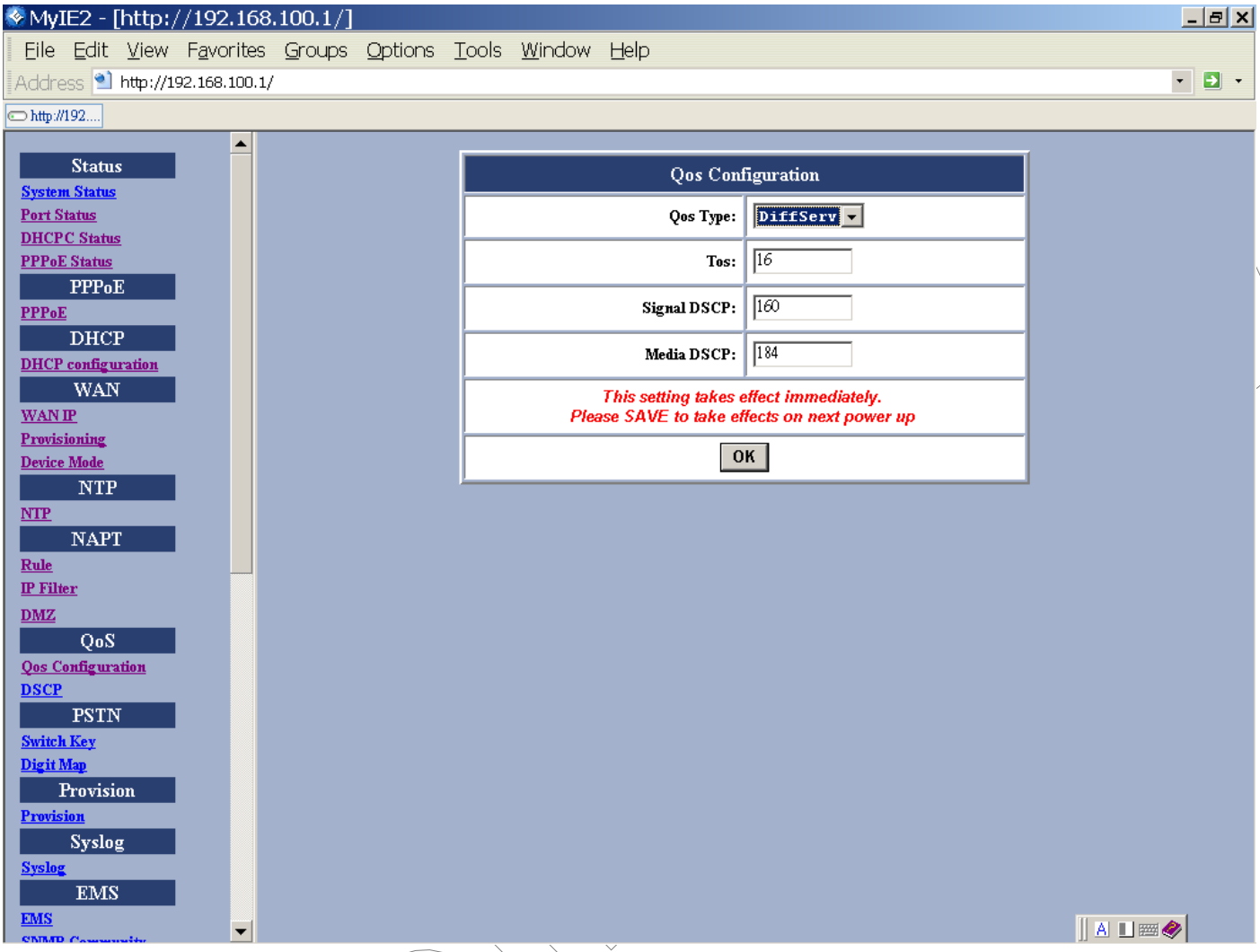


Figure 19 – Qos Configuration Window

Item	Description
Qos Type	The type of Qos can be disabled, DiffServ or Tos.
Tos	The value of Tos is usually between 0~15.
Signal DSCP	The DSCP value for Signalling traffic.

Media DSCP	The DSCP value for Media traffic

DSCP Configuration

You can set the **DSCP mode** to **Trusted** or **Un-Trusted**. This **DSCP mode** of operations is supported for PCs traffic from LAN interface. If it is set to **Trusted Mode**, the TA will preserve DSCP settings from LAN interface. If it is set to **Un-Trusted mode**, the TA will remark to DSCP DE before forwarding to Uplink interface.

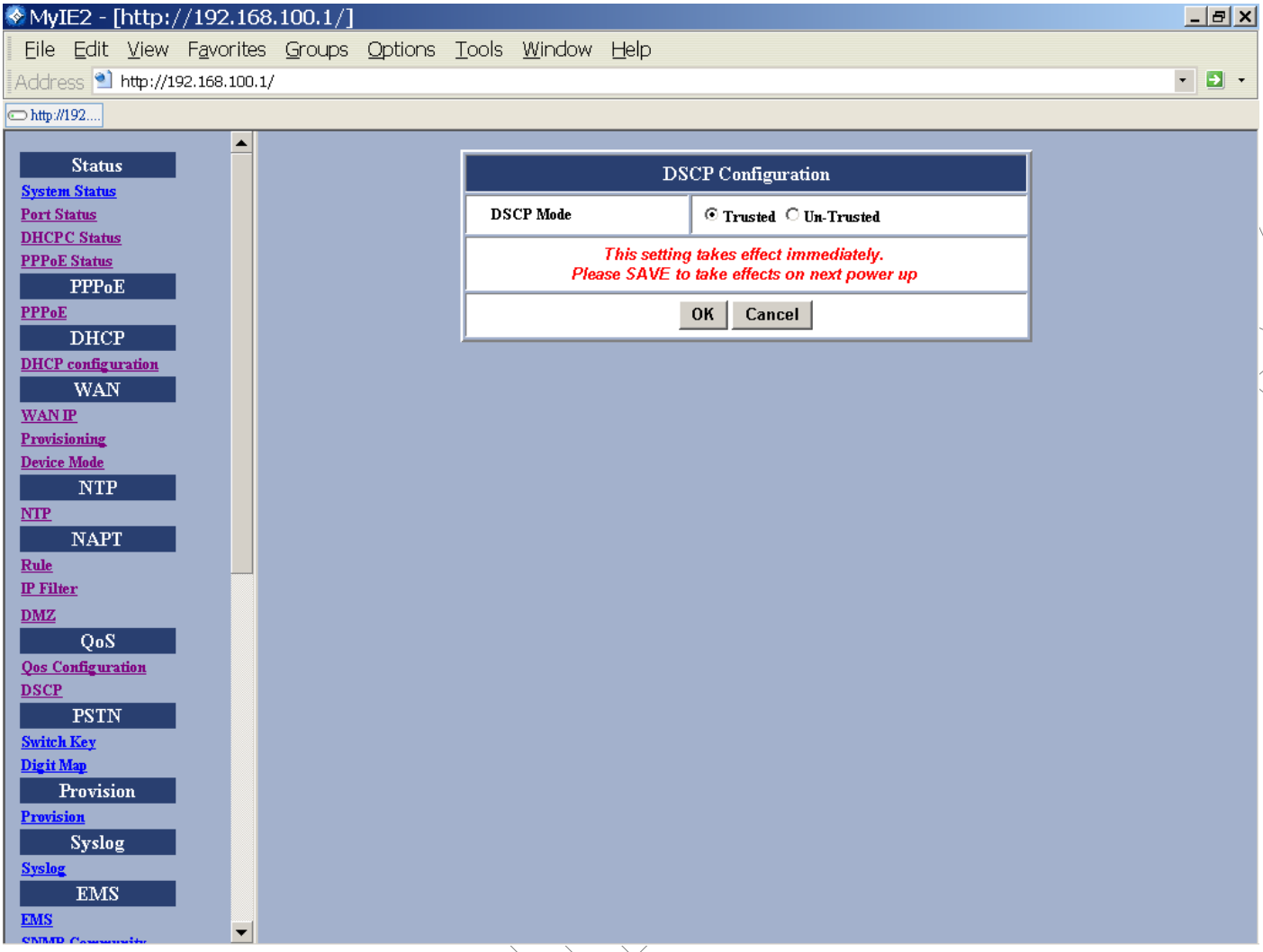


Figure 20 – DHCP Configuration Window

PSTN Configuration

Switch Key

. Normally, your telephone will be using VOIP service, except when the VOIP service itself is not available. In this case the phone will be directly connected to the line in order to use the standard PSTN service However, you can force your phone to switch from VoIP mode to PSTN mode by entering a 4-digit PSTN switch key . "0000" is the default value.

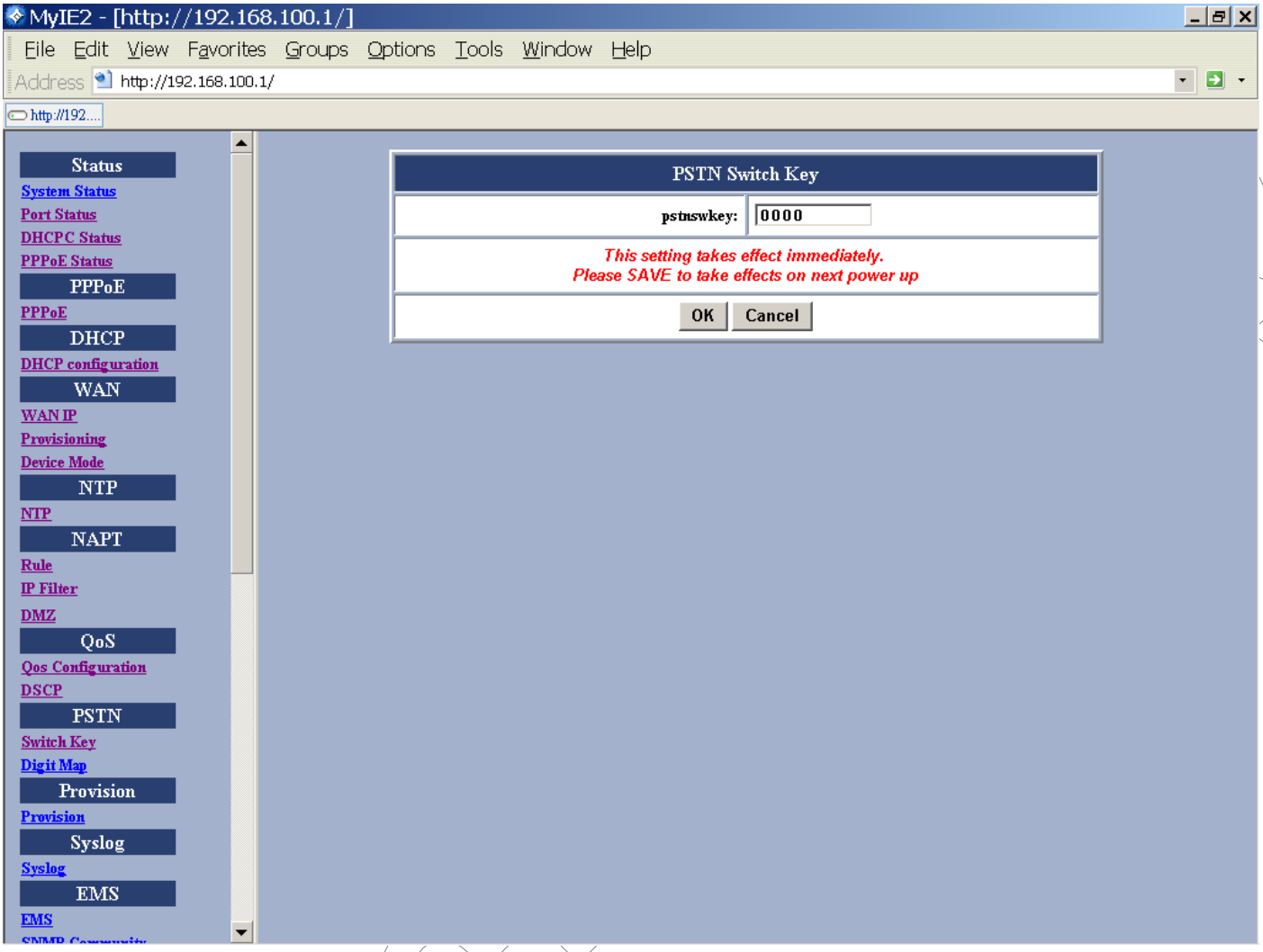


Figure 21 –PSTN Switch Key Window

Digit Map

This function allows user to set up a list of phone numbers with specific prefixes and total lengths. When you dial a number in this list, the phone will not be using the VoIP service but it will be directly connected to the PSTN line.

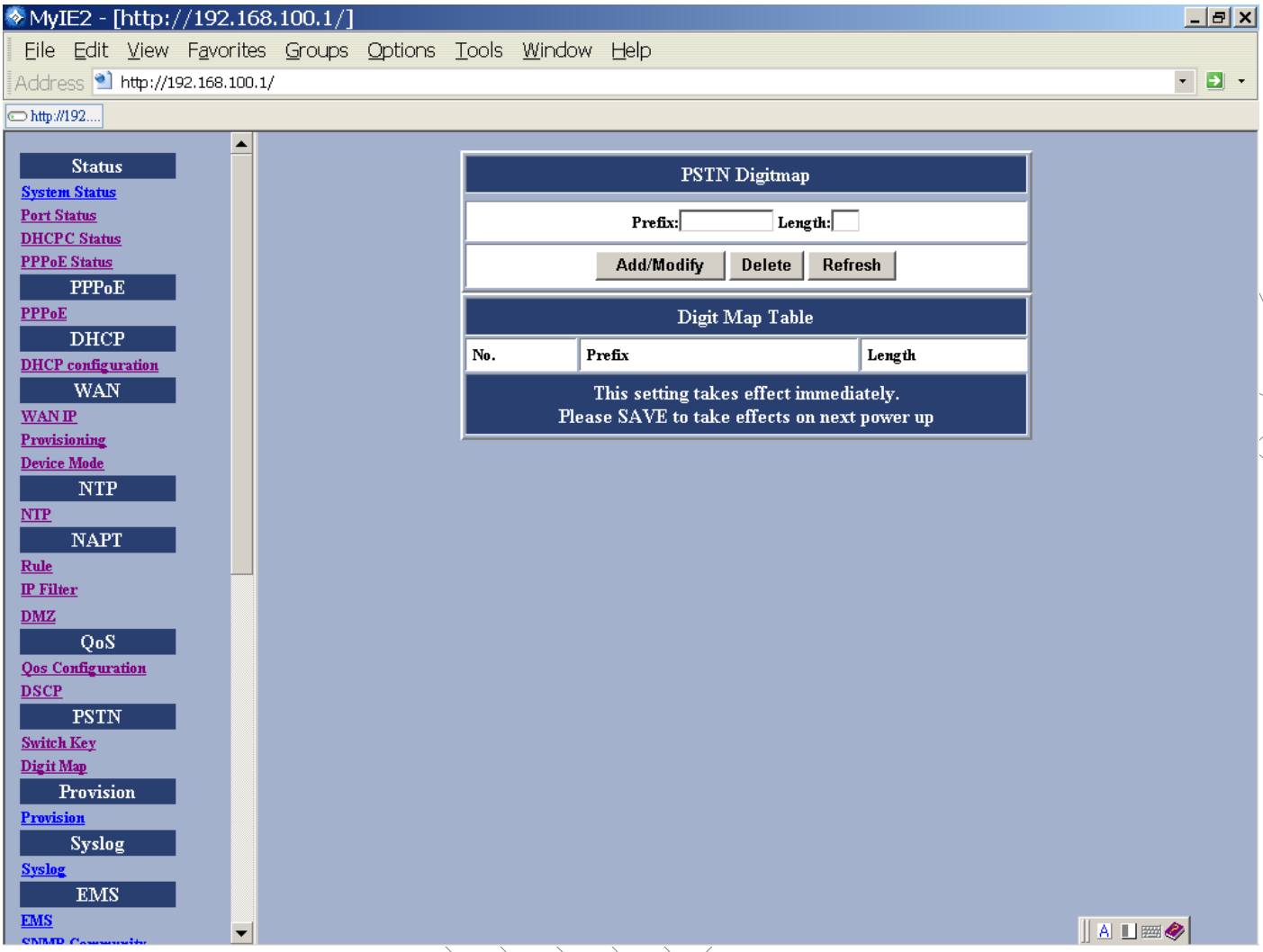


Figure 22 –PSTN Digitmap Window

Item	Description
Prefix	Enter the prefix of the telephone number. The maximum length is 5 digits.
Length	Enter the total length of the telephone number. The length is ranged from 0~64. "0" means the length is irrelevant.
Add/Modify	Add or modify your desired prefix and length of the telephone number.

Delete	Delete an existing prefix and length of the telephone number from the Digit Map Table
Refresh	Press this button will show new changes

Provisioning Configuration

ViP 3000 Gateways support Provisioning Configuration mechanism to get and set the Gateway configuration parameters. When the Gateway downloads the configuration file from a Provision server, it will compare the downloaded parameters with the existing local parameters. If the former paramaters are more recent, the existing local setting parameters will be overwritten and the downloaded parameters will be written into the FLASH memory. This feature sets provision configurations including server address, server port number, group and expiry time. After you make the settings, click **OK** and then **Reset** for the new settings to take effect.

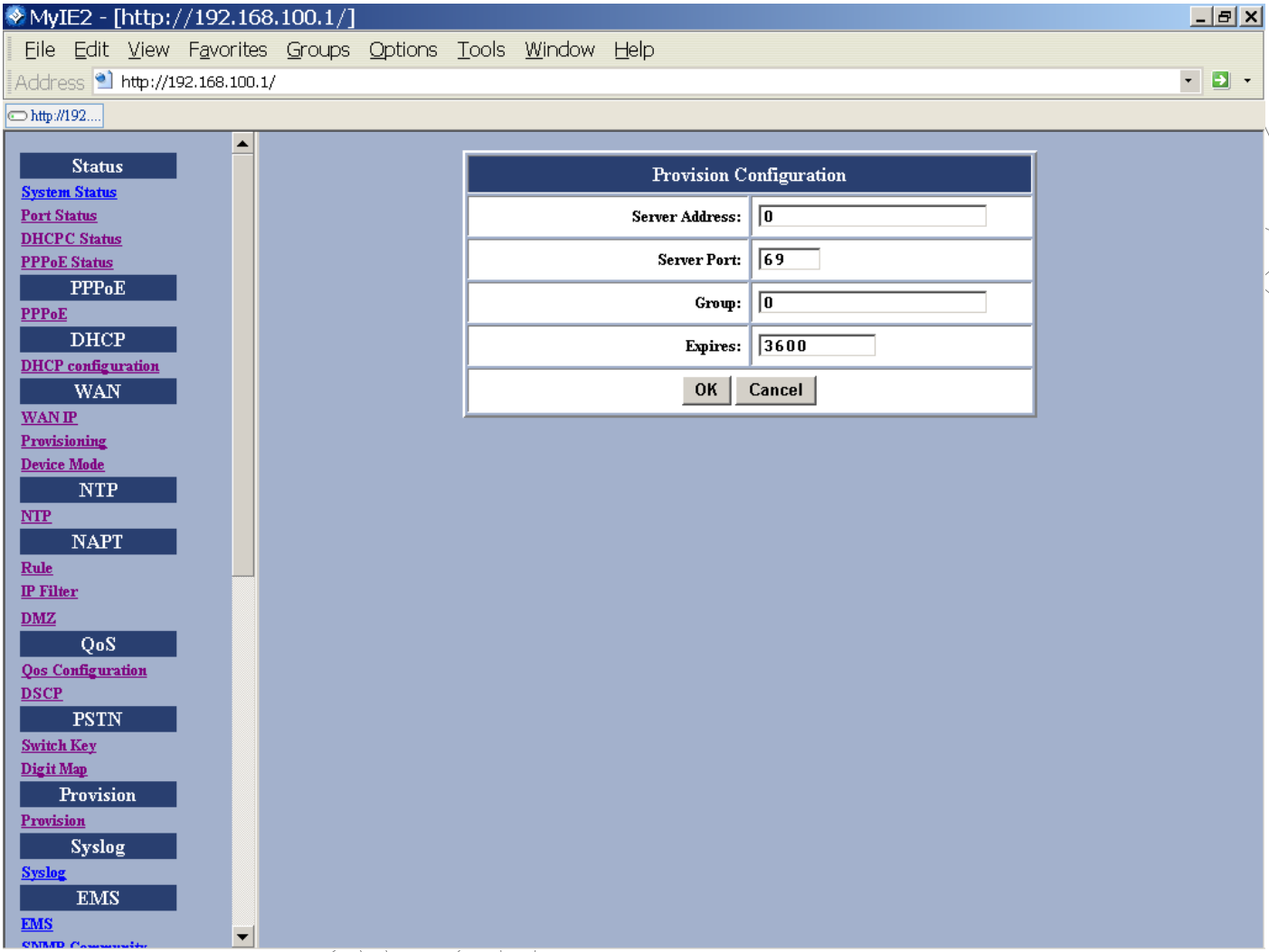


Figure 23 –Provision Configuration Window

Item	Description

Server Address	The IP Address of Provision Server. Enter the value provided by your ISP.
Server Port	The receiving port number of Provision Server. Enter the value provided by your ISP.
Group	Enter the string for different user group. The maximum length is 64. Enter the value provided by your ISP.
Expires	The valid period for this device's IP Address assigned by DHCP server or PPPoE server. The unit is second. Enter the value provided by your ISP.

Syslog Configuration

The ViP 3000 Gateways support **Syslog**. Syslog is used to send UDP packets via Syslog port (514) and keep messages in the Log Server.

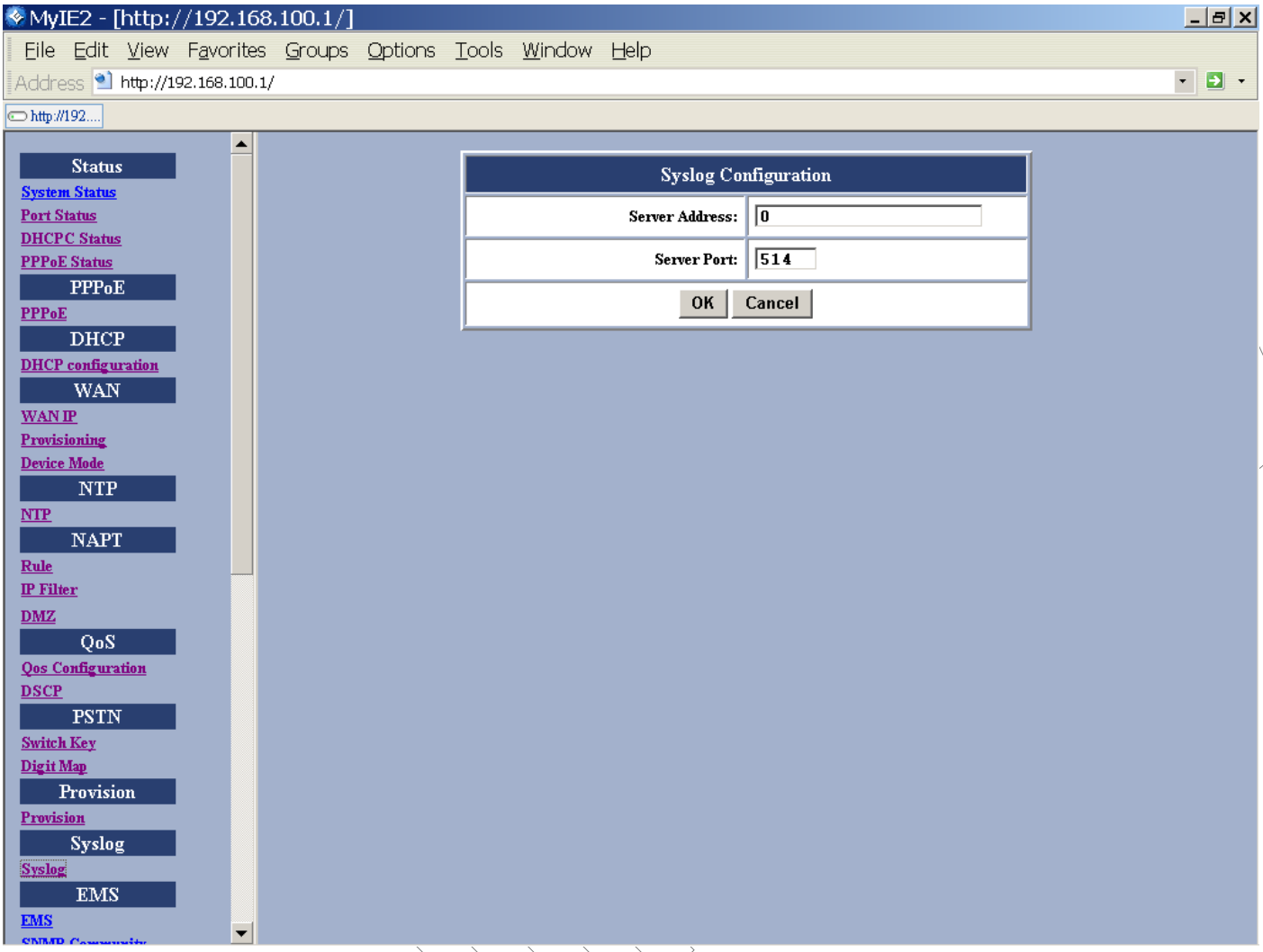


Figure 24 –Syslog Configuration Window

Item	Description
Server Address	Specify the IP Address of Syslog server.
Server Port	Specify the port number of Syslog server.

EMS Configuration

EMS

This VoIP gateway supports EMS management function. Users can set the EMS configuration including Server Address, Server Port, Community and expiration time.

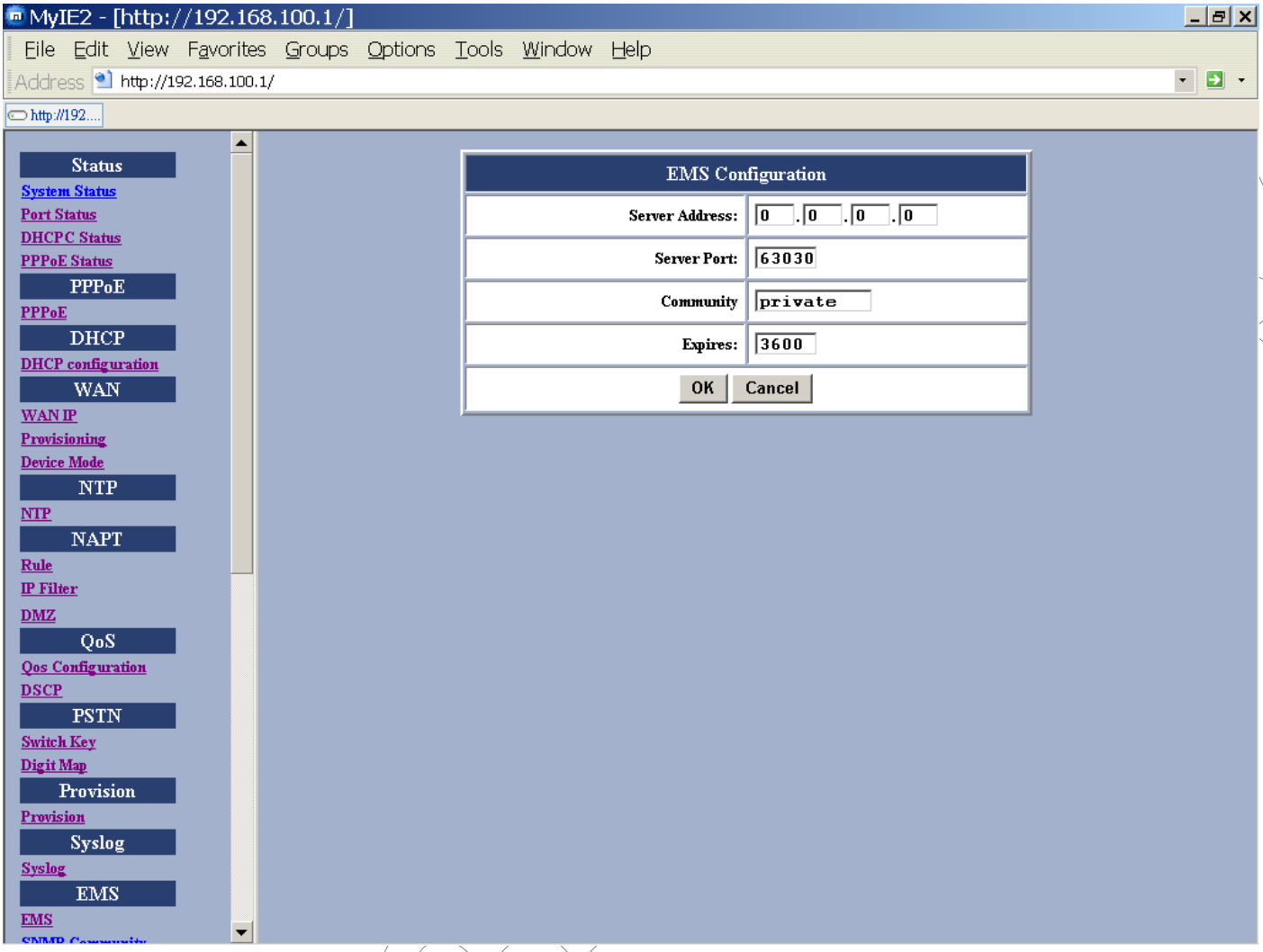


Figure 25 –EMS Configuration Window

Item	Description
Server Address	Specifies the IP address of EMS server
Server Port	Specifies the Port number of EMS Server
Community	Specifies the Community used to EMS Server

Expires	Specifies the valid period of the VoIP gateway managed by EMS Server. The unit is second.

SNMP Community

The ViP 3000 Gateways support SNMP agent. Users can use EMS to manage the VoIP Gateway via SNMP protocol. After you make the settings, click **OK** and then **Reset** the device for the settings to take effect.

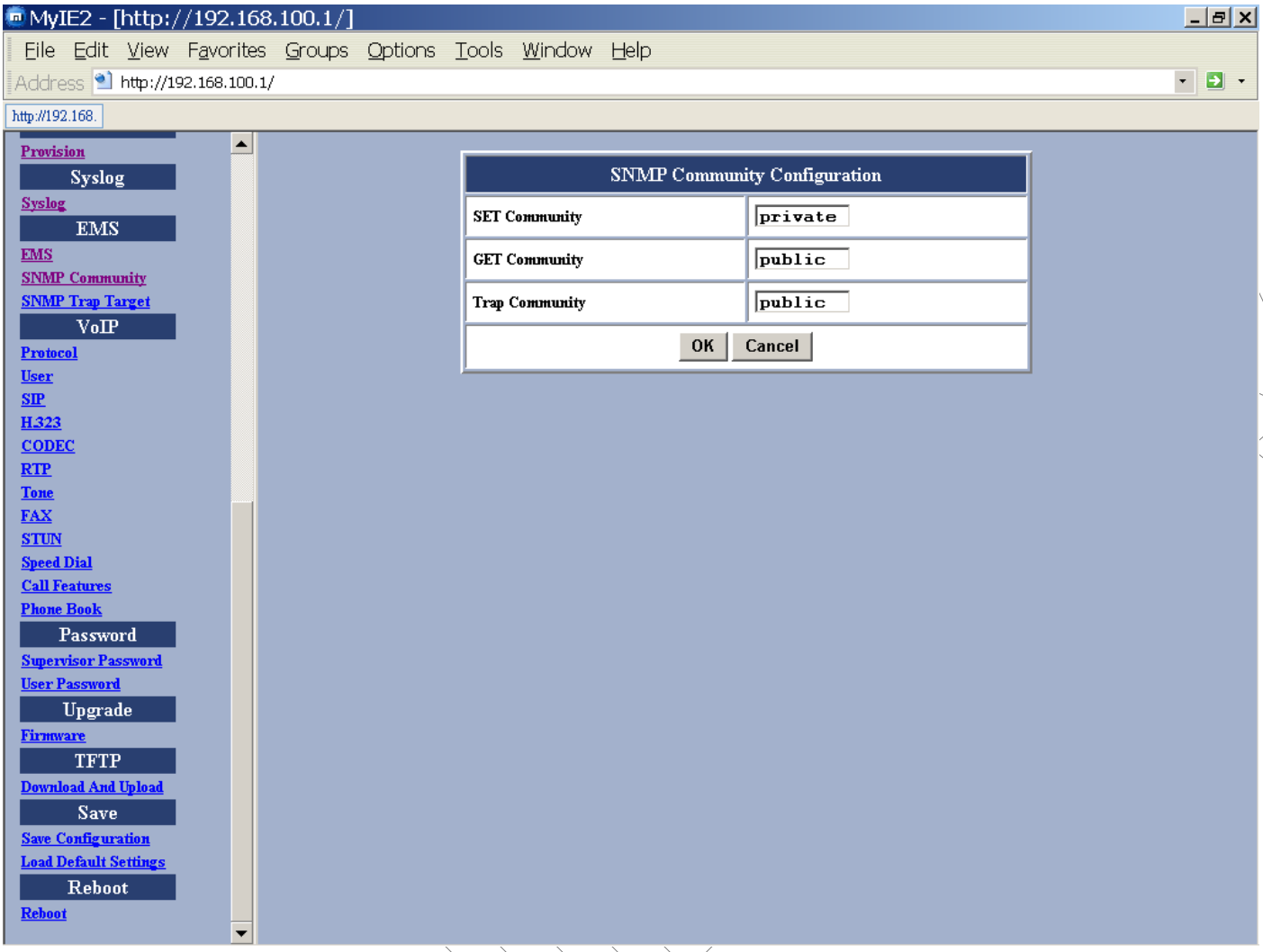


Figure 26 –SNMP Community Configuration Window

Item	Description
Set Community	The Community is used when the user sets some oids.
Get Community	The Community is used when the user gets some oids.
Trap Community	The Community is used when the user process the traps.

SNMP Trap Target

The ViP 3000 family support 4 Trap targets. You can specify different IP and Port to receive the traps sent from the VoIP Gateway. After you make the settings, click **OK** and then **Reset** the device for the new settings to take effect.

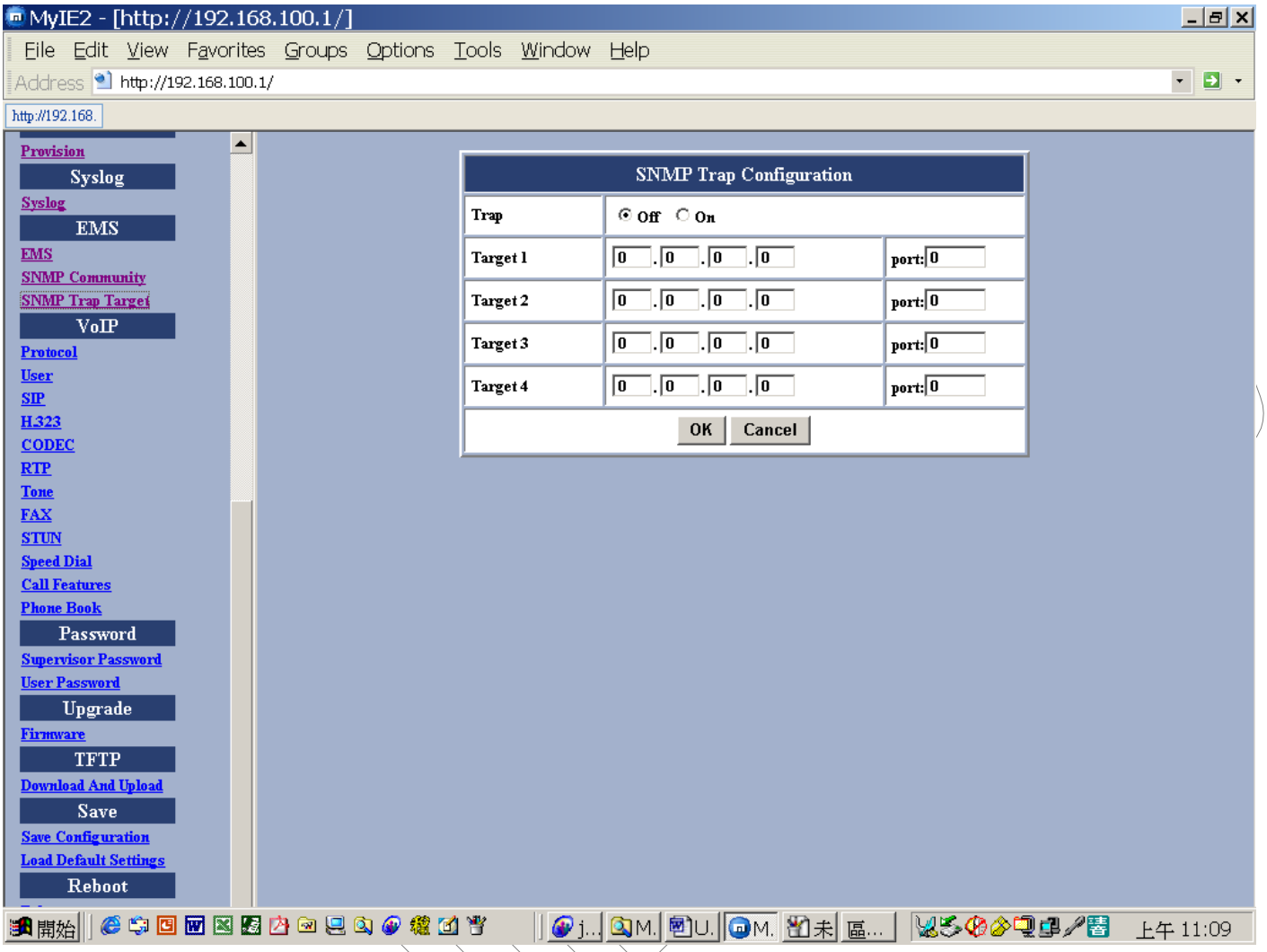


Figure 27 –SNMP Trap Configuration Window

Item	Description
Trap	The traps will be sent or not.
IP	Specify the IP Address to which the traps of the VoIP Gateway will send.
Port	Specify the Port to which the traps of the VoIP Gateway will send.

VoIP Configuration

Protocol

This screen allows you to set the protocol you would use,

NOTE : in this release, only H.323 is supported

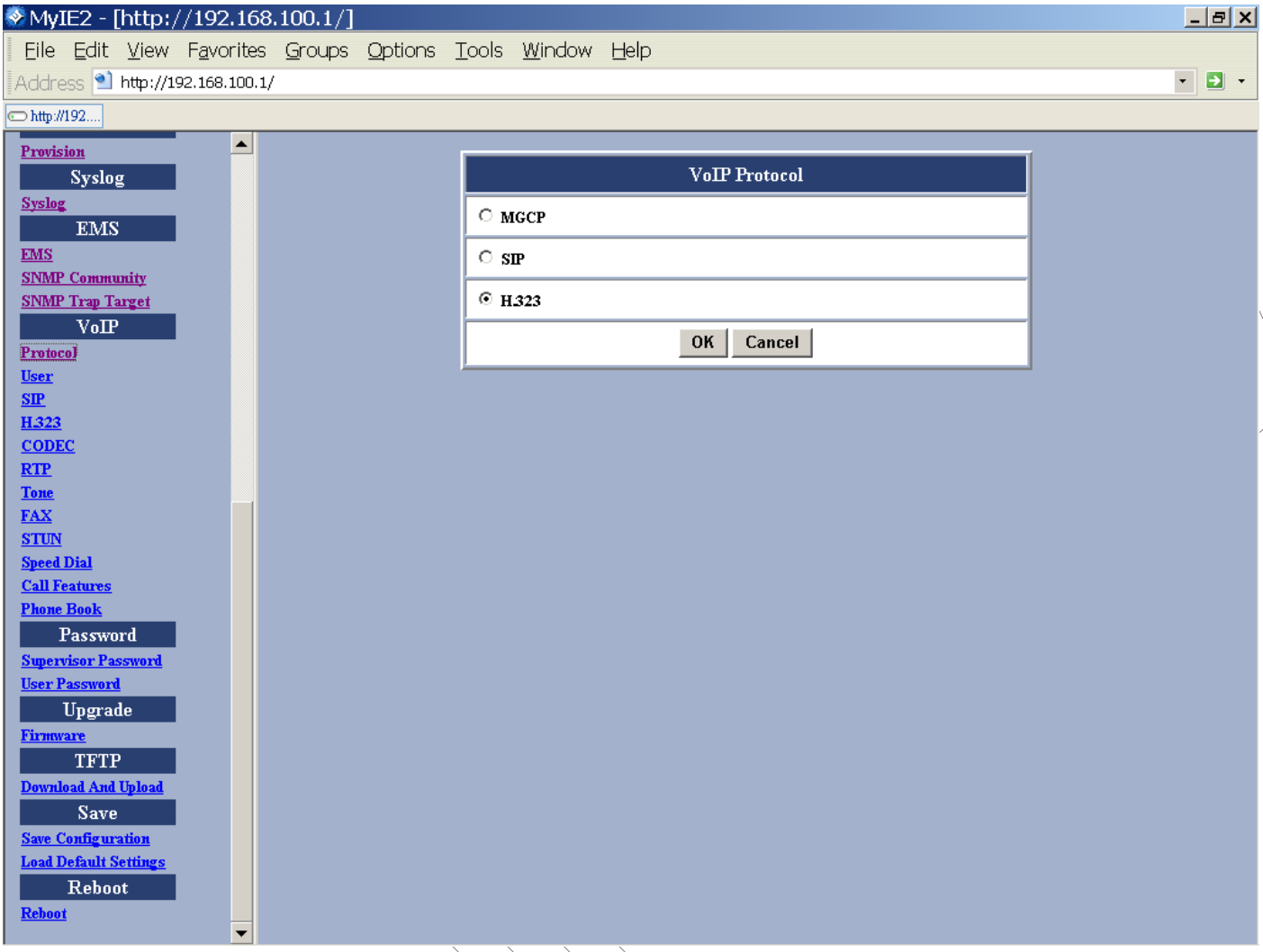


Figure 28 – VoIP User Configuration Window

User

This screen allows you to configure user information such as username, password and display name. You should obtain these values from your service provider. After entering or changing any settings, click **OK** and then **Reset** for the new settings to take effect.

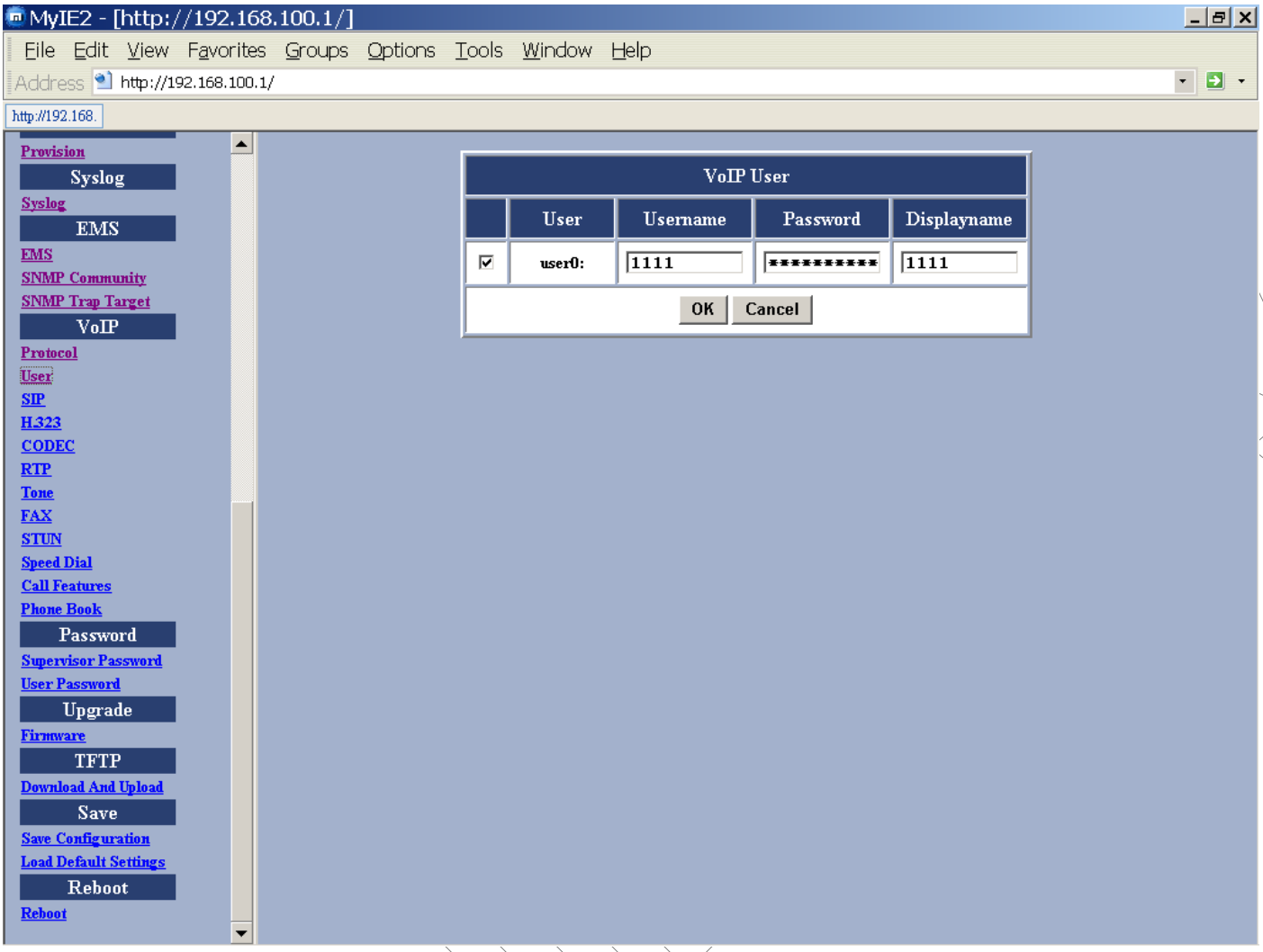


Figure 28 – VoIP User Configuration Window

Item	Description
Username	Specifies the name (or phone name) of the user
Password	Specifies the password of the user
Display name	Specifies the displayed user name

SIP

This screen allows you to make SIP configurations including local port, SIP proxy server address and port number, Registrar server address and port number, expiry time, SIP domain and subject. After you make the settings, click **OK** and then **Reset** for the new settings to take effect.

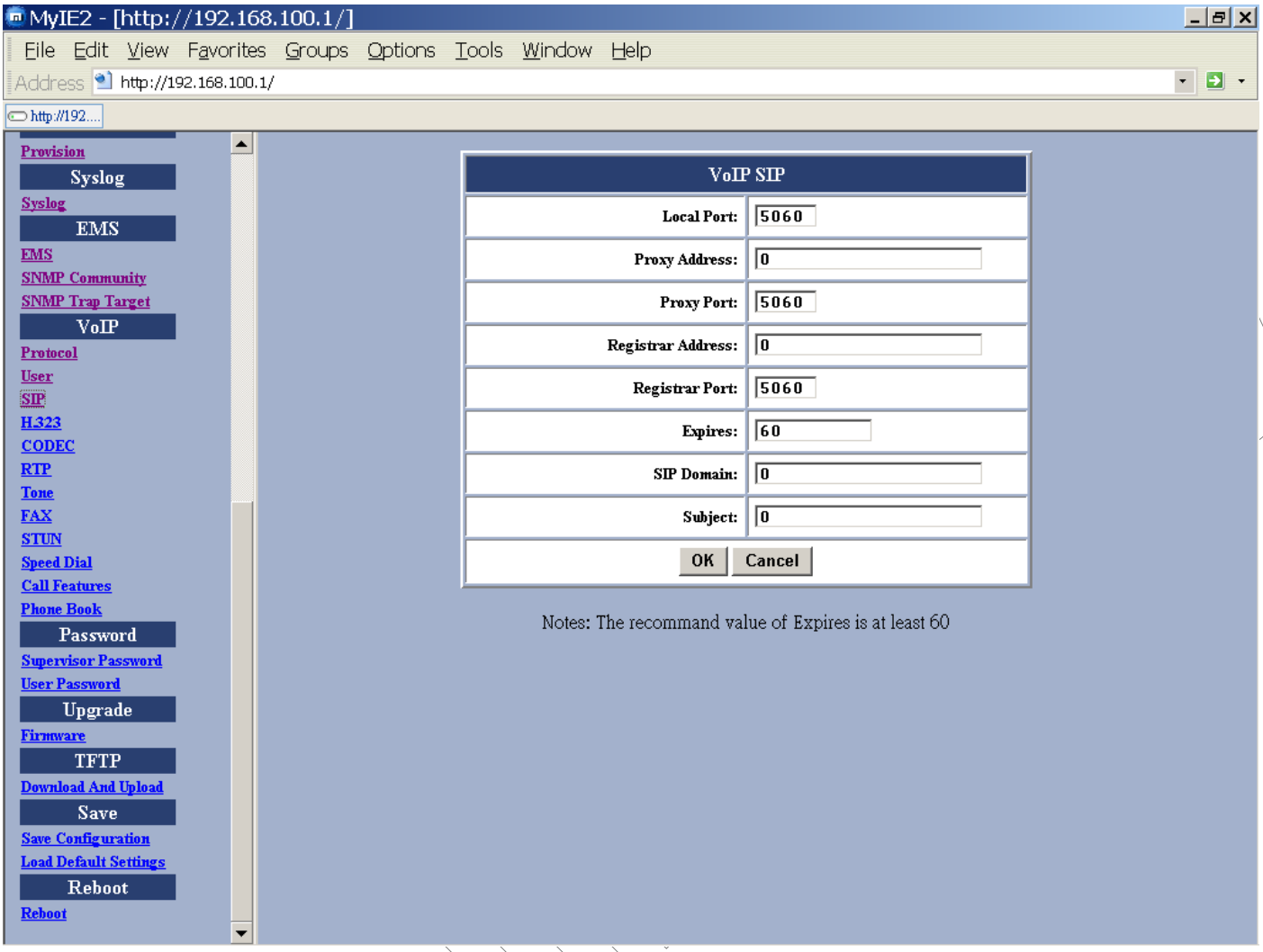


Figure 29 – VoIP SIP Configuration Window

Item	Description
Local Port	Specifies the port number of the SIP stack. 5060 is the default port number.
Proxy Address	Specifies the IP address of SIP proxy server
Proxy Port	Specifies the port number of SIP proxy server
Registrar Address	Specifies the IP address of Registrar server. Registrar server is often the same as SIP proxy server
Registrar Port	Specifies the port number of Registrar server.

Expires	"Expires" specifies the period (in seconds) that the VoIP Gateway sends Registration message to Registrar server. This is to help check the connection status in case the VoIP Gateway is accidentally disconnected from the Registrar server.
SIP Domain	Specifies the domain name to which the TA is assigned to by the service provider
Subject	Specifies the content of the subject header in outgoing INVITE message. This is used to indicate the title of the call.

H.323

This screen allows you to make H.323 configurations including local port, H.323 Gatekeeper ID, Katekeeper Address and port number, H235 Password for optional, expiry time, Fast Start mode and DTMF signal type. After you make the settings, click **OK** and then **Reset** for the new settings to take effect.

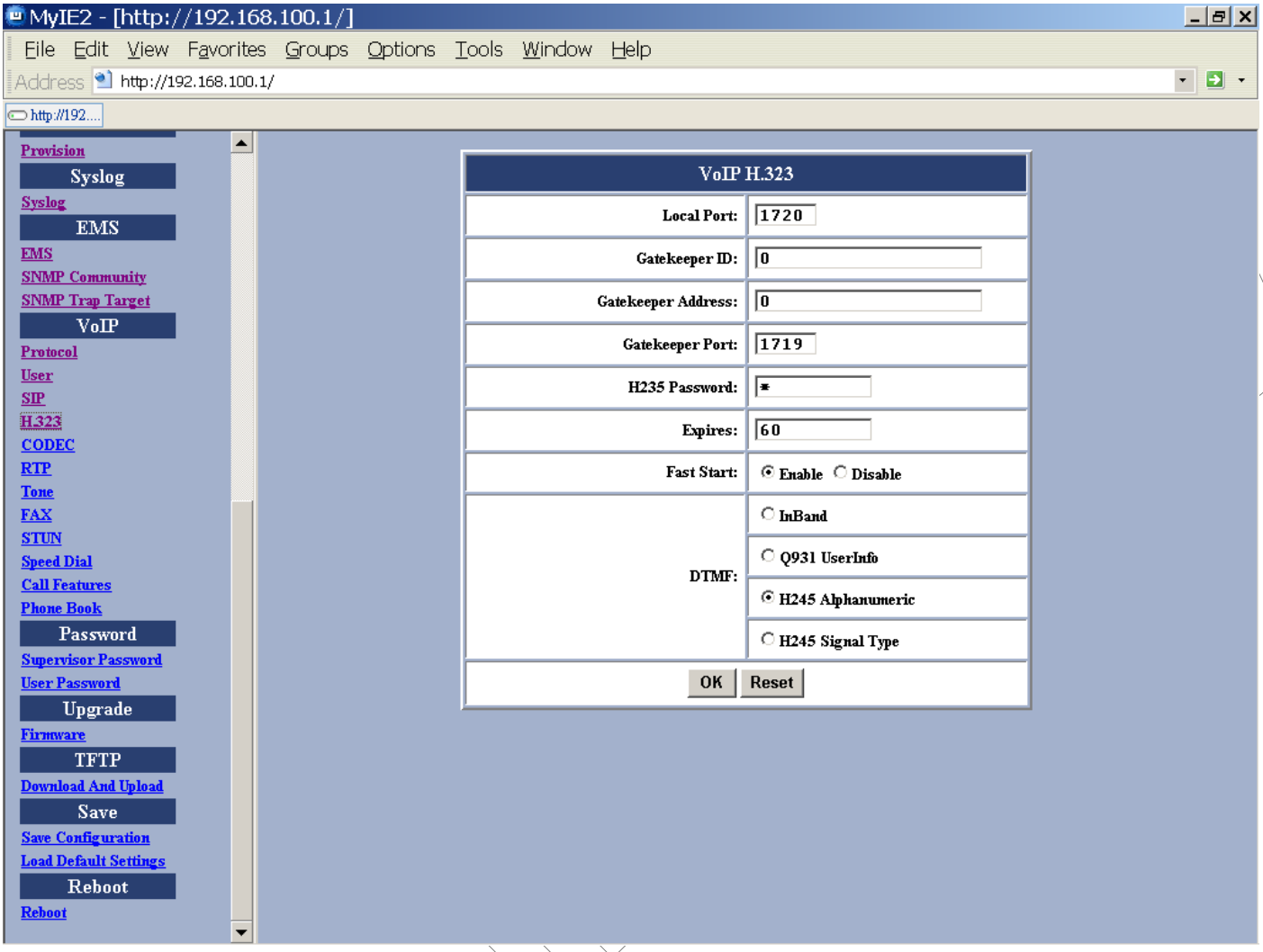


Figure 29 – VoIP SIP Configuration Window

Item	Description
Local Port	Specifies the port number of the H.323 stack. 1720 is the default port number.
Gatekeeper ID	Specifies the ID of Gatekeeper, this content will be carried in the H.323 PVT/PST frame for Gatekeeper to check.
Gatekeeper Address	Specifies the IP address of Gatekeeper

Gatekeeper port	Specifies the port number of Gatekeeper, default is 1719.
H235 Password	For authentication check, now this function is not available.
Expires	"Expires" specifies the period (in seconds) that the VoIP Gateway sends Registration message to Registrar server. This is used to maintain the connection between VoIP Gateway and Gatekeeper.
Fast Start	To specify the fax operation mode; Enable is the Fast start mode or Disable is set to Normal start mode.
DTMF	Specifies the DTMF signal types.

CODEC

This screen allows you to set CODEC configurations including Codec Rate, Preferred Codec, and VAD. After making any setting, click **OK** and then **Reset** for the new settings to take effect.

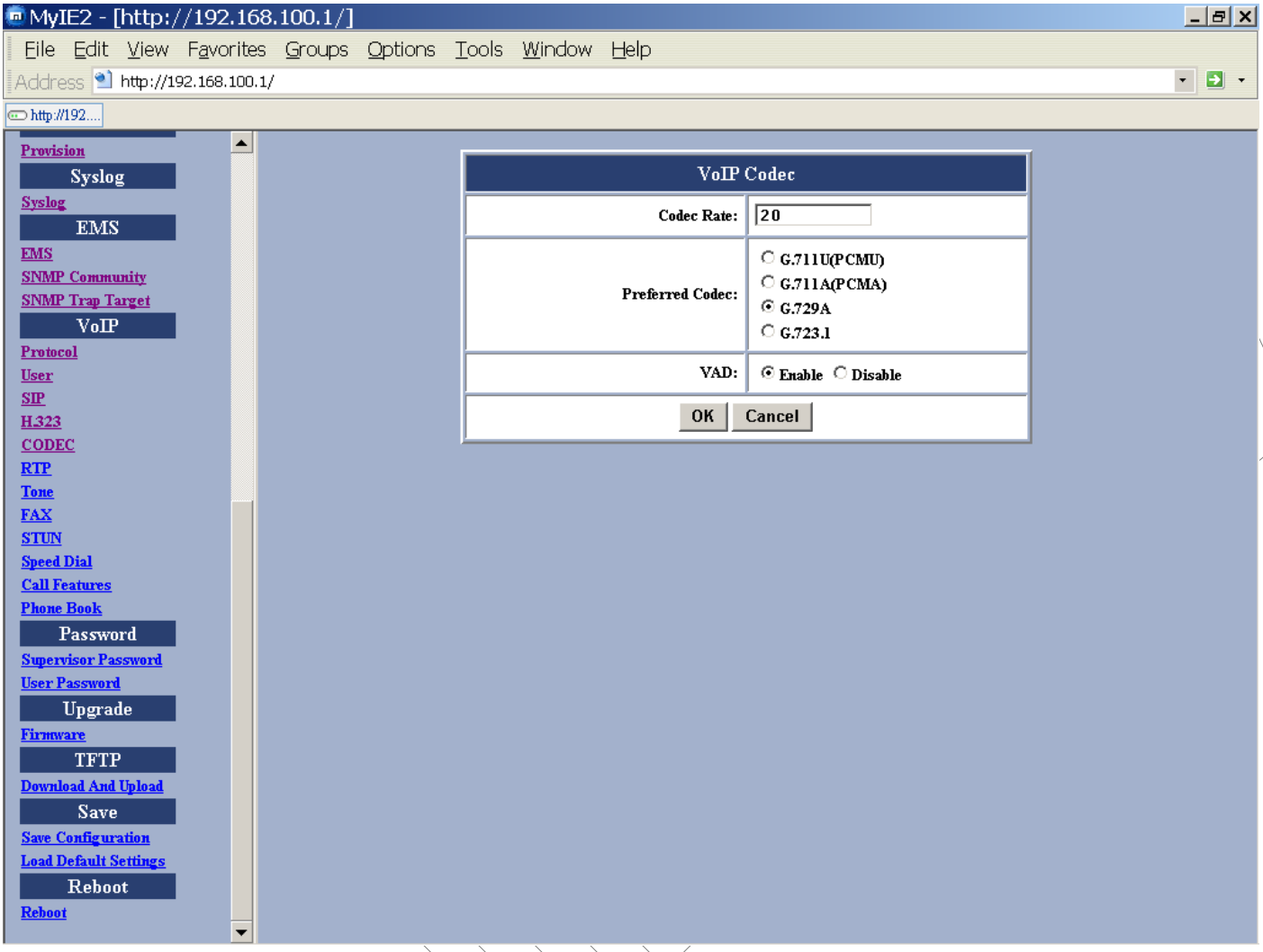


Figure 30 – VoIP Codec Configuration Window

Item	Description
CODEC Rate	"CODEC rate" specifies how long to send a voice packet(RTP), in milliseconds. G.711u/A: from 10 to 40 ms in 10 ms increment, default is 20ms. G.729: from 10 to 80 ms, in 10ms increment, default is 20 ms. G.723.1: from 30 to 90 ms, in 30ms increment, default is 30 ms.
Preferred	To specify the preferred method of voice compression.

CODEC	
VAD	Voice Activity Detection feature. Enabled: sending the silent packets while the user is no speaking. This will save the bandwidth but cause the time delay. Disabled: Normal RTP packet is sent no matter the user is speaking or not. This will improve the voice quality to be more smoothly but increase more traffic load.

RTP

This page allows the user to set RTP port number. After making any setting, click **OK** and then **Reset** for the new settings to take effect.

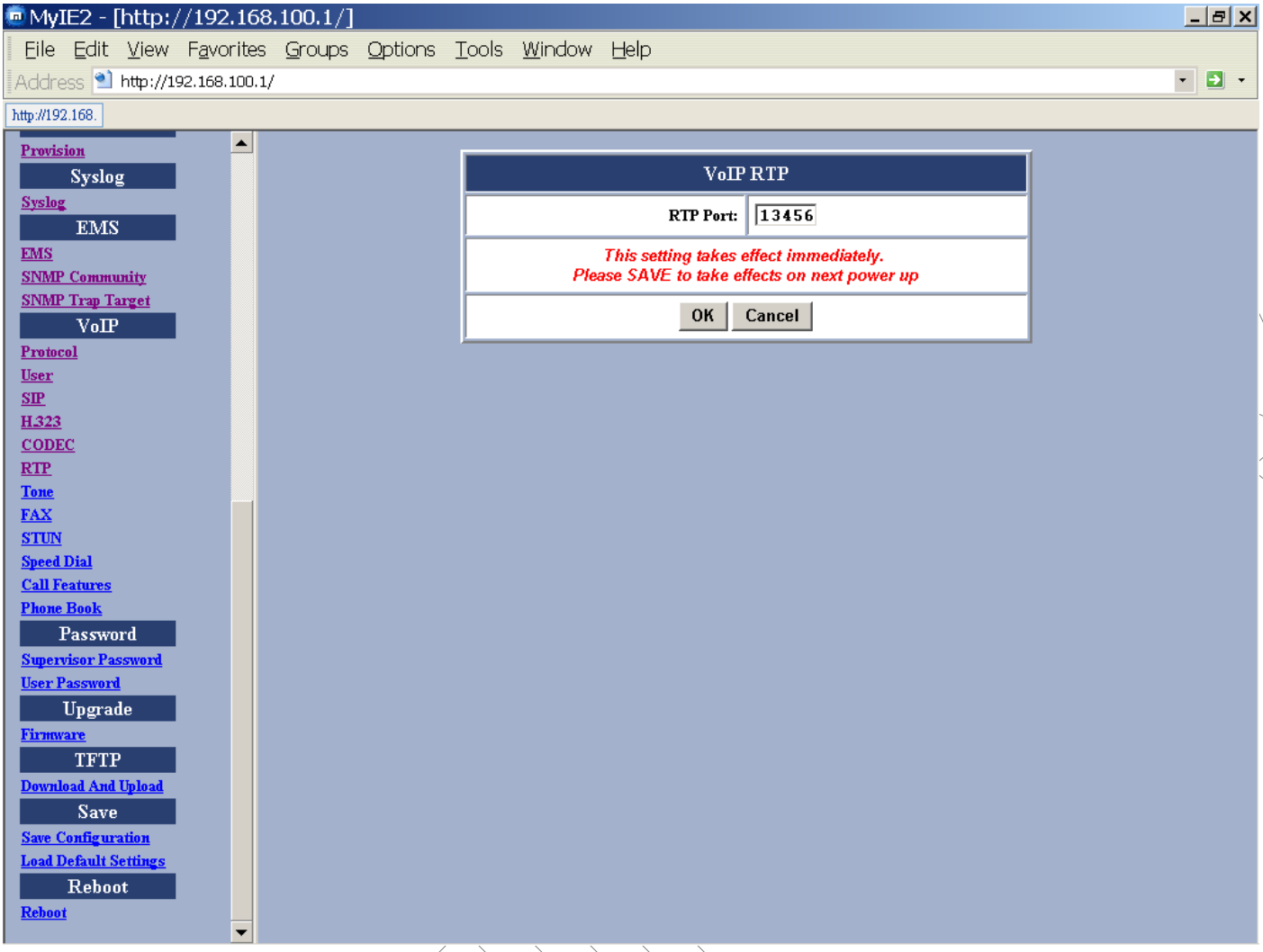


Figure 31 – VoIP RTP Configuration Window

Item	Description
RTP port	Specifies a port number. This port will be used to send and receive voice packets..

Tone

This page allows the user to configures the ringing pattern and the tones heard during call establishment, (dial, busy, ring back and call waiting tones)and also to set receive and transmit levels,. You can either choose country specific tone or you can freely configure the pattern and the pitch used for each tone. After making any setting, click **OK** and then **Reset** for the new settings to take effect.

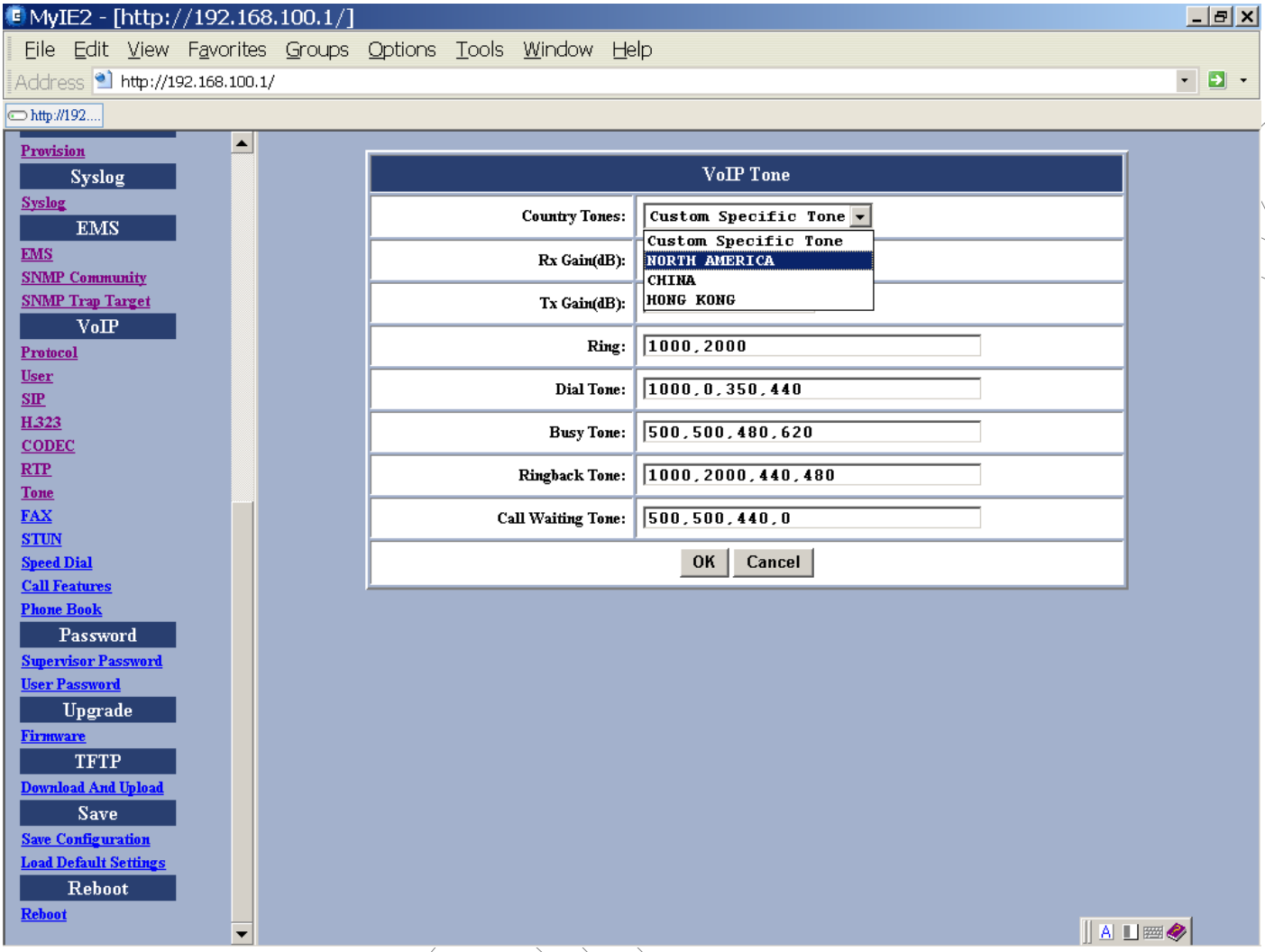


Figure 32 – VoIP Tone Configuration Window

Item	Description
Rx Gain	Adjusts the receiving audio gain to be higher or lower
Tx Gain	Adjust the transmitting audio gain to be higher or lower
Ring	Sets the ringing cadence (in milliseconds).

	<ontime, offtime>
Dial Tone	Sets the dial tone pattern <ontime, offtime (in milliseconds), freq1, freq2 (in Hz)>
Busy Tone	Sets the busy tone pattern <ontime, offtime (in milliseconds), freq1, freq2 (in Hz)>
Ring Back Tone	Sets the ring back tone pattern <ontime, offtime (in milliseconds), freq1, freq2 (in Hz)>
Call Waiting Tone	Sets the call waiting tone pattern <ontime, offtime (in milliseconds), freq1, freq2 (in Hz)>

FAX

This screen allows user to set the port number for sending/receiving T.38 packets. T.38 protocol supports data-resending mechanism in case of any missing data during transmission. After making any setting, click **OK** and then **Reset** for the new settings to take effect.

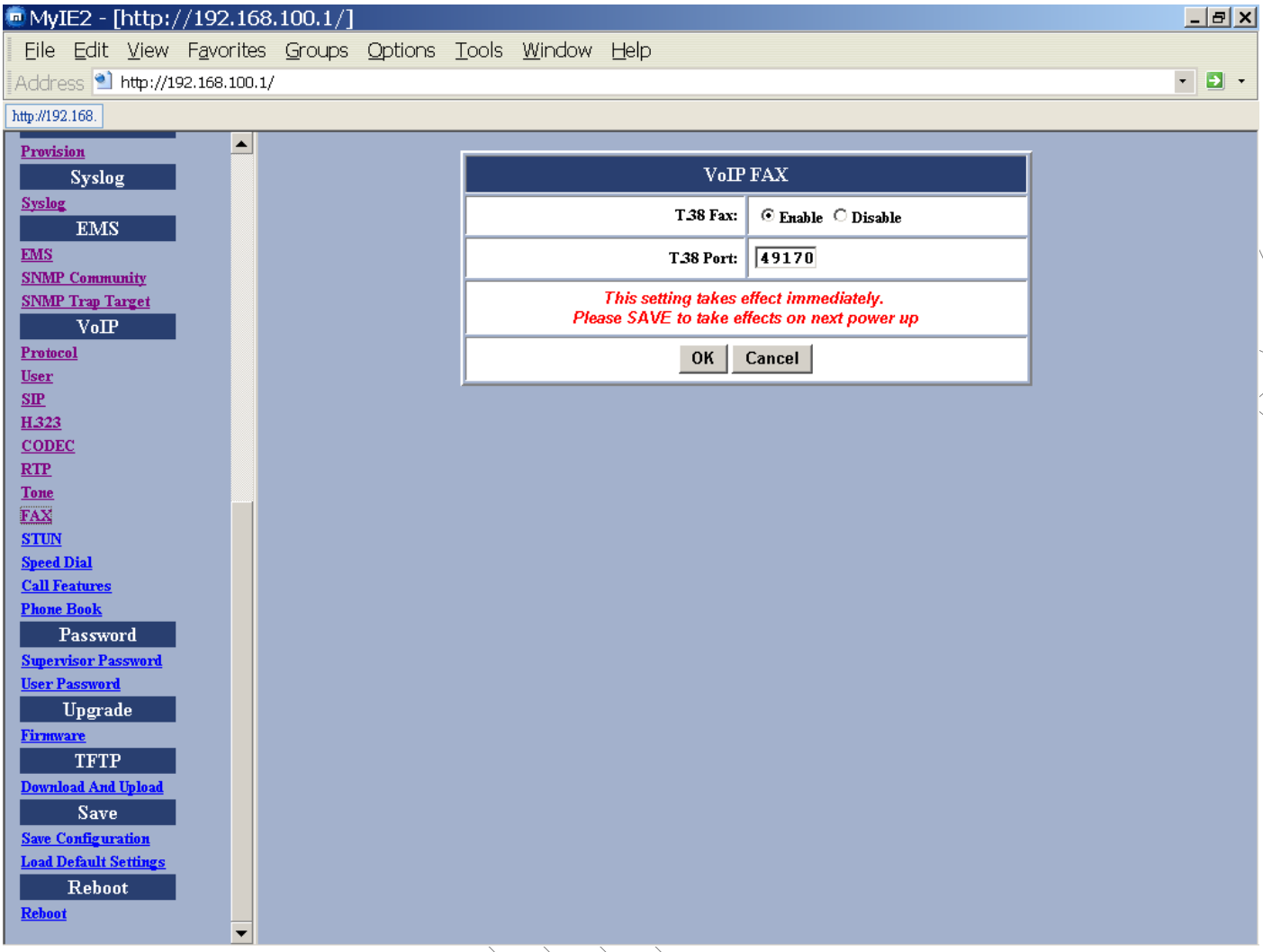


Figure 33 – VoIP Fax Configuration Window

Item	Description
T.38 port	Specifies the T.38 port number for sending/receiving T.38 packets

STUN

This screen allows user to set NAT address, STUN server address, STUN server port, local port and expiry time. After making any setting, click **OK** and then **Reset** for the new settings to take effect.

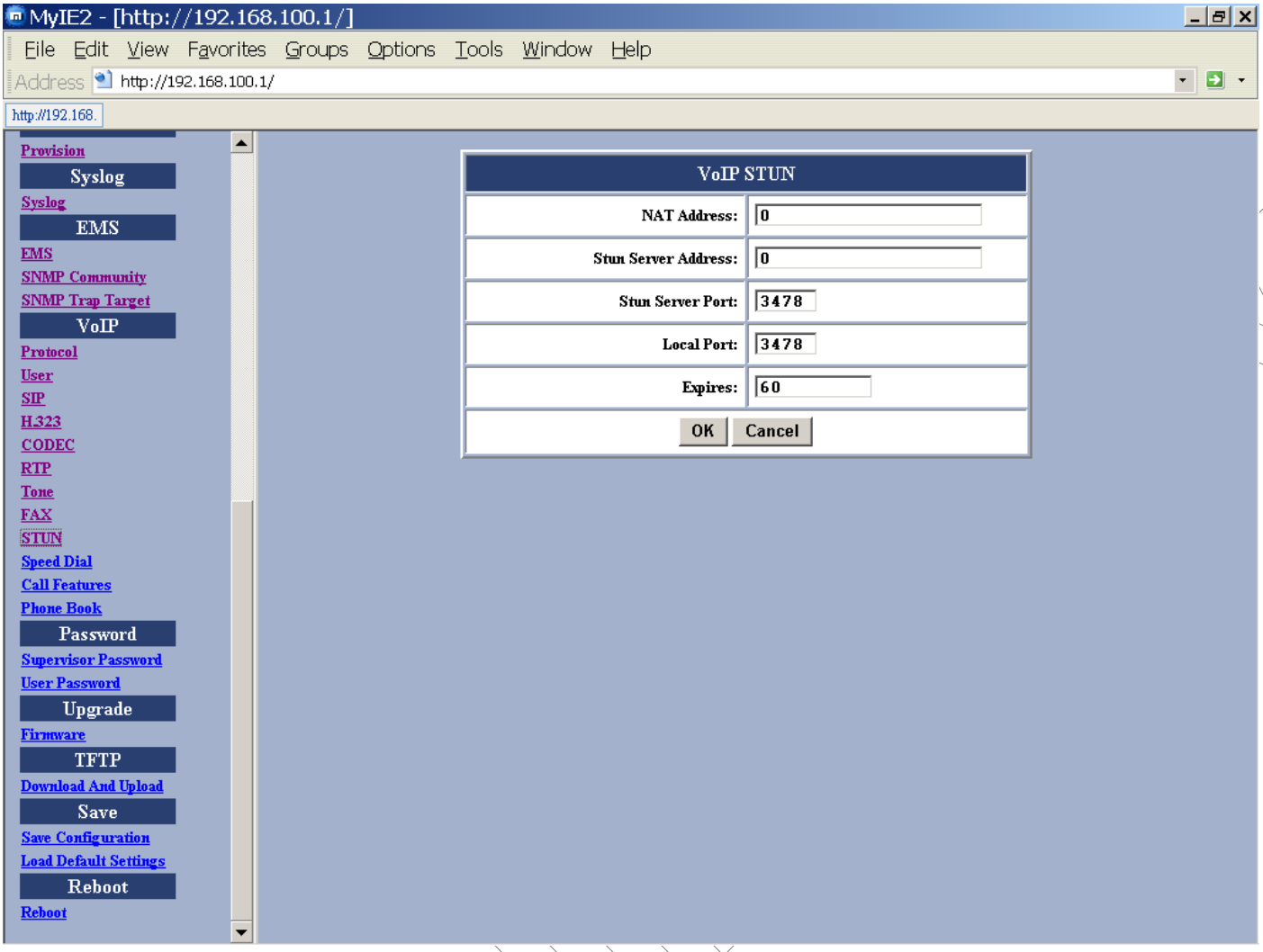


Figure 34 – VoIP STUN Configuration Window

Item	Description
NAT Address	Statically specifies the IP address of the TA for VoIP if it is installed behind a NAT. The IP address is the WAN side IP address from the NAT device.
STUN Server Address	Specifies the IP address of STUN server (Simple Traversal of User Datagram)
STUN Server Port	Specifies the port number of STUN server

Local Port	Specifies the local port number of STUN client
Expires	"Expires" specifies the period (in seconds) that the VoIP Gateway sends STUN message to STUN server. This is to help check the connection status in case the VoIP Gateway is accidentally disconnected from STUN server.

Note

User can dynamically set the IP address for VoIP using STUN. Please set the NAT address to 0 if STUN method is used. Vice versa, if NAT address is used, set the STUN Server Address to 0.

Speed Dial

The speed dial is used to set up a list of most frequently used telephone numbers and SIP addresses.

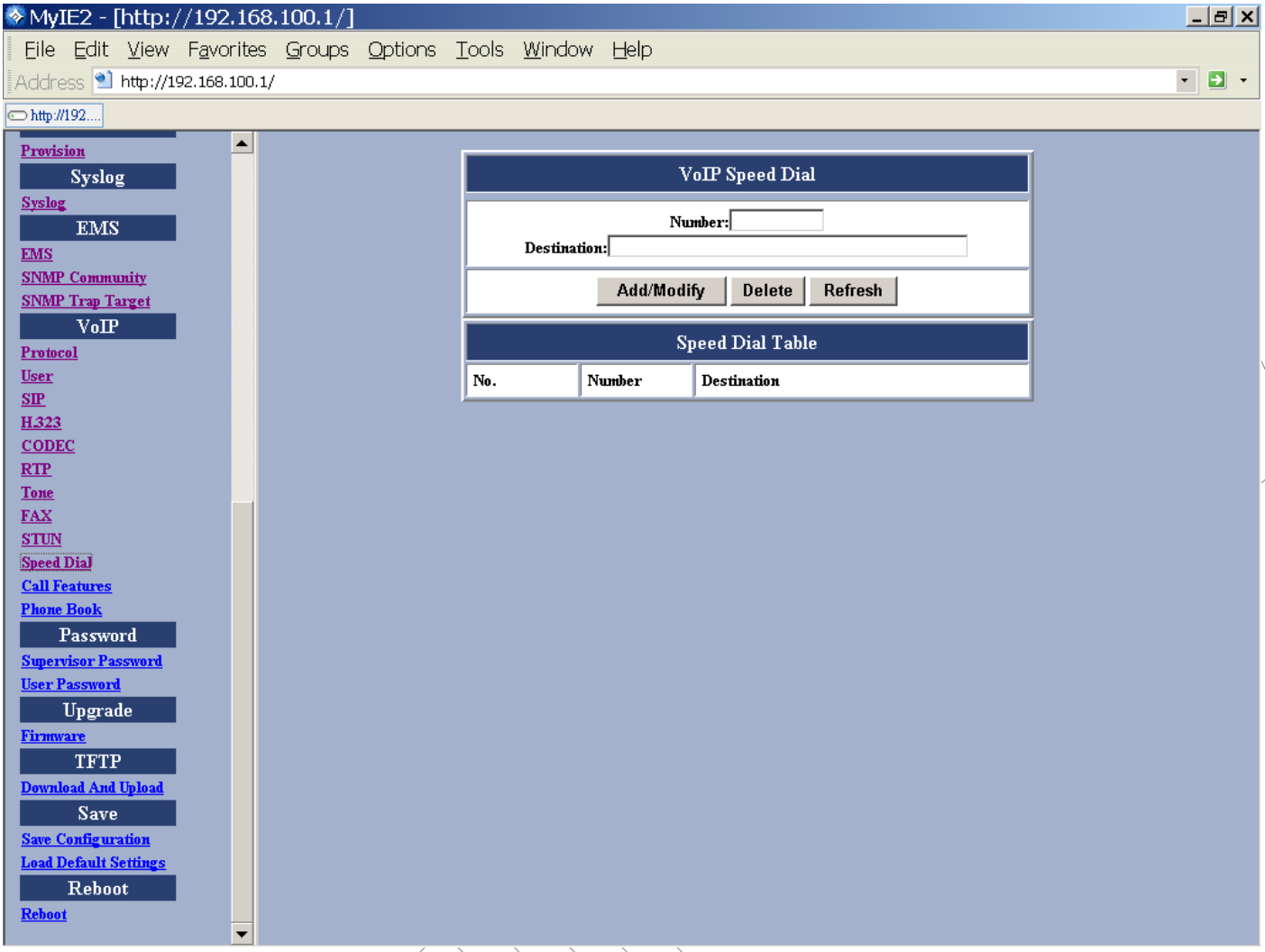


Figure 35 – VoIP Speed Dial Configuration Window

Item	Description
Number	Specifies the abbreviated number of the call party.
Destination	Enter the SIP address (or PSTN number) of the call party (Example: leon.tung@172.11.123.20)
Add/Modify	Add or modify the telephone number and SIP address of the call party.
Delete	Delete an existing telephone number and SIP address of the call party from the Speed Dial Table.
Refresh	Pressing this button will show new changes.

Call Features

This page sets Call features for SIP IAD including call Hold, call Waiting, call Forwarding. After you make the settings, click **OK** and then **Save**, **Reset** to take effect.

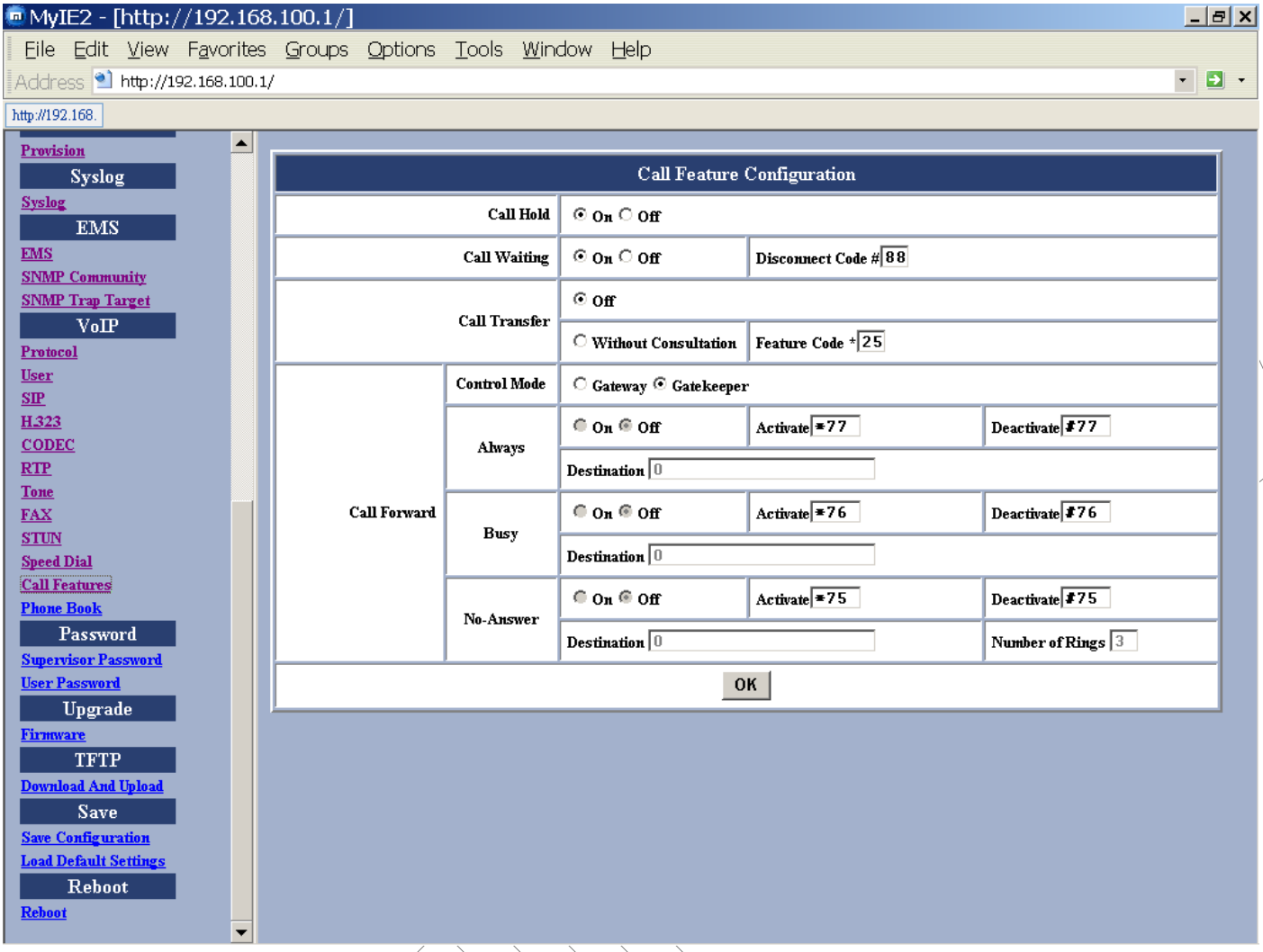


Figure 36 – VoIP Call Feature Configuration Window

Item	Description
Port	Configure port 1 or port 2 – For **ViP 3002** Only
Call Hold	Enable or Disable Call Hold feature. User may use flash key to hold the other party. Once call hold is disabled, call waiting is also disabled.

Call Waiting	Enable or Disable Call Waiting feature. If a user is talking with one party and the other call come in, a user can use flash key to switch to either party. If a user want to disconnect with one party and talk with the other one, a user need to enter disconnect code
Call Transfer	Enable or Disable Call Transfer feature, User could congfigure the feature code to perform call transfer function without consultation.
Call Forwarding	Enable or Disable Call Forwarding. There are 3 types Call Forwarding. **Always:** Unconditionally forward a call to the destination that user configured. **Busy:** Forward a call to the destination that user configured only when the line is busy **No-Answer:** Forward a call to the destination that user configured when nobody answer this call after # of rings

Phone book

You can edit the phone book to map the IP and phone number.

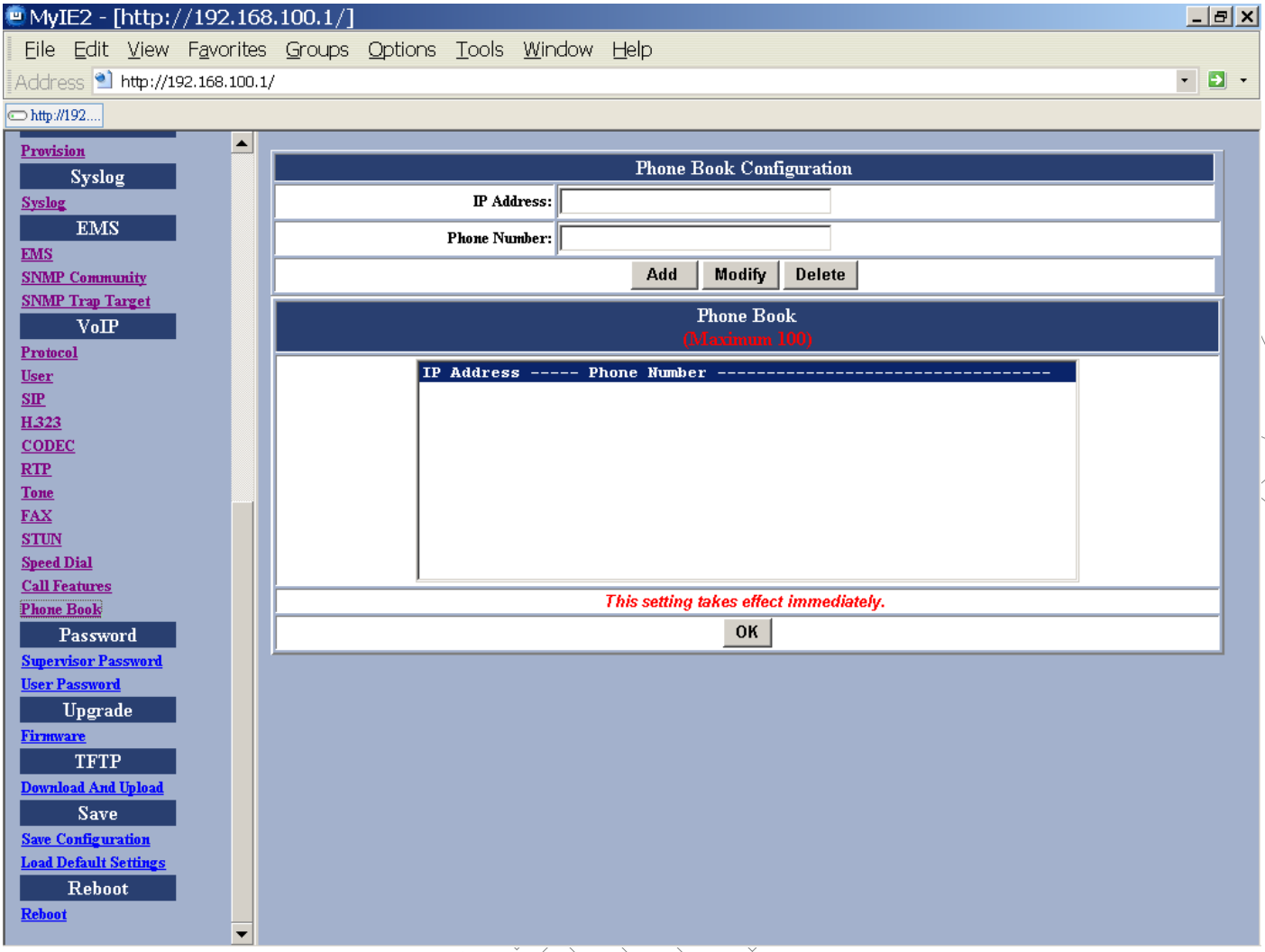


Figure 37 – VoIP Phone Book Window

Password Configuration

Supervisor Password

The password will be used for authentication. It is recommended that you reset the password for administrator security.

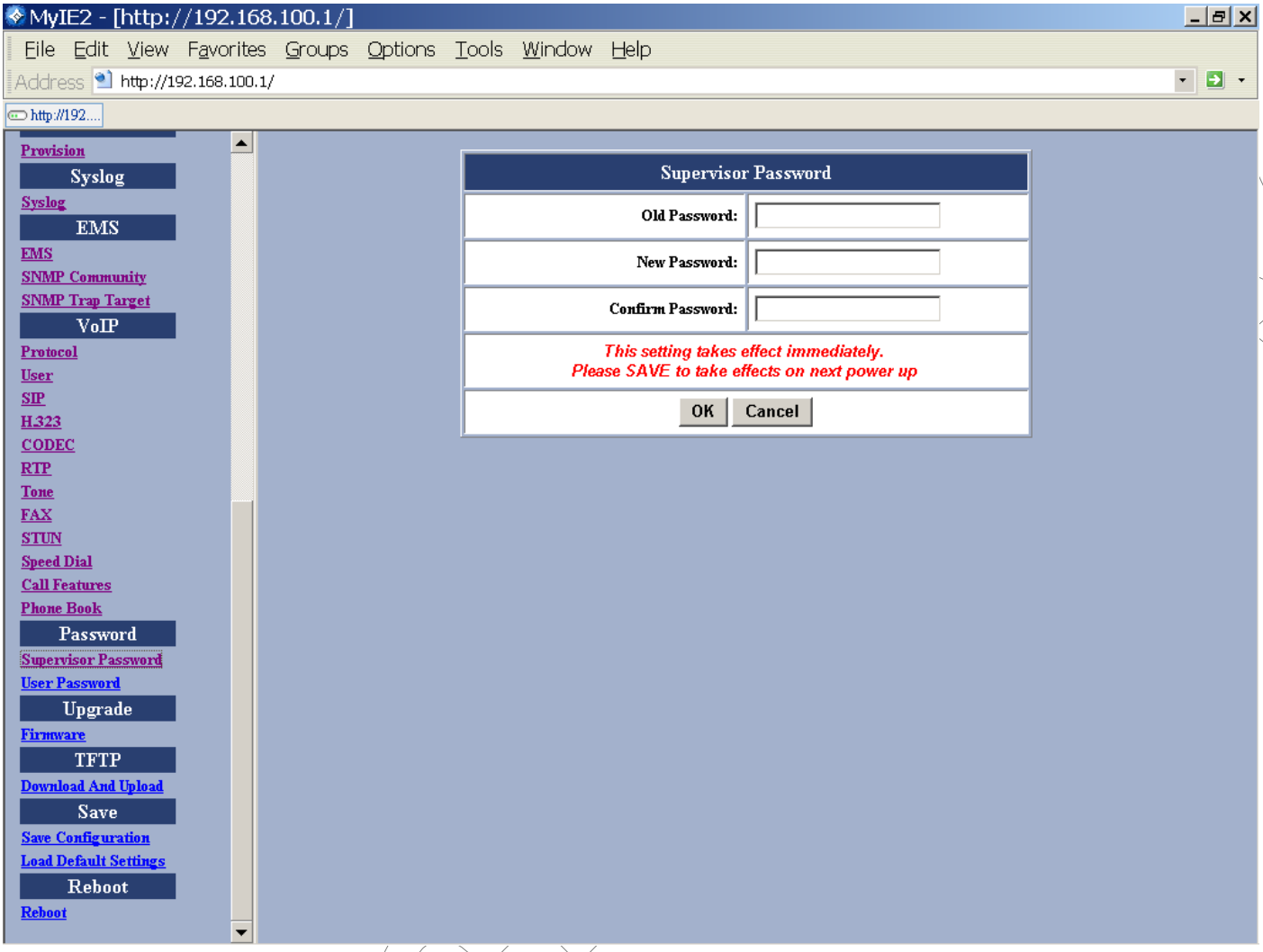


Figure 38 –Supervisor Password Window

Item	Description
Old Password	Enter the predefined password.
New Password	Enter the new password.
Confirm Password	Re-enter the new password in this field to ensure it is correct..

User Password

The password will be used for authentication. It is recommended that you reset the password for user security.

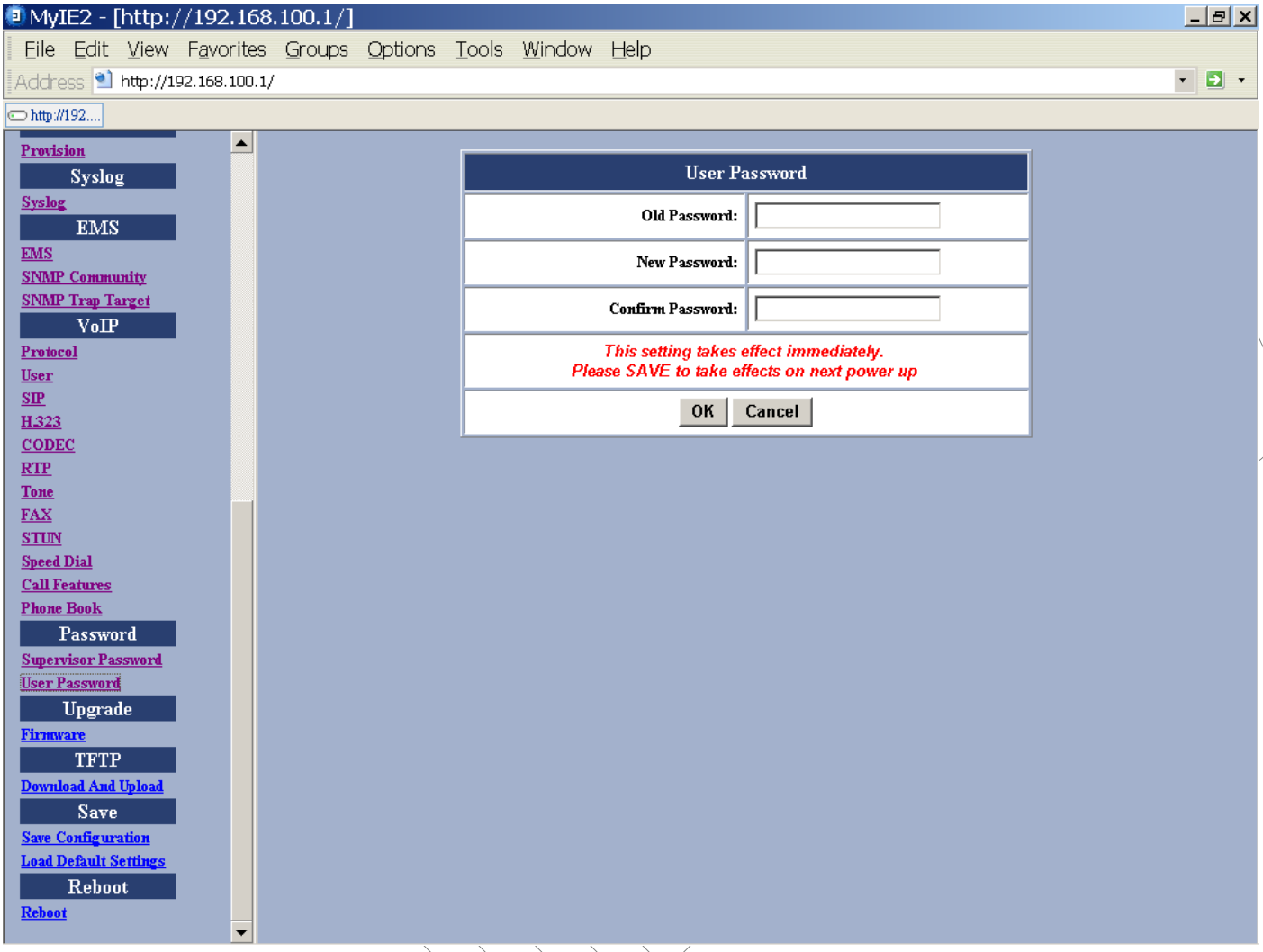


Figure 39 – User Password Window

Item	Description
Old Password	Enter the predefined password.
New Password	Enter the new password.
Confirm Password	Re-enter the new password in this field to ensure it is correct..

Upgrade Configuration

Firmware

This feature allows you to upgrade the firmware on the VoIP Gateway from the web browser. The firmware on the VoIP Gateway is stored on FLASH ROM. To upgrade firmware, you need to download the firmware to your local computer first. Once the new firmware is downloaded, click **Browse** to locate the new firmware on your computer. Then click **Upgrade** to complete the process.

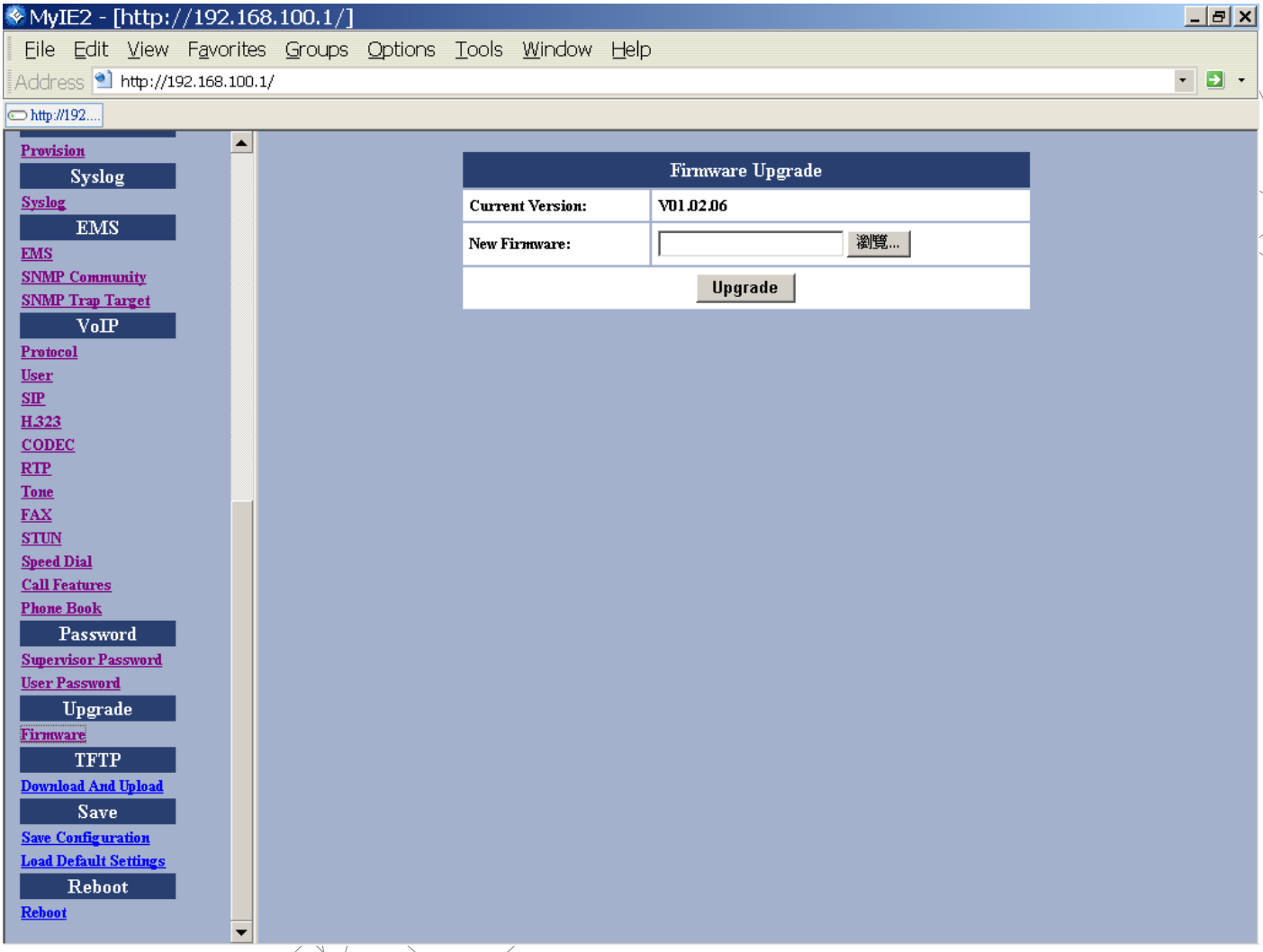


Figure 40 – Firmware Upgrade Window

Configuration

The upgrade process is the save as firmware upgrade but here is the **configuration file**.

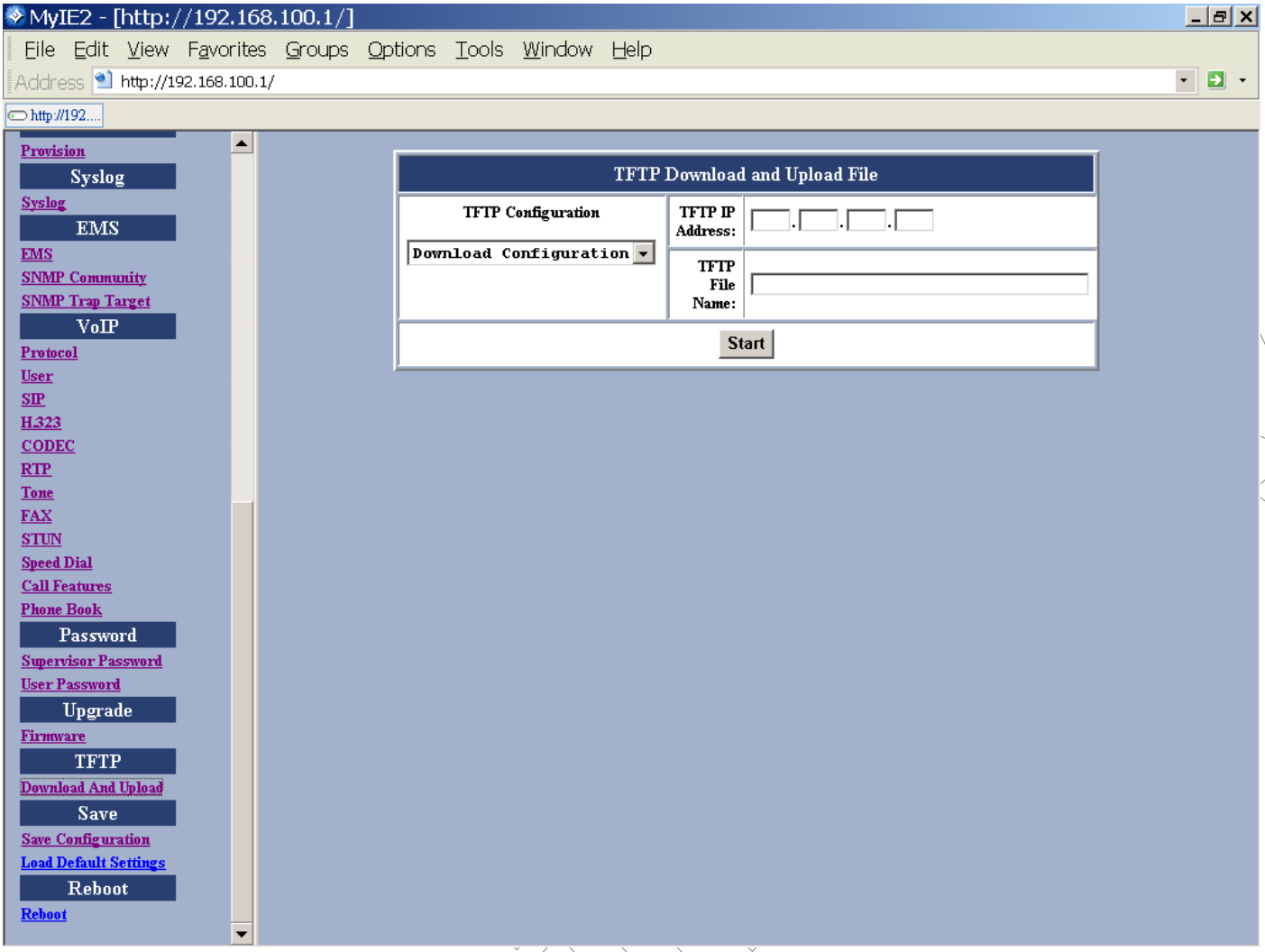


Figure 41 – Configuration Upgrade Window

Save

Save Configuration

Whenever you change into a new configuration, you need to save the new configuration data and then restart this device to have new settings take effect. Once you click on the "**Save**" button from the window below, the new configuration data is automatically written into the FLASH memory and the system will be refreshed with new data on your next reboot (refer to the following section "Reboot").

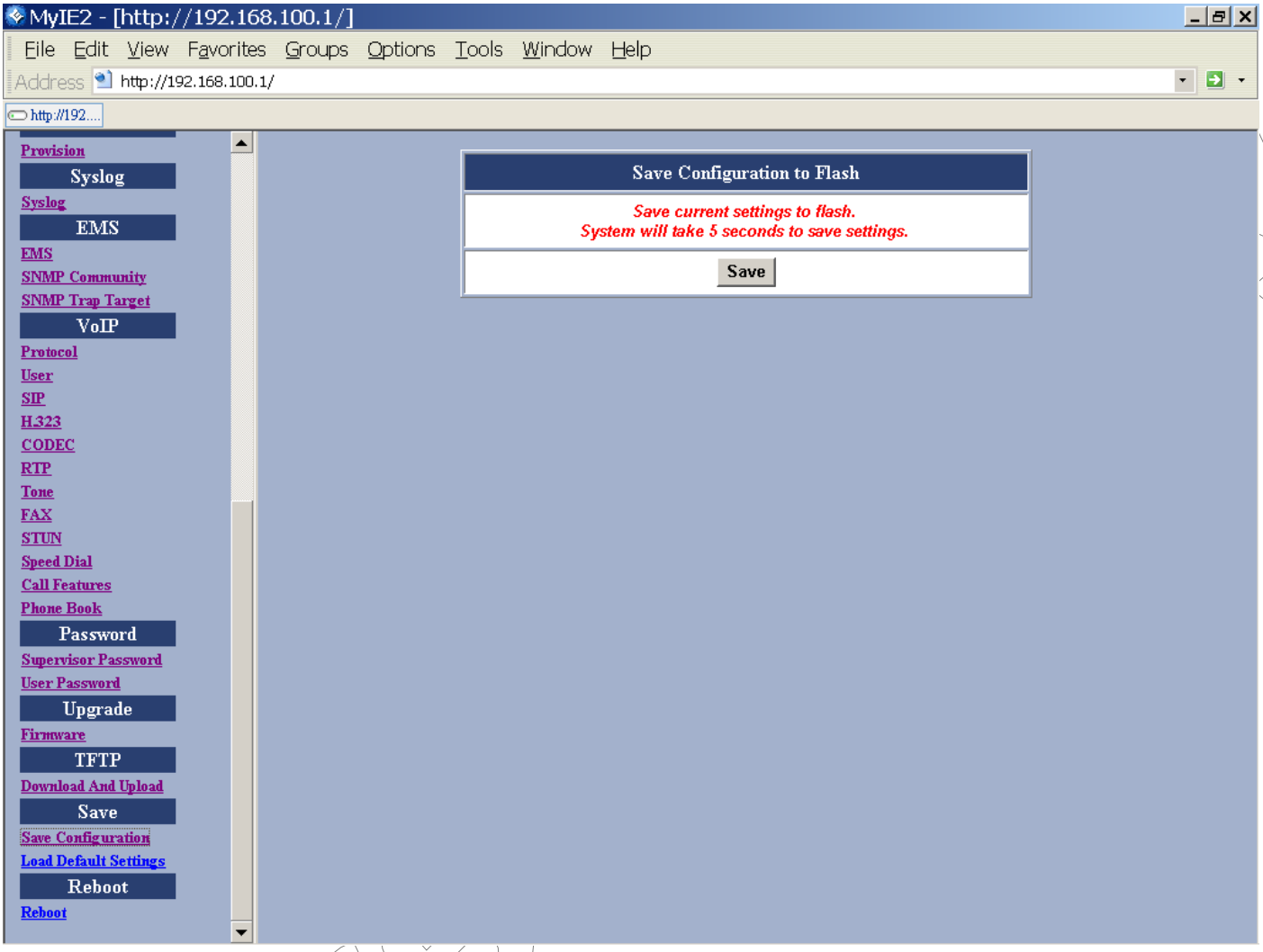


Figure 42 – Save Configuration to Flash Window

Load Default Settings

Click on the "**Load**" button if you would like to restore all default settings of the ViP 3001/2. Restart the device for the new settings to take effect. See the next section "Reboot" for more information.

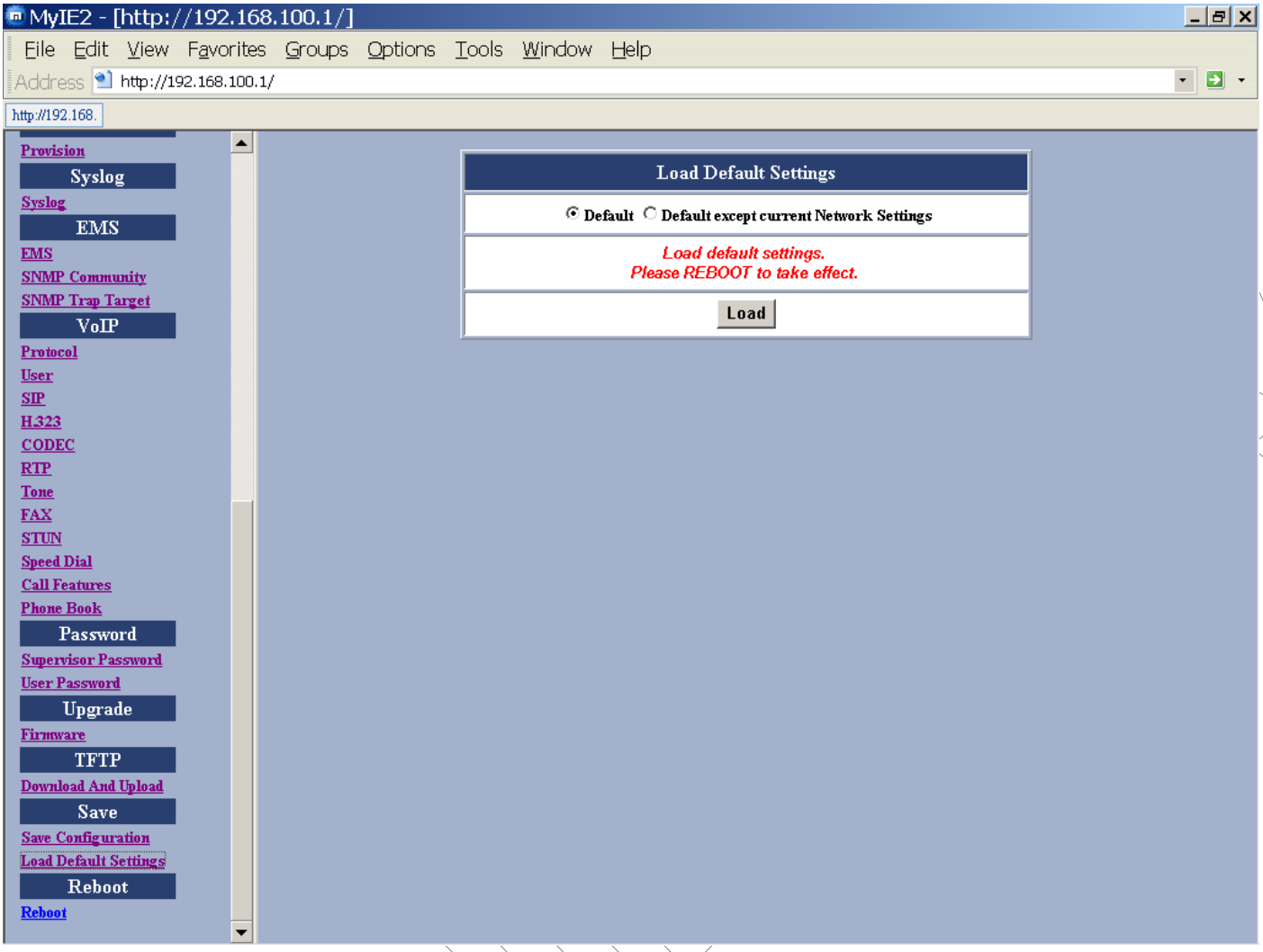


Figure 43 –Load Default Settings Window

Item	Description
Default	All value will reset to default
Default except current Netowrk Settings	All value will reset to default while the network setting (WAN IP) still remain.

Reboot

Once you click **Reboot**, the system will restart and be updated with new configuration data stored in the flash.

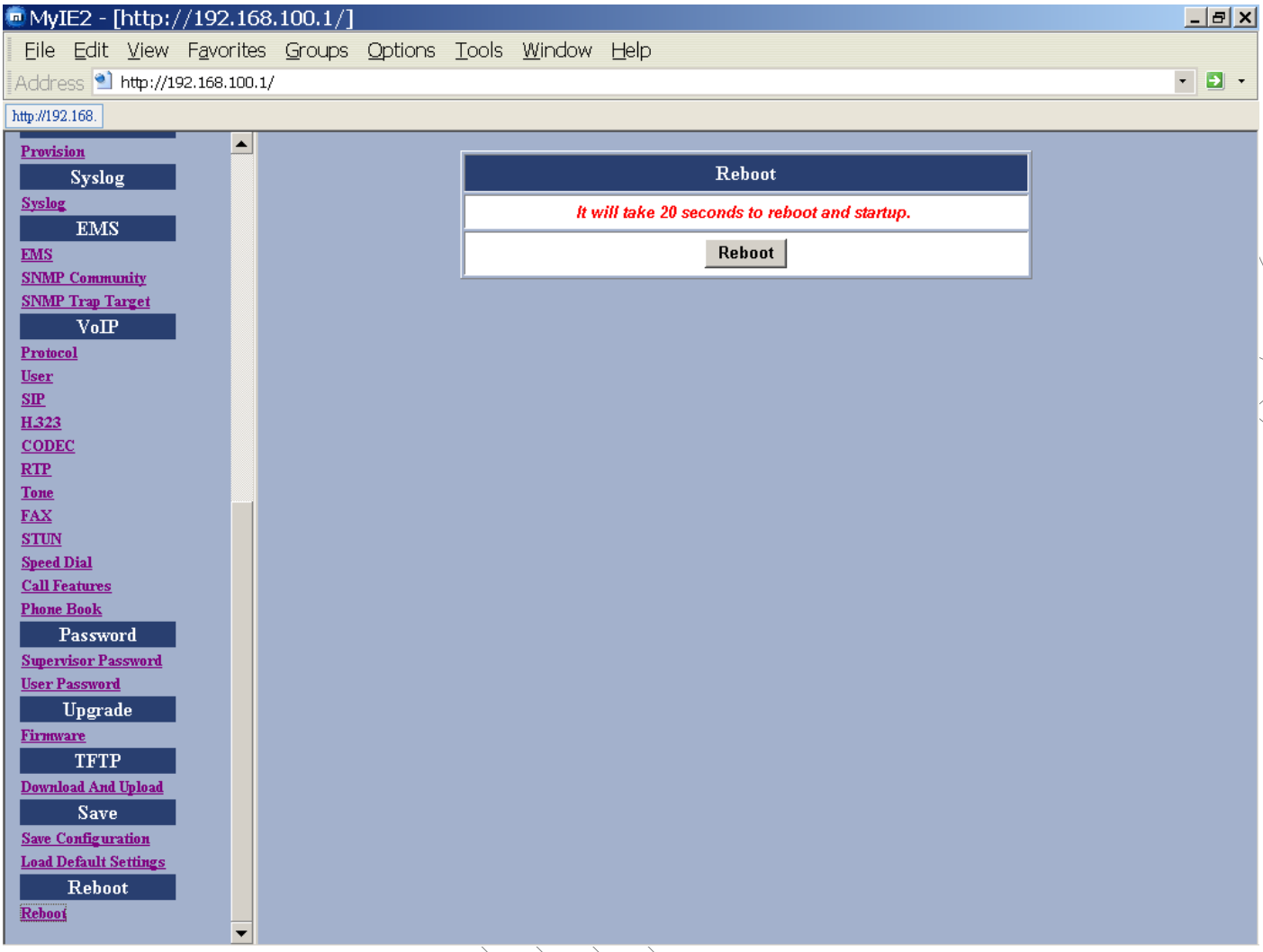


Figure 44 – Reboot Window

4. Appendix A: Troubleshooting

This section covers possible problems that may be encountered while using the ViP 3001 and 3002 Voice Gateways and suggested solutions to them. If you follow the suggested solutions below but the ViP 3001 and 3002 Voice Gateways still does not work properly, contact technical support for further advice.

Q: Power LED does not light up.

S: First check the AC adapter rating. The input rating must meet the specification of the country. The AC adapter must be DC 10V/1.2A output.

S: If the AC adapter output is correct. The problem will be on the VoIP Gateway. Please replace the VoIP Gateway.

Q: Ethernet interface cannot work.

S: Make sure the Ethernet adapter card installed in the PC is workable. The technician can use Hub/Switch to test it.

S: Make sure the Ethernet cable is workable, and the connection between PC and the VoIP Gateway is secure.

Q: Broadband access cannot work.

S: Make sure the Ethernet cable is workable, and the connection between Broadband device and the VoIP Gateway is secure.

S: Check the DHCP or PPPoE server setting. You have to enter correct username and password for PPPoE registration.

Q: Cannot download the proper configuration file.

S: Check if the connection between Provisioning Server and the VoIP Gateway is secure.

S: Check if the file name and setting of Provisioning file are correct.

Q: VoIP LED does not light up.

S: Check if configuration file indicates correct IP address and informatiosn of Soft-Switch.

S: Check if the VoIP Gateway is able to connect to Soft-Switch.

S: Check if the authorization content between the VoIP Gateway and Soft-Switch are the same.

Q: Cannot use PSTN backup line.

S: Disconnect the VoIP Gateway from the power supply and then check if PSTN backup line is workable.

S: Check the settings of "PSTN switch key and digit map" are correct.

5. Appendix B: Specifications

Call Control Protocols Compliance	MGCP, SIP, H.323
Voice Compression	G.711 μ , G.729a, G.723
Bandwidth Management	Voice Activity Detection (VAD), Comfort Noise Generation (CNG)
Analog Voice Ports	Type: Loop-Start FXS interfaces Echo Cancellation: G.168
Fax Support	T.30: Auto Switch T.38: no Auto Switch
Ethernet Ports	WAN: 10/100Base-TX Fast Ethernet port LAN: 10/100Base-TX Fast Ethernet port
Quality of Service	Type of Service (TOS) supported Internal voice & data priority queue supported
Network Protocols	TCP/IP, ICMP, ARP, UDP, TFTP, RTP, RTCP, HTTP, Telnet, SNMP PPPoE Client DHCP Client
Network Management	Web-based configuration software TFTP firmware upgrade and configuration back and restore TELNET server for remote management
Security	Password protected system management terminal
LEDs	Ethernet LAN: 10/100M, Link/Activity WAN: 10/100M, Link/Activity VoIP Ready LINE1: Activity LINE2: Activity Power
Number of Ports	WAN: One 10/100BASE-TX Fast Ethernet port LAN: One 10/100BASE-TX Fast Ethernet port PHONE: Two loop-start FXS RJ-11 ports LINE: PSTN Loop through backup for power outage POWER
Power Supply	Output voltage: 12VDC/1.1A
Operating Temperature	0° C~40° C
Storage Temperature	-10° C ~70° C
Operating Humidity	10%~90%, non-condensing
Storage Humidity	5%~95%, non-condensing
Dimensions	190 (W) x 130 (D) x 30 (H) mm
Weight	335.45g
EMI	FCC Part 15 Class B

Safety	CSA International (UL 60950 3rd, EN 60950, IEC 60950 3rd)