TW-VoIP-S2
User manual

1. Features

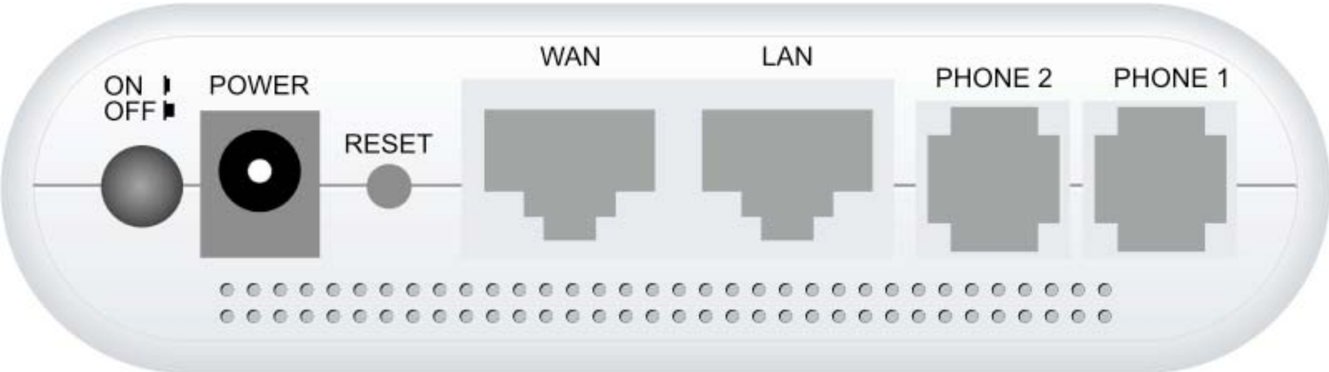
Network Protocol	Tone
● SIP v1 (RFC2543), v2(RFC3261) ● IP/TCP/UDP/RTP/RTCP ● IP/ICMP/ARP/RARP/SNTP ● TFTP Client/DHCP ● Client/ PPPoE Client ● Telnet/HTTP Server ● DNS Client ● NAT/DHCP Server	● Ring Tone ● Ring Back Tone ● Dial Tone ● Busy Tone ● Programming Tone
Codec	**Phone Function**
● G.711: 64k bit/s (PCM) ● G.723.1: 6.3k / 5.3k bit/s ● G.726: 16k / 24k / 32k / 40k bit/s (ADPCM) ● G.729A: 8k bit/s (CS-ACELP) ● G.729B: adds VAD & CNG to G.729	● Volume Adjustment ● Speed dial key ● Phone book ● Flash
Voice Quality	**IP Assignment**
● VAD: Voice activity detection ● CNG: Comfortable noise generator ● LEC: Line echo canceller ● Packet Loss Compensation ● Adaptive Jitter Buffer	● Static IP ● DHCP ● PPPoE
Call Function	**Security**
● Call Hold ● Call Waiting ● Call Forward ● Caller ID ● 3-way conference	● HTTP 1.1 basic/digest authentication for Web setup ● MD5 for SIP authentication (RFC2069/ RFC 2617)
DTMF Function	**NAT Traversal**
● In-Band DTMF ● Out-of Band DTMF ● SIP Info	● STUN
SIP Server	**Configuration**
● Registrar Server ● Outbound Proxy	● Web Browser ● Console/Telnet ● IVR/Keypad

Firmware Upgrade	**Auto Provisioning**
● TFTP ● Console ● HTTP	● HTTP ● FTP ● TFTP
Interface	**Modem & Fax modes**
1 WAN port interface 1 LAN port interface 1 PSTN port interface (FXO) (Optional) 1 VOIP port interface (FXS)	● G.711 fax/modem pass-through with fax/modem detection ● T.38 support

2. VoIP Gateway Overview

VoIP Gateway has many ports, switches and LEDs. VoIP Gateway may have some or all of the features listed below

2.1 Ports and Buttons



1WAN + 1 LAN + 2 FXS

POWER: Connect the power adapter that came with the VoIP Gateway. Using a power supply with a different voltage rating will damage this product. Make sure to observe the proper power requirements. The power requirement is DC12 volts/0.6 A.
POWER Switch: Power on/off the VoIP Gateway.
WAN Port: Connect to Broadband devices, such as a ADSL or Cable modem.
LAN Port: Connect to Ethernet network devices, such as a PC, hub, switch, or router. Depending on the connection, you may need a cross over cable or a strait through cable.
RESET: The RESET button will set the VoIP Gateway to its factory default setting and reset the VoIP Gateway. You may need to place the VoIP Gateway into its factory defaults if the configuration is changed, you loose the ability to enter the VoIP Gateway via the web interface, or following a software upgrade, and you loose the ability to enter the VoIP Gateway. To reset the VoIP Gateway, simply press the reset button for more than 10 seconds. The VoIP Gateway will be reset to its factory defaults and after about 30 seconds the VoIP Gateway will become operational again.

LINE Jack: Connect a telephone cable between the VoIP Gateway line jack and a wall jack.
PHONE Jack: Connect a standard telephone handset to the VoIP Gateway phone jack using a telephone cable.

2.2 LED Description

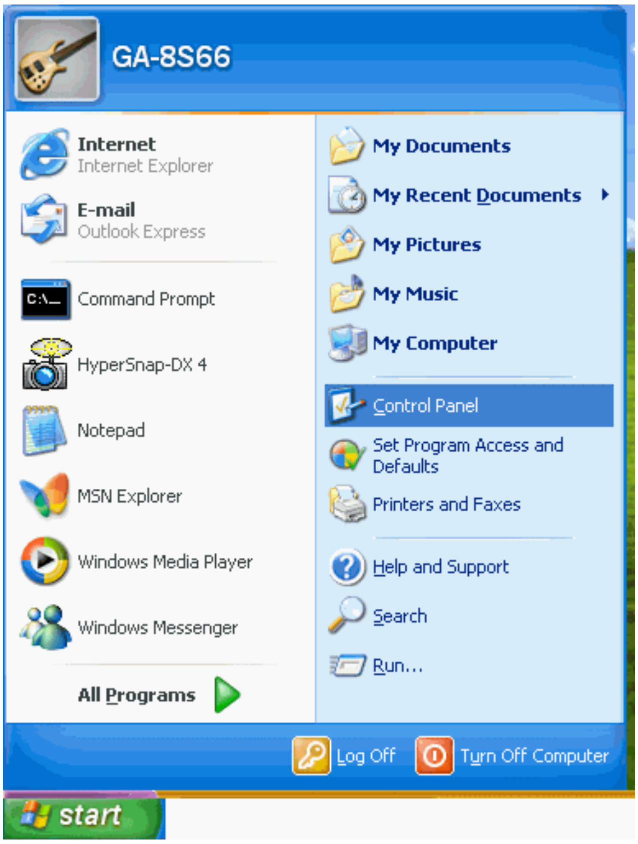
PWR LED: The LED stays lighted to indicate the system is power on properly.
SIP LED: This LED is lighted when the VoIP Gateway is REGISTERED successfully to the SIP Server.
ETH LED: The LED is lighted when a connection is established to WAN/LAN port and flashes when WAN/LAN port is sending/receiving data.

3. Installing VoIP Gateway

3.1 To check what the Internet/WAN access of your own Network is - DHCP Client, Static IP or PPPoE Client

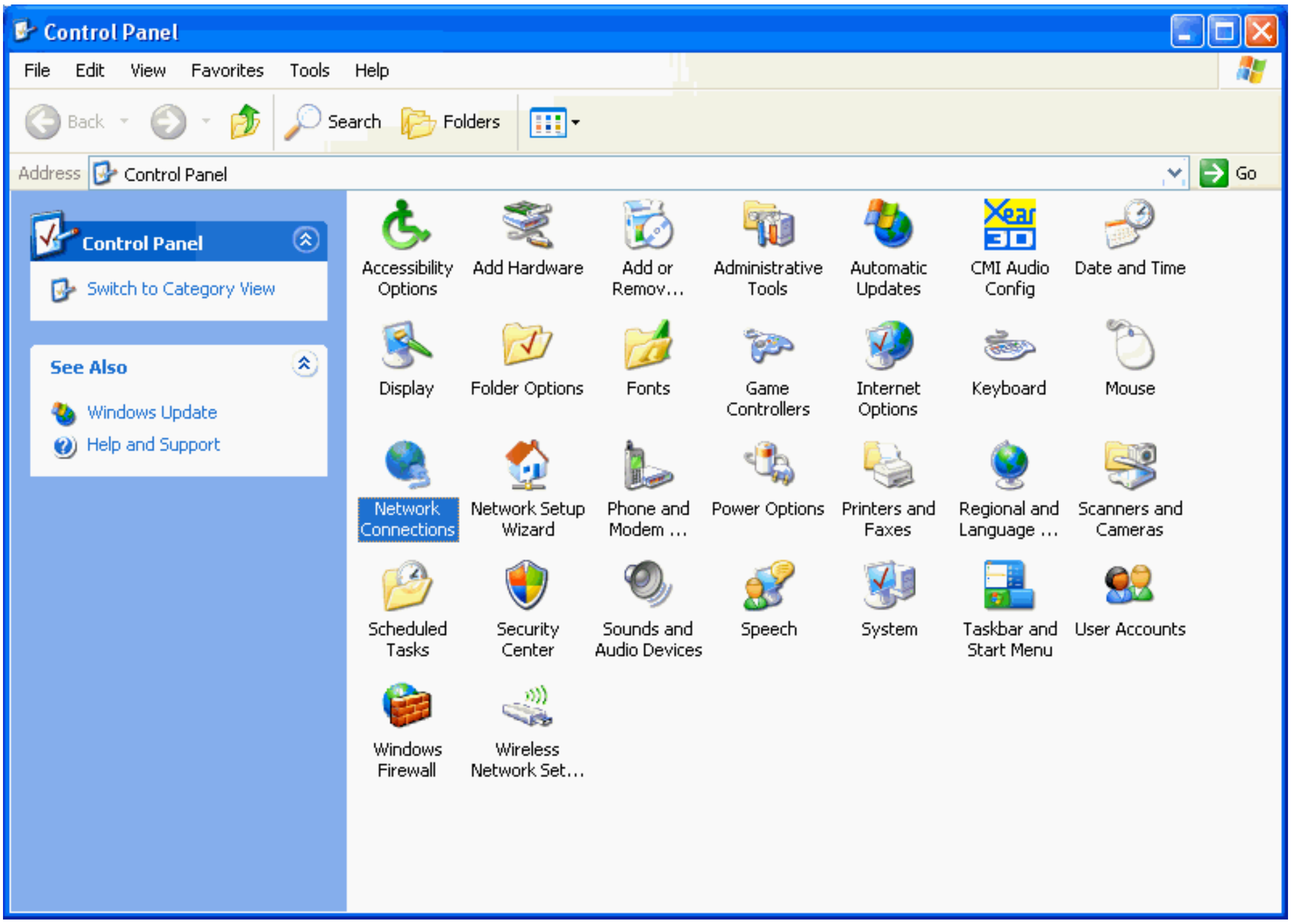
Please follow the steps below to check what the Internet/WAN access if your own Network is DHCP Client, Static IP or PPPoE Client.

Step 1:
Click " **Start -> Control Panel** "



Step 2:

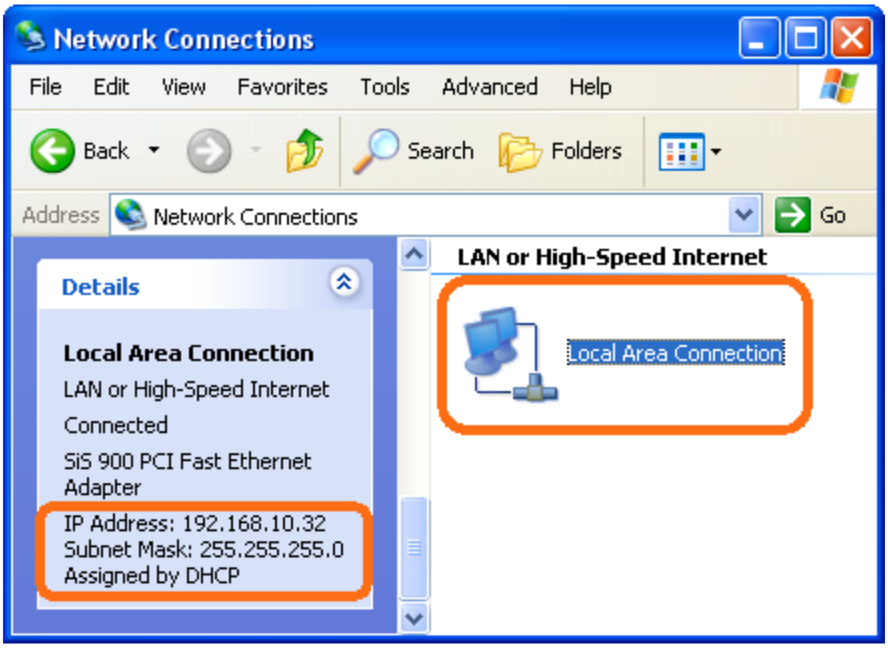
Double click " **Network Connections** "



Step 3-1 Internet/WAN access is the DHCP client:

If you cannot see any **Broadband Adapter** in **the Network Connections**, your Internet/WAN access is **DHCP Client** or **Static IP**.

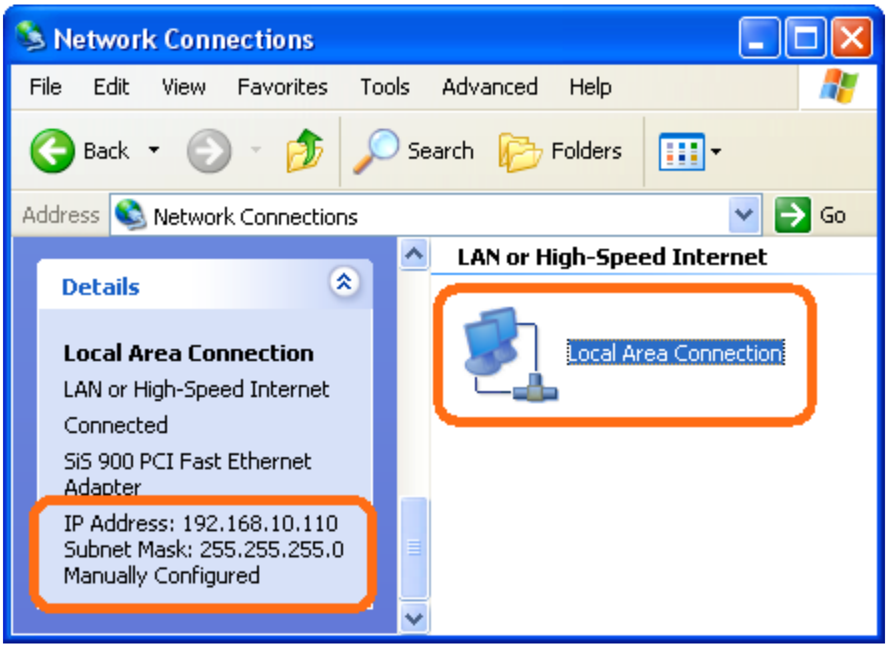
Click **" Local Area Connection "** in **LAN or High-Speed Internet** and you could see string **Assigned by DHCP** in **Details**.



Step 3-2 Internet/WAN access is the Static IP:

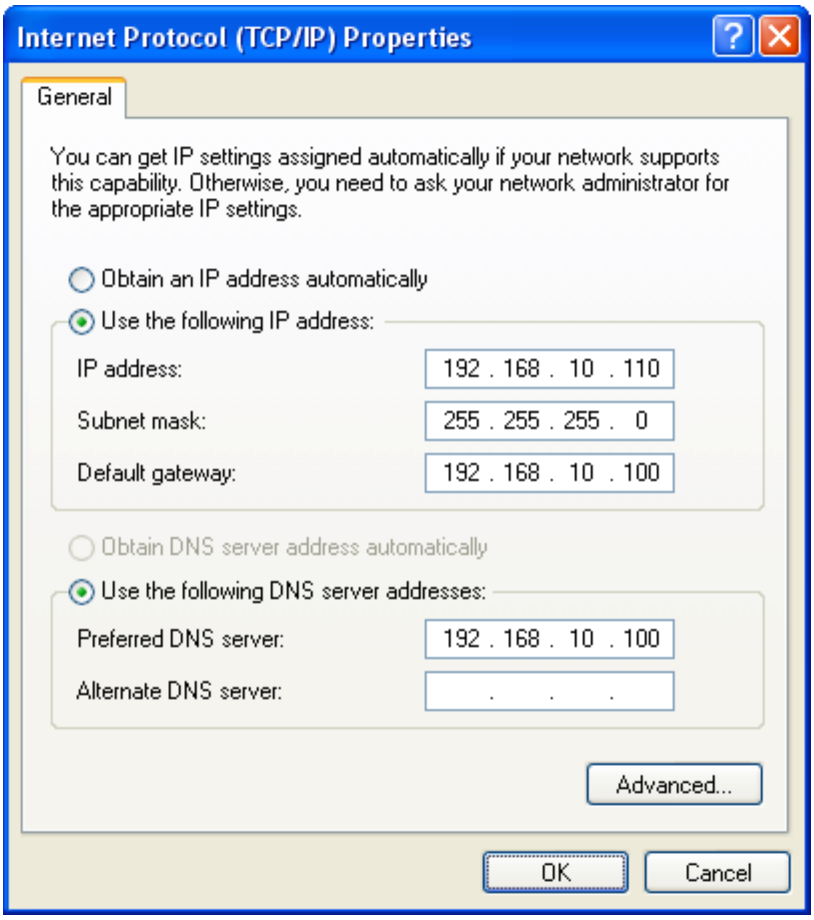
If you cannot see any **Broadband Adapter** in **the Network Connections**, your Internet/WAN access is **DHCP Client** or **Static IP**.

Click **" Local Area Connection "** in **LAN or High-Speed Internet** and you could see string **Manually Configured** in **Details**.



Right click " **Local Area Connection** " and click " **Properties** " and then you could get the IP settings in detail and write down the IP settings as follow:

IP Address: 192.168.10.110
Subnet mask: 255.255.255.0
Default gateway: 192.168.10.100
Preferred DNS server: 192.168.10.100
Alternate DNS Server: If you have it, please also write it down.



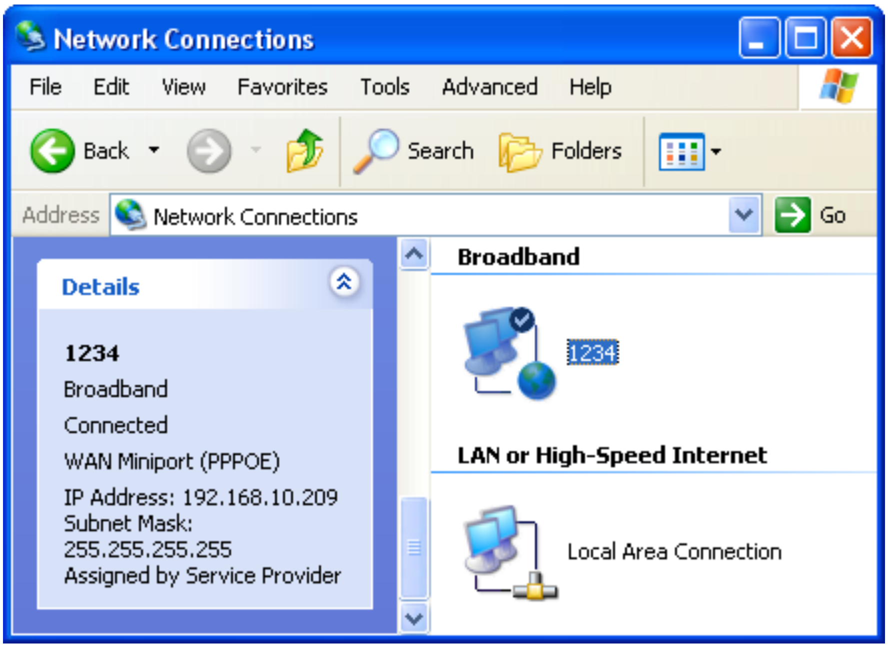
Step 3-3 Internet/WAN access is the PPPoE client:

If you can see any **Broadband Adapter** in **the Network Connections**, your Internet/WAN access is **PPPoE Client**.

Click **" Broadband Adapter "** in **Broadband** and you could see string **Assigned by Service Provider** in **Details**.

For PPPoE configuration on VoIP Gateway, you'll need following information that you could get from your Internet Service Provider.

Username of PPPoE: 1234 for example
Password of PPPoE: 1234 fpr example

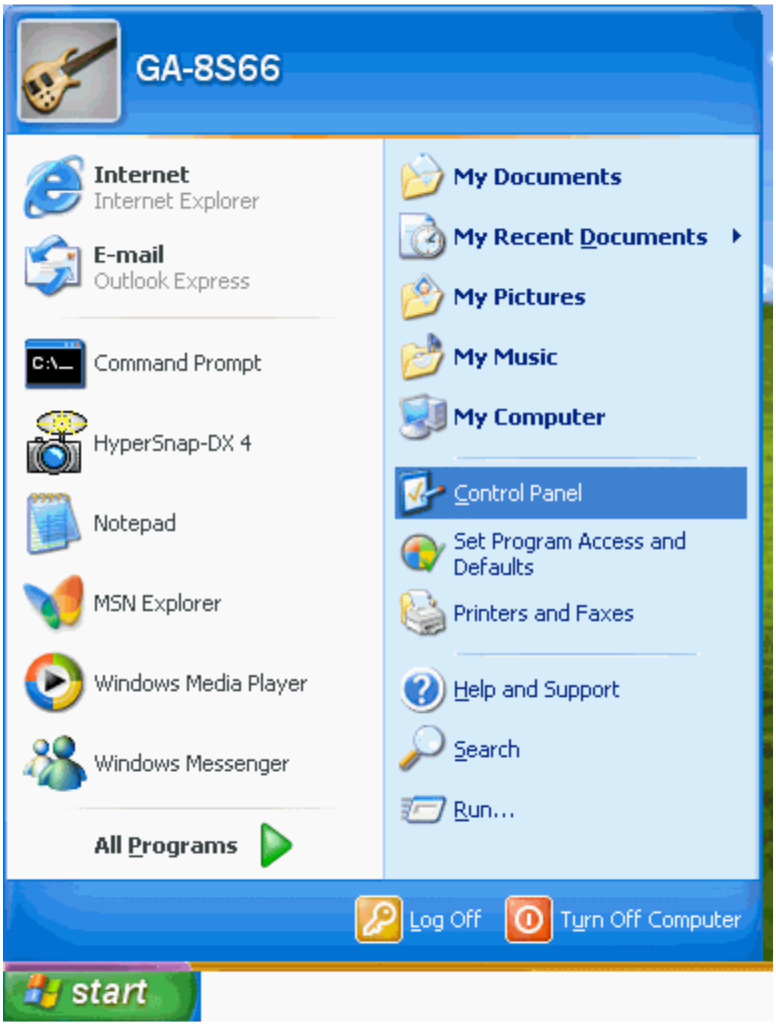


3.2 Configure the Obtain an IP Address automatically for LAN Card

To configure the VoIP Gateway by Easy Setup utility or Web page, please follow steps below to configure your LAN Card to obtain an IP Address automatically (DHCP Client).

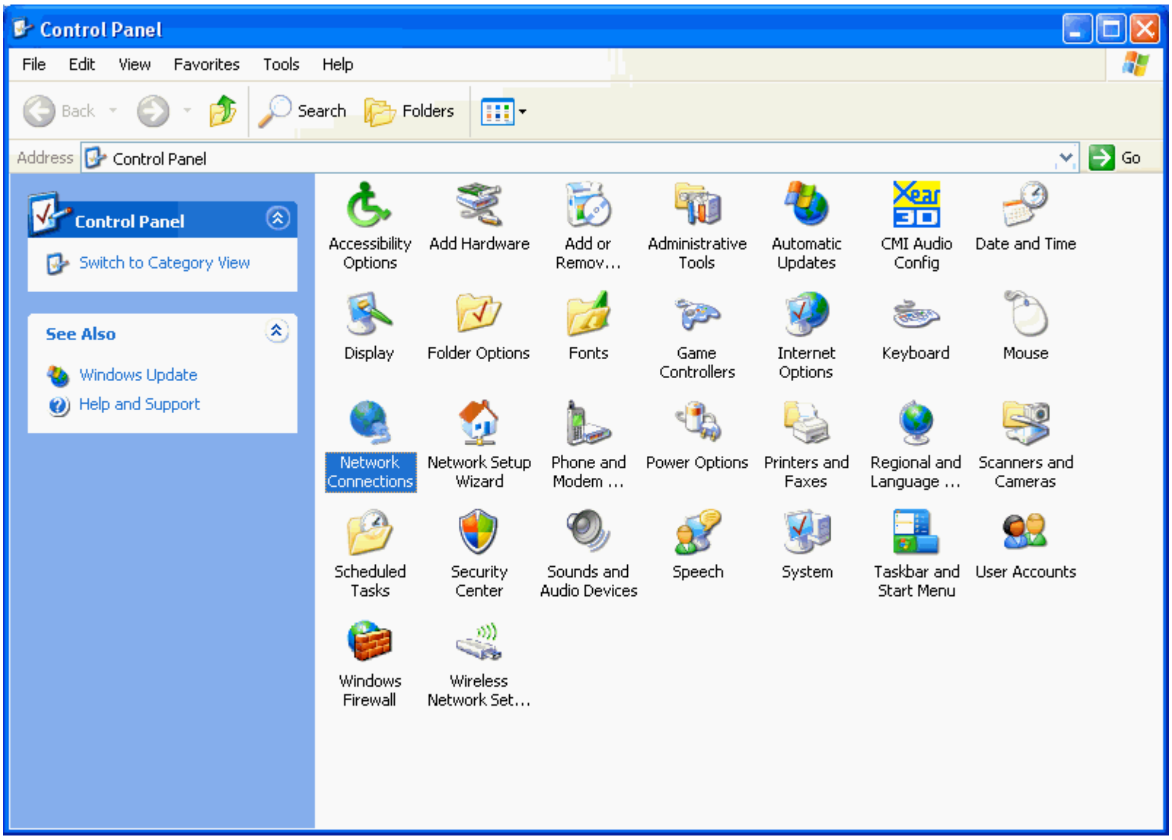
If your LAN Card is configured to obtain an IP Address automatically (DHCP Client), just skip this chapter.

Step 1:
Click " **Start -> Control Panel** "



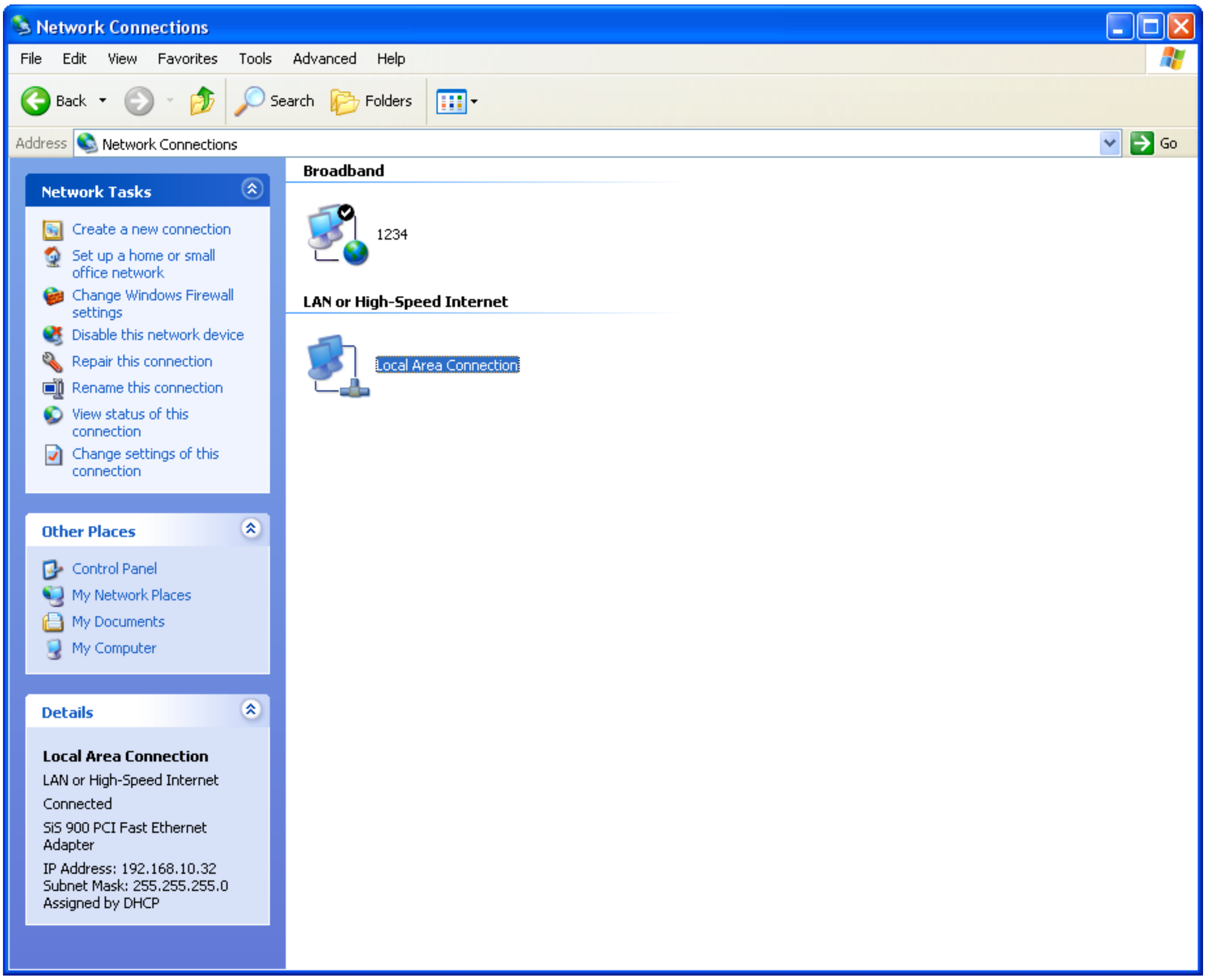
Step 2:

Double click " **Network Connections** "



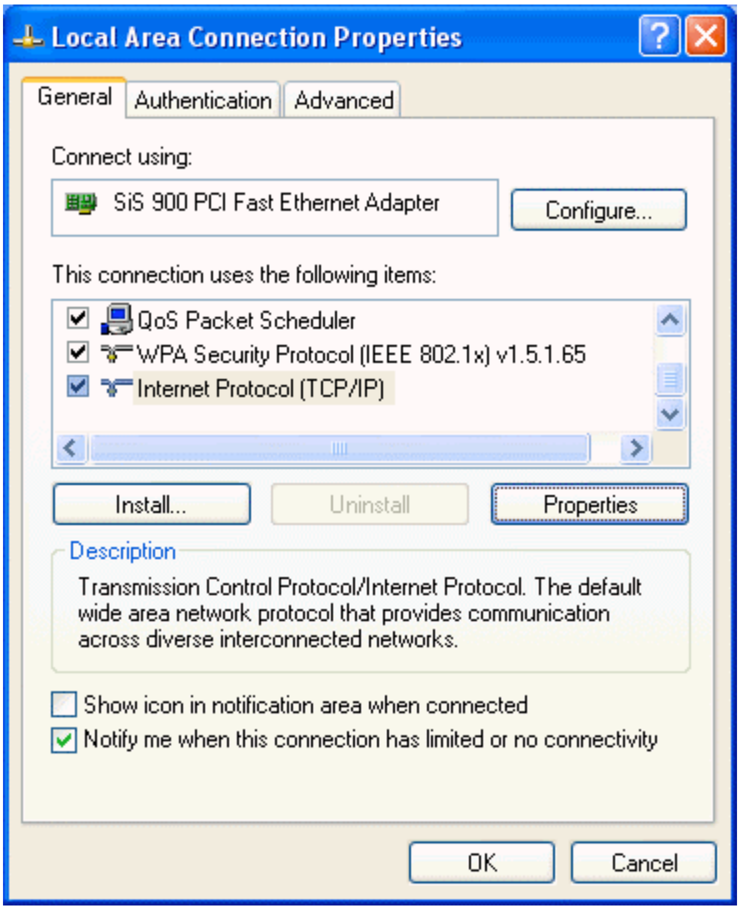
Step 3:

Right click " **Local Area Connection** " and then click " **Properties** "



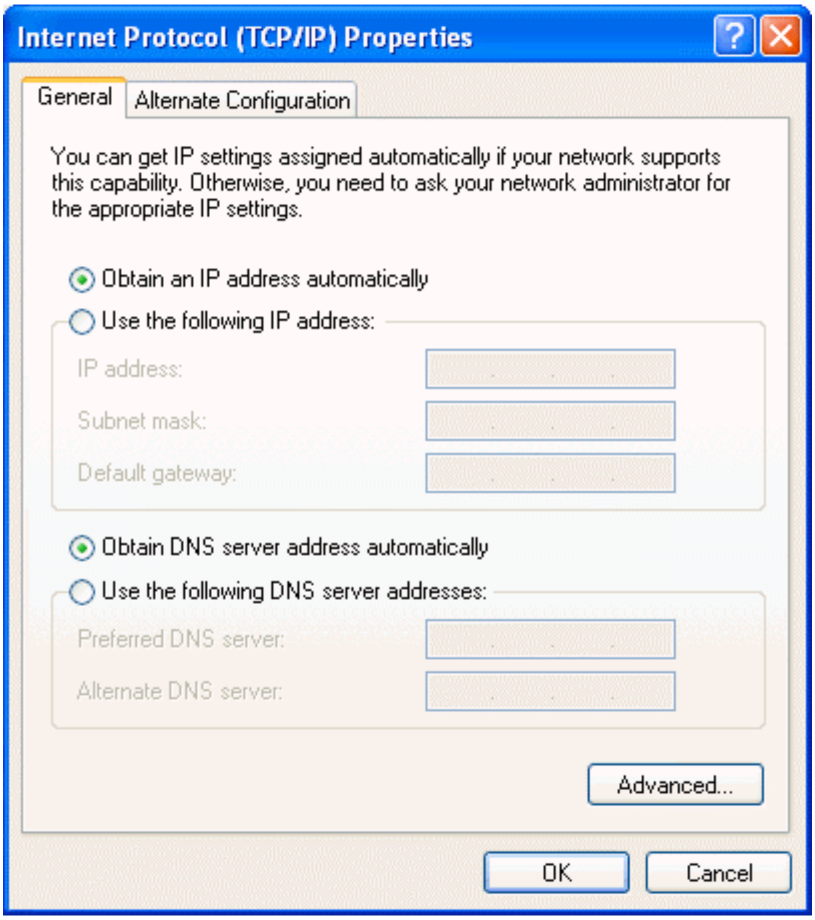
Step 4:

Click " **Internet Protocol [TCP/IP]** " and then click " **Properties** "



Step 5:

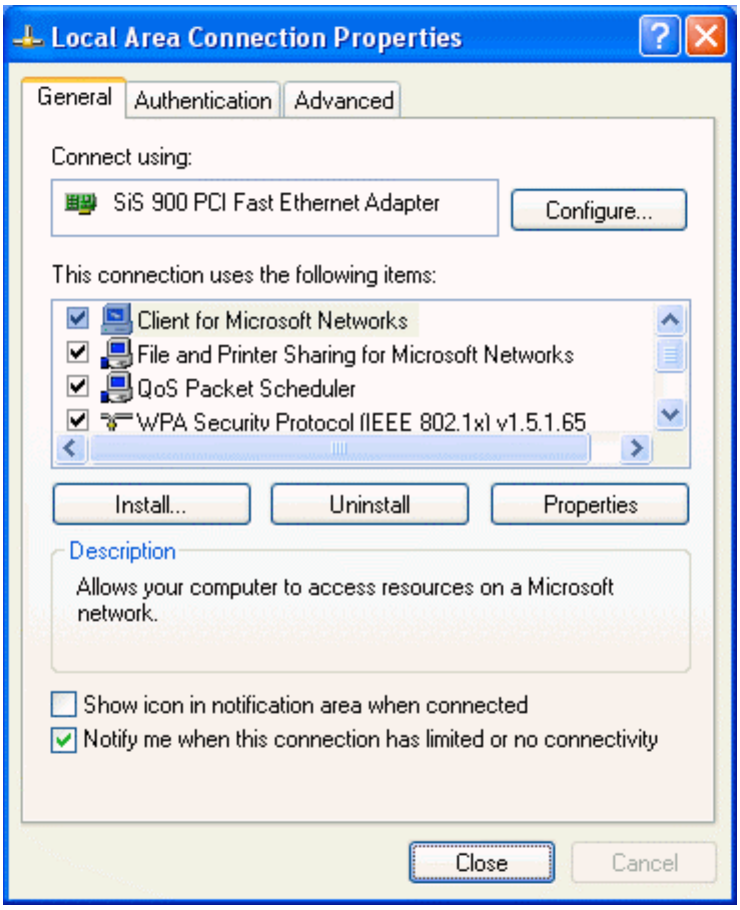
Select " **Obtain and IP Address automatically** " and then click " **OK** "



Step 6:

Click " **Close** "

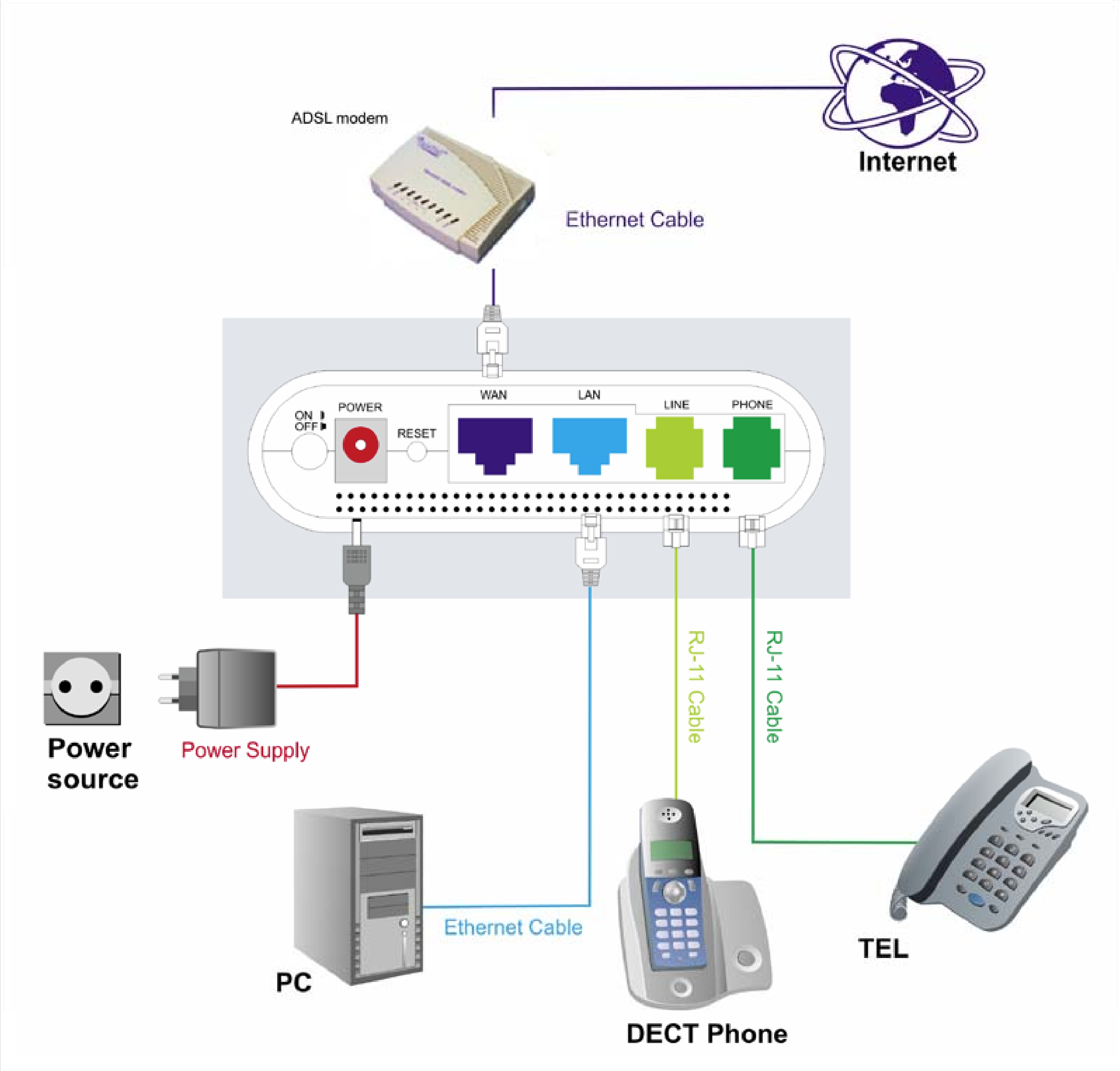
Now you've already configured the LAN to obtain an IP Address automatically (DHCP Client), just follow reset steps to finish the installation of VoIP Gateway.



3.3 Hardware Installation

1. Locate an optimum location for the VoIP Gateway.
2. For connections to all interfaces, refer to figure below.
3. Connect the AC Power Adapter. Depending upon the type of network, you may want to put the power supply on an uninterruptible supply. Only use the power adapter supplied with the VoIP Gateway. A different adapter may damage the product.

Now that the hardware installation is complete, proceed to reset Chapters to set up VoIP Gateway.



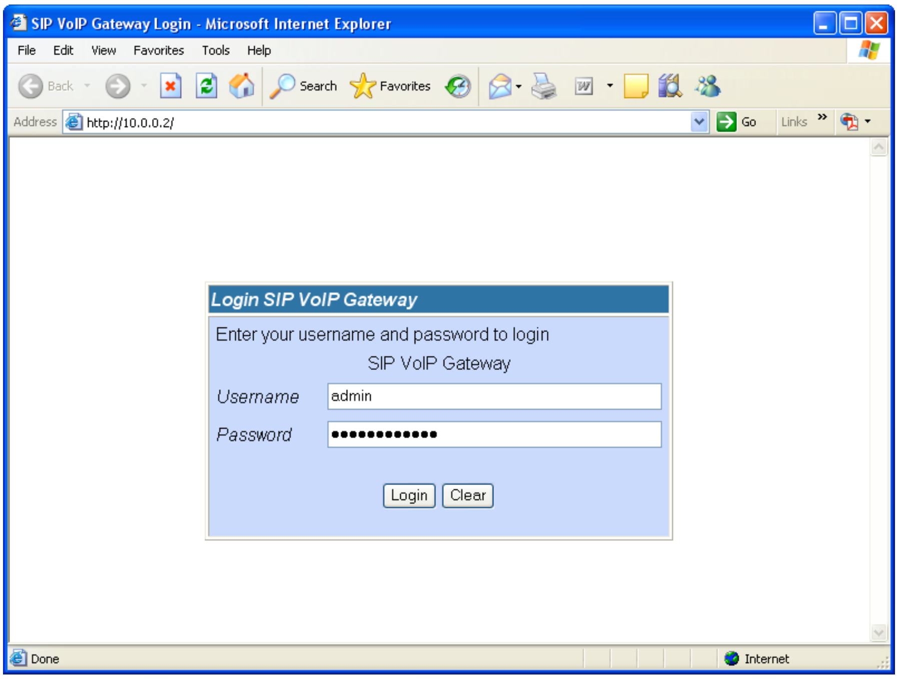
2 FXS

3.5 Basic VoIP Configuration

3.5.1 Access to the web configuration of VoIP Gateway

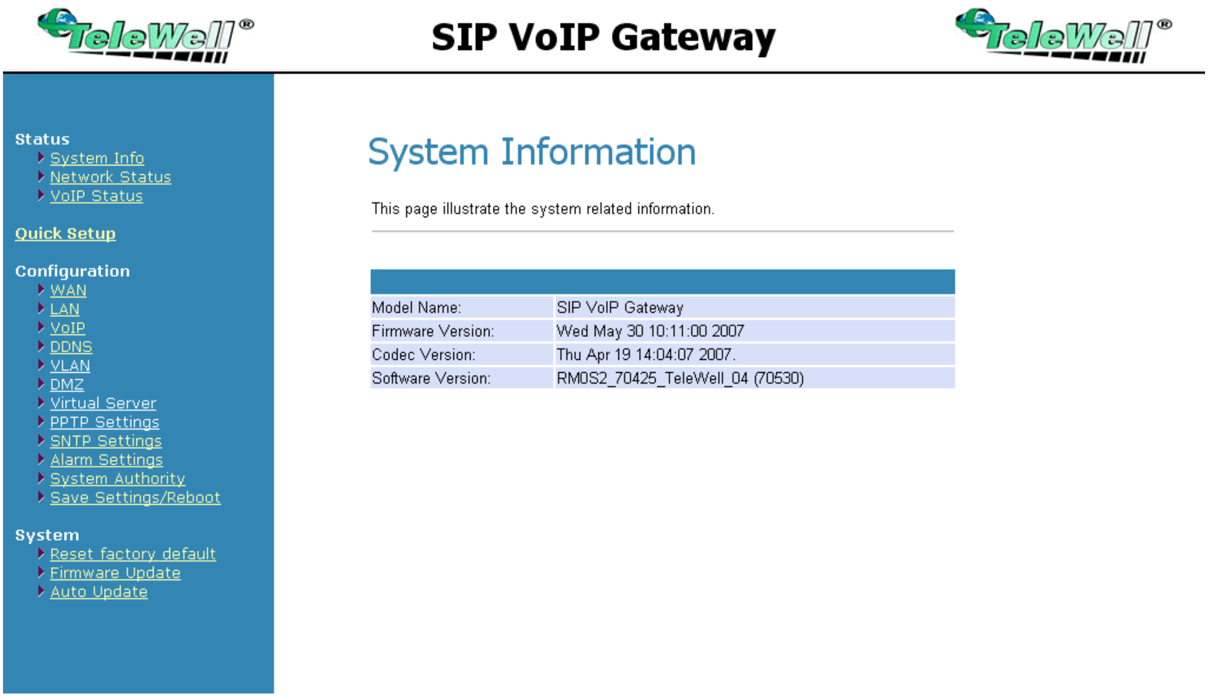
Step 1:
1. Launch the Web browser (Internet Explorer, Netscape, etc.).
2. Enter the LAN port default IP address (default gateway) http://10.0.0.2 in the address bar.
3. Entry of the username and password will be prompted. Enter the default login User Name and Password:
The default login User Name of the administrator is **admin**, and the default login Password is **admin**.

Remember my password checkbox: By default, this box is not checked. Users can check this box so that Internet Explorer will remember the User name and Password for future logins. It is recommended to leave this box unchecked for security purposes.



Step 2:

Now you could configure the VoIP Gateway in detail.



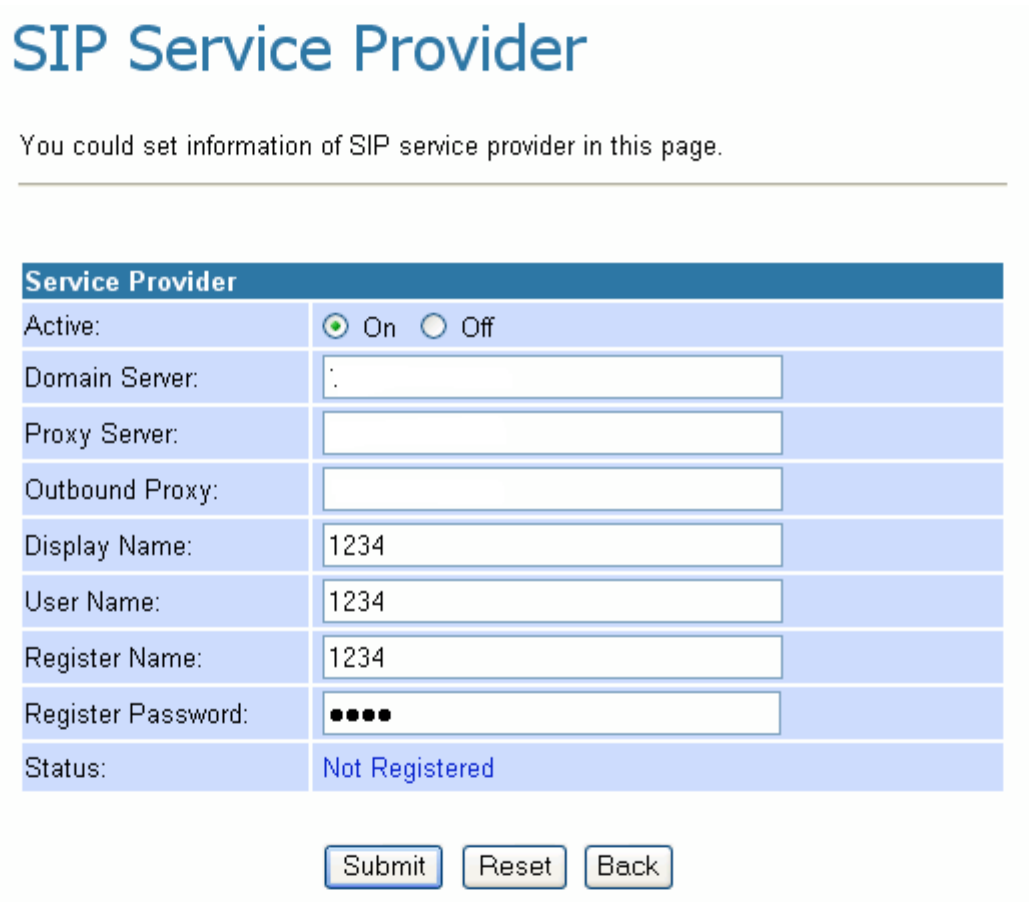
3.5.2 VoIP Configuration

Step 1:

Click " **Configuration -> VoIP -> SIP Service Provider** "

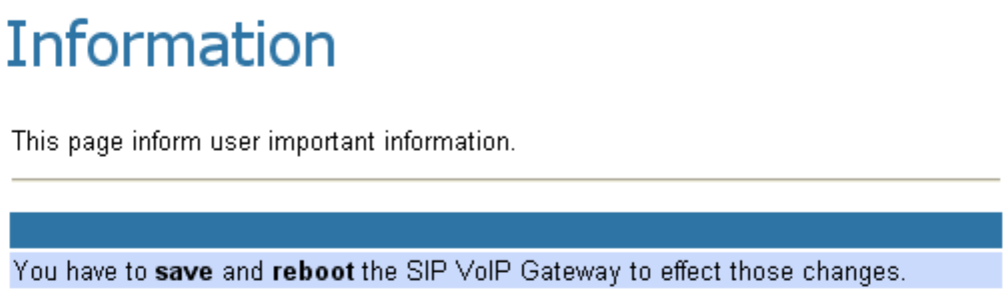
Step 2:

Click **On** ratio in Active, enter the information of **"Domain Server / Proxy Server / OutboundProxy / Display Name / User Name / Register Name / Register Password "** , which was provided by your VoIP Service Provider and then click **"Submit "**.



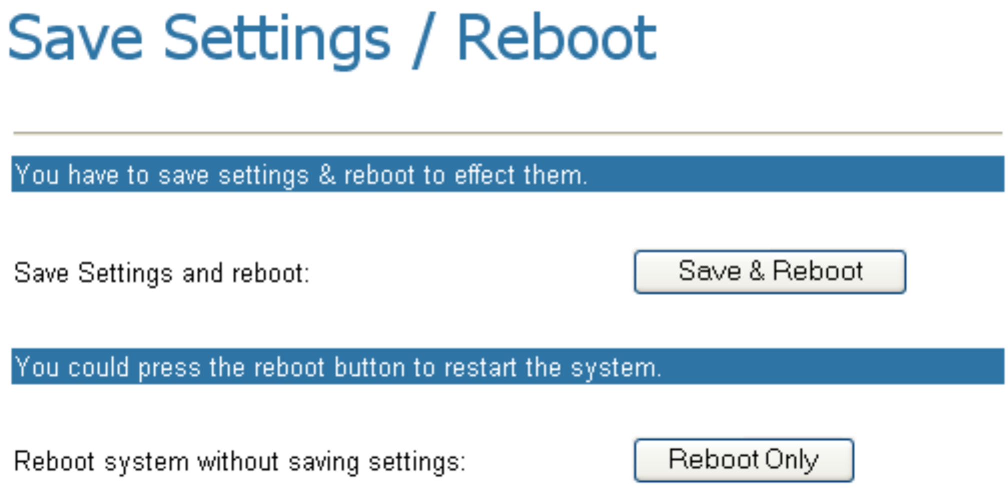
Step 3:

You have to **save** and **reboot** the SIP VoIP Gateway to effect those changes.



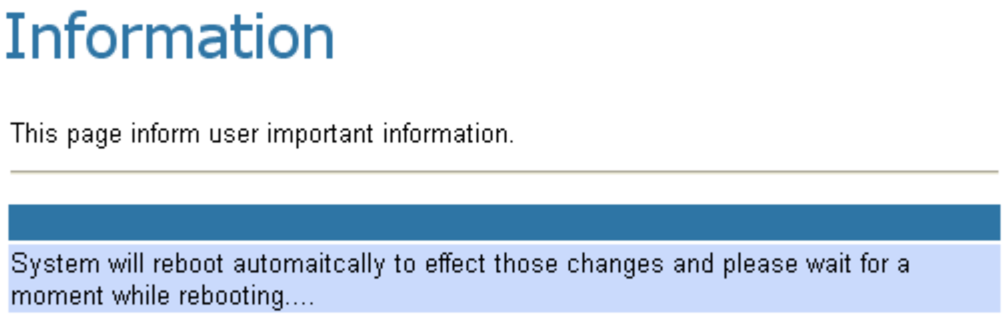
Step 4:

Click **" Configuration -> Save Settings/Reboot "** and then click **" Save & Reboot "** button.



Step 5:

System will reboot automatically to effect those changes and please wait for a moment while rebooting....



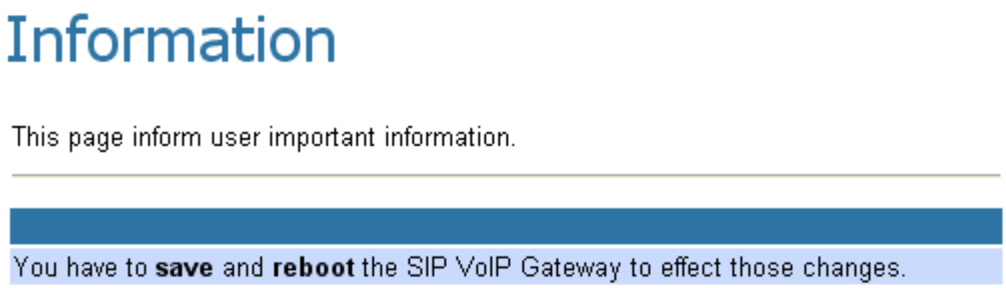
3.5.3 WAN Configuration

3.5.3.1 Static IP Configuration

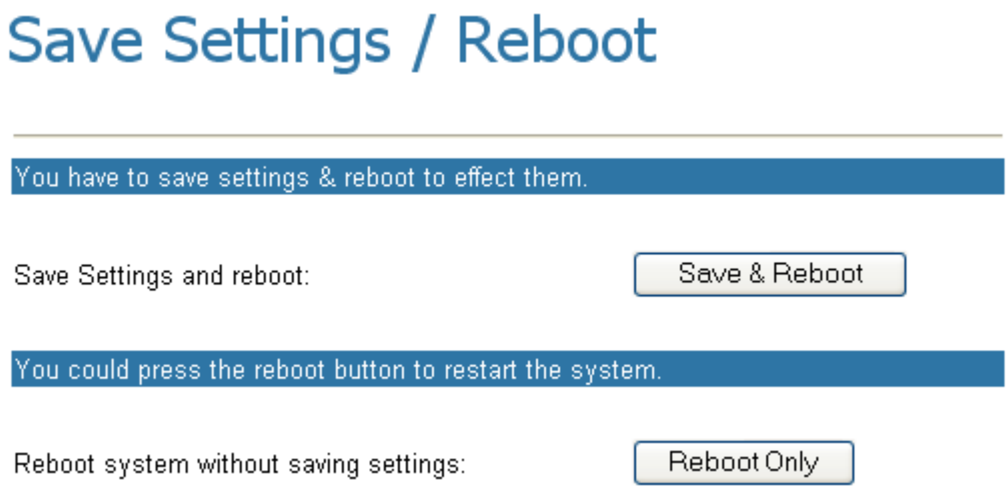
Step 1:

Click **" WAN -> Fixed IP "** and then enter the **" IP Address / Subnet Mask / Gateway / DNS Server1 / DNS Server2 "** and then click **" Submit "**

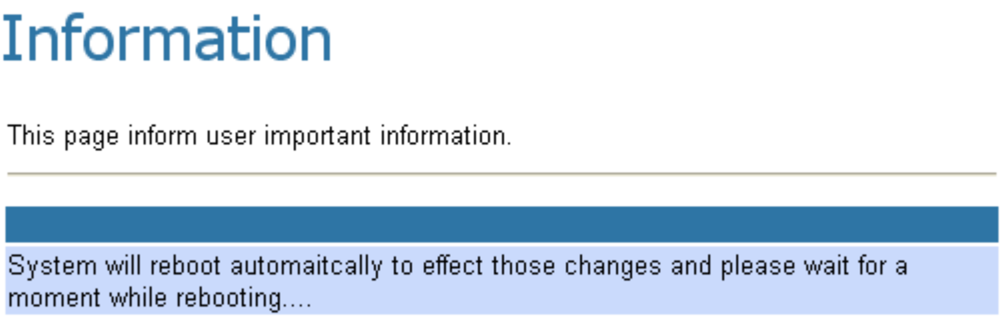
Step 2:
You have to **save** and **reboot** the SIP VoIP Gateway to effect those changes.



Step 3:
Click **" Configuration -> Save Settings/Reboot "** and then click **" Save & Reboot "** button.



Step 4:
System will reboot automatically to effect those changes and please wait for a moment while rebooting….
Please check the SIP LED is lighted or not. If the SIP LED is lighted, the VoIP Gateway is REGISTERED successfully to the SIP Server. If not, please press reset button and reconfigure configuration again.



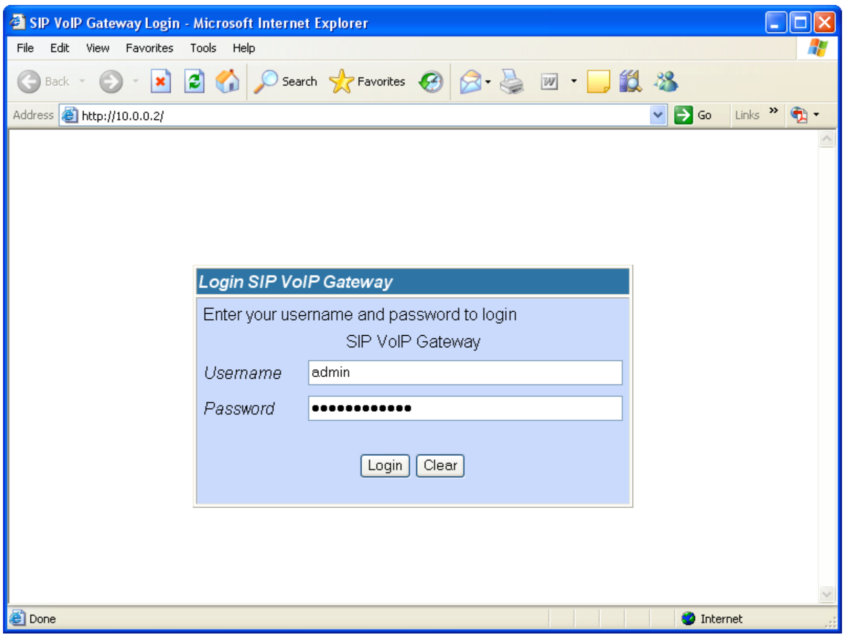
3.5.3.2 DHCP Client Mode Configuration

Step 1:

1. Launch the Web browser (Internet Explorer, Netscape, etc.).
2. Enter the LAN port default IP address (default gateway) http://10.0.0.2 in the address bar.
3. Entry of the username and password will be prompted. Enter the default login User Name and Password:

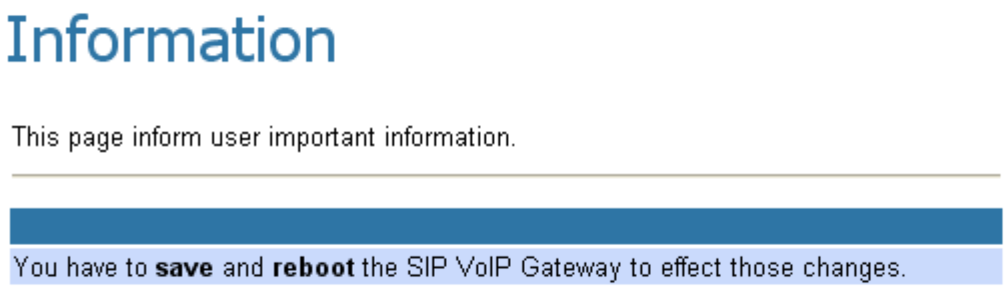
The default login User Name of the administrator is **admin**, and the default login Password is **admin**.

Remember my password checkbox: By default, this box is not checked. Users can check this box so that Internet Explorer will remember the User name and Password for future logins. It is recommended to leave this box unchecked for security purposes.

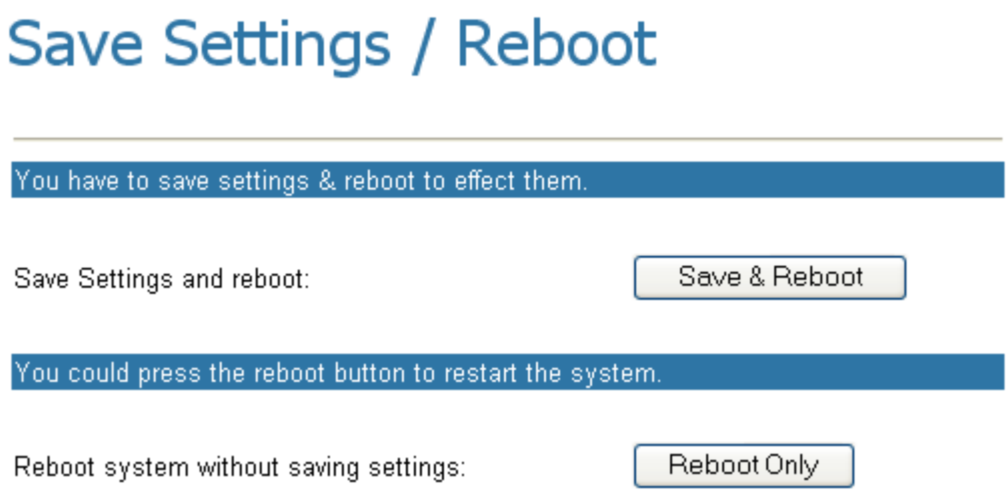


Step 2:
Click " **WAN -> DHCP client** " and then click " **Submit** "

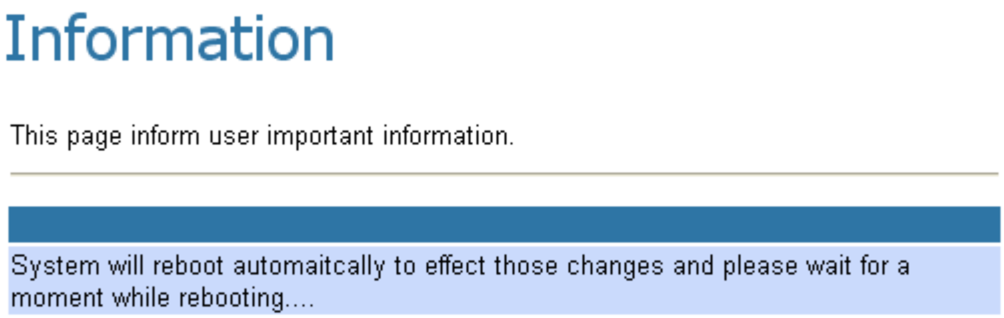
Step 3:
You have to **save** and **reboot** the SIP VoIP Gateway to effect those changes.



Step 4:
Click " **Configuration -> Save Settings/Reboot** " and then click " **Save & Reboot** " button.



Step 5:
System will reboot automatically to effect those changes and please wait for a moment while rebooting….
Please check the SIP LED is lighted or not. If the SIP LED is lighted, the VoIP Gateway is REGISTERED successfully to the SIP Server. If not, please press reset button and reconfigure configuration again.



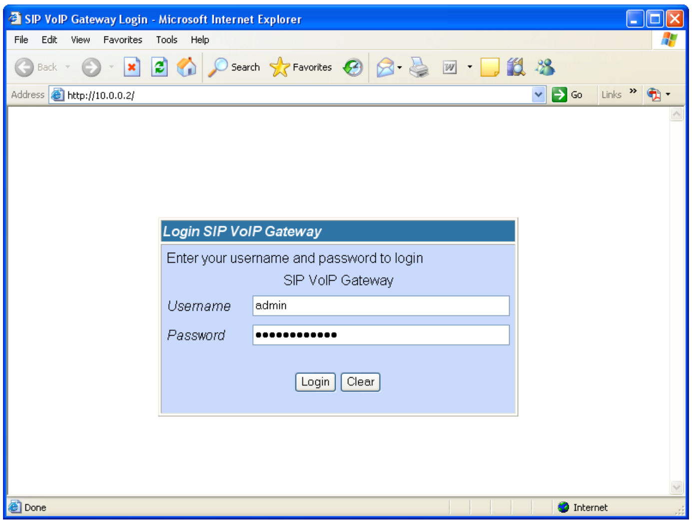
3.5.3.3 PPPoE Client Mode Configuration

Step 1:

1. Launch the Web browser (Internet Explorer, Netscape, etc.).
2. Enter the LAN port default IP address (default gateway) http://10.0.0.2 in the address bar.
3. Entry of the username and password will be prompted. Enter the default login User Name and Password:

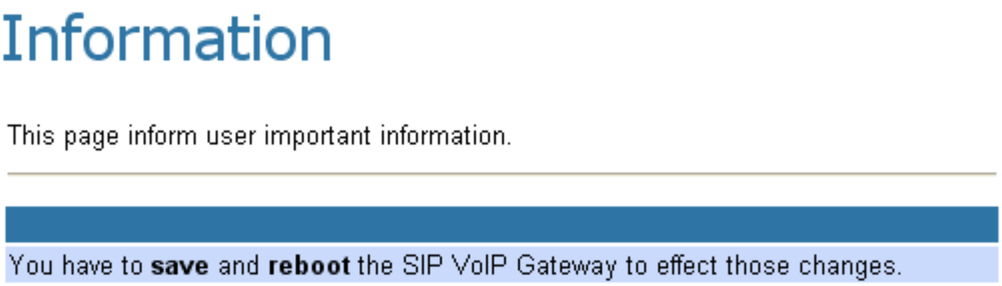
The default login User Name of the administrator is **admin**, and the default login Password is **admin**.

Remember my password checkbox: By default, this box is not checked. Users can check this box so that Internet Explorer will remember the User name and Password for future logins. It is recommended to leave this box unchecked for security purposes.

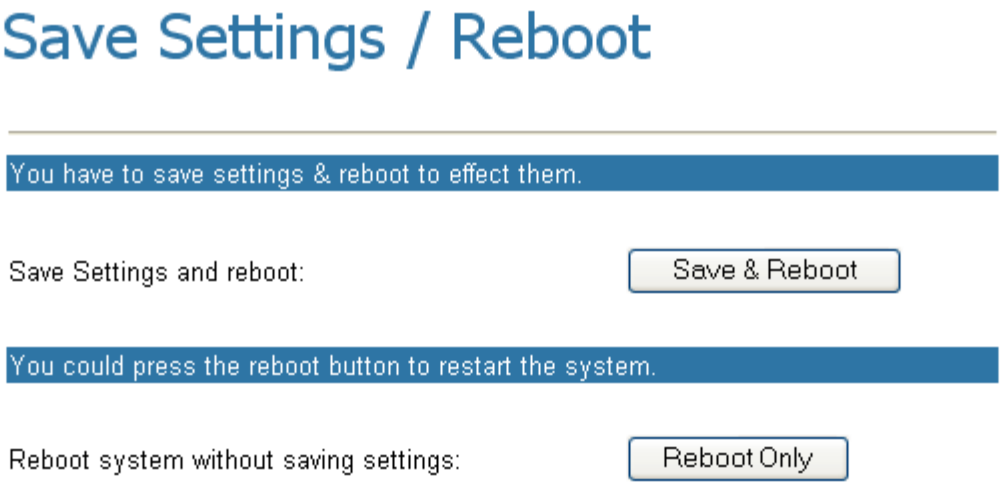


Step 2:
Click **" WAN -> PPPoE "**, enter the **" User Name and Password "** which was given by your Internet Service Provider (ISP) and then click **" Submit "**

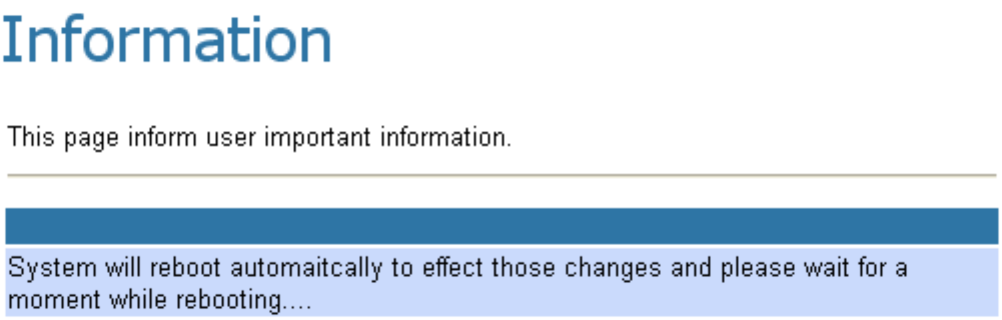
Step 3:
You have to **save** and **reboot** the SIP VoIP Gateway to effect those changes.



Step 4:
Click **" Configuration -> Save Settings/Reboot "** and then click **" Save & Reboot "** button.



Step 5:
System will reboot automatically to effect those changes and please wait for a moment while rebooting....
Please check the SIP LED is lighted or not. If the SIP LED is lighted, the VoIP Gateway is REGISTERED successfully to the SIP Server. If not, please press reset button and reconfigure configuration again.



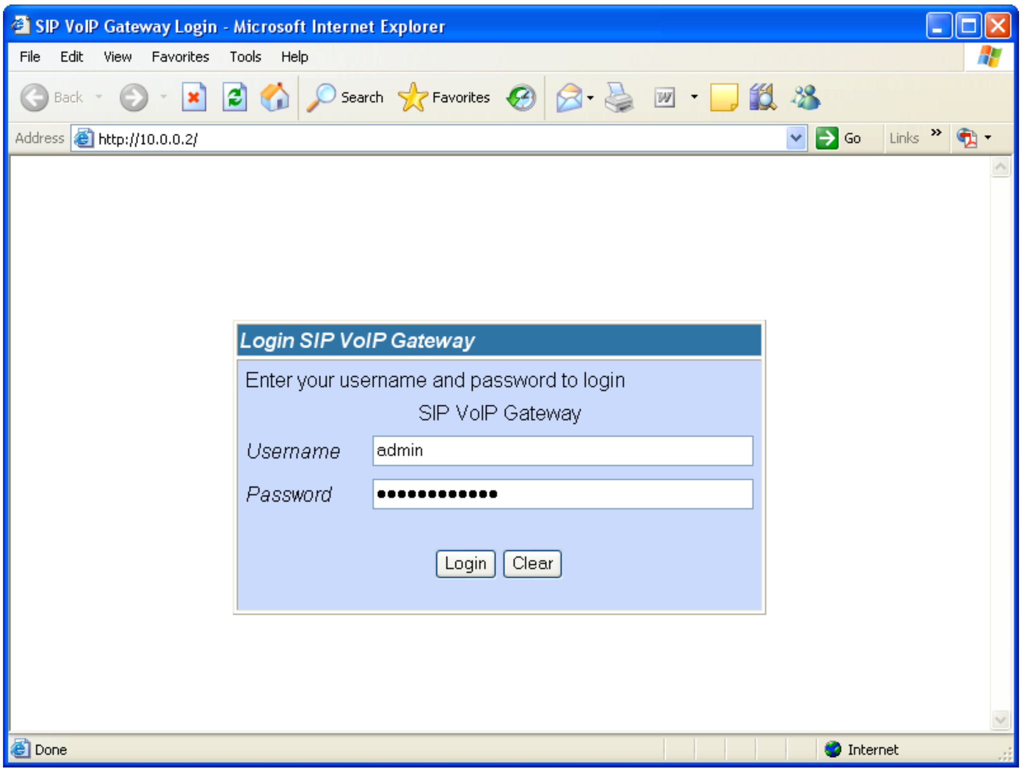
4. Advanced VoIP Configuration

The VoIP Gateway is configured using the web interface. The **VoIP Gateway Configuration** page can be reached as follows:

1. Launch the Web browser (Internet Explorer, Netscape, etc.).
2. Enter the LAN port default IP address (default gateway) http://10.0.0.2/ in the address bar.
3. Entry of the username and password will be prompted. Enter the default login User Name and Password:

The default login User Name of the administrator is **admin**, and the default login Password is **admin**.

Remember my password checkbox: By default, this box is not checked. Users can check this box so that Internet Explorer will remember the User name and Password for future logins. It is recommended to leave this box unchecked for security purposes.



4. On the router **Home Page**, click the **VoIP** link on the left frame to view the **VoIP Gateway Configuration** page.

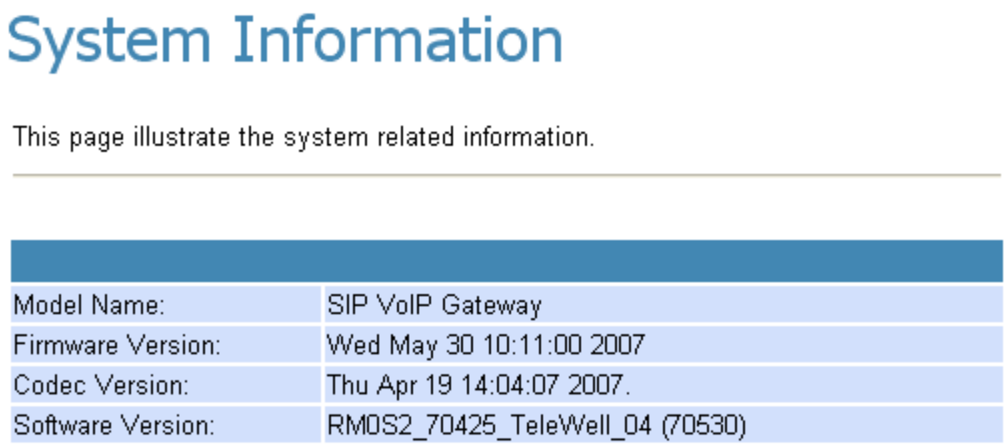
In general, configuration changes made using the web interface will be activated only upon clicking **Save & Reboot** button on the **Save Savings / Reboot** page.

Note: Certain Voice Parameters do not require a **Save & Reboot** to take effect. These Voice Parameters will take effect on the next voice call after the Voice Parameter is entered and submitted. If **Save & Reboot** is not done, then these Voice Parameters will not be saved over a power cycle. The Voice Parameters that can be changed **"on the fly"** are noted in the respective sections.

4.1 Status Page

4.1.1 System Information Page

This page illustrate the system related information



4.1.2 Network Status Page

You can check the current Network setting in this page.

Network Status

This page shows current status of network interfaces of the system.

WAN Status	
Type:	DHCP Client
IP:	192.168
Mask:	255.255.255.0
Gateway:	0.0.0.0
DNS Server 1:	16€
DNS Server 2:	168.

LAN Status	
Type:	DHCP Server
IP:	10.0.0.2
Mask:	255.255.255.0
Gateway:	10.0.0.2
DNS Server 1:	16€
DNS Server 2:	16€

4.1.3 VoIP Status Page

The page shows current status of VoIP SIP Service provider.

VoIP Status

The page shows current status of VoIP SIP Service provider.

VoIP SIP Provider Status	
Domain Server:	192.166.
Display Name:	1234
User Name:	1234
Status:	Registered

DNS Server 2:	168.

LAN Status	
Type:	DHCP Server
IP:	10.0.0.2
Mask:	255.255.255.0
Gateway:	10.0.0.2
DNS Server 1:	16€
DNS Server 2:	16€

4.2 Configuration Page

4.2.1 WAN Configuration Page

You could configure the WAN settings in this page.

4.2.1.1 The **TCP/IP Configuration item** is to setup the LAN port's network environment. You may refer to your current network environment to configure the VoIP Gateway properly.

4.2.1.2 The **PPPoE Configuration item** is to setup the PPPoE Username and Password. If you have the PPPoE account from your Service Provider, please input the Username and the Password correctly.

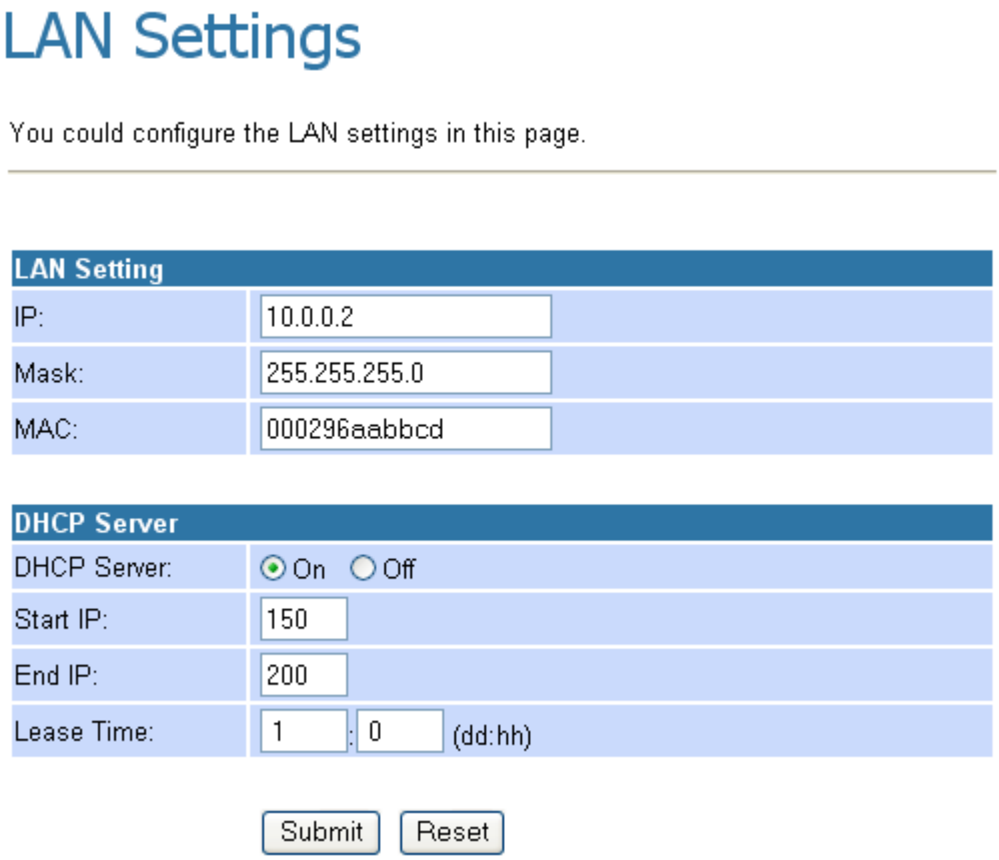
4.2.1.3 The **Bridge Item** is to setup the VoIP Gateway Bridge mode

Enable/Disable. If you set the Bridge On, then the two Fast Ethernet ports will be transparent.

4.2.1.4 When you finished the setting, please click the **Submit** button.

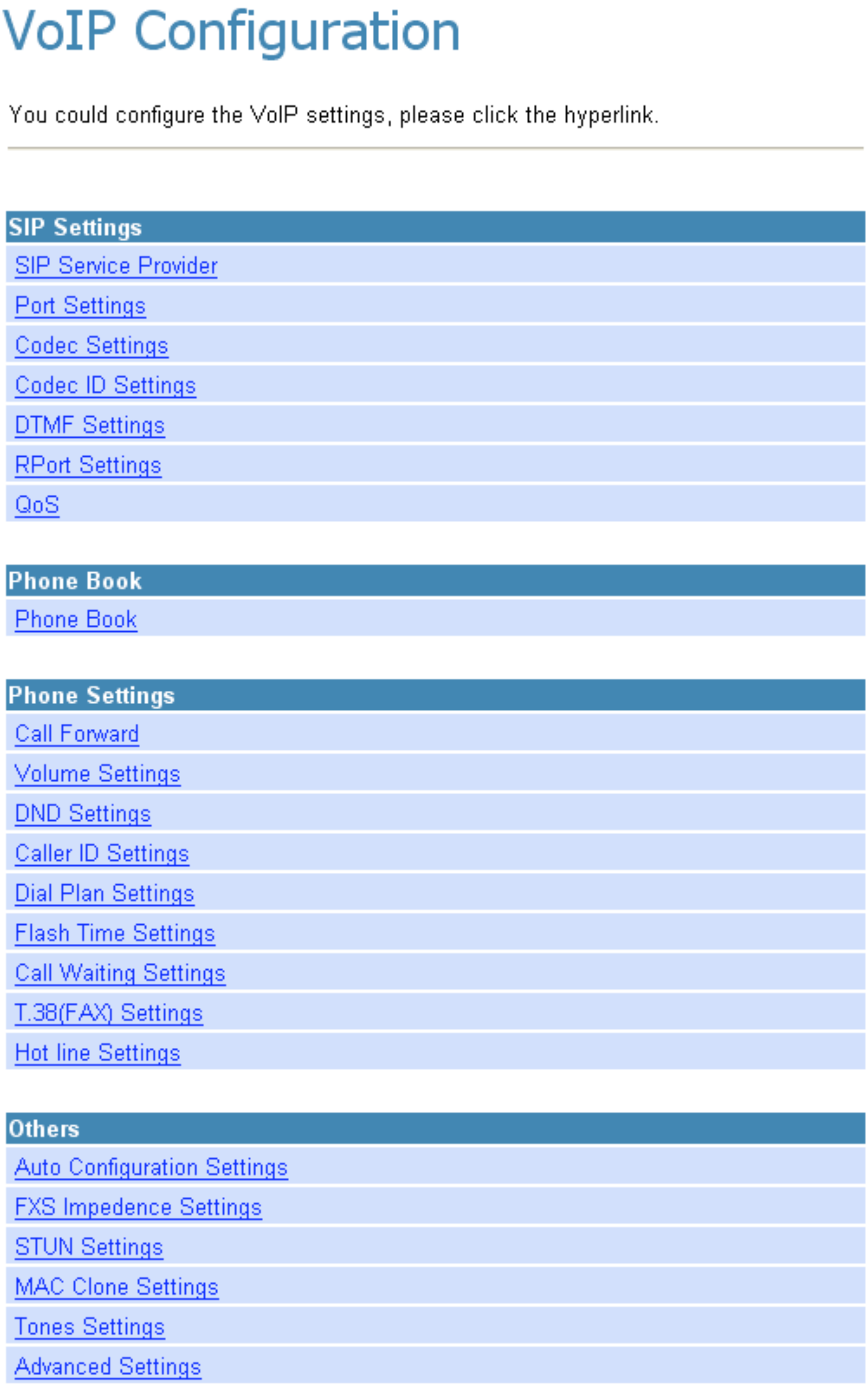
4.2.2 LAN Configuration Page

You could configure the LAN settings/DHCP Server in this page.



4.2.3 VoIP Gateway Configuration Page

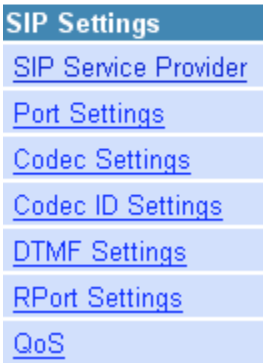
The **VoIP Gateway Configuration** page sets parameters for the VoIP application.



The **VoIP Gateway Configuration** page is divided into three general categories: **SIP Setting**, **Phone Book**, **Phone Setting**, and **Others**.

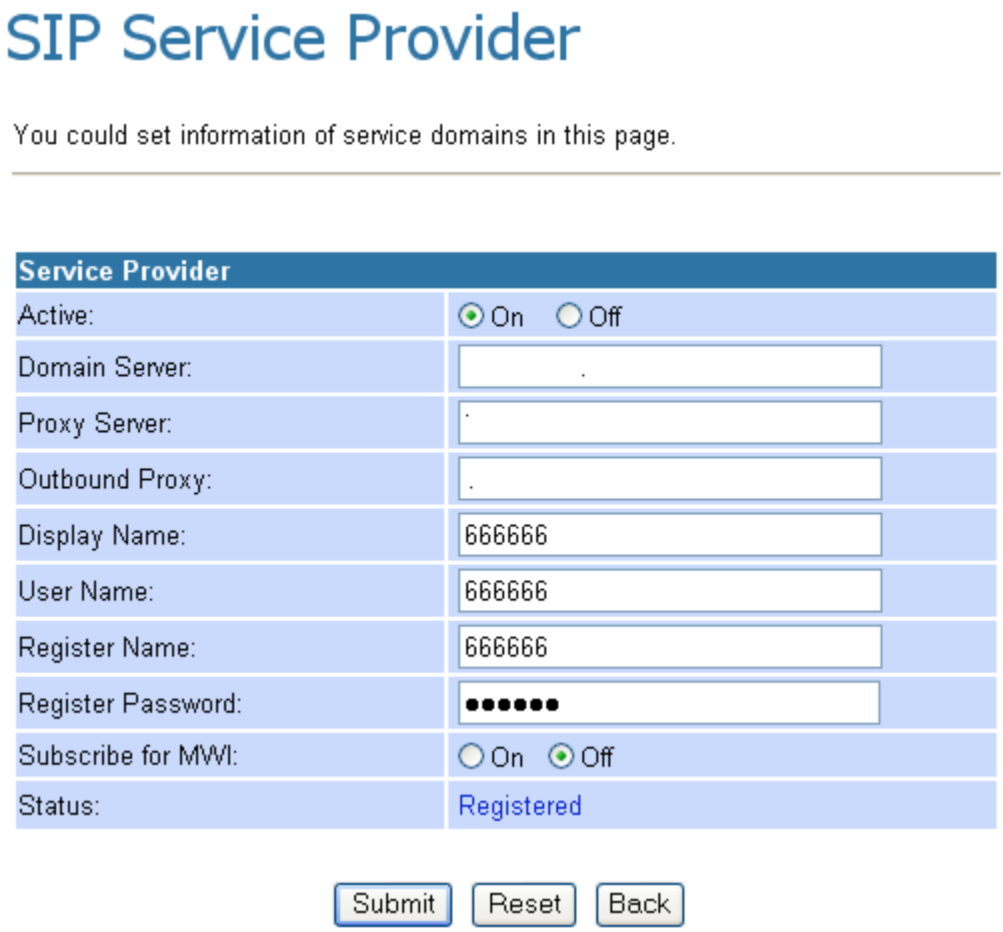
4.2.3.1 SIP Setting Configuration

In SIP Settings you can setup the Service Domain, Port Settngs, Codec Settings, RTP Setting, RPort Setting and Other Settings. If the VoIP service is provided by ISP, you need to setup the related information correctly then you can register to the SIP Proxy Server correctly.



4.2.3.1.1 SIP Service Provider

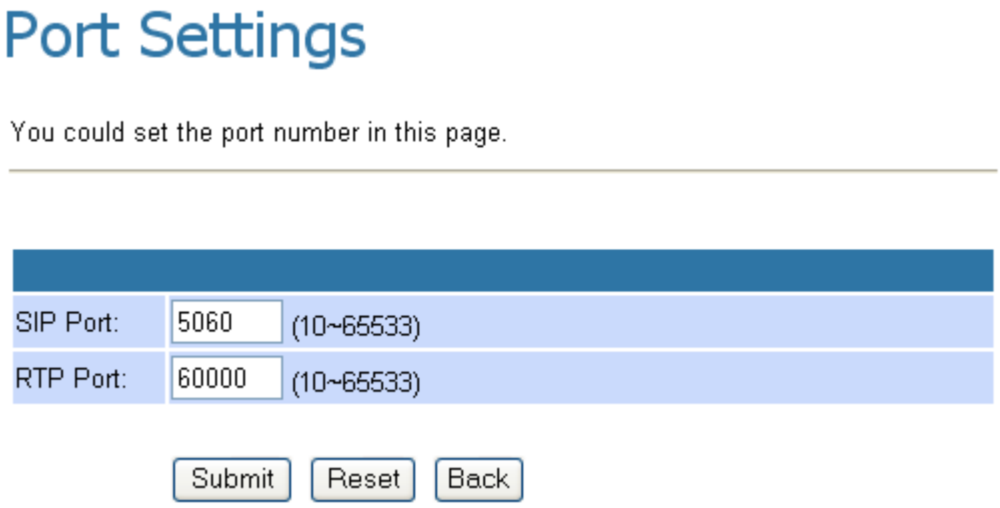
In Service Domain Function you need to input the account and the related information in this page please refer to your ISP provider. You can register three SIP account in the VoIP Gateway. You can dial the VoIP phone to your friends via first enable SIP account and receive the phone from these three SIP accounts.



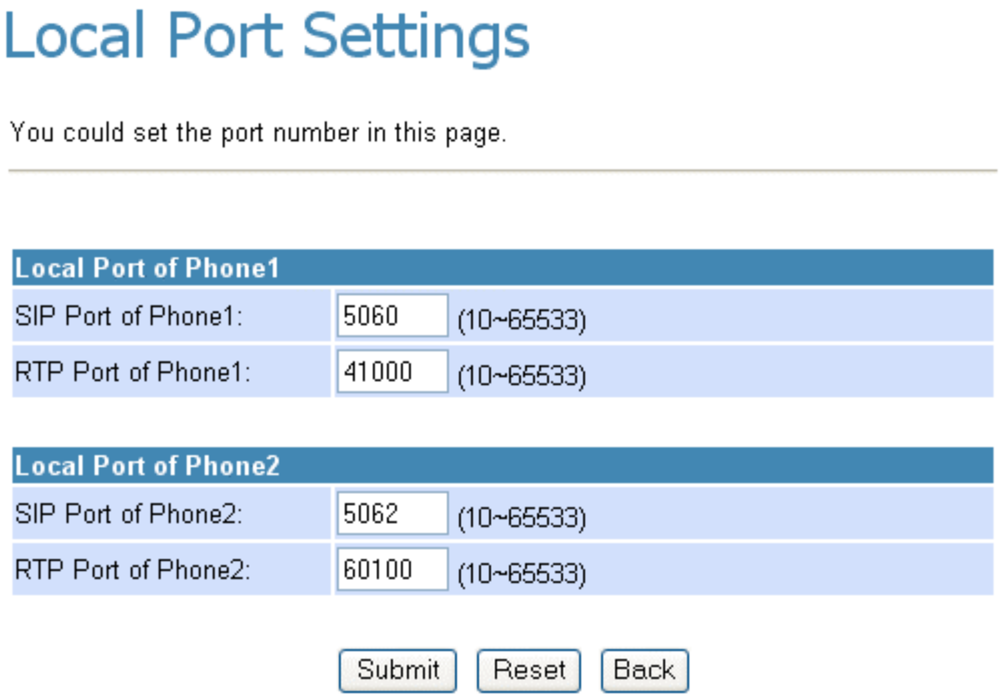
SIP Service Provider	
Active	First you need click **On** to enable the Service Domain, then you can input the following items:
Domain Server	For example, in test@domain.com, the domain is “domain.com”. Provided by your VoIP Service Provider.
Proxy Server	If your VoIP service provider has an proxy address and requires that you provide the address to VoIP Gateway. For the address enter a domain name (for example, domain.com) or an IP address (for example, 123.456.789.012).
Outbound Proxy	If your VoIP service provider has an outbound proxy address and requires that you provide the address to VoIP Gateway. For the address enter a domain name (for example, domain.com) or an IP address (for example, 123.456.789.012).
Display Name	This name is displayed in the VoIP Gateway display. Other parties will see this name they are when connected to you.
User Name	Typically the account number for the SIP account. For example, in test@domain.com, the user name is “test”. Provided by your VoIP Service Provider.
Register Name	May not be required. If it is required, it will be provided by your VoIP Service Provider.
Register Password	Provided by the VoIP Service Provider.
Subscribe for MWI	When set to **On** a Subscribe for **Message Waiting Indication** will be sent periodically.
Register Status	You can see the **Register Status** in the **Status** item. If the item shows **“Registered”**, then your VoIP Gateway is registered to the ISP, you can make a phone call directly.
Submit Button	When you finished the setting, please click the **Submit** button.
Reset Button	You can reset the configured parameters before you submit
Back Button	Go back to the previous web page

4.2.3.1.2 Port Setting

You can setup the **SIP** and **RTP** port number in this page. Each ISP provider will have different SIP/RTP port setting, please refer to the ISP to setup the port number correctly. When you finished the setting, please click the **Submit** button.



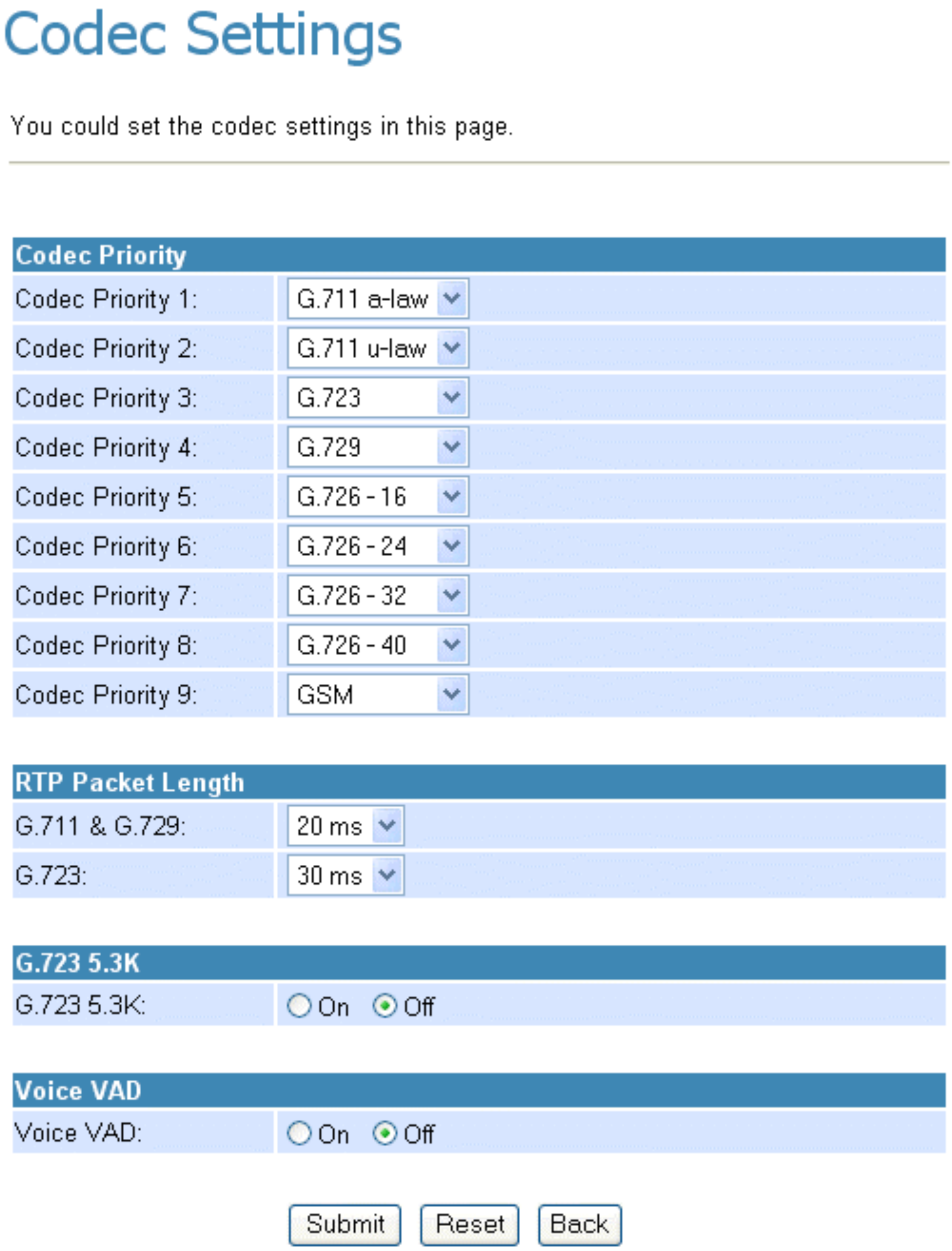
For 1 FXS Port



For 2 FXS Port

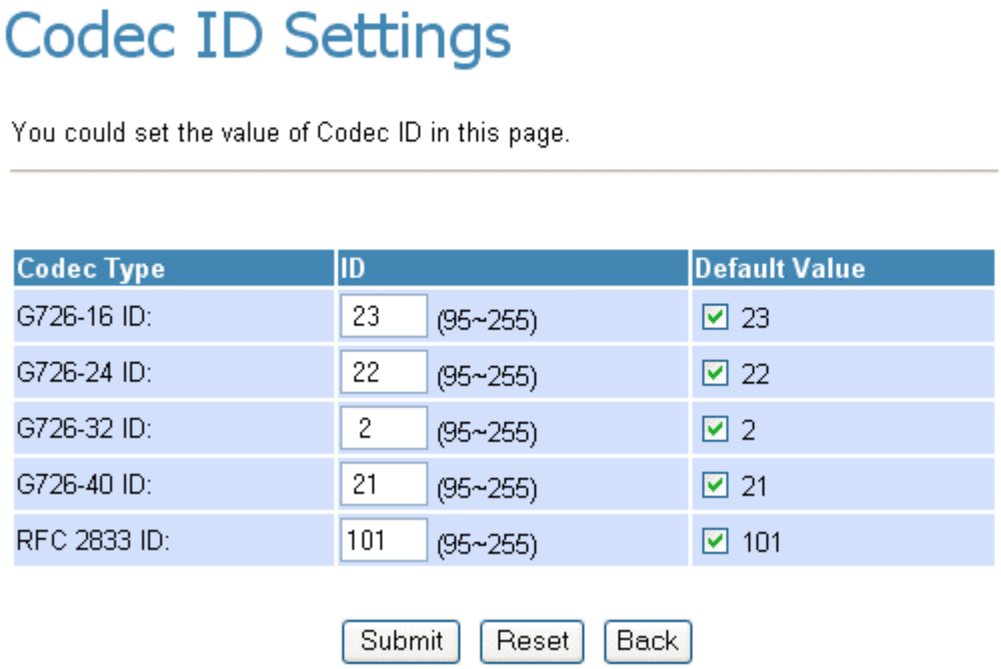
4.2.3.1.3 Codec Settings

You can setup the **Codec priority**, **RTP packet length**, and **VAD(Voice Activity Detection)** function in this page.
You need to follow the ISP suggestion to setup these items. When you finished the setting, please click the **Submit** button.



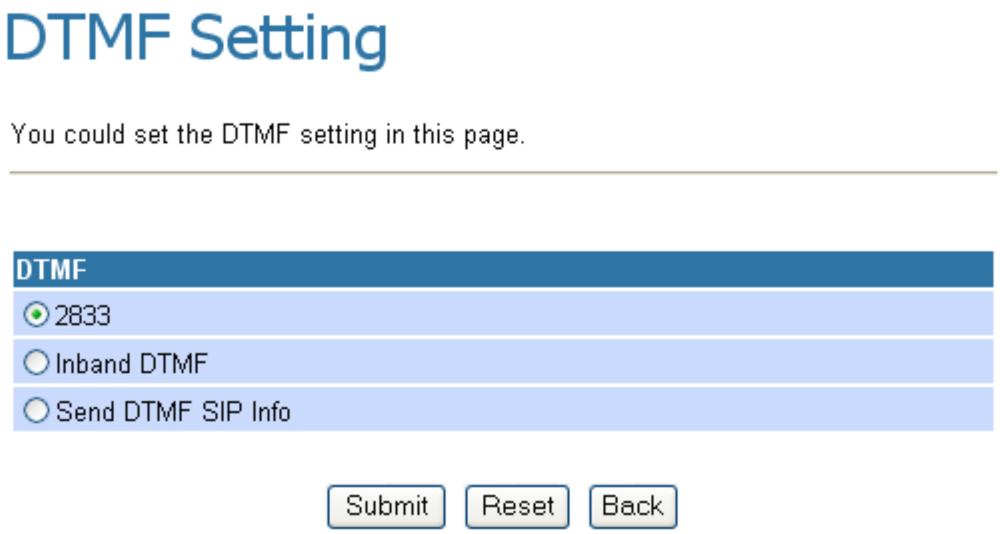
4.2.3.1.4 Codec ID Setting

You could set the value of Codec ID in this page.



4.2.3.1.5 DTMF Setting

You can setup the Out-Band DTMF and Send DTMF SIP Info Enable/Disable in this page. To change this setting, please follow your VoIP Service Provider's information. When you finished the setting, please click the **Submit** button.



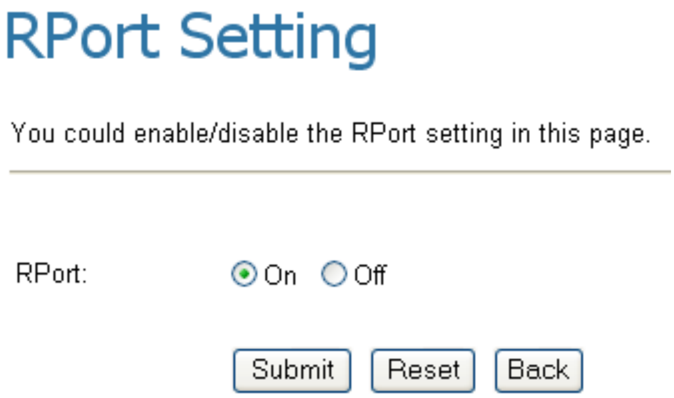
• **RFC 2833:** Click this button to send Mid-Call DTMF tones in RTP packets ?separately
using RFC2833, i.e., dynamic negotiation of RTP payload for DTMF digits will be done.

• **Inband DTMF (IN AUDIO):** Click this button to send Mid-Call DTMF tones in RTP ?packets with the same payload as voice, i.e., dynamic payload

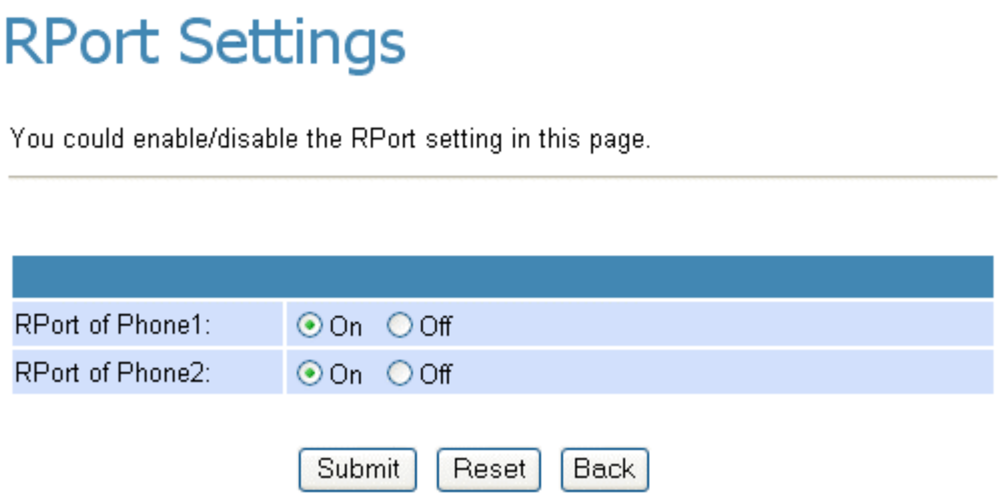
negotiation for DTMF digits will not be done.
• **Send DTMF SIP Info:** This field is configurable when RFC 2833 is ?selected as the
DTMF Relay mechanism. Specify the payload number that needs to be used for DTMF
information negotiated in SDP during SIP signaling.

4.2.3.1.6 RPort Function

You can setup the RPort Enable/Disable in this page. To change this setting, please
follow your VoIP Service Provider's information. When you finished the setting, please click the **Submit** button.



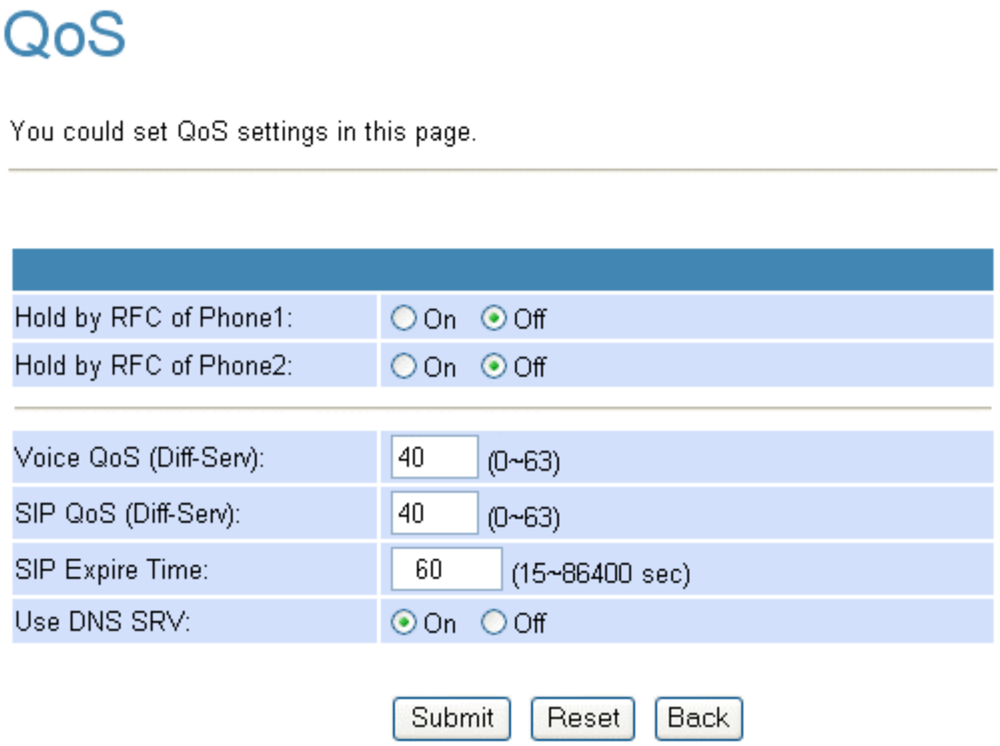
For 1 FXS Port



For 2 FXS Port

4.2.3.1.7 QoS

You can setup the **Hold by RFC**, **Voice/SIP QoS**, **SIP expire time** and **Use DNS SRV** in this page. To change these settings please following your ISP information. When you finished the setting, please click the **Submit** button. The QoS setting is to set the voice packets' priority. If you set the value higher than 0, then the voice packets will get the higher priority to the Internet. But the QoS function still needs to cooperate with the others Internet devices.



4.2.3.2 Phone Book Configuration

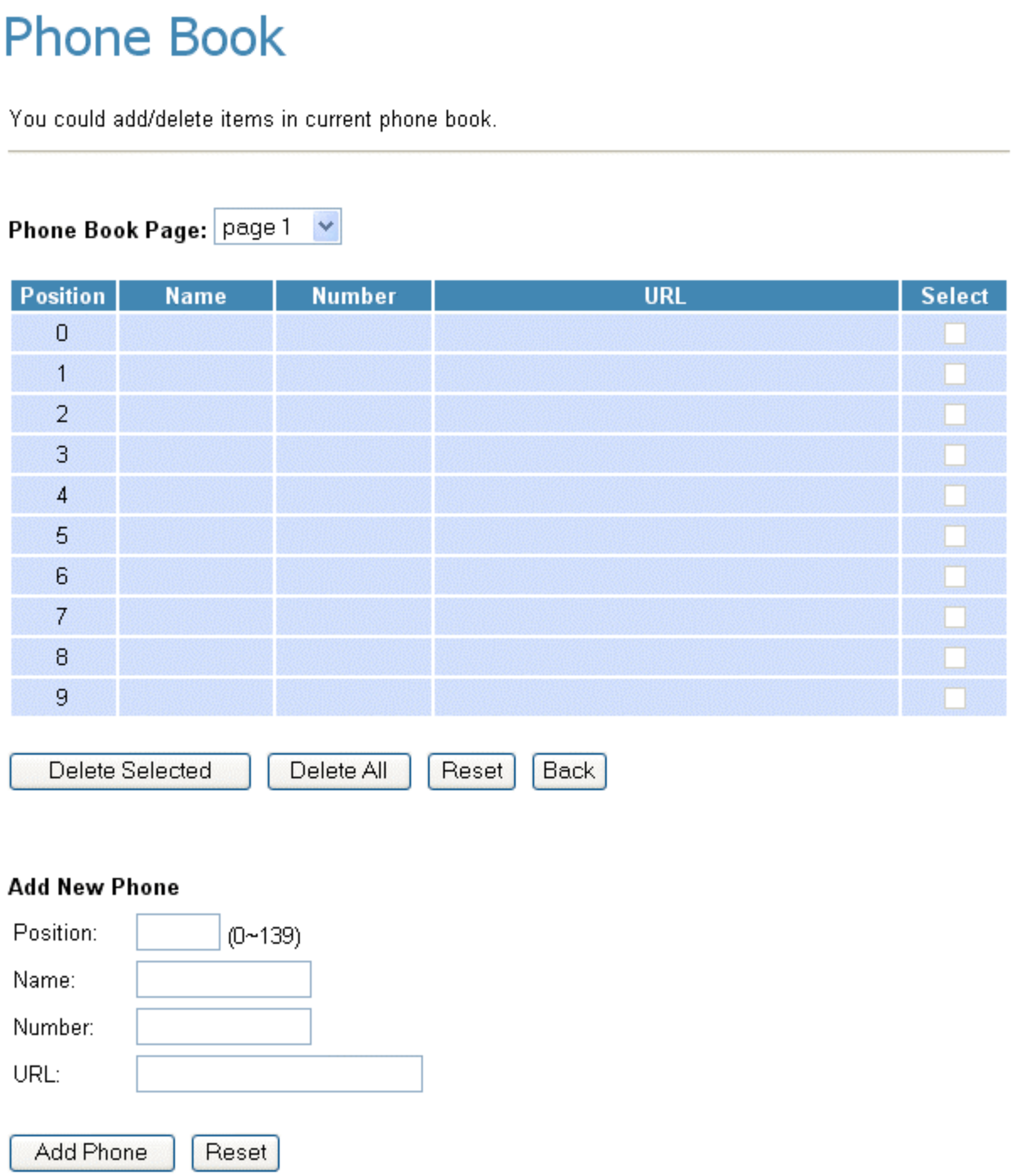
4.2.3.2.1 In Phone Book contains **Speed Dial Settings**. You can setup the Speed Dial number. If you want to use Speed Dial you just dial the speed dial number then press **"#"**.

4.2.3.2.2 In Phone Book setting function you can **add/delete** Speed Dial number. You can input maximum **140** entries speed dial list.

4.2.3.2.2.1 If you need to add a phone number into the Speed Dial list, you need to input the position, the name (Speed Dial Number), and the phone number (by URL type). When you finished a new phone list, just click the **"Add Phone"** button.

4.2.3.2.2.2 If you want to delete a phone number, you can select the phone number you want to delete then click **"Delete Selected"** button.

4.2.3.2.2.3 If you want to delete all phone numbers, you can click **"Delete All"** button.



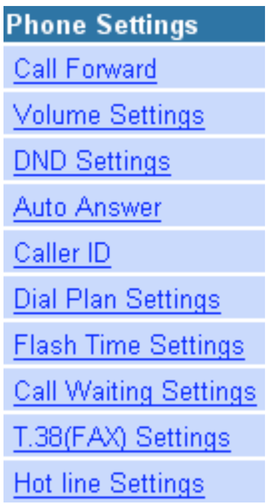
Phone Book Page	
Book Page	Default page is Page1. There are total 14 pages from Page 1 to Page 14
Phone	Show the phone number by sequence. There are total 140 phone numbers from Phone 0 to Phone 139 can be set
Name	Enter the Name
Number	Enter the Speed Dial Number
URL	Display the URL that you configured
Select	Select the item of the phone number

Delete Selected [Button]	Delete selected item
Delete All [Button]	Delete all items
Reset [Button]	Reset selected item

Add New Phone	
Position	Enter the phone number from 0 to 139
Name	Enter the Name
Number	Enter the Speed Dial Number
URL	Enter the URL, VoIP Phone Number, Remote WAN IP Address of VoIP Gateway
Add Phone [Button]	Add the new Phone which you configured
Reset [Button]	Reset configured items

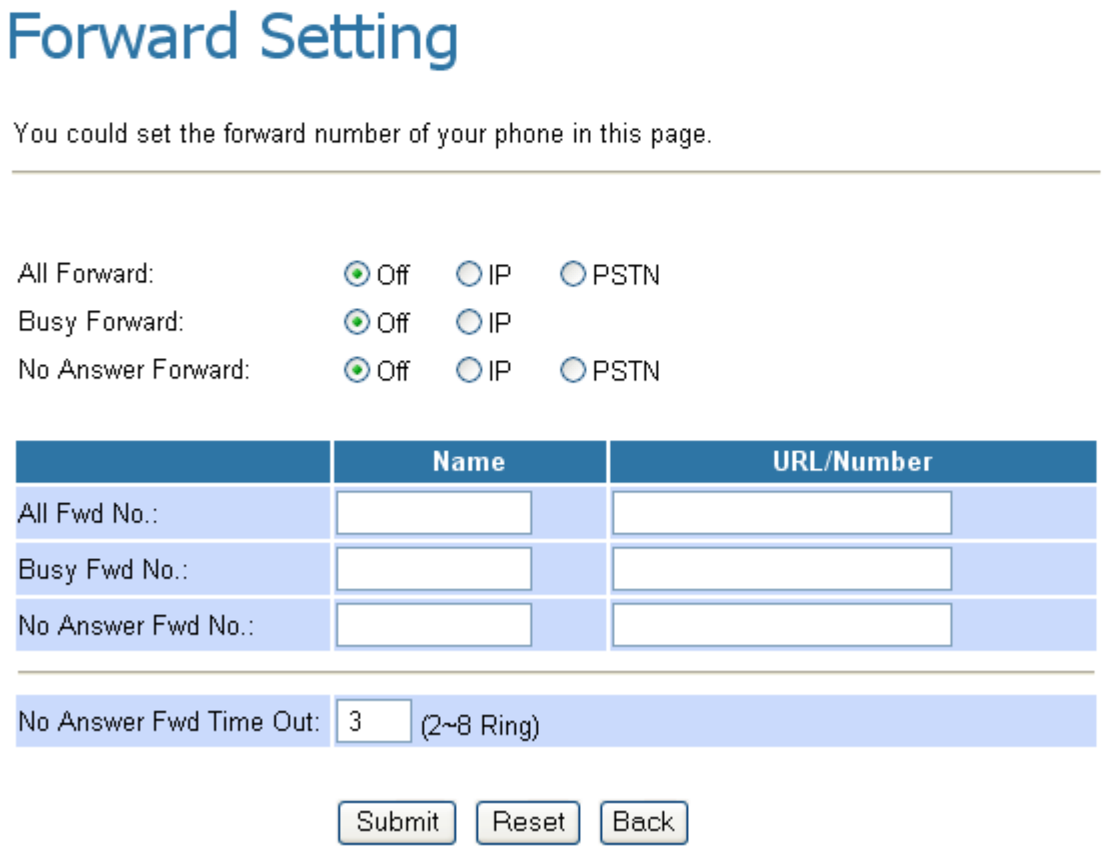
4.2.3.3 Phone Setting

In Phone Setting contains **Call Forward**, **Volume Settings**, **DND Settings**, **Auto Answer**, **Caller ID**, **Dial Plan Settings**, **Flash Time Settings**, **Call Waiting Settings**, **T.38(FAX) Settings** and **Hot line Settings** functions.



4.2.3.3.1 Call Forward function

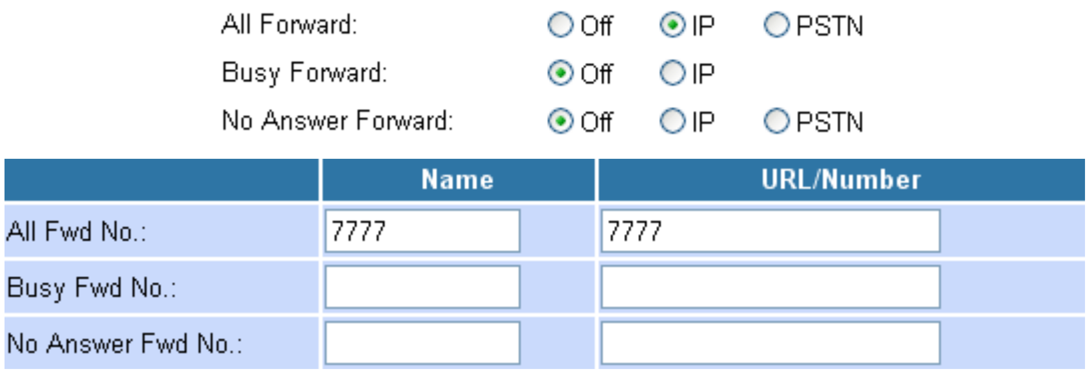
You can setup the phone number you want to forward in this page. There are three type of Forward mode. You can choose **All Forward**, **Busy Forward**, and **No Answer Forward** by click the icon.



All Forward	All incoming call will forward to the URL/number you configured.
Busy Forward	If you are on the phone, the new incoming call will forward to the URL/number you configured.
No Answer Forward	If you can not answer the phone after a specific ring you configured, the incoming call will forward to the URL/number you configured.
Off	Disable call forward.
IP	Enable call forward for URL/number.
PSTN (Optional)	Enable call forward for PSTN phone number. **Only the for 1 FXO +1 FXS**
All Fwd No.	The URL/number you configured will be forward to for **All Forward**
Busy Fwd No.	The URL/number you configured will be forward to for **Busy Forward**
No Answer	The URL/number you configured will be forward to for **No**

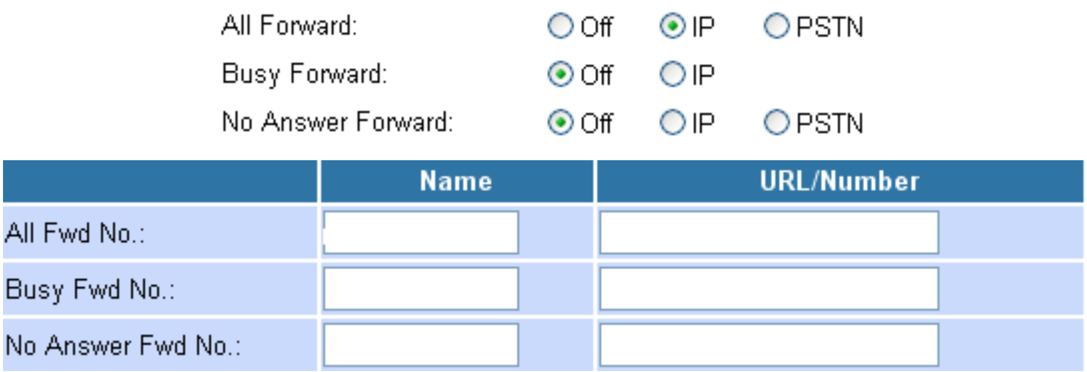
Fwd No.	Answer Forward
Name	Display the name of URL/number that you configured
URL	Enter the URL, VoIP Phone Number, Remote WAN IP Address of VoIP Gateway which you want forward to.
No Answer Fwd Time Out	You can set the Time Out time for system to start to forward the call to the number you configured for **No Answer Forward**
Submit Button	When you finished the setting, please click the **Submit** button.
Reset Button	You can reset the configured parameters before you submit
Back Button	Go back to the previous web page

Example 1: All Forward: **IP**, Name.: **7777**, URL/Number: **7777**



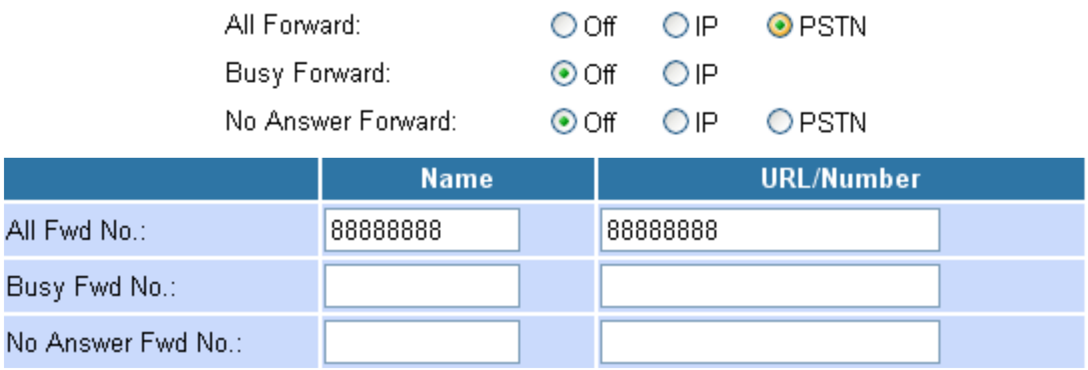
All incoming call will forward to the VoIP phone number **7777**.

Example 2: All Forward: **IP**, Name: **192.168.10.36**, URL/Number: **192.168.10.36**



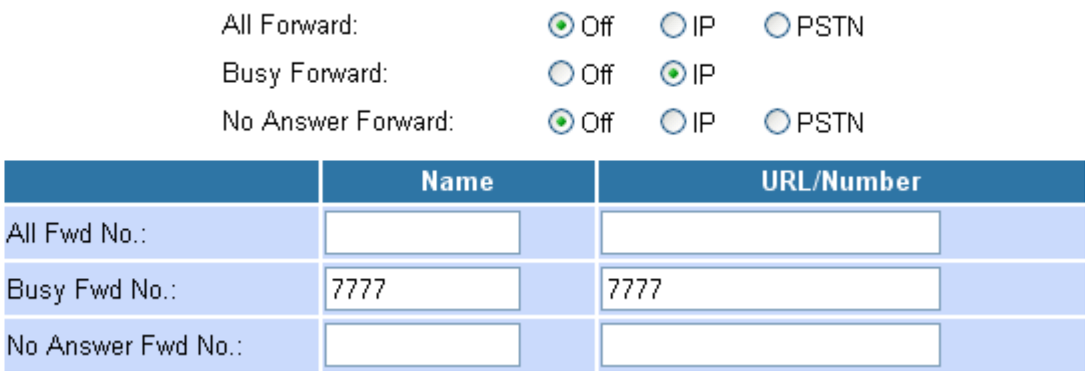
All incoming call will forward to the VoIP IP Gateway's WAN IP Address **192.168.10.36**.

Example 3: All Forward: **PSTN**, Name.: **88888888**, URL/Number: **88888888**



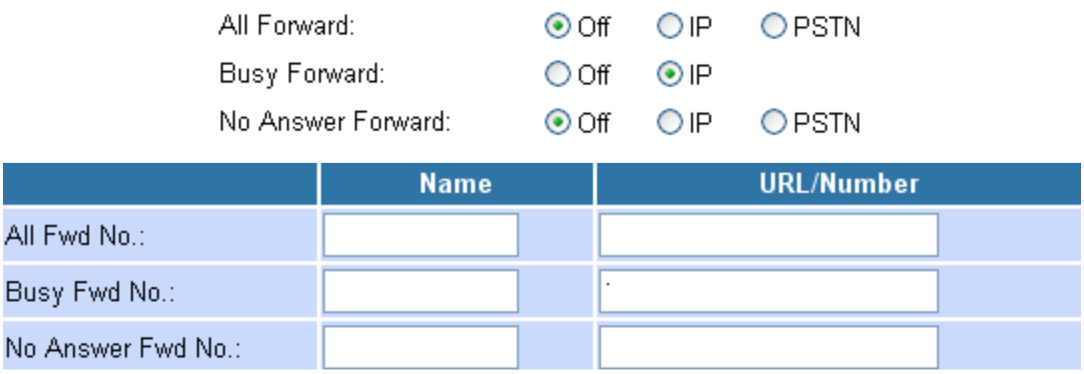
All incoming call will forward to the PSTN phone number **88888888**.

Example 4: All Forward: **IP**, Name.: **7777**, URL/Number: **7777**



If you are on the phone, the new incoming call will forward to the VoIP phone number **7777**.

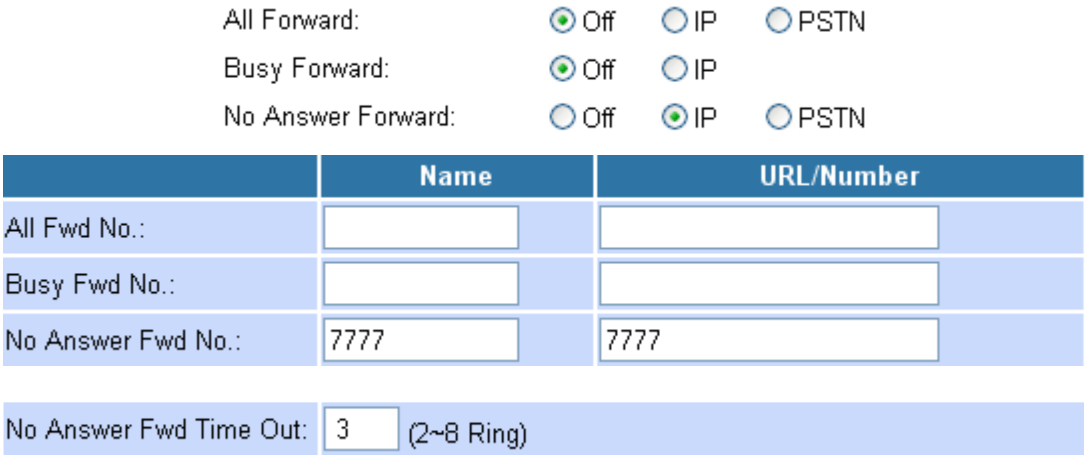
Example 5: All Forward: **IP**, Name: **192.168.10.36**, URL/Number: **192.168.10.36**



If you are on the phone, the new incoming call will forward to the VoIP IP

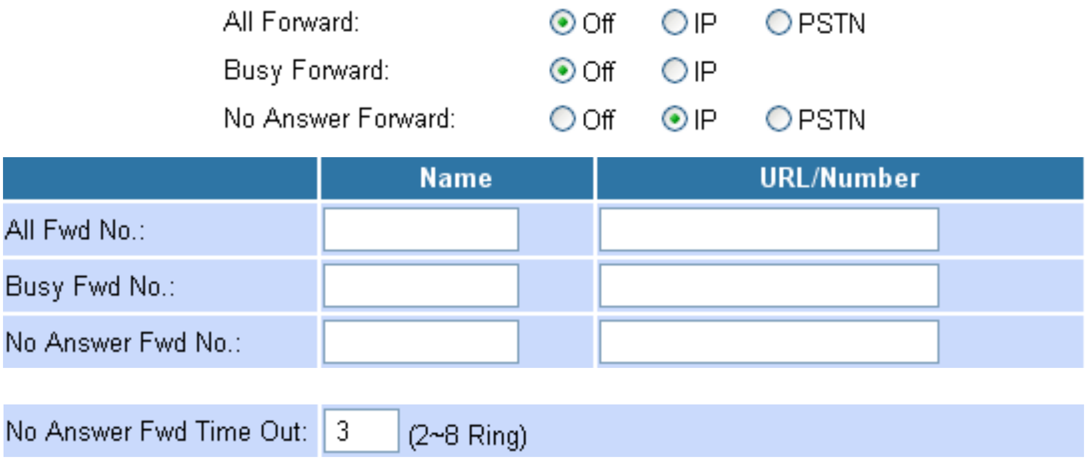
Gateway's WAN IP Address **192.168.10.36**.

Example 6: All Forward: **IP**, Name.: **7777**, URL/Number: **7777**



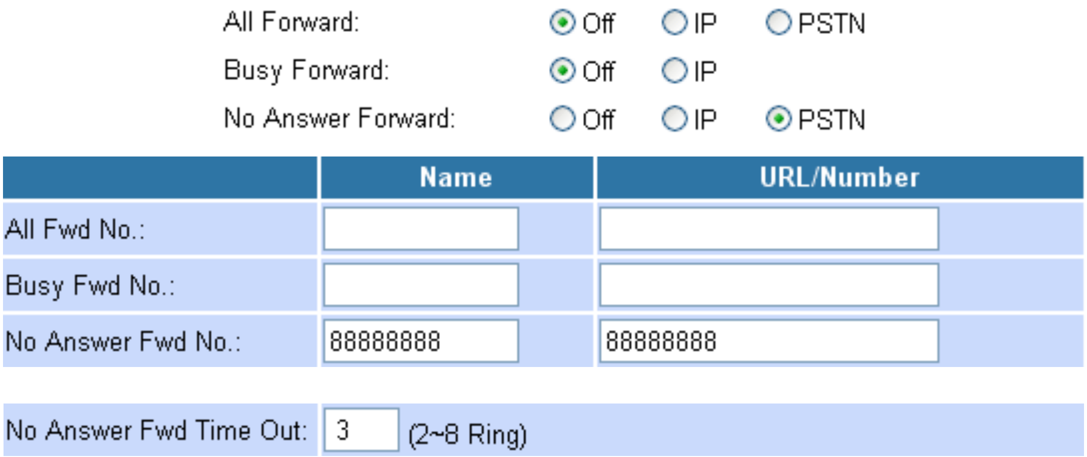
If you can not answer the phone after 3 rings, the incoming call will forward to the VoIP phone number **7777**.

Example 7: All Forward: **IP**, Name: **192.168.10.36**, URL/Number: **192.168.10.36**



If you can not answer the phone after 3 rings, the incoming call will forward to the VoIP IP Gateway's WAN IP Address **192.168.10.36**.

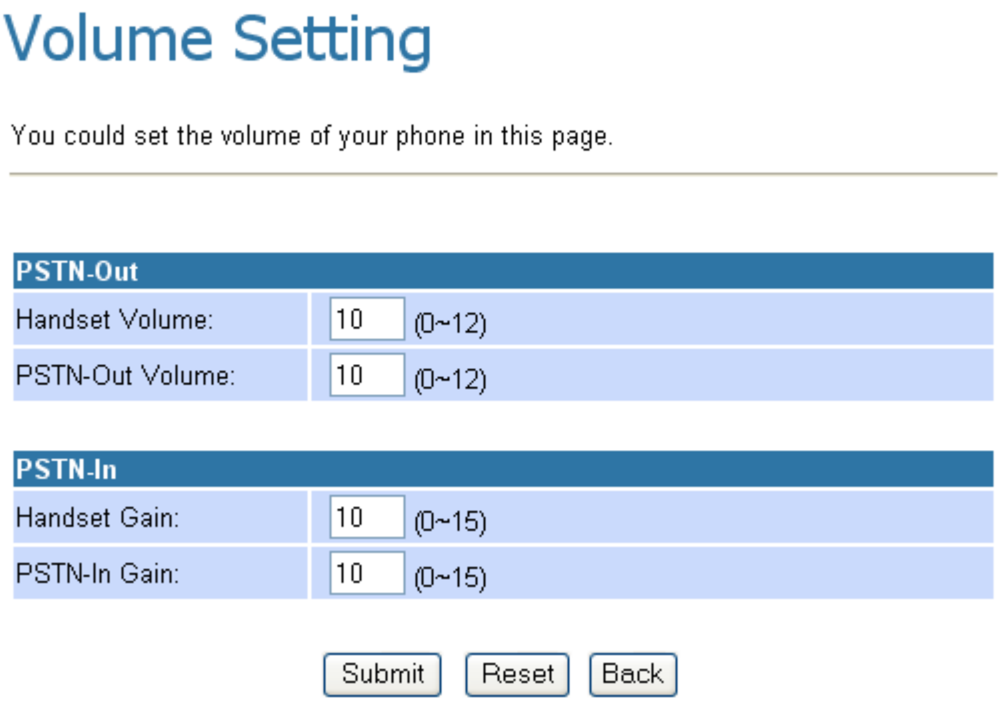
Example 8: All Forward: **PSTN**, Name.: **88888888**, URL/Number: **88888888**



If you can not answer the phone after 3 rings, the incoming call will forward to the PSTN phone number **88888888**.

4.2.3.3.2 Volume Setting function

You can setup the **Handset Volume**, **PSTN-Out Volume**, **Handset Gain** and the **PSTN-In Gain**.
When you finished the setting, please click the **Submit** button.



4.2.3.3.2.1 **Handset Volume** is to set the volume for you can hear from the earphone of your handset.

4.2.3.3.2.2 **PSTN-Out Volume** is to set the PSTN volume from the microphone of your handset send out to the other side's earphone of handset.

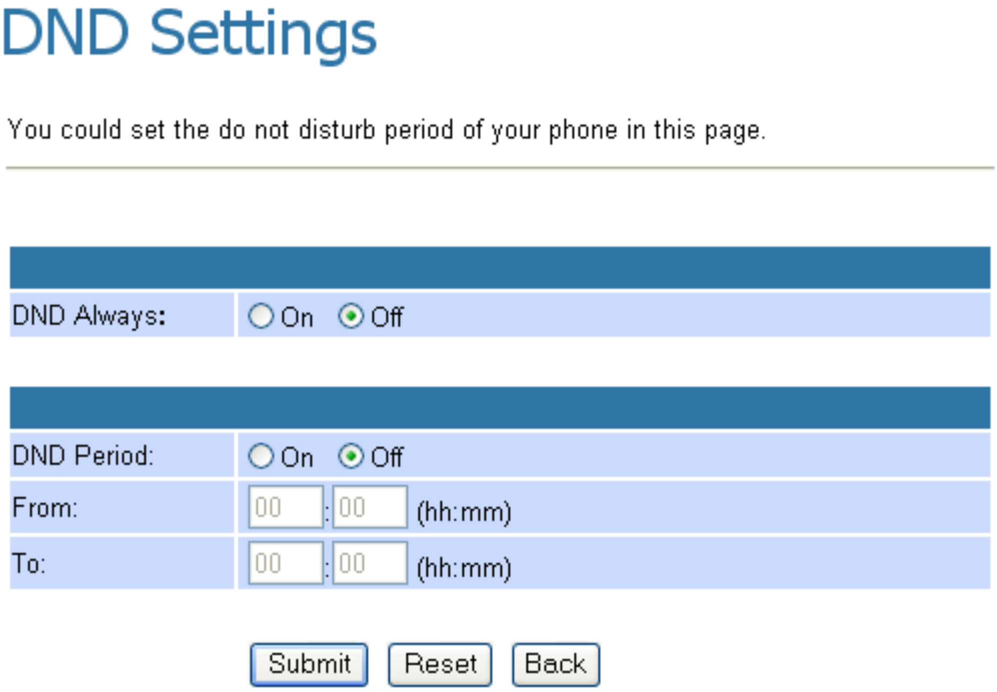
4.2.3.3.2.3 **Handset Gain** is to set the volume from the microphone of your handset send out to the other side's earphone of handset.

4.2.3.3.2.4 **PSTN-In Gain** is to set the PSTN volume for you can hear from the earphone of your handset.

4.2.3.3.2.5 When you finished the setting, please click the **Submit** button.

4.2.3.3.3 DND Setting function

You could set the do not disturb period of your phone in this page.



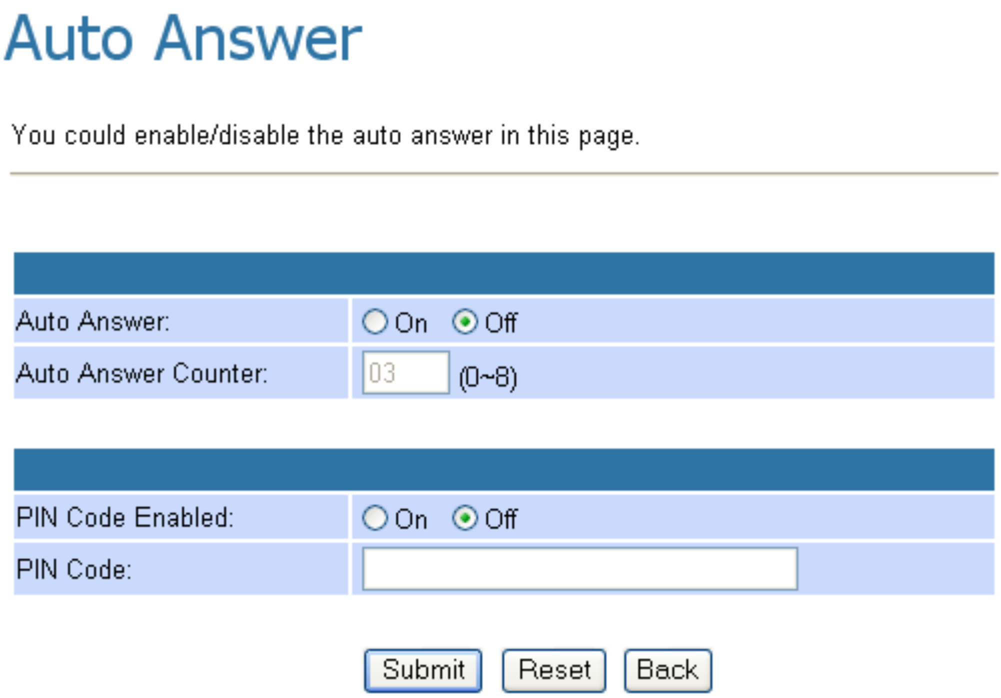
DND Always	Default is **Off (disable)**. When it was **On (enable)**. All incoming call will be blocked and the caller will hear the busy tone any time when place a call until disable this feature.
DNS Period	Default is **Off (disable)**. When it was **On (enable)**. All incoming call will be blocked and the caller will hear the busy tone any time when place a call during the time period until disable this feature. If the **"From"** time is large than the **"To"** time, the Block time will from Day 1 to Day 2.
From	Input the start time of the time period. (24 hours format, hh:mm)
To	Input the end time of the time period. (24 hours format, hh:mm)
Submit Button	When you finished the setting, please click the **Submit** button.
Reset Button	You can reset the configured parameters before you submit
Back Button	Go back to the previous web page

4.2.3.3.4 Auto Answer function (Only the for 1 FXO +1 FXS)

You can set the **Auto Answer** function to answer the incoming call by the phone.
If the call is come from the VoIP, then the VoIP Gateway can let user to redial the call to PSTN phone number.
If the call is coming from PSTN, then the VoIP Gateway can let user to redial to VoIP Phone number.
Once VoIP Gateway received specific rings in the **Auto Answer Counter** and you authenticated by entering the correct **PIN Code** as configured, you can make a call on the other network. For example, you can call in on the VoIP network and make a PSTN call or you can call in on the PSTN line and make a VoIP call.

In order to make a call, simply enter the desired number or speed dial number followed by the '**#**' key and you call will be placed.

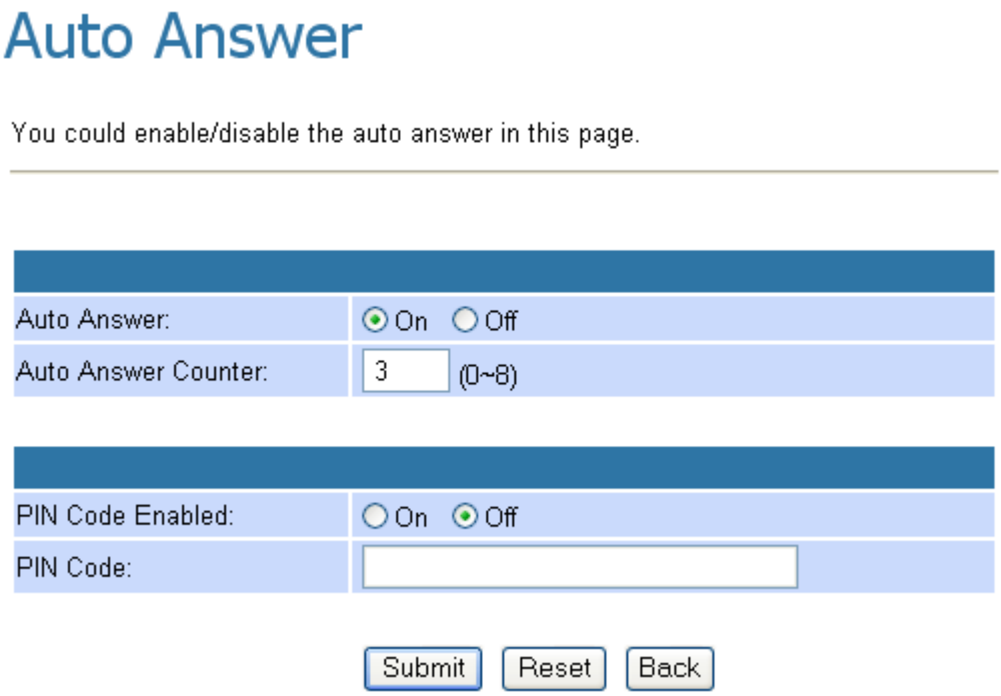
If you call in via PSTN and make a VoIP Toll Bypass call, you can terminate the call by simply hanging up. If you call in via the VoIP network and make a PSTN Toll Bypass call, you can terminate the call by ending the VoIP session.



Auto Answer	Default is **Off (disable)**. When it was **On (enable)**, It'll enable **Auto Answer**.
Auto Answer	Default is **3**. It is to set after the ring count meets the

Counter	number you set then the auto answer will enable.
PIN Code Enabled	Default is **Off (disable)**. When it was **On (enable)**, It'll detect the Caller ID Type 2.
PIN Code	This can be changed and will be required in order to redial the call to VoIP or PSTN phone number. The range of PIN Code is 1 to 30. If you've configured the PIN Code and the call is come from the VoIP, then the VoIP Gateway can let user to dial the PIN Code first and then redial the call to PSTN phone number. If you've configured the PIN Code and the call is come from the PSTN, then the VoIP Gateway will ask user to dial the PIN Code first and then redial the call to VoIP phone number.
Submit Button	When you finished the setting, please click the **Submit** button.
Reset Button	You can reset the configured parameters before you submit
Back Button	Go back to the previous web page

Example 1: Auto Answer: **On**, Auto Answer Counter: **3**



How to Use - PSTN to VoIP Call:

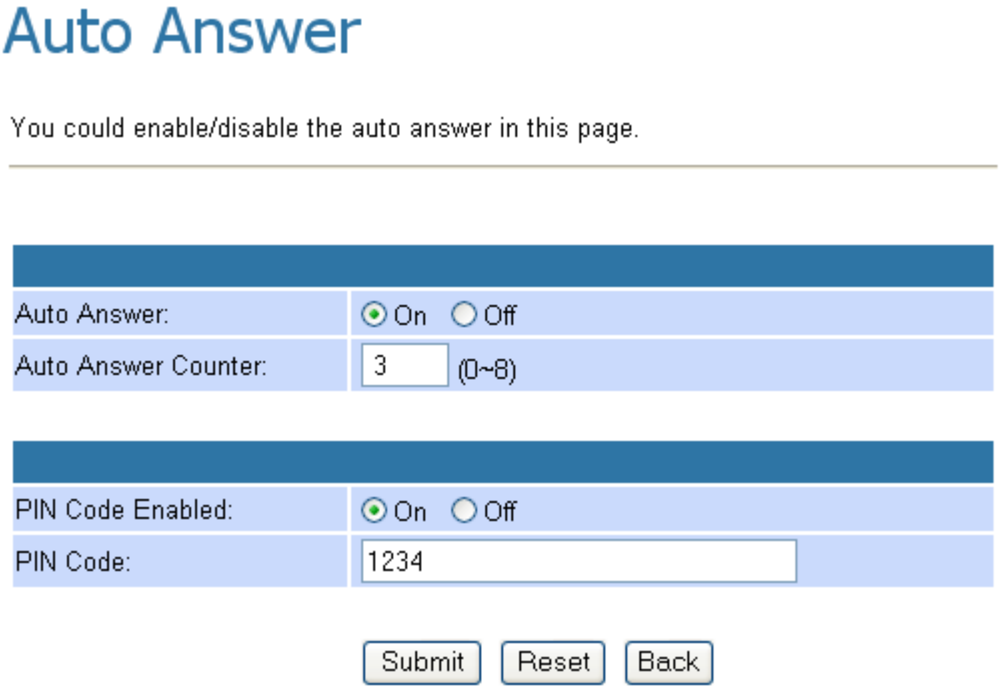
1. Call in via PSTN.
2. When you hear the dial tone indicating that the VoIP Gateway is receiving **3 rings** and expecting a number, dial the VoIP phone number to which you

want to call, then press **#** (optional) to make a PSTN to VoIP call.

How to Use - VoIP to PSTN Call:

1. Call in via VoIP.
2. When you hear the dial tone indicating that the VoIP Gateway is receiving **3 rings** and expecting a number, dial the VoIP phone number to which you want to call, then press **#** (optional) to make a VoIP to PSTN call.

Example 2: Auto Answer: **On**, Auto Answer Counter: **3**, PIN Code Enabled: **On**, PIN Code: **1234**



How to Use - PSTN to VoIP Call:

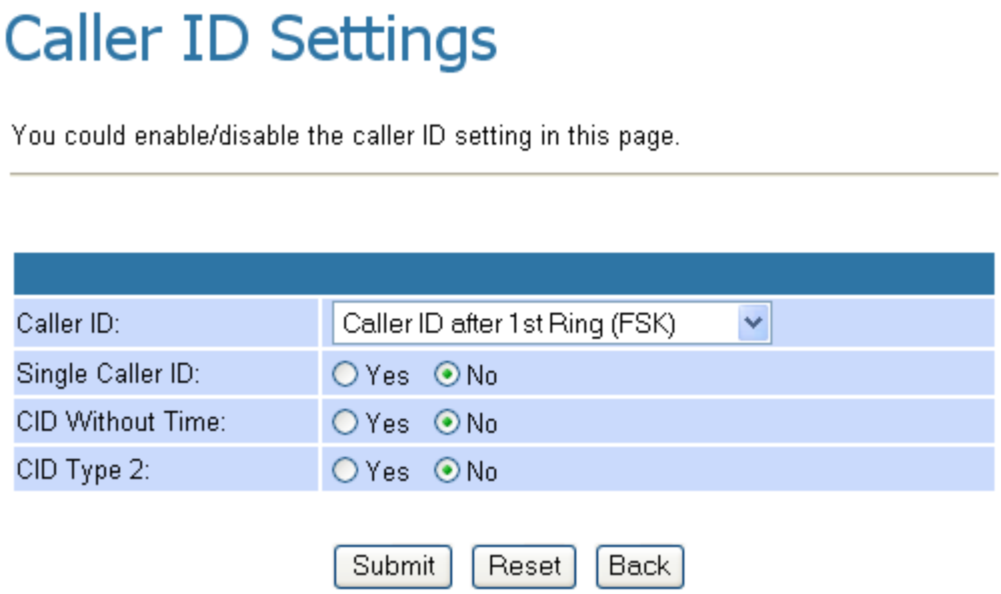
1. Call in via PSTN.
2. When you hear the continued **BEEP BEEP** indicating that the VoIP Gateway is asking you to enter the **PIN Code**.
3. Enter the correct PIN Code **1234**, then press **#**.
4. When you hear the dial tone indicating that the VoIP Gateway is expecting a number, dial the VoIP phone number to which you want to call, then press **#** (optional) to make a PSTN to VoIP call.

How to Use – VoIP to PSTN Call:

1. Call in via VoIP.
2. When you hear the continued **BEEP BEEP** indicating that the VoIP Gateway is asking you to enter the **PIN Code**.
3. Enter the correct PIN Code **1234**, then press **#**.
4. When you hear the dial tone indicating that the VoIP Gateway is expecting a number, dial the PSTN phone number to which you want to call, then press **#** (optional) to make a VoIP to PSTN call.

4.2.3.3.5 Caller ID function

You can set the device to show Caller ID in your PSTN Phone or IP Phone. There are four selections of Caller ID. You need to base on your environment to set the Caller ID function for FSK or DTMF.

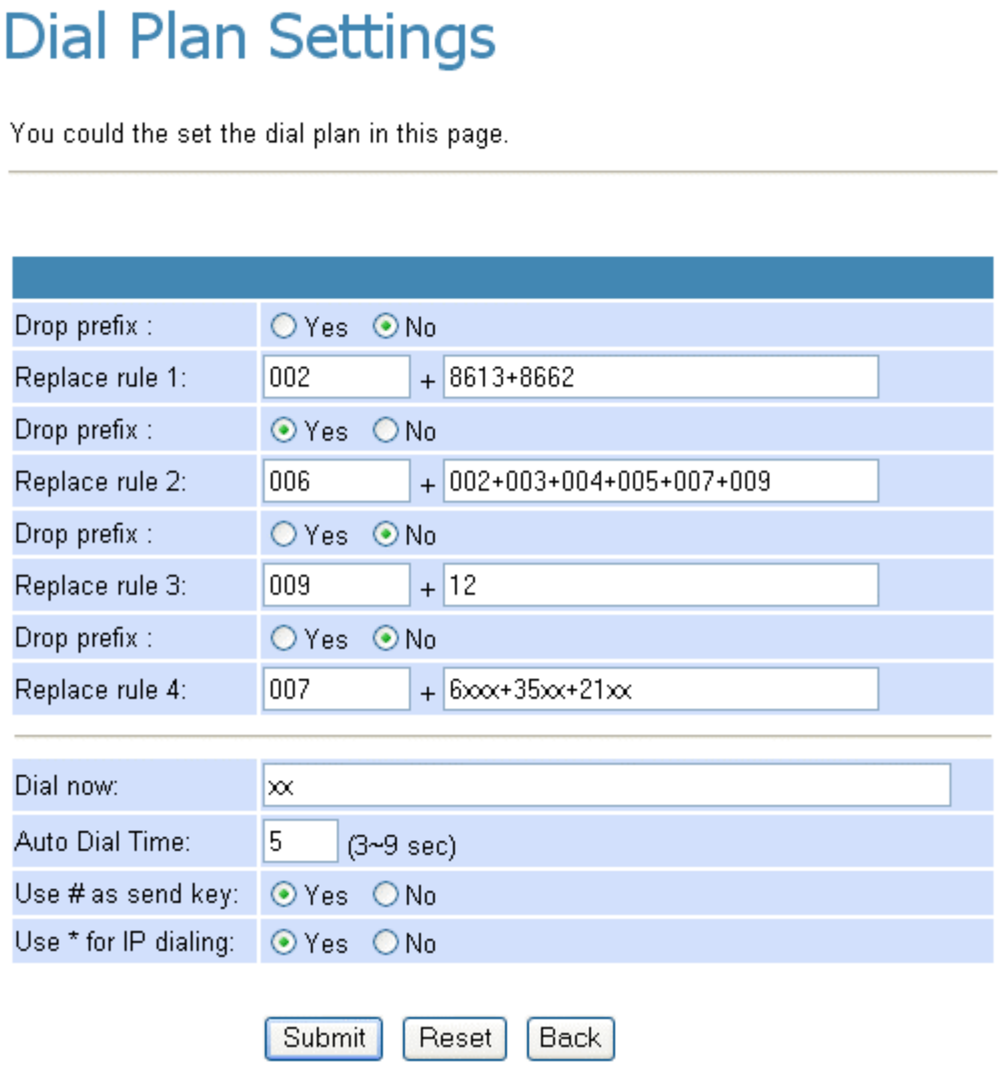


Single Caller ID	Default is **Off (disable)**. When it was **Yes (enable)**, It'll detect the Singel Caller ID.
CID Without Time	Default is **Off (disable)**. When it was **Yes (enable)**, It'll detect the Caller ID without time.
CID Type 2	Default is **Off (disable)**. When it was **Yes (enable)**, It'll

	detect the Caller ID Type 2.
Submit Button	When you finished the setting, please click the **Submit** button.
Reset Button	You can reset the configured parameters before you submit
Back Button	Go back to the previous web page

4.2.3.3.6 Dial Plan function

Number for add or replace before dial the phone number.

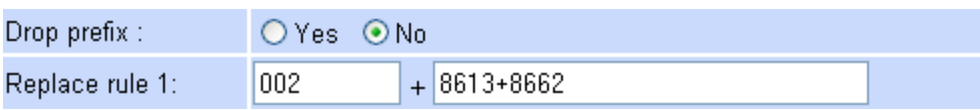


Drop Prefix	Default is **NO (Add the Prefix)**. When it was **Yes (Drop the Prefix)**, It'll drop the prefix. **NO (Add the Prefix):** When it meets the rule which you configured, it'll add the prefix. **Maximum input digits are 7**. **Yes (Drop the Prefix):** When it meets the rule which you configured, it'll drop the prefix and replace the number which you configured. **Maximum input digits are 31**.
Replace rule1 **Replace rule2** **Replace rule3** **Replace rule4**	There are total 4 replace rules for use. **+:** or **xxx**: Define the length of digits.
Dial now	If the numbers which you dialed met this rule, it will dial out with its dial plan immediately. Be noted that the first digit cannot be 0 due to 0 in the first digit is to ignore this rule. If you set the rule 0xxxxx and this rule is invalid due to the first digit is 0.
Auto Dial Time	Default is **5 (Seconds)**. How long the phone number will be dialed out after finishing dialing the digits.
Use # as send key	Default is **Yes**. When it was **No**, It'll wait for the setting of Auto Dial Time and then dial out after dialing the phone numbers.
Use * for IP dialing	Default is **Yes**. When it was **No**, the * key will not be as **.** for IP Dialing.
Submit Button	When you finished the setting, please click the **Submit** button.
Reset Button	You can reset the configured parameters before you submit
Back Button	Go back to the previous web page

Symbol explain:

x or X	0,1,2,3,4,5,6,7,8,9
+	or

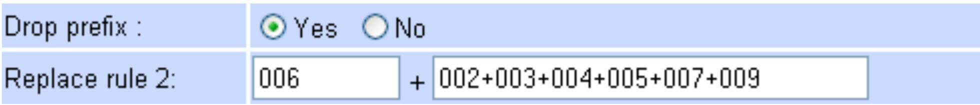
Example 1: Drop prefix: **No**, Replace rule 1: **002, 8613+8662**



When the number **8613** has been dialed, the prefix **002** will be added and the real phone number [**002+8613+xxx**] will be dialed out.
For example, when you dial the number **86315555** and the prefix **002** will be added and the real phone number **00286135555** will be dialed out.

When the number **8662** has been dialed, the prefix **002** will be added and the real phone number [**002+8662+xxx**] will be dialed out.
For example, when you dial the number **86625555** and the prefix **002** will be added and the real phone number **00286625555** will be dialed out.

Example 2: Drop prefix: **Yes**, Replace rule 2: **006, 002+003+004+005+007+009**



When the number **002** has been dialed, the digits **002** will be replaced to **006** and the whole digits [**006+xxx**] will be dialed out.
For example, when you dial the number **0025555** and the digits **002** will be replaced to **006** and then the real phone number **0065555** will be dialed out.

When the number **003** has been dialed, the digits **003** will be replaced to **006** and the real phone number [**006+xxx**] will be dialed out.
For example, when you dial the number **0035555** and the digits **003** will be replaced to **006** and then the real phone number **0065555** will be dialed out.

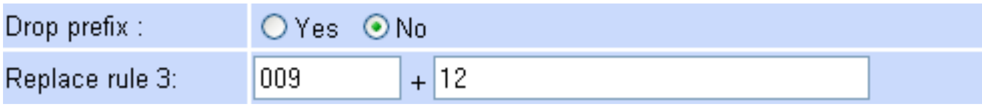
When the number **004** has been dialed, the digits **004** will be replaced to **006** and the real phone number [**006+xxx**] will be dialed out.
For example, when you dial the number **0045555** and the digits **004** will be replaced to **006** and then the real phone number **0065555** will be dialed out.

When the number **005** has been dialed, the digits **005** will be replaced to **006** and the real phone number [**006+xxx**] will be dialed out.
For example, when you dial the number **0055555** and the digits **005** will be replaced to **006** and then real phone number digits **0065555** will be dialed out.

When the number **007** has been dialed, the digits **007** will be replaced to **006** and the real phone number [**006+xxx**] will be dialed out.
For example, when you dial the number **0075555** and the digits **007** will be replaced to **006** and then the real phone number **0065555** will be dialed out.
When the number **009** has been dialed, the digits **009** will be replaced to **006** and the real phone number [**006+xxx**] will be dialed out.

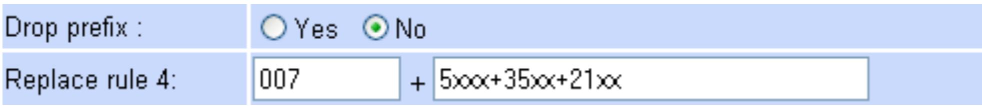
For example, when you dial the number **0095555** and the digits **009** will be replaced to **006** and then the real phone number **0065555** will be dialed out.

Example 3: Drop prefix: **No**, Replace rule 3: **009, 12**



When the number **12** has been dialed, the prefix **009** will be added and the whole digits [**009+12+xxx**] will be dialed out.
For example, when you dial the number **125555** and the prefix **009** will be added and the real phone number **009125555** will be dialed out.

Example 4: Drop prefix: **No**, Replace rule 4: **007, 5xxx+35xx+21xx**



When the number **5xxx** has been dialed, the prefix **007** will be added and the whole digits [**007+5xxx**] will be dialed out. Be note that the range of **xxx** is from **000** to **999**.
For example, when you dial the number **5000** and the prefix **007** will be added and the real phone number **0075000** will be dialed out.
For example, when you dial the number **5999** and the prefix **007** will be added and the real phone number **0075999** will be dialed out.

When the number **35xx** has been dialed, the prefix **007** will be added and the whole digits [**007+35xx**] will be dialed out. Be note that the range of **xx** is from **00** to **99**.
For example, when you dial the number **3500** and the prefix **007** will be added and the real phone number **0073500** will be dialed out.
For example, when you dial the number **3599** and the prefix **007** will be added and the real phone number **0073599** will be dialed out.

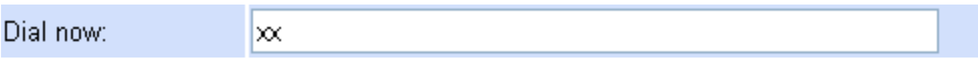
When the number **21xx** has been dialed, the prefix **007** will be added and the whole digits [**007+21xx**] will be dialed out. Be note that the range of **xx** is from **00** to **99**.
For example, when you dial the number **2100** and the prefix **007** will be added and the real phone number **0072100** will be dialed out.

For example, when you dial the number **2199** and the prefix **007** will be added and the real phone number **0072199** will be dialed out.

When the number **534** have been dialed, the prefix **007** will not be added and the real phone number **534** will be dialed out due to **534 (3 digits)** is not in the rule **5xxx (4 digits)**.

When the number **358822** have been dialed, the prefix **007** will not be added and the real phone number **358822** will be dialed out due to **358822 (6 digits)** is not in the rule **35xx (4 digits)**.

Example 5: Dial Now: **xx**

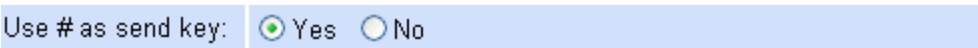


When the two digits in the range from **00 to 99** has been dialed, it will be dial out immediately.

Auto Dial Time function

This function is when you input the phone number by the keypad but you don't need to press **"#"**. After time out the system will dial directly.

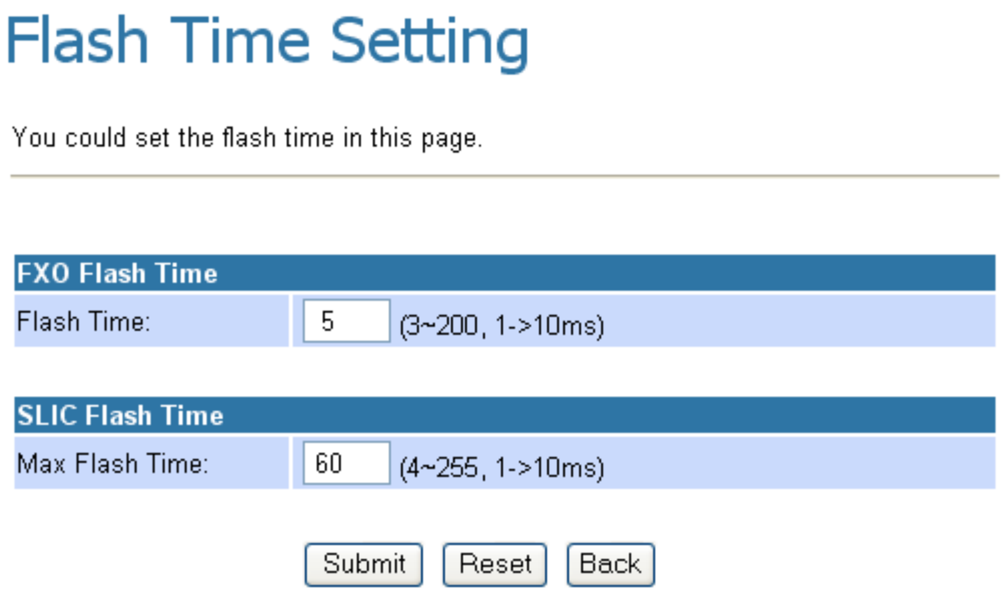
Auto Dial Time function



The ***** key will not be as **.** for IP Dialing. If you want to dial the IP Dialing, you need to know the WAN IP Address of the remote VoIP Devices. For example if the WAN IP Address of Remote VoIP Device is 222.222.222.222 and then you need to dial 222*222*222*222# to make a IP Dialing.

4.2.3.3.7 Flash Time Settings function

When you use the PSTN Phone and you need to press the Hook to do the Flash (Switch to the other phone line or HOLD), this function is for you to set the time you press the Hook to represent the Flash function.

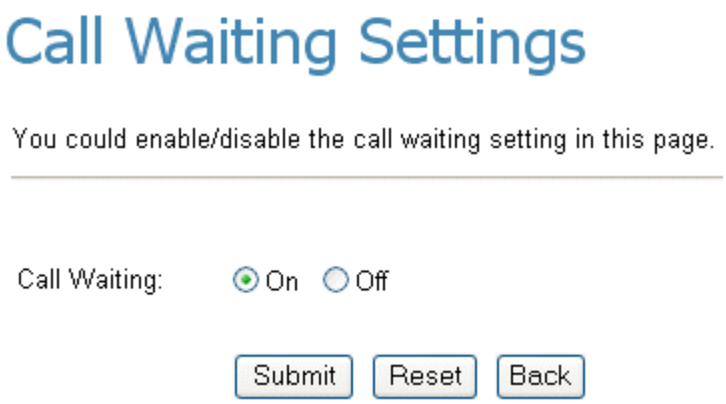


4.2.3.3.8 Call Waiting Settings

You could enable/disable the call waiting setting in this page.

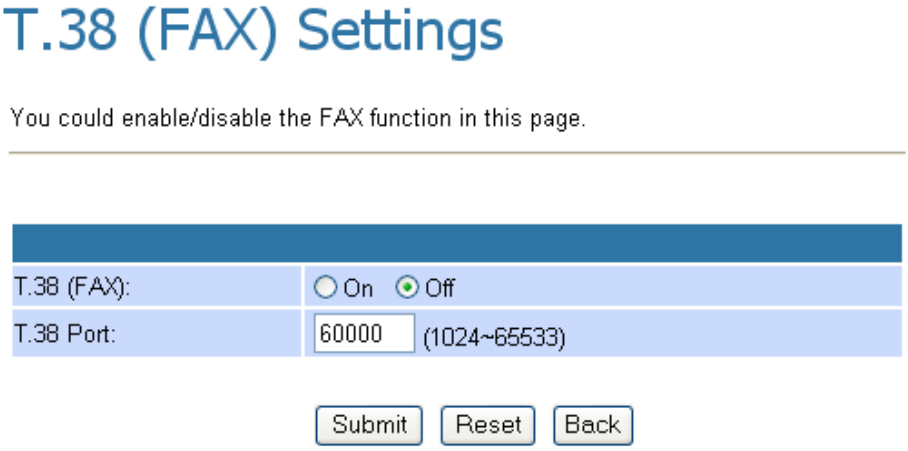
When a new call is coming while you are talking, you can push the Flash button to switch to the new call. You can push the Flash button to switch between the two calls.

Hang up the phone to end the first call and then the phone will ring, please pick it up to talk to the second call. Hang up again to end the call.

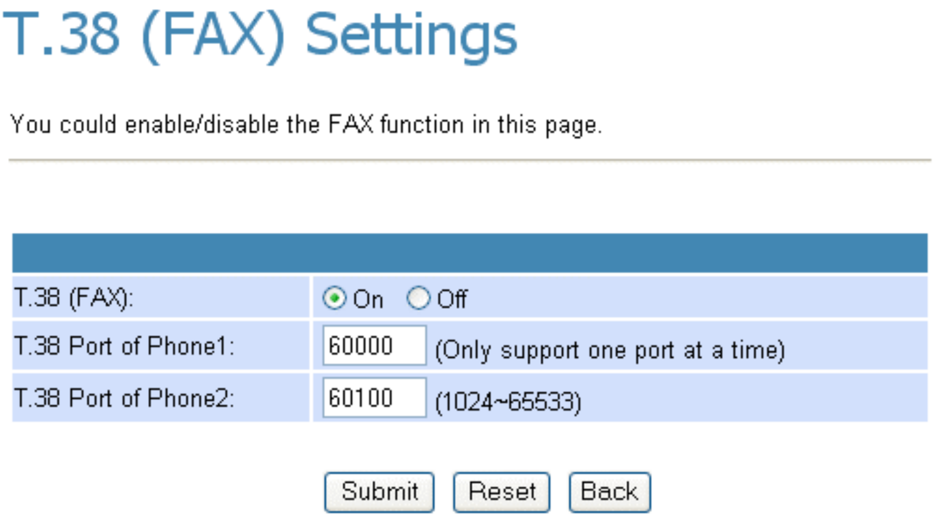


4.2.3.3.9 T.38 (FAX) Setting

You could enable/disable the FAX function in this page.



For 1 FXS Port



For 2 FXS Port

T.38 (FAX)	When it was **On (Enable)**, It'll enable the T.38 Fax function.
T.38 Port/ T.38 Port of Phone1	Default is **60000**. (Only support one port at a time)
T.38 Port of Phone2	Default is **60100**. (Only support one port at a time)
Submit Button	When you finished the setting, please click the **Submit** button.
Reset Button	You can reset the configured parameters before you submit
Back Button	Go back to the previous web page

T.38 support

Fax Pass-through

In fax pass-through mode, UDPTL packets are not used. Fax communication between the two fax machines is carried in its entirety in-band over a voice call (over RTP). The VoIP Gateway is aware that the call in progress is a fax call and not voice call. If during a voice call, the CED/CNG fax tones are recognized, then the VoIP Gateway will change the voice codec to G.711, if necessary, turn off echo cancellation (EC) and voice activity detection (VAD) and fix the jitter and reorder buffers to fix the network delay for the duration of the call.

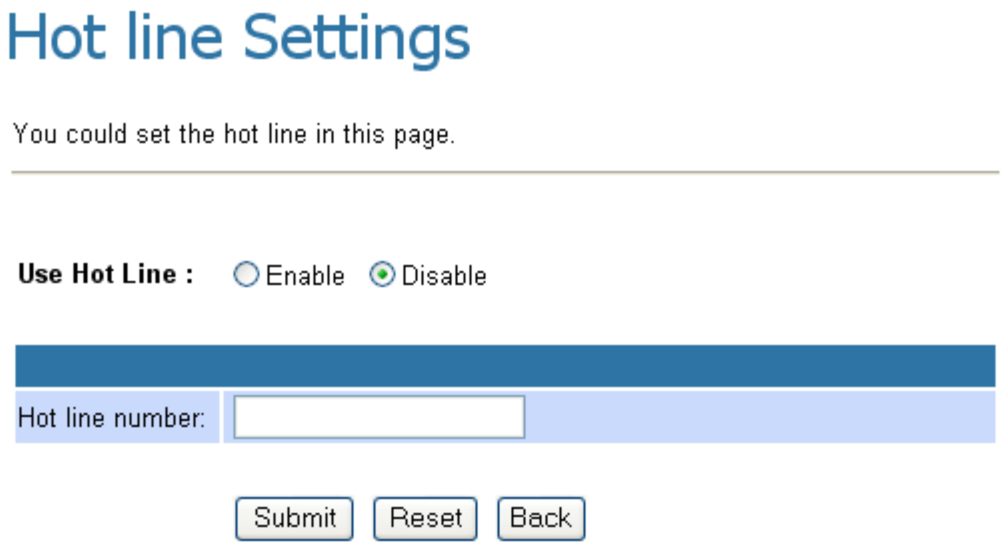
T.38 support mode

T.38 provides an ITU-T standards-based method and protocol for fax. Annexure D describes the system level requirements and procedures for establishing fax calls between two SIP based endpoints. In this mode, the VoIP Gateway will establish a normal voice call and switch to fax based on the detection of Fax tones from the PTM. It will then renegotiate the session parameters with new T.38 parameters. The rest of the fax signaling and data is then encapsulated and sent in IFP packets. The IFP packets can be sent over TCP or UDP (VoIP Gateway supports only UDP). On call disconnect, SIP signaling is used to end the call.

The ITU-T T.38 defines the behavior for both Internet Aware Fax Devices (IAF, network aware fax machine) and Gateways connected to G3FE (Group 3 Fax equipment). The VoIP Gateway supports both kinds of behaviors.

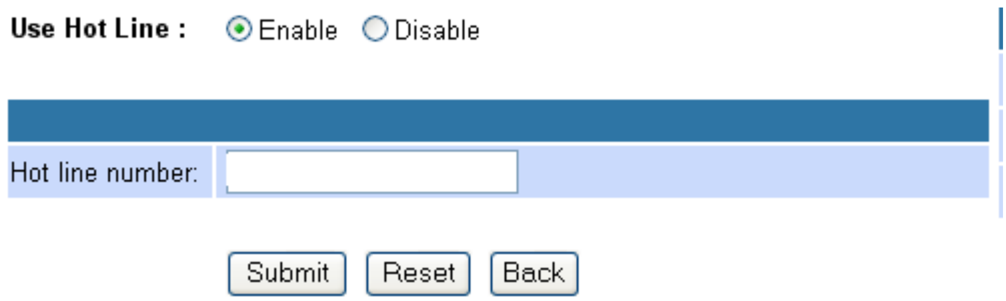
4.2.3.3.10 Hot line Settings

Provide the Hot Ling function.
It'll dial to the configured URL, VoIP Phone Number, Remote WAN IP Address of VoIP Gateway automatically every time when you pick up the phone.



Use Hot Line	Default is **Disable**. When it was **Enable**, It'll enable the **Hot Line** function.
Hot Line Number	Enter the URL, VoIP Phone Number, Remote WAN IP Address of VoIP Gateway which you want to use for **Hot Line**.
Submit Button	When you finished the setting, please click the **Submit** button.
Reset Button	You can reset the configured parameters before you submit
Back Button	Go back to the previous web page

Example 1: Use Hot Line: **Enable**, Hot line number



Every time when you pick up the phone, it'll dial to the VoIP Phone Number **2468013579** automatically.

Example 2: Use Hot Line: **Enable**, Hot line number: **voiptest**

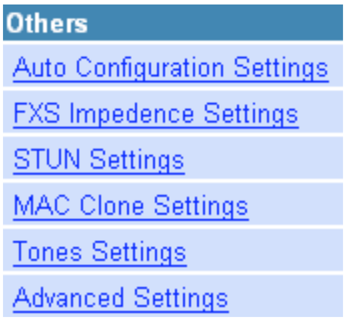
Every time when you pick up the phone, it'll dial to the VoIP Phone Number **voiptest** automatically.

Example 3: Use Hot Line: **Enable**, Hot line number

Every time when you pick up the phone, it'll dial to the WAN IP Address **192.168.10.63** of Remote VoIP Gateway automatically.

4.2.3.4 Others

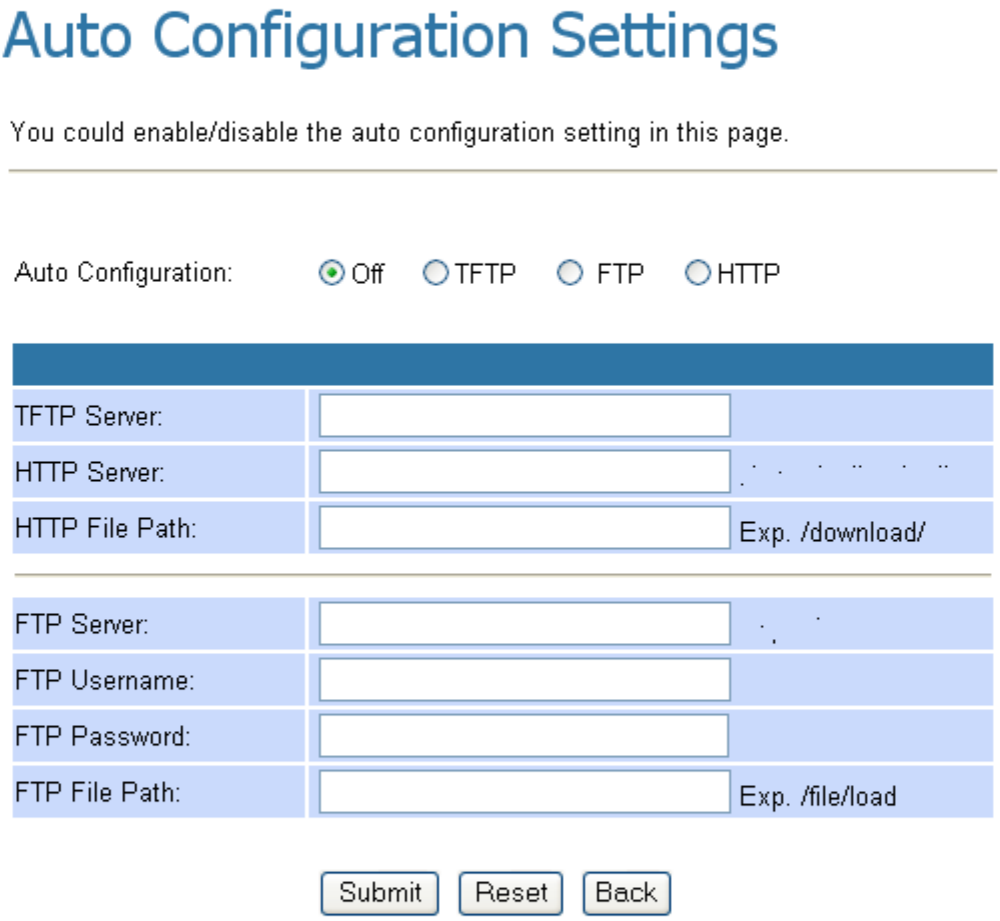
In Others contains **Auto Configuration Settings**, **FXO & FXS Impedence Setting**, **MAC Clone Settings** and **Advanced Settings functions**.



4.2.3.4.1 Auto Configuration Settings

You could enable/disable the auto configuration/provisioning setting in this page.

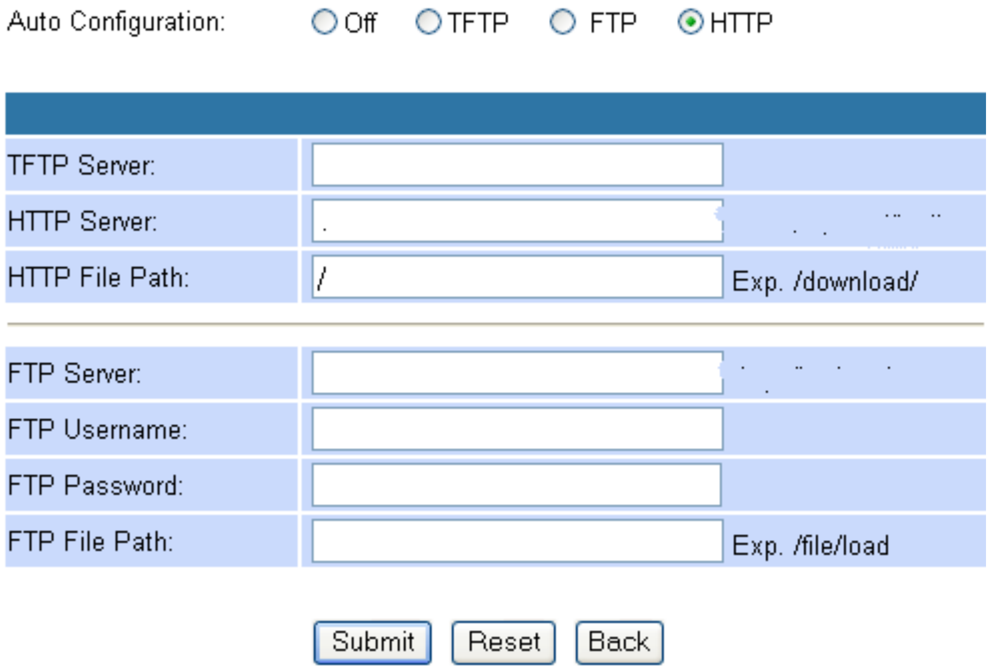
The VoIP Gateway provides for secure provisioning and remote upgrade. Provisioning is achieved through configuration profiles transferred to the device via TFTP, HTTP or FTP. The VoIP Gateway can be configured to update its VoIP Configuration from a remote profile on power up or reboot.



Auto Configuration	Default is **Off(Disable)**. When it was **Enable**, there are 3 types of Auto Configuration: **TFTP, FTP and HTTP**.
TFTP Server	Enter IP or Domain Name of **TFTP Server**.
HTTP Server	Enter IP or Domain Name of **HTTP Server**.
HTTP Path	Enter File Path where the provisioning file is.
FTP Server	Enter IP or Domain Name of **FTP Server**.
FTP Username	Enter Username which provided by **FTP Server**.
FTP Password	Enter Password which provided by **FTP Server**.
File Path	Enter File Path where the provisioning file is.
Submit Button	When you finished the setting, please click the **Submit** button.
Reset Button	You can reset the configured parameters before you submit
Back Button	Go back to the previous web page

Example 1: Auto Configuration for HTTP Server

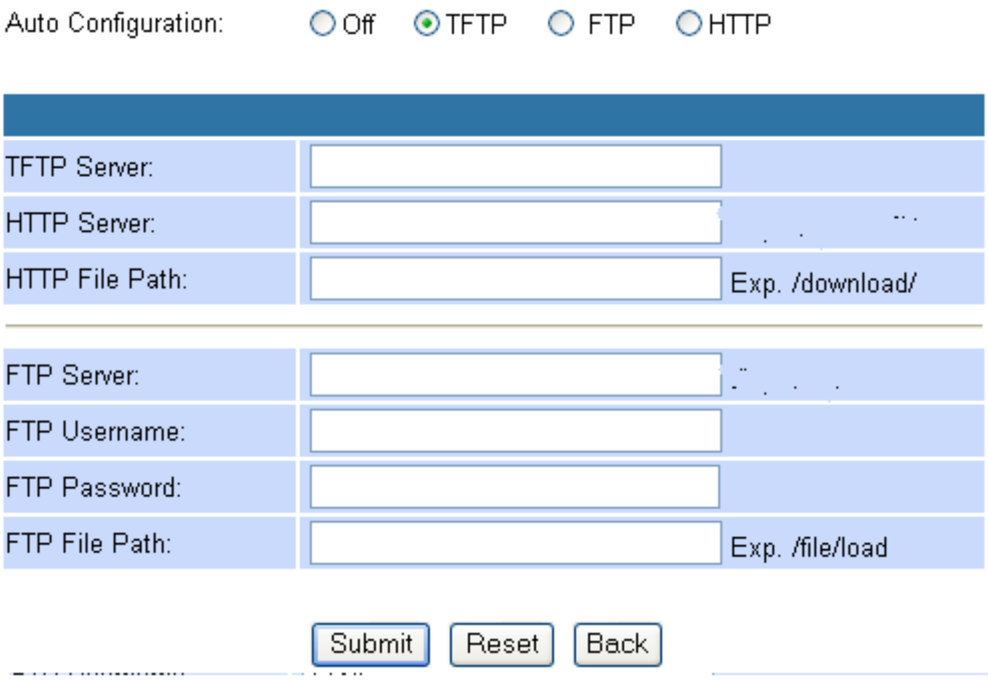
Auto Configuration: **HTTP**, HTTP Server: **192.168.10.100**, HTTP Path: **/**



Every time when you power on the VoIP Gateway, it'll update its VoIP configuration to the latest one from **Auto Provisioning Server (HTTP Server)** automatically.

Example 2: Auto Configuration for TFTP Server

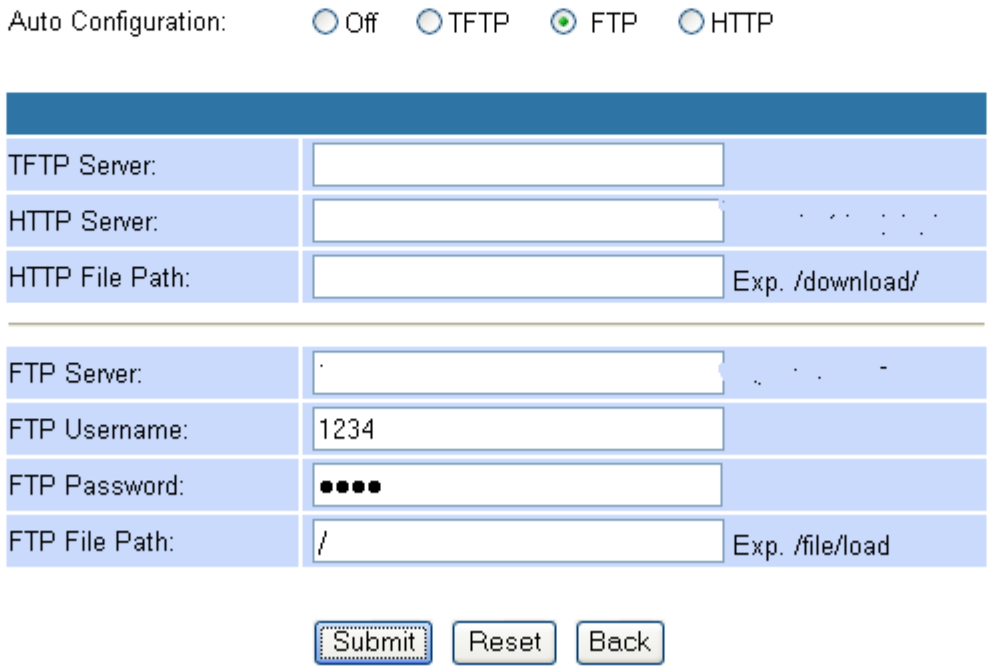
Auto Configuration: **TFTP**, TFTP Server: **192.168.10.100**



Every time when you power on the VoIP Gateway, it'll update its VoIP configuration to the latest one from **Auto Provisioning Server (TFTP Server)** automatically.

Example 3: Auto Configuration for FTP Server

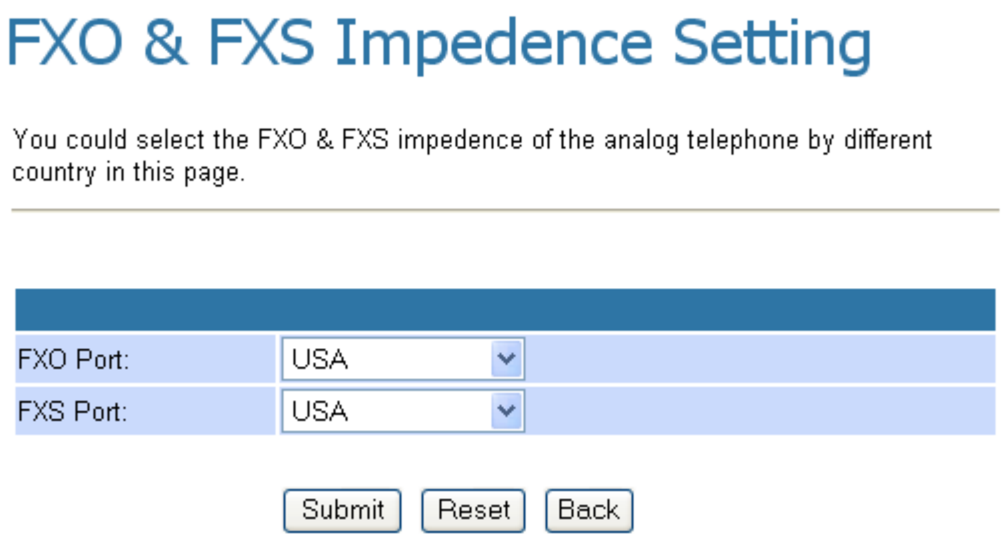
Auto Configuration: **FTP**, FTP Server: **192.168.10.100**, FTP Username: **1234**, FTP Password: **1234**, FTP Path: **/**



Every time when you power on the VoIP Gateway, it'll update its VoIP configuration to the latest one from **Auto Provisioning Server (FTP Server)** automatically.

4.2.3.4.2 FXO & FXS Impedence Setting

You could select the FXO & FXS Impedence Setting for different country in this page.

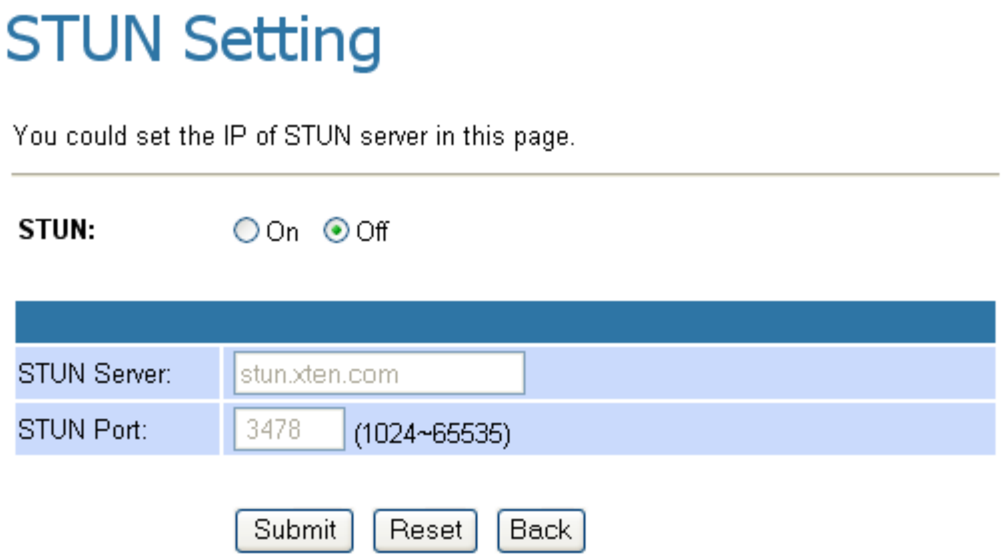


FXO Port	Default is **USA**. You could select the FXO Impedence Setting for different country here.
FXS Port	Default is **USA**. You could select the FXS Impedence Setting for different country here.
Submit Button	When you finished the setting, please click the **Submit** button.
Reset Button	You can reset the configured parameters before you submit
Back Button	Go back to the previous web page

4.2.3.4.3 STUN Setting

You can setup the STUN Enable/Disable and STUN Server IP address in this page.

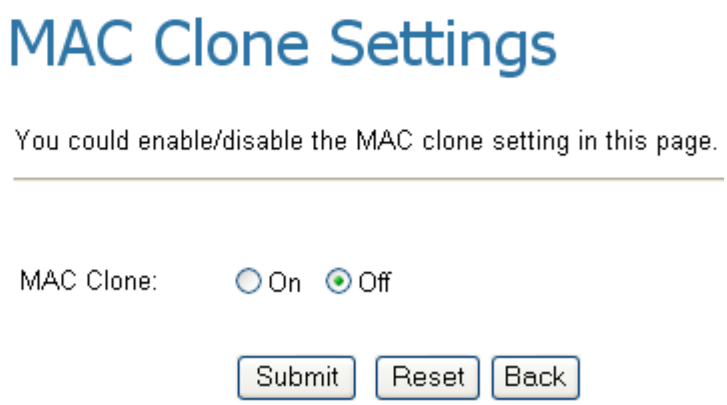
This function can help your VoIP Gateway working properly behind NAT. To change these settings please follow your VoIP Service Provider's information. When you finished the setting, please click the **Submit** button.



STUN	Default is **Off (disable)**. When it was **On (enable)**. It enables STUN (Simple Transversal of UDP through NAT) if the VoIP Gateway is behind a NAT enabled router and the router has no ALG for SIP, or NONE to disable STUN (VoIP Gateway is not to use STUN for NAT traversal). VoIP Gateway also supports a proprietary implementation of NAT traversal where the Service provider is expected to provide some relay support. If NONE is selected, then based on the responses received, the VoIP Gateway will dynamically determine if the SIP Server supports the proprietary implementation. Note: Even when STUN is enabled, the VoIP Gateway does an automatic detection of the presence of SIP ALG and disables the use of STUN. This is to avoid some media problems arising out of the behavior of some ALGs when STUN is used at the user end.
STUN Server	Enter the IP address or Domain Name of the STUN Server. The default is ***stun.xten.com***. This field is applicable only if USE STUN is selected as the NAT traversal technique.
STUN Port	Enter the port number on which the STUN server listens for requests from the STUN Client on VoIP Gateway. The range is 1024 to 65535. The default is ***3478***. This field is applicable only if USE STUN is selected as the NAT traversal technique.
Submit Button	When you finished the setting, please click the **Submit** button.
Reset Button	You can reset the configured parameters before you submit
Back Button	Go back to the previous web page

4.2.3.4.4 MAC Clone Settings

Some particularly ISPs do not want you to have a home network and have a DSL/Cable modem that allows only 1 MAC to talk on the internet. If you change network cards, you have to call them up to change the MAC. The VoIP Gateway can clone the computer's MAC that was originally set up for such an ISP.



MAC Clone	Default is **Off (disable)**. When it was **On (enable)**. The VoIP Gateway clones the computer's MAC that was originally set up for such an ISP.
Submit Button	When you finished the setting, please click the **Submit** button.
Reset Button	You can reset the configured parameters before you submit
Back Button	Go back to the previous web page

4.2.3.4.5 Tones settings

You could configure your tones settings in this page.

Tones Settings

You could configure your tones settings in this page.

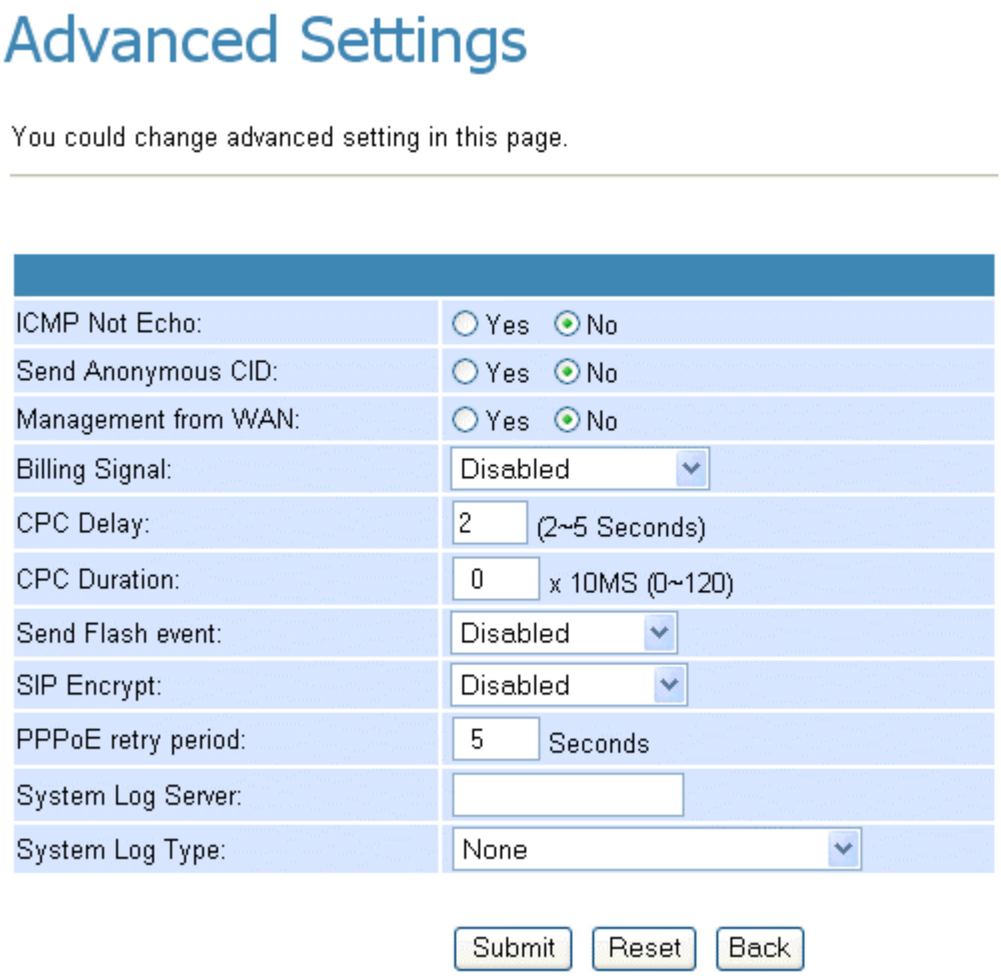
	Dial Tone	Ring Back Tone	Busy Tone	Error Tone	Ring Tone	Insert Tone
Cadence On:	☐	☑	☑	☑	☑	☑
Hi-Tone Freq.:	440	480	620	620	480	440
Lo-Tone Freq.:	350	440	480	480	440	350
Hi-Tone Gain:	4522	2261	2261	2261	15360	2261
Lo-Tone Gain:	2261	2261	2261	2261	15360	1130
On Time 1:	0	200	50	30	200	30
Off Time 1:	0	400	50	20	400	20
On Time 2:	0	0	0	0	0	30
Off Time 2:	0	0	0	0	0	400
On Time 3:	0	0	0	0	0	0
Off Time 3:	0	0	0	0	0	0

Submit Reset Back

4.2.3.4.6 Advanced Settings

You could change advanced setting in this page.

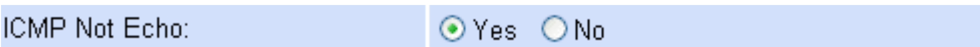
CPC (Calling Party Control) is a signal sent from most modern electronic COs to indicate that the "**Calling Party**" has hung up. The CPC signal tells the phone equipment that the outside party has hung-up, so it can stop recording to an answering machine or voice mail, drop the call off hold, or just release a line that might be used for dictation or announcements. Please refer to http://www.sandman.com/cpcbull.html in detail.



ICMP Not Echo	Default is **Off (disable)**. When it was **On (enable)**. The VoIP Gateway will not echo the ICMP request.
Send Anonymous	The ***Anonymous*** Caller ID to display when you make a call to others VoIP Gateways.

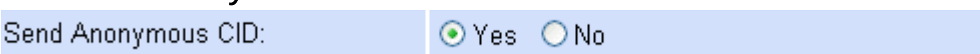
CID	
Billing Signal	Default is **Off (disable)**. When it was **On (enable)**. Polarity Reversal is enabled to inform the charge/billing system (Polarity Reversal, Tone_12K, Tone_16K). **Support FXS Port only**
CPC Delay	Default is **2**. The VoIP Gateway will send the CPC after the delay time which you configuration. **Support FXS Port only**
CPC Duration	When VoIP Gateway is the called party, CPC duration is the "voltage drop" duration, before it plays dial tone again. **Support FXS Port only**
Send Flash event	Default is **Disable**. There are two types of Flash event: ***DTMF Event*** and ***SIP Info***.
SIP Encrypt	Default is **Disable**. There are four types of SIP Encrypt: ***INFINET, AVS, WALKERSUN1, WALKERSUN2, CSF1, CSF2*** and ***GX***.
PPPoE retry period (*)	Default is **5** seconds. The range is 5 to 255. When PPPoE failed to connect to ISP, it will wait for the period which you configured to redial.
System Log Server	To upload the system log on the specified Server
System Log Type	Default is None. There are 7 types: **Call Statistics**, **General Debug**, **Call Statistics + General Debug**, **SIP Debug**, **Call Statistics + SIP Debug**, **General Debug + SIP Debug**, **All**.
Submit Button	When you finished the setting, please click the **Submit** button.
Reset Button	You can reset the configured parameters before you submit
Back Button	Go back to the previous web page

Example 1: ICMP Not Echo: **Yes**



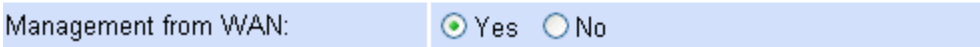
The ICMP will not echo no matter you request from LAN side or WAN side.

Example 2: Send Anonymous CID: **Yes**



Every time when you make a call to others VoIP Gateways, it'll send the **Anonymous** as Caller ID out automatically.

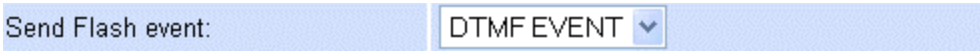
Example 3: Management from WAN: **Yes**



You can remote managed from the WAN IP Address of the VoIP Gateway.

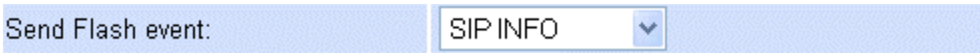
Example 4:

Send Flash event: **DTMF EVENT**



It'll send the DTMF EVENT as Flash event.

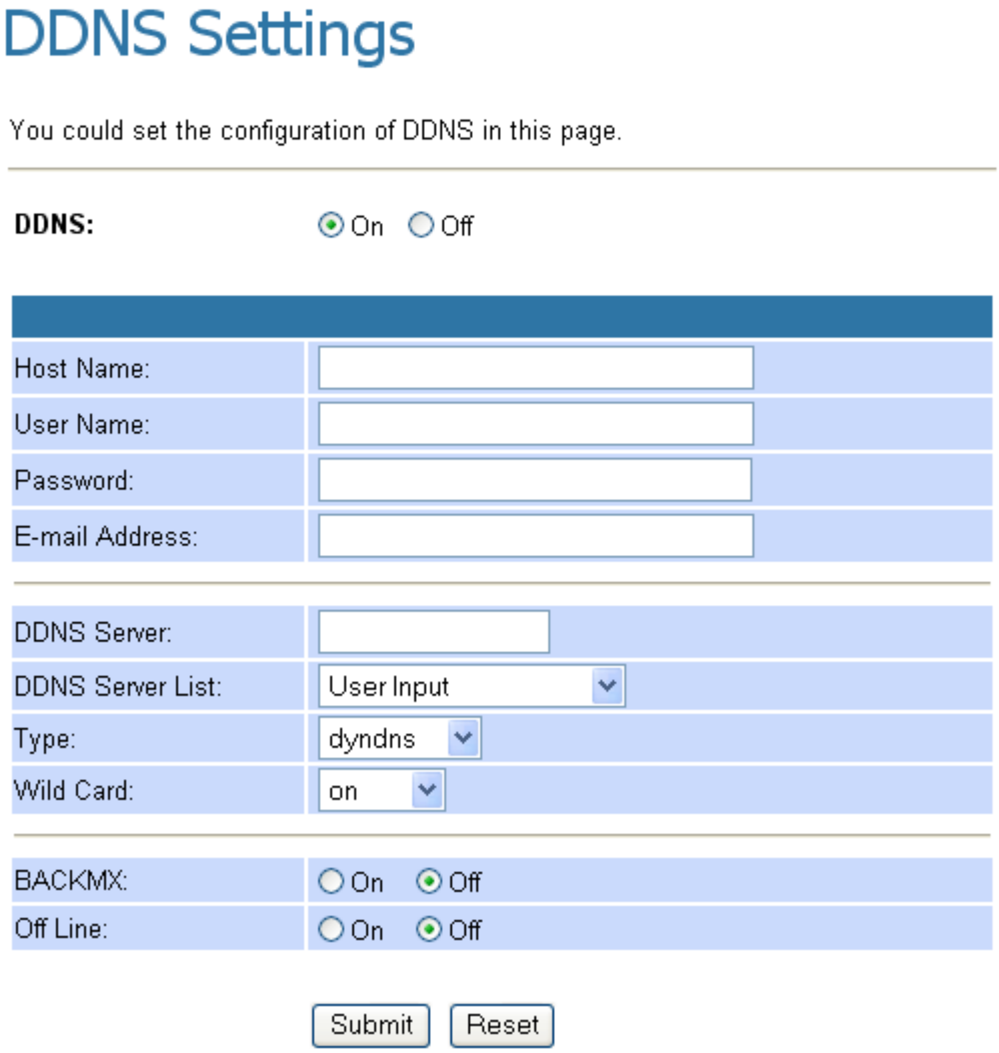
Send Flash event: **SIP INFO**



It'll send the SIP INFO as Flash event.

4.2.4 DDNS Configuration Page

You can configure the DDNS setting in this page. You need to have the DDNS account and input the information properly. You can have a DDNS account with a public IP address then others can call you via the DDNS account. But now most of the VoIP applications are work with a SIP Proxy Server. When you finished the setting, please click the **Submit** button.

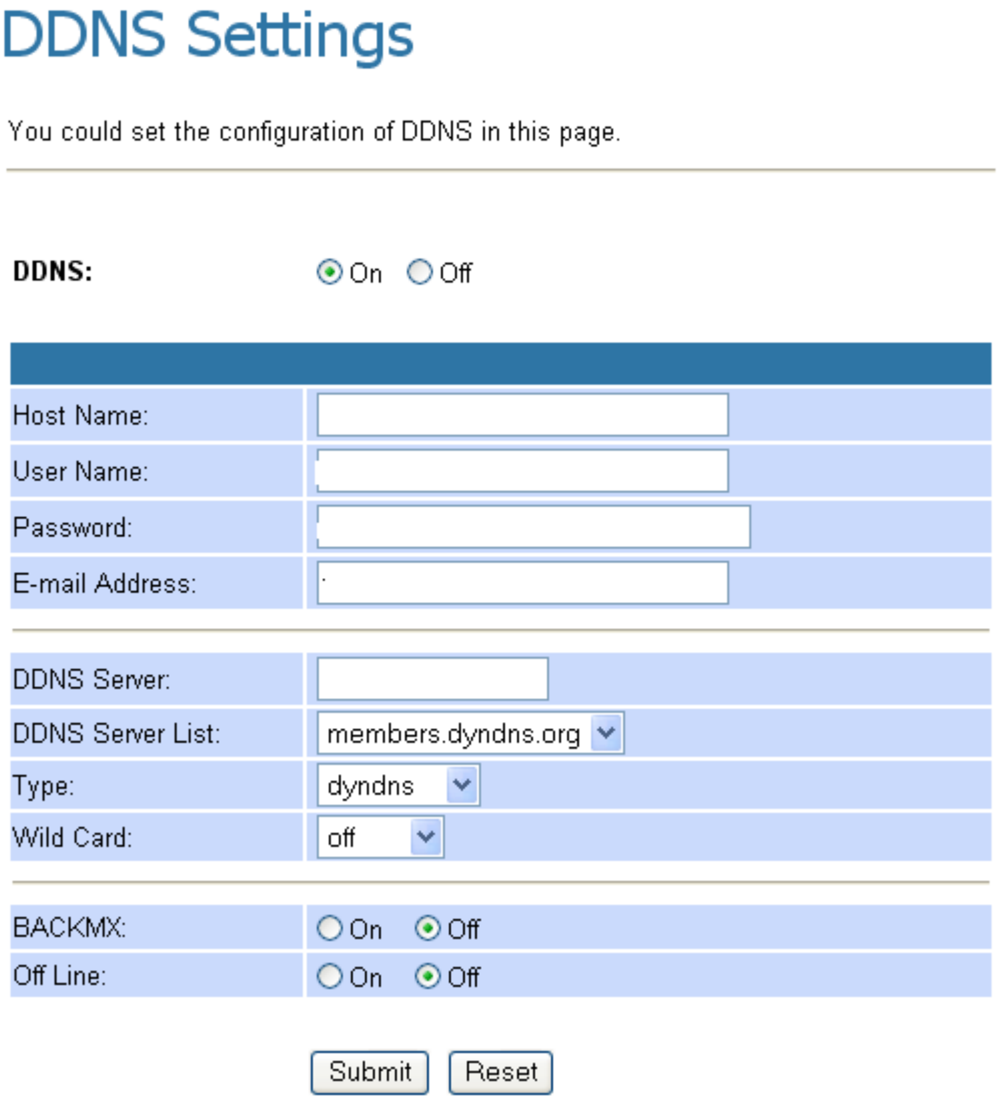


Example 1:

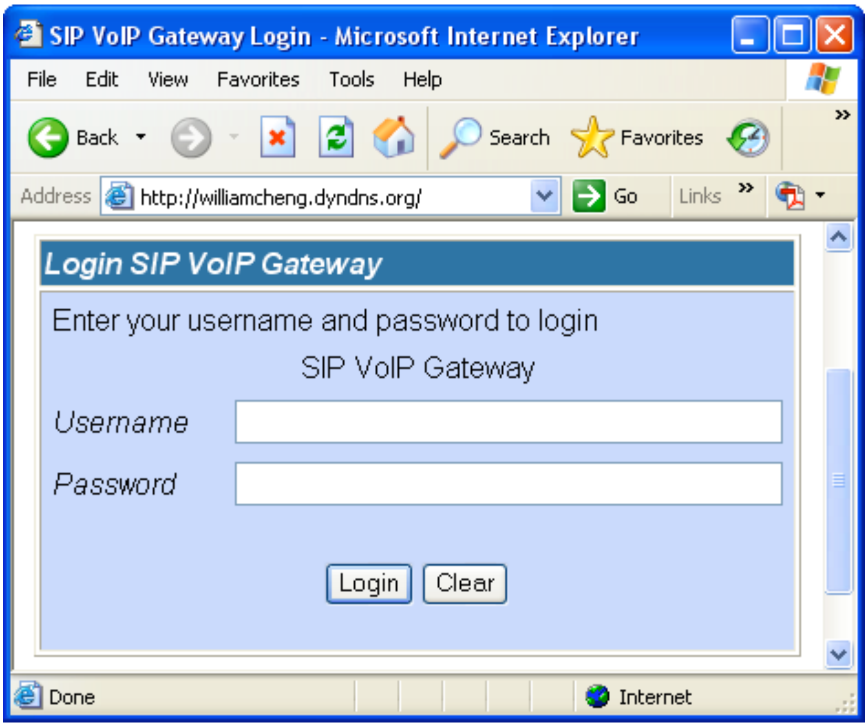
Configure the WAN to PPPoE Client and make sure you got the WAN IP Address (Public IP Address).


WAN Settings
You could configure the WAN settings in this page.
LAN Mode:
Bridge
NAT
WAN Setting
IP Type:
Fixed IP
DHCP Client
PPPoE
IP:
Mask:
Gateway:
DNS Server1:
DNS Server2:
MAC:
PPPoE Setting
User Name:
Password:
Submit
Reset

Configure the **Host Name**, **User Name**, **Password**, and **E-mail Address**.

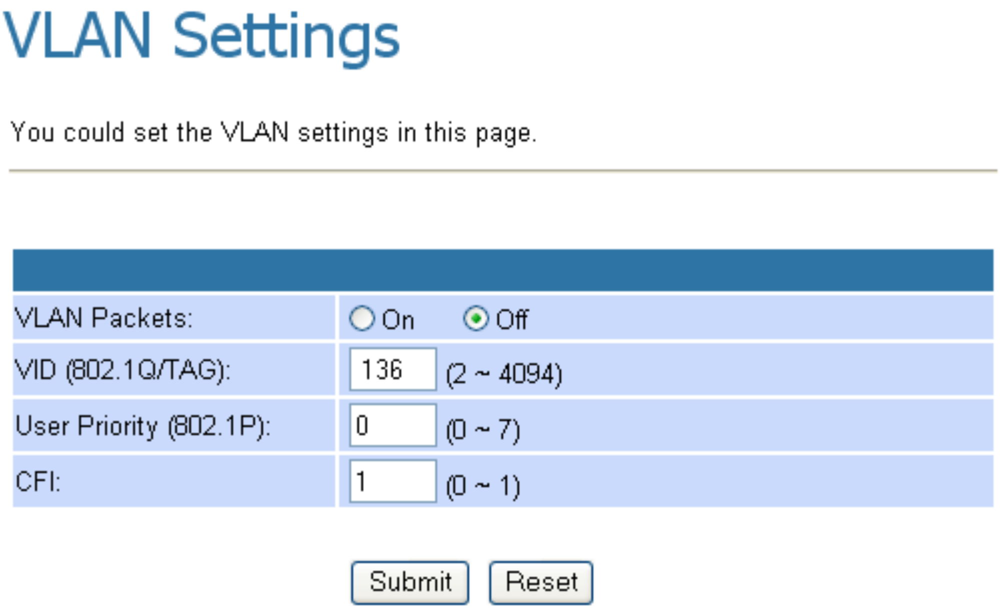


If every parameter was configured correctly, you could visit the home page of the VoIP Gateway by enter the **DDNS Host Name** as follow.



4.2.5 VLAN Settings Page

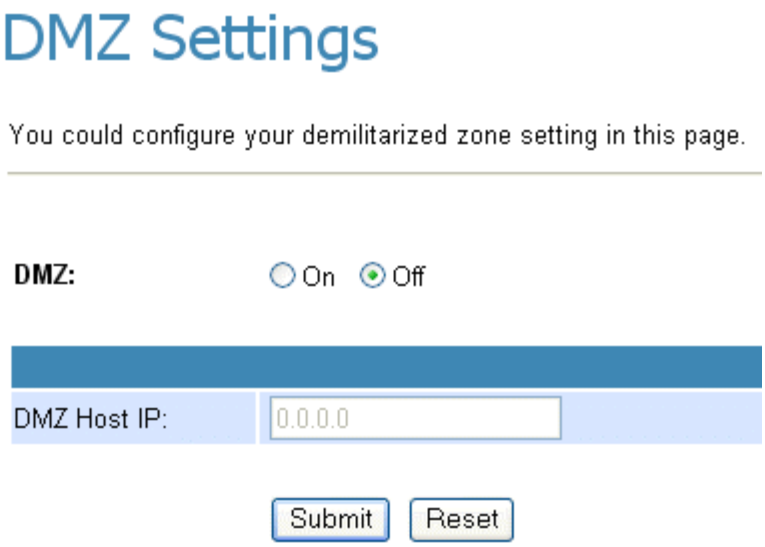
You could set the VLAN settings in this page.



VLAN Packets	Default is **Off(Disable)**. When it was **On(Enable)**, It'll enable to receive **VLAN Packets** function.
VID (802.1Q/TAG)	Default is **136**. Configure the **Virtual LAN ID (VLAN ID or VID)** for VLAN Server. The VLAN Identifier is a 12-bit field. It uniquely identifies the VLAN to which the frame belongs. The field can have a value between 2 and 4094.
User Priority (802.1P)	Default is **0**. Configure user priority. Also known as user priority, this 3-bit field refers to the IEEE 802.1p priority. The field indicates the frame priority level which can be used for the prioritization of traffic. The field can represent 8 levels (0 through 7).
CFI	The *Canonical Format Indicator* is a 1-bit field. If the value of this field is 1, the MAC address is in non-canonical format. If the value is 0, the MAC address is in canonical format.
Submit Button	When you finished the setting, please click the **Submit** button.
Reset Button	You can reset the configured parameters before you submit
Back Button	Go back to the previous web page

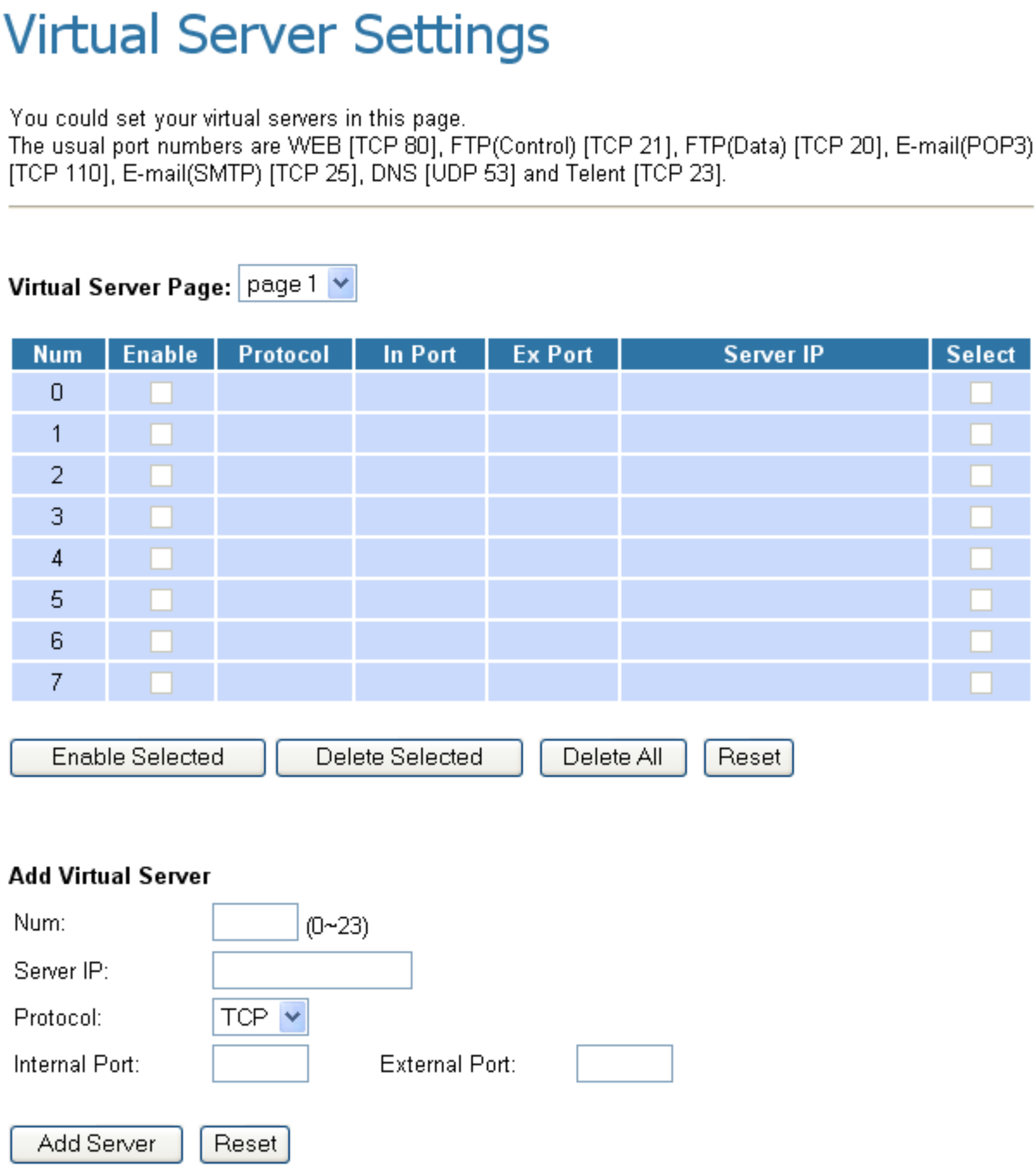
4.2.6 Virtual Server Page

You could configure your demilitarized zone setting in this page.



4.2.7 Virtual Server Page

Virtual Servers are used for port forwarding from the WAN to LAN networks. The Virtual Server Configuration page allows you to set the configuration of the Virtual Server. All UDP/TCP ports are protected from intrusion. If any specific local PCs need to be mapped to the UDP/TCP port on WAN side, please input the mappings here.
There can be up to 24 different Virtual Server Configurations.

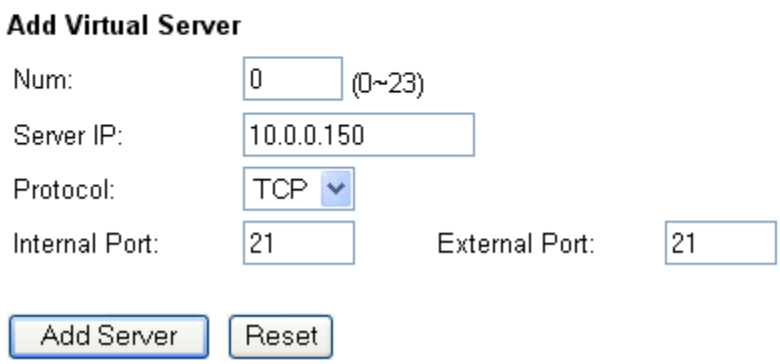


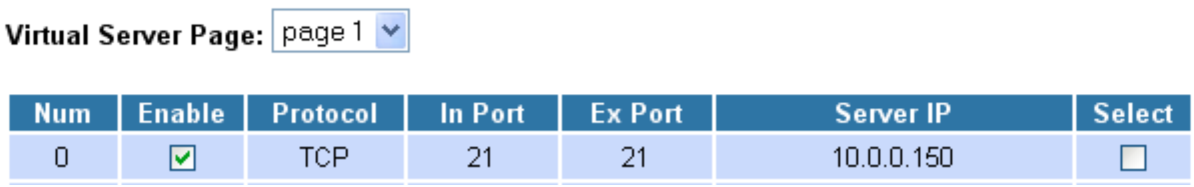
Virtual Server Page	
Virtual Server Page	Default page is Page1. There are total 3 pages from Page 1 to Page 3
Num	Show the number by sequence. There are total 24 numbers from Phone 0 to Phone 23 can be set This is the number corresponding to the Virtual Server configuration.
Enable	Default is **Disable**. When it was **Enable**, It'll enable the Virtual Server
Protocol	Select **TCP** or **UDP**.
In Port (Internal Port)	Display the Internal Port that you configured
Ex Port (External Port)	Display the External Port that you configured
Server IP	Display the private network IP address for the particular server.
Select	Select the item of the Virtual Server
Enable Selected [Button]	Enable selected item
Delete Selected [Button]	Delete selected item
Delete All [Button]	Delete all items
Reset [Button]	Reset selected item

Add Virtual Server	
Num	Enter the number corresponding to the Virtual Server configuration.
Server IP	Enter the private network IP address for the particular server.
Protocol	Select **TCP** or **UDP**.
Internal Port	Enter the port number of the Private Network (LAN or internal network). In most cases, the private port number is same as public port number. This port number cannot be seen from the WAN side.

External Port	Enter the port number of the Public Network (WAN or external network).
Add Server [Button]	Add the new Server which you configured
Reset [Button]	Reset configured items

Example 1 (FTP Server):
Num: **0**, Server IP: **10.0.0.150**, Protocol: **TCP**, Internal Port: **21**, External Port: **21**





Other people can visit your FTP Server by entering the WAN IP Address of VoIP Gateway and then the VoIP Gateway will re-directly it to your LAN IP 10.0.0.150.

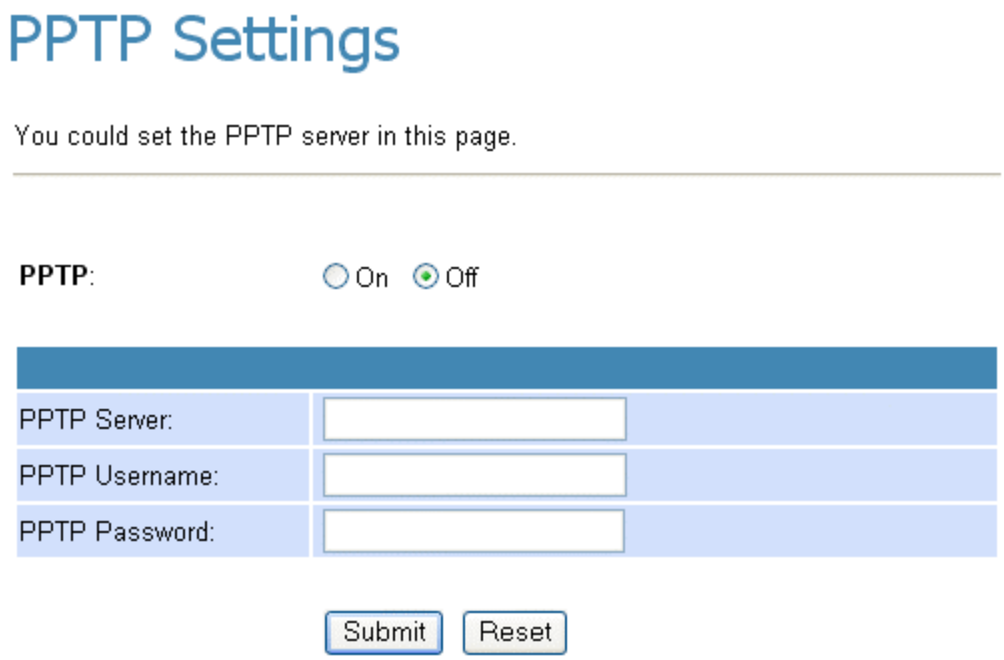
Table 4-3. Well Known TCP/UDP Ports

Port	Protocol	UDP	TCP
20	File Transfer Protocol (FTP) Data		X
21	FTP Commands		X
23	Telnet		X
25	SMTP		X
43	Whois		X
53	Domain Name System (DNS)	X	X
69	Trivial File Transfer Protocol (TFTP)	X	
70	Gopher		X
79	Finger		X
80	HTTP		X
110	POP3		X
111	SUN Remote Procedure Call (RPC)	X	
115	SFTP		X
119	Network News Transfer Protocol (NNTP)		X
123	Network Time Protocol (NTP)		X
144	News	X	X
161	Simple Network Management Protocol (SNMP)	X	

162	SNMP traps	X	
179	Border Gateway Protocol (BGP)		X
443	Secure HTTP (HTTPS)		X
513	rlogin		X
514	rexec		X
517	talk	X	X
518	ntalk	X	X
520	Routing Information Protocol (RIP)	X	
1701	Layer 2 Tunneling Protocol (L2TP)	X	
2000	Open Windows	X	X
2049	Network File System (NFS)		X
6000	X11	X	X

4.2.8 PPTP Settings Page

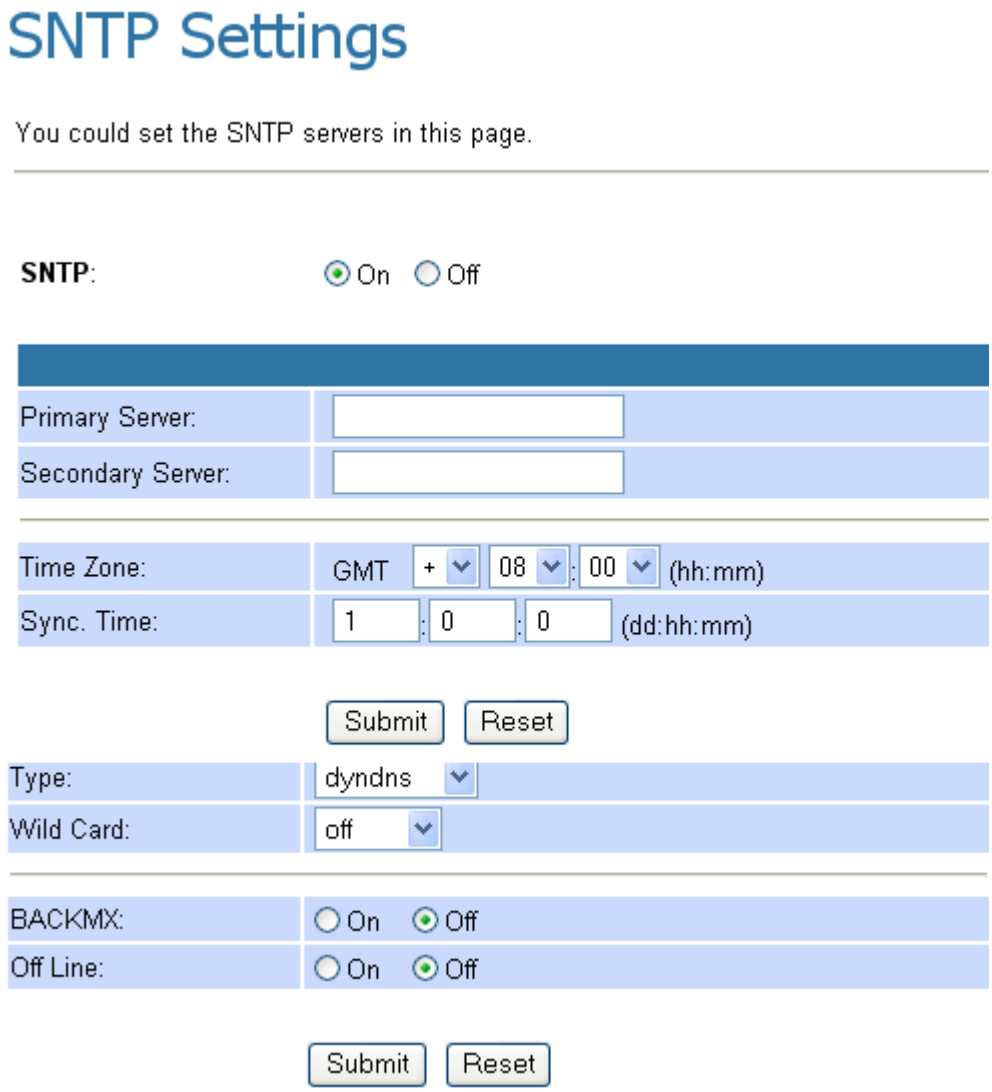
A VPN is a private network of computers that uses the public Internet to connect some nodes. Because the Internet is essentially an open network, the Point-to-Point Tunneling Protocol (PPTP) is used to ensure that messages transmitted from one VPN node to another are secure. With PPTP, users can dial in to their corporate network via the Internet.



PPTP Settings Page	
PPTP	Default is **Off**. When it was **On**, It'll enable the PPTP client.
PPTP Server	Enter the IP Address of PPTP Server.
PPTP Username	Enter the Username of PPTP client.
PPTP Password	Enter the Pasword of PPTP client.
Submit Button	When you finished the setting, please click the **Submit** button.
Reset Button	You can reset the configured parameters before you submit

4.2.9 SNTP Settings Page

You can setup the primary and second SNTP Server IP Address, to get the date/time information. Also you can base on your location to set the Time Zone, and how long need to synchronize again. When you finished the setting, please click the Submit button.



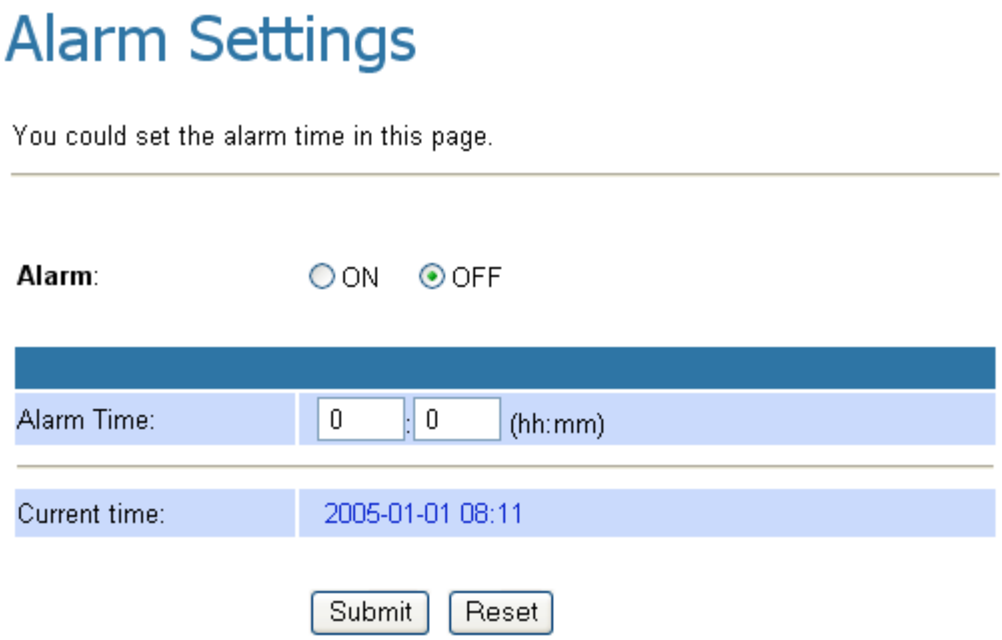
If synchronization is enabled, your VoIP Gateway clock is synchronized with an Internet time server once a day. However, if you don't have a continuous Internet connection through a cable modem or DSL modem, the automatic synchronization might not always occur.

If time synchronization fails, it might be for one of the following reasons:

- You are not connected to the Internet. Establish an Internet connection before you attempt to synchronize your clock.
- Your personal or network firewall prevents clock synchronization. Most corporate and organizational firewalls will block time synchronization
- The Internet time server is too busy or is temporarily unavailable. If this is the case, try synchronizing your clock later, or update it manually by powering off and then on the VoIP Gateway. You can also try using a different time server.
- The time shown on your VoIP Gateway is too different from the current time on the Internet time server. Internet time servers might not synchronize your clock if your VoIP Gateway's time is off by more than 15 hours.

4.2.10 Alarm Settings Page

Provide the alarm function.
The alarm will sound when it reached the **Alarm Time** that you configured.

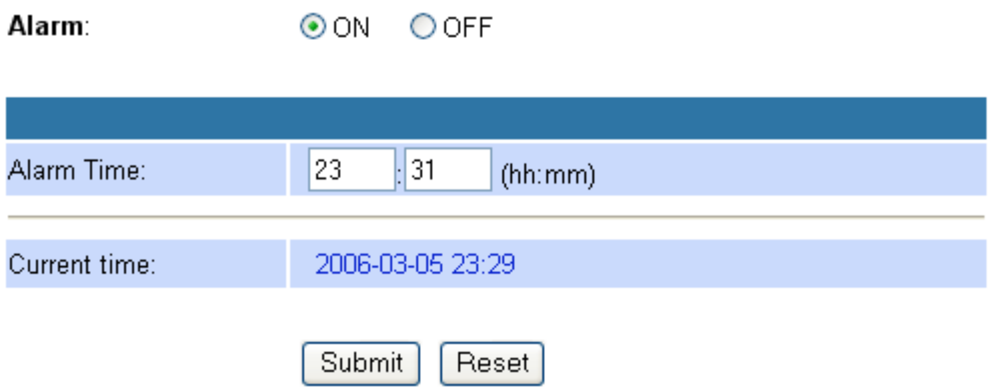


Alarm	Default is **OFF (Disable)**. When it was **ON(Enable)**, It'll enable the **Alarm** function.
Alarm Time	Default is **0:0 (hh:mm)**. Set the **Alarm Time**. (24 hours format, hh:mm)
Current time	It's the current time of the VoIP Gateway.
Submit Button	When you finished the setting, please click the **Submit** button.
Reset Button	You can reset the configured parameters before you submit

Example 1: Alarm: **ON**, Alarm Time: **8:1(hh:mm)**

The alarm will sound when it reached the current time **08:01**.

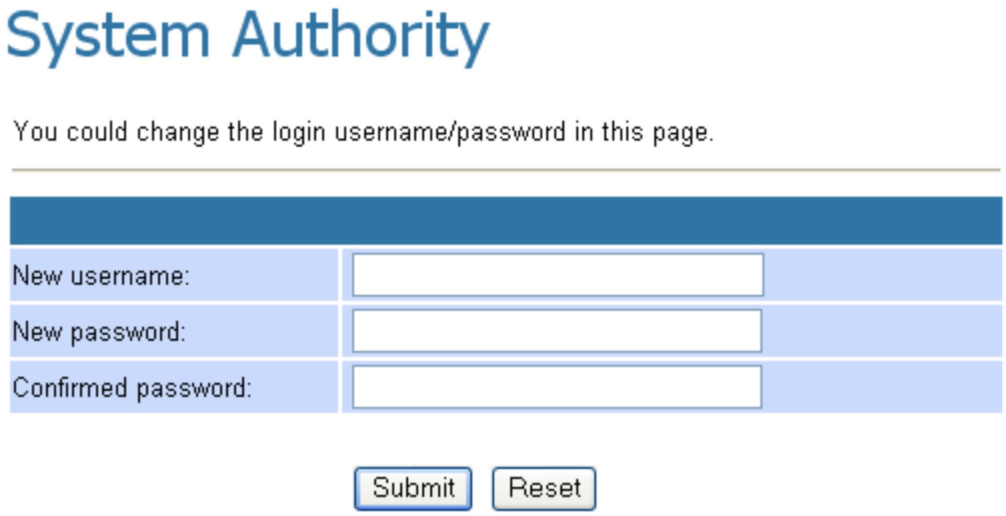
Example 2: Alarm: **ON**, Alarm Time: **23:31(hh:mm)**



The alarm will sound when it reached the current time **23:31**.

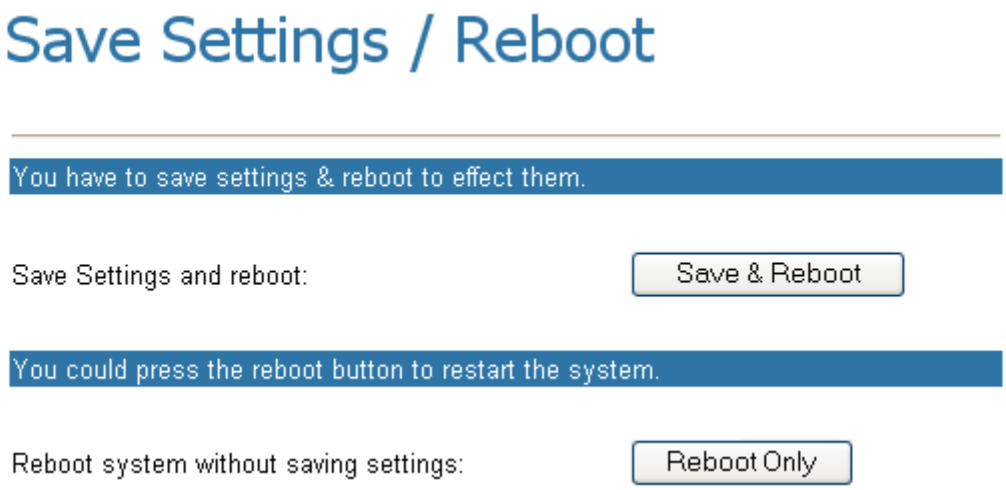
4.2.11 System Authority Page

In System Authority you can change your login name and password.



4.2.12 Save Settings/Reboot Page

In Save Settings/Reboot you can save the changes you have done or reboot only. If you want to use new setting in the VoIP Gateway, You have to click the **Save & Reboot** button. After you click the **Save & Reboot** button, the VoIP Gateway will automatically restart and the new setting will effect.
If you want to reboot the VoIP Gateway, You have to click the **Reboot Only** button. After you click the **Reboot Only** button, the VoIP Gateway will automatically restart.



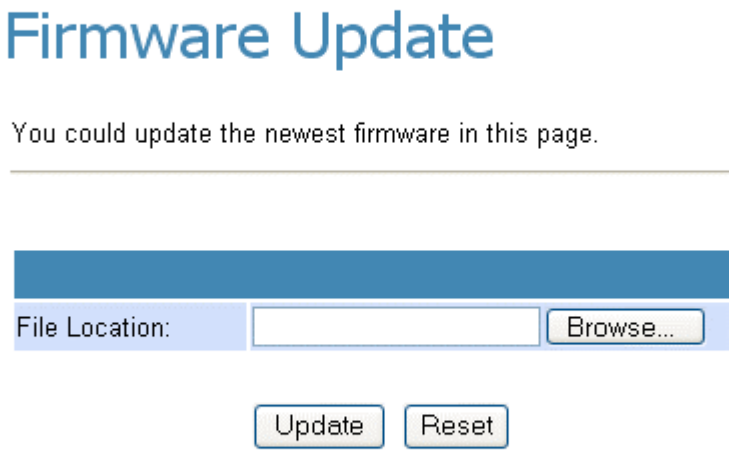
4.3 System Page

4.3.1 Reset factory default Page

In Reset to Factory Default setting you can restore the VoIP Gateway to factory default in this page. You can just click the Restore button, then the VoIP Gateway will restore to default and automatically restart again.

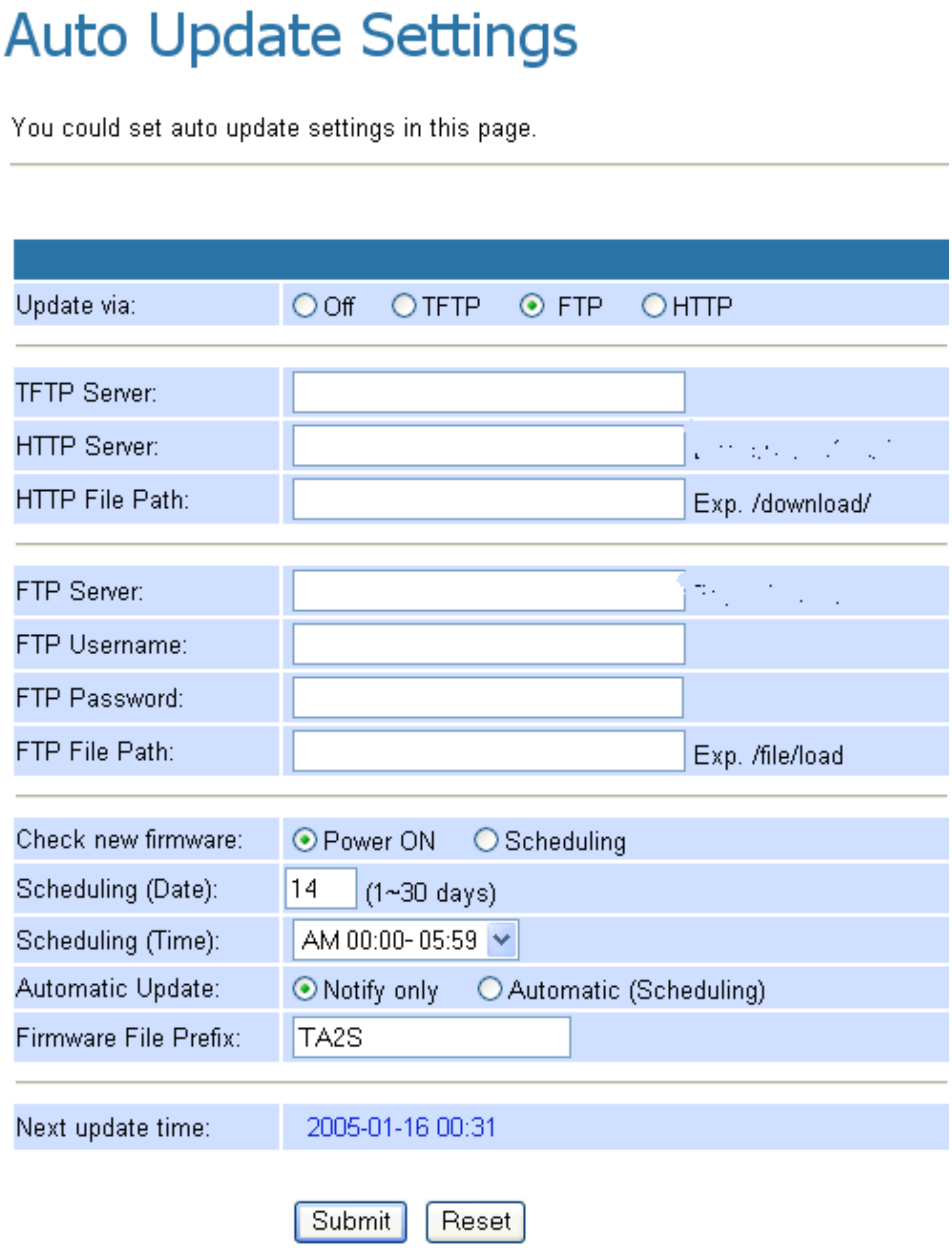
4.3.2 Firmware Update Page

In Update you can update the VoIP Gateway's firmware to the new one or do the factory reset to let the VoIP Gateway back to default setting.
Click the **"Browse"** button in the right side of the File Location or you can type the correct path and the filename in File Location blank and then click the **Update** button.



4.3.3 Auto Update Page

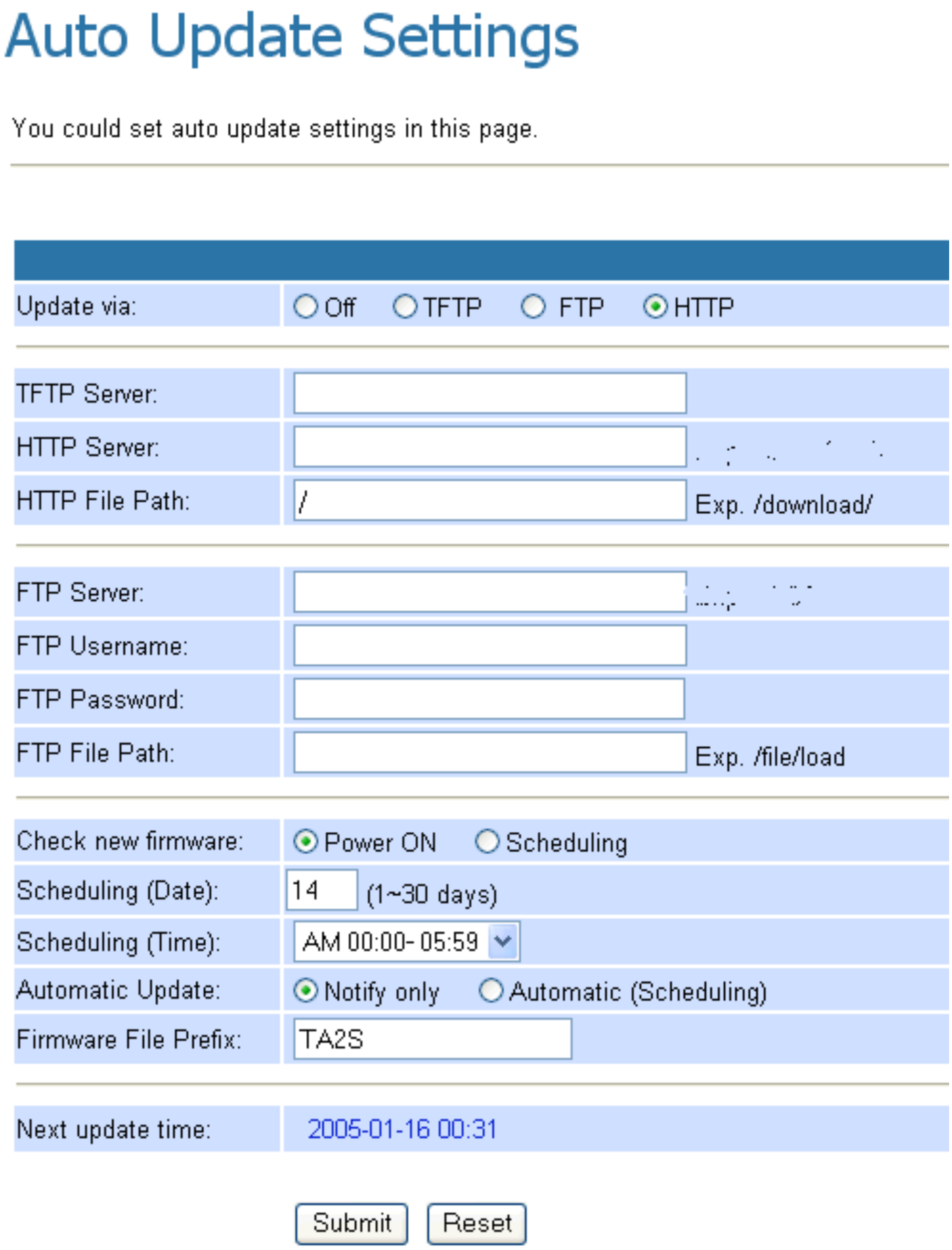
To have the firmware up to date by powering on the VoIP Gateway or Scheduling.



Update via	Default is **OFF (Disable)**. When it was **TFTP/FTP/HTTP(Enable)**, it'll enable the **auto update** function and request from the **TFTP/FTP/HTTP Server**.
TFTP Server	Enter IP or Domain Name of **TFTP Server**.
HTTP Server	Enter IP or Domain Name of **HTTP Server**.
HTTP Path	Enter File Path where the file is.
FTP Server	Enter IP or Domain Name of **FTP Server**.
FTP Username	Enter Username which provided by **FTP Server**.
FTP Password	Enter Password which provided by **FTP Server**.
File Path	Enter File Path where the file is.
Check new firmware	**Power ON**: It'll check if there is a new firmware on the **TFTP/FTP/HTTP Server** by powering on the VoIP Gateway. **Scheduling**: It'll check if there is a new firmware on the **TFTP/FTP/HTTP Server** by scheduling.
Scheduling (Date)	Default is **14**. It'll check if there is a new firmware on the **TFTP/FTP/HTTP Server** periodically. The range of the **Scheduling Date** is **1 - 30**.
Scheduling (Time)	Default is **AM 00:00- 05:59**. It'll check if there is new firmware on the **TFTP/FTP/HTTP Server** periodically. There are four **Scheduling Time**: ***AM 00:00- 05:59***, ***AM 06:00- 11:59***, ***PM 12:00- 17:59***, ***PM 18:00- 23:59***
Automatic Update	**Notify only**: When there is a newer firmware, it will only notify by "***BEEP BEEP BEEP***" you when you pick up the phone. **Automatic (Scheduling)**: When there is a newer firmware, it will update the firmware automatically.
Firmware File Prefix	The file prefix of the firmware
Next update time	It's the next update or check time.
Submit Button	When you finished the setting, please click the **Submit** button.
Reset Button	You can reset the configured parameters before you submit

Example 1: HTTP - Firmware update by notification when powered on

Auto Update Settings
Update via: **HTTP**
HTTP Server: **192.168.10.100**
HTTP Path: **/**
Check new Firmware: **Power ON**
Automatic Update: **Notify only**
Firmware File Prefix**: TA2S**



RULE of AUTO UPDATE:

Every time when you power on the VoIP Gateway, it'll notify you by "***BEEP BEEP BEEP***" there is an up to date firmware available on **HTTP Server** after you pick up the phone and you can update the firmware manually.

Create the Auto Update files on HTTP Server:

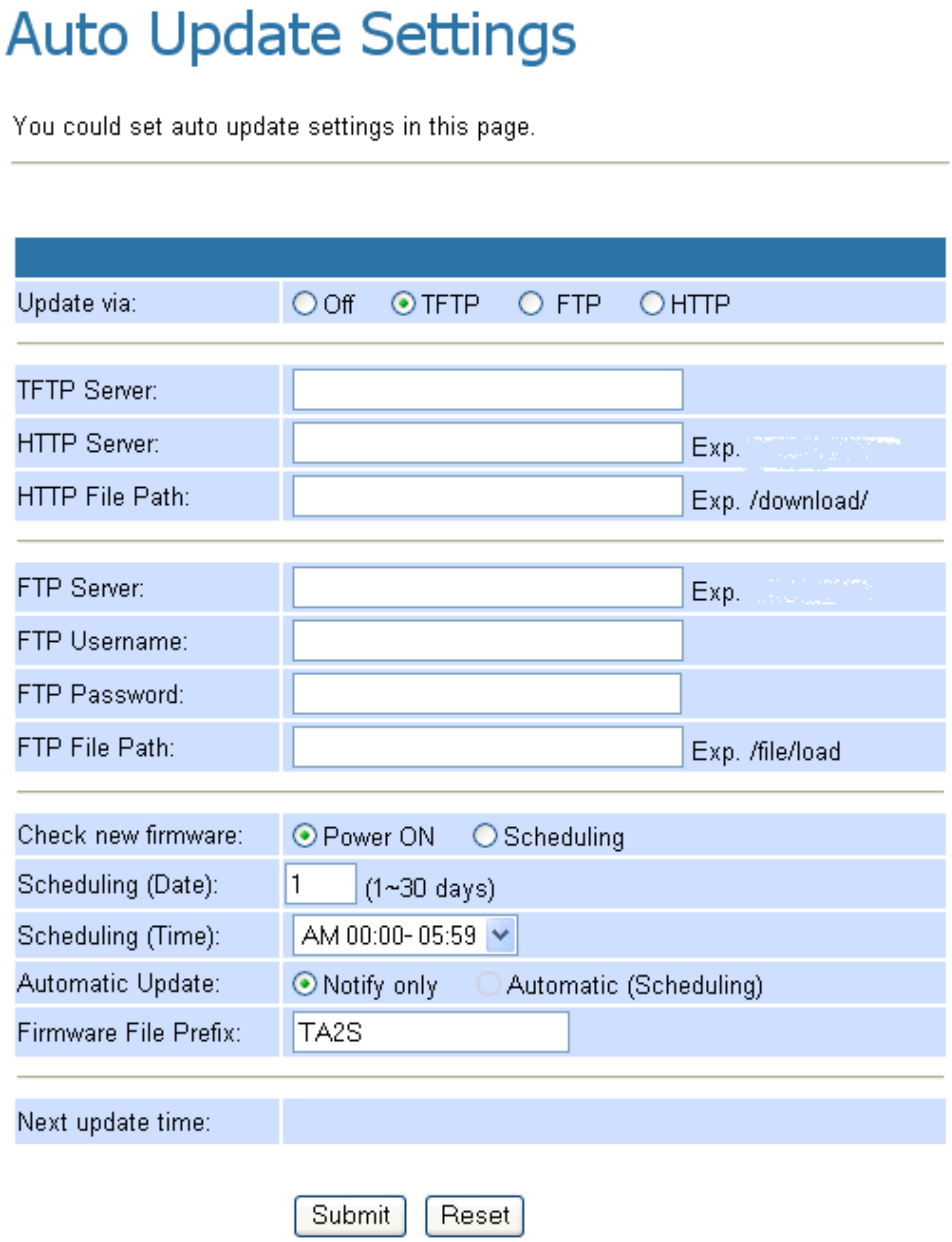
1. To check the current firmware version of the VoIP Gateway:
   a. **Telnet 10.0.0.2**
   b. Enter the login name **admin** and password **admin**.
   c. **ver**
   d. You will get the firmware version as follow:
      Firmware Version: V701240
2. Create a ***TA2S_ver.dat*** due to format of the file is **Firmware File Prefix_ *ver.dat*** and edit the content as follow:
   Version: **701250** NAME:***TA2S_***
3. Change the new firmware ***voip.gz*** to **TA2S_701250*.gz***
4. Put the **TA2S_701250*.gz*** and ***TA2S_ver.dat*** in Server

AUTO UPDATE PROCEDURES:

1. Power on the VoIP Gateway and it will check if there is any update firmware is newer one on the Server. When there is a newer firmware, it will only notify you by "***BEEP BEEP BEEP***" after you pick up the phone.
2. Please press ***#190#*** and then hang up the phone to unlock the special key on keypad.
3. Pick up the phone and then press ***#160#*** and then hang up the phone to have VoIP Gateway to update the firmware immediately.
4. It takes around **3 minutes** for updating the new firmware and the SIP LED starts blinking while updating the firmware.
5. Once the SIP LED stop blinking, please power off and then power on the VoIP Gateway to active the new firmware.

Example 2: TFTP - Firmware update by notification when powered on

Auto Update Settings
Update via: **TFTP**
TFTP Server: **192.168.10.100**
Check new Firmware: **Power ON**
Automatic Update: **Notify only**
Firmware File Prefix**: TA2S**



RULE of AUTO UPDATE:

Every time when you power on the VoIP Gateway, it'll notify you by "***BEEP BEEP BEEP***" there is an up to date firmware available on **TFTP Server** after you pick up the phone and you can update the firmware manually.

Create the Auto Update files on TFTP Server:

1. To check the current firmware version of the VoIP Gateway:
   a. **Telnet 10.0.0.2**
   b. Enter the login name **admin** and password **admin**.
   c. **ver**
   d. You will get the firmware version as follow:
      Firmware Version: V701240
2. Create a ***TA2S_ver.dat*** due to format of the file is **Firmware File Prefix_ *ver.dat*** and edit the content as follow:
   Version: **701250** NAME: ***TA2S_***
3. Change the new firmware ***voip.gz*** to **TA2S_701250*.gz***
4. Put the **TA2S_701250*.gz*** and ***TA2S_ver.dat*** in Server

AUTO UPDATE PROCEDURES:

1. Every time when power on the VoIP Gateway reaches the, it'll check if there is an up to date firmware available on **TFTP Server** and update the firmware manually. When there is a newer firmware, it will only notify you by "***BEEP BEEP BEEP***" after you pick up the phone.
2. Please press ***#190#*** and then hang up the phone to unlock the special key on keypad.
3. Pick up the phone and then press ***#160#*** and then hang up the phone to have VoIP Gateway to update the firmware immediately.
4. It takes around **3 minutes** for updating the new firmware and the SIP LED starts blinking while updating the firmware.
5. Once the SIP LED stop blinking and the VoIP Gateway will reboot itself to active the new firmware.

Example 3: FTP - Firmware update by notification when powered on

Auto Update Settings

Update via: **FTP**

FTP Server: **192.168.10.100**

FTP Username: **1234**

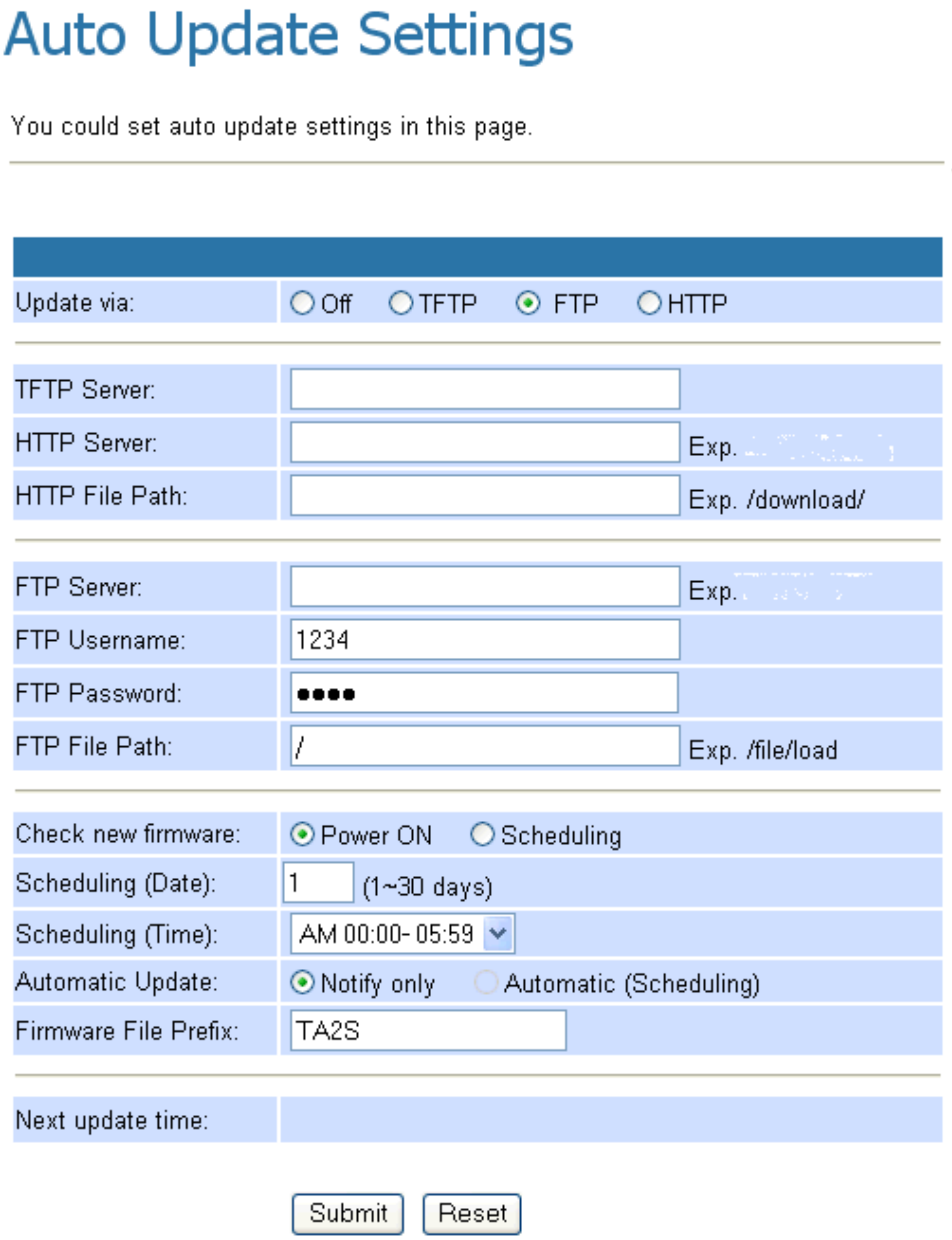
FTP Password: **1234**

File Path: **/**

Check new Firmware: **Power ON**

Automatic Update: **Notify only**

Firmware File Prefix**: TA2S**



RULE of AUTO UPDATE:

Every time when you power on the VoIP Gateway, it'll notify you by "***BEEP BEEP BEEP***" there is an up to date firmware available on **FTP Server** after you pick up the phone and you can update the firmware manually.

Create the Auto Update files on FTP Server:

1. To check the current firmware version of the VoIP Gateway:
   a. **Telnet 10.0.0.2**
   b. Enter the login name **admin** and password **admin**.
   c. **ver**
   d. You will get the firmware version as follow:
      Firmware Version: V701240
2. Create a ***TA2S_ver.dat*** due to format of the file is **Firmware File Prefix_ *ver.dat*** and edit the content as follow:
   Version: **701250** NAME: ***TA2S_***
3. Change the new firmware ***voip.gz*** to **TA2S_701250*.gz***
4. Put the **TA2S_701250*.gz*** and ***TA2S_ver.dat*** in Server

AUTO UPDATE PROCEDURES:

1. Every time when power on the VoIP Gateway reaches the, it'll check if there is an up to date firmware available on **FTP Server** and update the firmware manually. When there is a newer firmware, it will only notify you by "***BEEP BEEP BEEP***" after you pick up the phone.
2. Please press ***#190#*** and then hang up the phone to unlock the special key on keypad.
3. Pick up the phone and then press ***#160#*** and then hang up the phone to have VoIP Gateway to update the firmware immediately.
4. It takes around **3 minutes** for updating the new firmware and the SIP LED starts blinking while updating the firmware.
5. Once the SIP LED stop blinking and the VoIP Gateway will reboot itself to active the new firmware.

Example 3: FTP - Firmware update by notification when reached the Scheduling Date and Time

Auto Update Settings

Update via: **FTP**

FTP Server:

FTP Username: **1234**

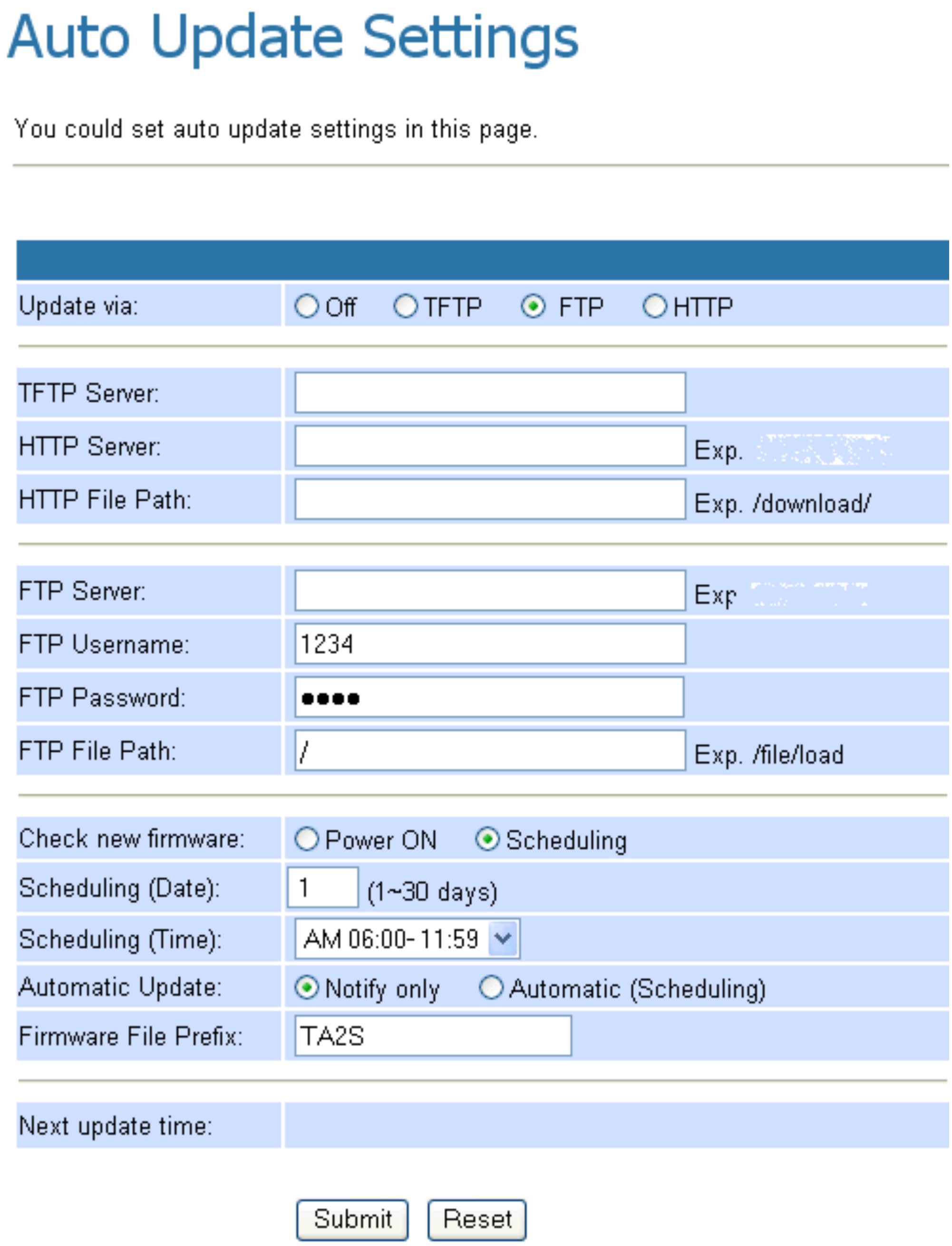
FTP Password: **1234**

File Path: **/**

Check new Firmware: **Scheduling**

Automatic Update: **Notify only**

Firmware File Prefix**: TA2S**



RULE of AUTO UPDATE:
It'll update its VoIP firmware to the latest one from **FTP Server** automatically when it reaches the **Scheduling Date** and **Scheduling Time (Next update time)**.

Create the Auto Update files on FTP Server:
1. To check the current firmware version of the VoIP Gateway:
   a. **Telnet 10.0.0.2**
   b. Enter the login name **admin** and password **admin**.

c. **ver**
d. You will get the firmware version as follow:
Firmware Version: V701240

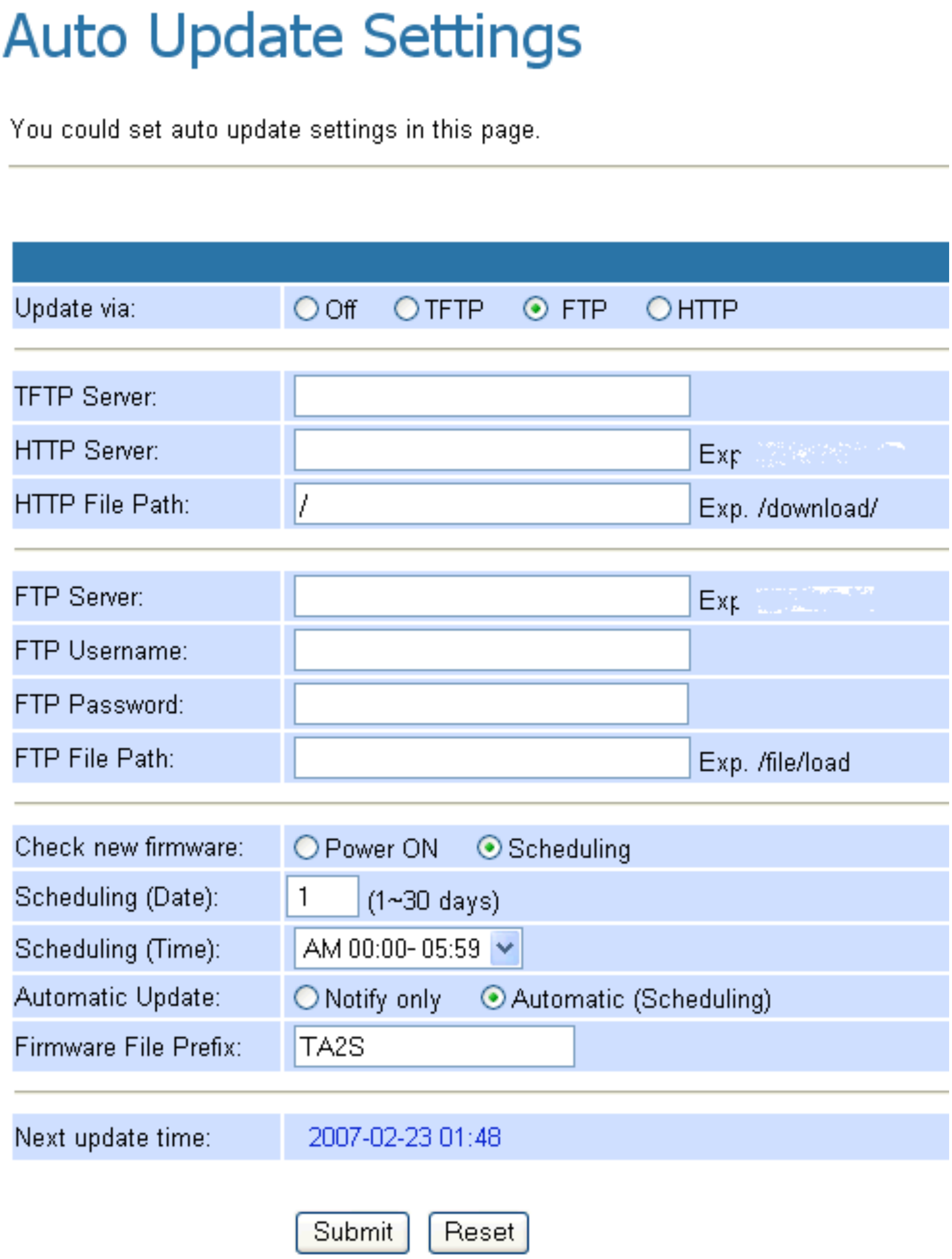
2. Create a ***TA2S_ver.dat*** due to format of the file is **Firmware File Prefix_ *ver.dat*** and edit the content as follow:
Version: **701250** NAME: ***TA2S_***
3. Change the new firmware ***voip.gz*** to **TA2S_701250*.gz***
4. Put the **TA2S_701250*.gz*** and ***TA2S_ver.dat*** in Server

AUTO UPDATE PROCEDURES:
Every time when the VoIP Gateway reaches the scheduling date and time, it'll notify you by "***BEEP BEEP BEEP***" there is an up to date firmware available on **FTP Server** after you pick up the phone and you can update the firmware manually.

Be noted:
1. If the VoIP Gateway is powered off and passed the **Next update time**, it'll not update the firmware after you power on the VoIP Gateway. It'll only update when the VoIP Gateway is power on and reaches **Next update time**.
2. If you are on the phone and have a conversation to others by VoIP and the **Next update time** is passing, it'll update the firmware immediately after you hang up the phone.

Example 3: Firmware update by notification when reached the Scheduling Date and Time

Auto Update Settings
Update via: **HTTP**
HTTP Server:
HTTP Path: ***/***
Check new Firmware: **Scheduling**
Automatic Update: **Automatic (Scheduling)**
Firmware File Prefix**: TA2S**



RULE of AUTO UPDATE:
It'll update its firmware to the latest one from **HTTP Server** automatically when it reaches the **Scheduling Date** and **Scheduling Time (Next update time)**.

Create the Auto Update files on HTTP Server:

1. To check the current firmware version of the VoIP Gateway:
   a. **Telnet 10.0.0.2**
   b. Enter the login name **admin** and password **admin**.
   c. **ver**
   d. You will get the firmware version as follow:
      Firmware Version: V701240
2. Create a ***TA2S_ver.dat*** due to format of the file is **Firmware File Prefix_ *ver.dat*** and edit the content as follow:
   Version: **701250** NAME: ***TA2S_***
3. Change the new firmware ***voip.gz*** to **TA2S_701250*.gz***
4. Put the **TA2S_701250*.gz*** and ***TA2S_ver.dat*** in Server

AUTO UPDATE PROCEDURES:

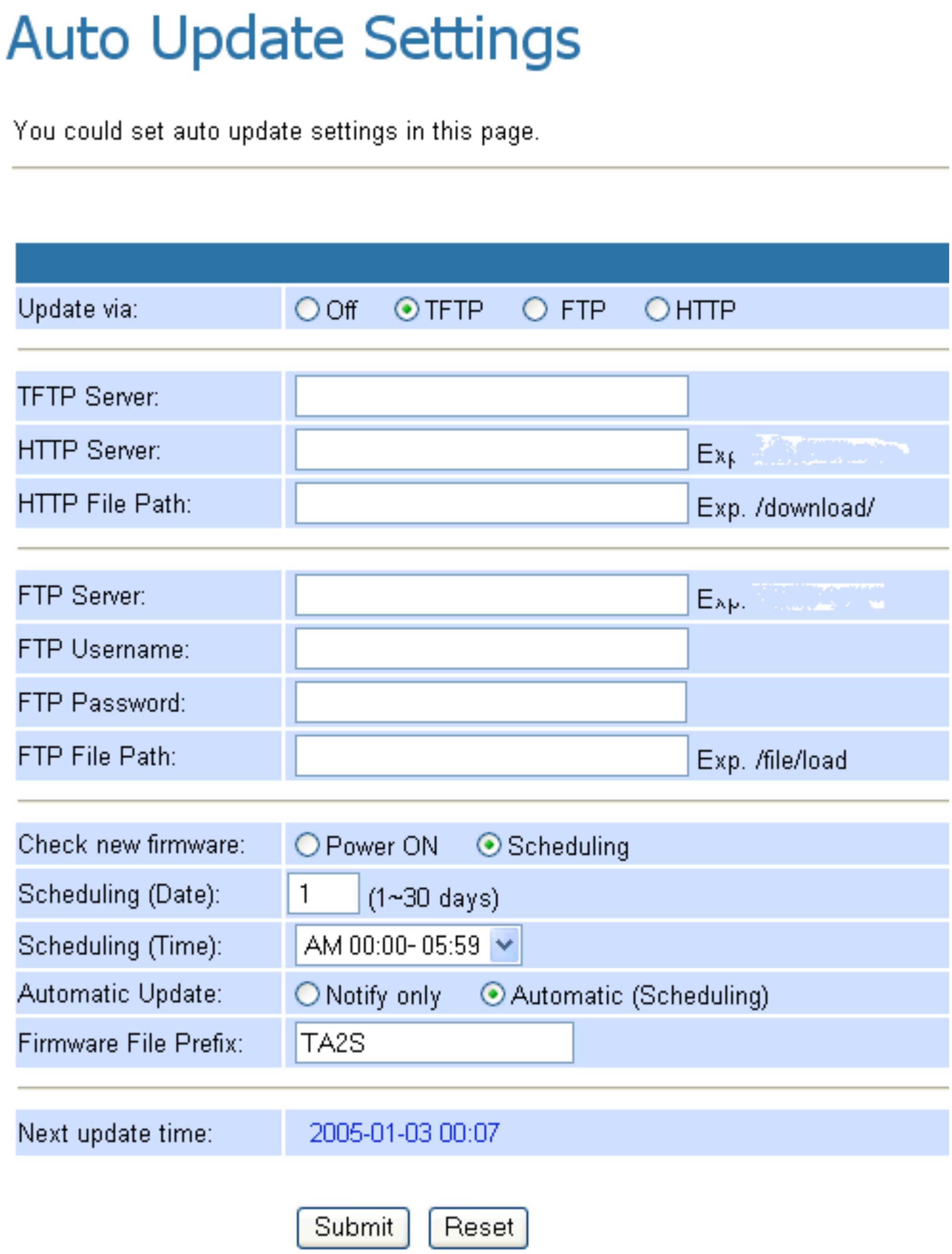
1. Every time when the VoIP Gateway reaches the Scheduling date and time, it'll check if there is an up to date firmware available on **HTTP Server** and update the firmware automatically.
2. It takes around **3 minutes** for updating the new firmware and the SIP LED starts blinking while updating the firmware.
3. Once the SIP LED stop blinking, please power off and then power on the VoIP Gateway to active the new firmware.

Be noted:

1. If the VoIP Gateway is powered off and passed the **Next update time**, it'll not update the firmware after you power on the VoIP Gateway. It'll only update when the VoIP Gateway is power on and reaches **Next update time**.
2. If you are on the phone and have a conversation to others by VoIP and the **Next update time** is passing, it'll update the firmware immediately after you hang up the phone.

Example 4: Firmware update automatically when reached the Scheduling Date and Time

Auto Configuration Settings
Update via: **TFTP**
TFTP Server:
Check new Firmware: **Scheduling**
Automatic Update: **Automatic (Scheduling)**
Firmware File Prefix**: TA2S**



RULE of AUTO UPDATE:
It'll update its firmware to the latest one from **TFTP Server** automatically when it reaches the **Scheduling Date** and **Scheduling Time (Next update time)**.

Create the Auto Update files on TFTP Server:

1. To check the current firmware version of the VoIP Gateway:
   a. **Telnet 10.0.0.2**
   b. Enter the login name **admin** and password **admin**.
   c. **ver**
   d. You will get the firmware version as follow:
      Firmware Version: V701240
2. Create a ***TA2S_ver.dat*** due to format of the file is **Firmware File Prefix_ *ver.dat*** and edit the content as follow:
   Version: **701250** NAME: ***TA2S_***
3. Change the new firmware ***voip.gz*** to **TA2S_701250*.gz***
4. Put the **TA2S_701250*.gz*** and ***TA2S_ver.dat*** in Server

AUTO UPDATE PROCEDURES:

6. Every time when the VoIP Gateway reaches the Scheduling date and time, it'll check if there is an up to date firmware available on **TFTP Server** and update the firmware automatically.
7. It takes around **3 minutes** for updating the new firmware and the SIP LED starts blinking while updating the firmware.
8. Once the SIP LED stop blinking and the VoIP Gateway will reboot itself to active the new firmware.

Be noted:

1. If the VoIP Gateway is powered off and passed the **Next update time**, it'll not update the firmware after you power on the VoIP Gateway. It'll only update when the VoIP Gateway is power on and reaches **Next update time**.
2. If you are on the phone and have a conversation to others by VoIP and the **Next update time** is passing, it'll update the firmware immediately after you hang up the phone.

Example 5:

Auto Configuration Settings (Firmware update by Scheduling)

Update via: **FTP**

Update via: **FTP**

FTP Server:

FTP Username: **1234**

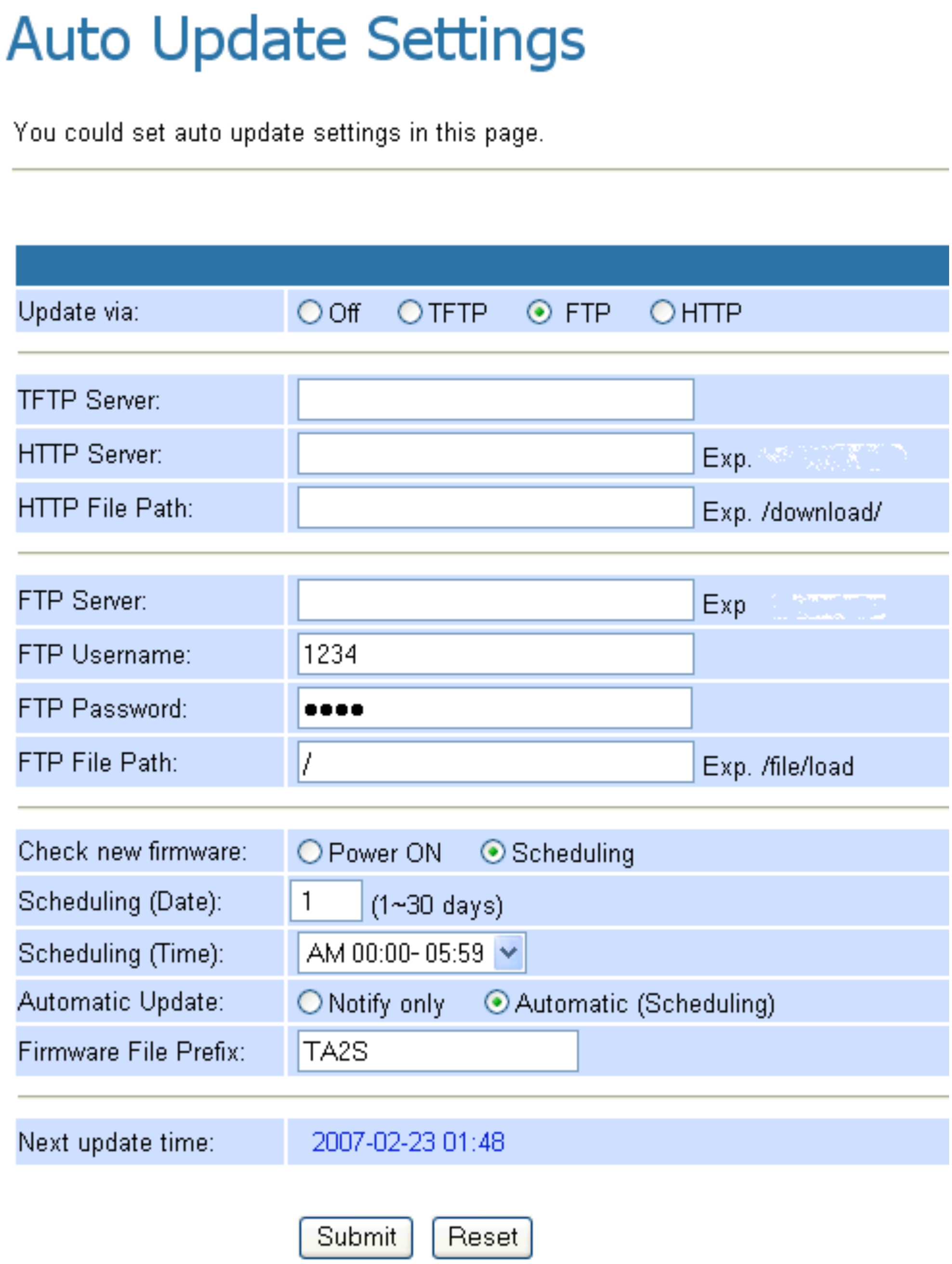
FTP Password: **1234**

File Path: **/**

Check new Firmware: **Scheduling**

Automatic Update: **Automatic (Scheduling)**

Firmware File Prefix**: TA2S**



RULE of AUTO UPDATE:

It'll update its firmware to the latest one from **FTP Server** automatically when it reaches the **Scheduling Date** and **Scheduling Time (Next update time)**.

Create the Auto Update files on FTP Server:

1. To check the current firmware version of the VoIP Gateway:
   a. **Telnet 10.0.0.2**

b. Enter the login name **admin** and password **admin**.

c. **ver**

d. You will get the firmware version as follow:

Firmware Version: V701240

2. Create a ***TA2S_ver.dat*** due to format of the file is **Firmware File Prefix_ *ver.dat*** and edit the content as follow:

Version: **701250** NAME: ***TA2S_***

3. Change the new firmware ***voip.gz*** to **TA2S_701250*.gz***

4. Put the **TA2S_701250*.gz*** and ***TA2S_ver.dat*** in Server

AUTO UPDATE PROCEDURES:

1. Every time when the VoIP Gateway reaches the Scheduling date and time, it'll check if there is an up to date firmware available on **FTP Server** and update the firmware automatically.
2. It takes around **3 minutes** for updating the new firmware and the SIP LED starts blinking while updating the firmware.
3. Once the SIP LED stop blinking and the VoIP Gateway will reboot itself to active the new firmware.

Be noted:

1. If the VoIP Gateway is powered off and passed the **Next update time**, it'll not update the firmware after you power on the VoIP Gateway. It'll only update when the VoIP Gateway is power on and reaches **Next update time**.
2. If you are on the phone and have a conversation to others by VoIP and the **Next update time** is passing, it'll update the firmware immediately after you hang up the phone.

5. IVR Interface for VoIP Gateway

You can use the PSTN phone to configure the VoIP Gateway. Please follow the instruction to configure your VoIP Gateway.

Group	IVR Action	IVR Menu Choice	Parameter(s)	Notes:
Function	Reboot	#195#	None	After you hear "Option Successful," hang-up. The system will reboot automatically.
Function	Factory Reset	#198#	None	System will automatically Reboot. WARNING: ALL "User-Changeable" NONDEFAULT SETTINGS WILL BE LOST! This will include network and service provider data.
Info	Check IP Address	#120#	None	IVR will report the LAN port IP address
Info	Check IP Type	#121#	None	IVR will report the WAN Port DHCP is enabled or disabled.
Info	Check the Phone Number	#122#	None	IVR will report current in use VoIP number
Info	Check Network Mask	#123#	None	IVR will report the WAN Port network mask
Info	Check Gateway IP Address	#124#	None	IVR will announce the current gateway IP address of the VoIP Gateway
Info	Check Primary DNS Server Setting	#125#	None	IVR will announce the current setting in the Primary DNS field.
Info	Check IP Address	#126#	None	IVR will report the WAN port IP address
Info	Check Firmware Version	#128#	None	IVR will announce the version of the firmware running on the VoIP Gateway.

6. How to make a phone call

When your VoIP Gateway is configured properly, you can make a phone call to your friend in the same Service provider. Please make sure all the cables are connected properly, like PSTN Line cable, Phone cable, Ethernet cable, Power cable.
If you want to make a phone VoIP call, you can dial the phone number and press "#" button to start to dial the phone number.

6.1 Dial a PSTN Phone call

Default the VoIP Gateway is set in VoIP Phone Call mode. If you want to make a phone PSTN call, you can press **"0*"**, dial the **phone number** and press **"#"** button to start to dial the phone number.
For example: **0* + phone number + #**

6.2 Dial a VoIP Phone call

When your VoIP Gateway is configured properly, you can make a phone call to your friend in the same Service provider.
If you want to make a phone call, you can dial the **phone number** and press **"#"** button to start to dial the phone number.

The VoIP Gateway also provides some functions that list as below:

6.2.1 Blind Transfer

This feature allows a user (transferor) to transfer an existing call to another telephone number (transfer target) without connecting to the transfer target number.

How to Use:
1. During an existing call, perform a hook flash to put the other party on hold and get a dial tone.
2. When you hear the dial tone, press ***#510#*** on your telephone dial-pad.
3. When you hear the dial tone indicating that the VoIP Gateway is expecting

a number, dial the phone number to which you want to transfer the other party, then press **#** (optional) and then hang up the phone.

6.2.2 Attendant Transfer

This feature allows a user to transfer an existing call to another telephone number after first consulting with the dialed party (transfer target) before hanging up.

How to Use:

1. During an existing call, perform a hook flash to put the other party on hold and get a dial tone.
2. When you hear the dial tone, press ***#511#*** on your telephone dial-pad.
3. When you hear the dial tone, dial the telephone number to which the existing party is to be transferred, then press **#** (optional).
4. When the target transfer answers the phone, you may consult with the target transfer, and then hang up your phone to transfer the call to the target transfer.

6.2.3 3-Way Conferencing

How to Use:

1. Dial the first number.
2. During connection to the first party, perform a hook flash to put the first party on hold.
2. When you hear the dial tone, press ***#512#*** on your telephone dial-pad.
3. When you hear the recall dial tone, dial another number and talk with the second person.
4. To conference with both callers at the same time, perform a hook flash.
5. To transfer the second call to first call, perform a hook flash after entering into conferencing mode.

Note: If you hang up during conferencing, it'll transfer the first call to the second call.

6.2.4 Call Waiting

How to Use:

1. When a new call is coming while you are talking, you can push the Flash button or perform a hook flash to switch to the new call.
2. You can push the Flash button to switch between the two calls.

or

1. Dial the first number to make a conversation.
2. During connection to the first party, push the Flash button or perform a hook flash to put the first party on hold.
3. When you hear the dial tone, dial another number and talk with the second person.
4. You can push the Flash button or perform a hook flash to switch between the two calls.

6.2.5 Call Hold

How to Use:

1. When a new call is coming while you are talking, you can push the Flash button or perform a hook flash to hold the current call for a while, then push Hold key again to keep talking.
2. You can push the Flash button to switch between the two calls.

7. Get a FWD account

1. The website is www.freeworlddialup.com; you can apply an account to use the VoIP communication. You can follow the instruction to input the information. After you finished, you will receive a mail sent by the FWD mail system, you will get the account information in the mail.
2. When you got the account, you can setup the related information into the VoIP Gateway.
3. You can setup the related information into the VoIP Gateway by web browser. You need to input the Proxy Name, Domain Name, Register Name, and password. The Display Name you can input what you want to let others see.

4. After you registered to the SIP Server, you can try to call your friends who also registered in the same SIP Server. You just need to dial your friend's user name (registered name) and press "#" then you can make a phone call to your friend.
5. If you want to make a phone call to the other in the internet, first you need to registered in a Proxy Server (with SIP Server IP, Domain IP, registered name, Password), make sure you already enable Stun function, then you can try

Appendix A Glossary

This glossary defines acronyms and keywords used in this document.

A.1 Acronyms

ATA	Analog Telephony Adaptor
BLAM	Background Logging Application Mechanism
Broadband	Broad or wide bandwidth. In data transmssion, the wider the band, the more data it is possible to transmit in a given time span. A cable, DSL and ADSL connection to the network provide broadband for data transmission. A dialup or ISDN connection typically provides a narrow bandwidth for data transmission.
Codec	The format by which audio or video streams are compressed for transmission over networks.
CPC	**CPC (Calling Party Control)** is a signal sent from most modern electronic COs to indicate that the "**Calling Party**" has hung up. It's usually called "**Open Loop Disconnect**" when you're programming telephone equipment. The CPC signal tells the phone equipment that the outside party has hung-up, so it can stop recording to an answering machine or voice mail, drop the call off hold, or just release a line that might be used for dictation or announcements.

	Generally speaking, if a human is using a phone line, it doesn't matter whether the phone equipment recognizes CPC or not, since the human will physically hang-up the phone when they're done with the call, or they'll pick the call up off of hold when the phone system rings back after X seconds / minutes. CPC is normally sent as an open (0 volts DC), ranging from 250 to 500 milliseconds. When the outside party hangs-up, either on an inbound or outbound call, the phone equipment sees this open on the line and hangs up. Most voice mail and phone systems have a timer setting for CPC (or Open Loop Disconnect). I generally set CPC at 500ms, unless I have a problem. If you set it at 800ms, and the CPC open loop signal is only 500ms, the system will never see the open loop (it never gets to 800ms). If you set it at 500ms, and the actual CPC duration is 800ms, the phone system will recognize the CPC since there was 0 volts (an open loop) for 500ms (it won't matter if the open loop lasted another 300ms). If you accidentally set it for 50ms you'll probably get cut-offs, especially during a lightning storm which sometimes results in very brief blips in the loop current. Setting this timer for 50ms means that if the phone equipment sees an open for 1/20th of a second (not very long), it will hang up. Setting it for 500ms means it will hang-up if it sees an open of half a second or longer. That's much more reliable. There's often a short open (0 volts DC) on a phone line just after you go off-hook, or just after you've finished dialing a phone number. These are usually very short opens, like 20 to 50ms. If your phone system Open Loop Disconnect timer is set at 50ms, you may never be able to make a call because every call would be cut-off as soon as you went off-hook or were finished dialing. ***That Open Loop Disconnect Timer is very important!***
DTMF	Dual-tone multifrequency. DTMF is the system that is used in interactive voice-response menu systems such as the menu system for accessing voicemail messages. The DTMF system allows the user to interact with the menu by

	pressing keys on a dialpad or keyboard.
FoIP	Fax over Internet Protocol
FXO	Foreign Exchange Office
FXS	Foreign Exchange Station
IP	Internet Protocol. A data-oriented protocol used for communicating data across a network. IP is the most common protocol used on the internet.
IP address	A unique number that devices use in order to identify and communicate with each other on a computer network using the IP standard.
MWI	Message Waiting Indicator. An indicator that there is a voicemail message for the owner of an account.
Narrowband	In data transmission, the wider the band, the more data it is possible to transmit in a given time span. A cable, DSL and ADSL connection to the network provide broadband for data transmission. A dialup or ISDN connection typically provides a narrow bandwidth for data transmission.
PSTN	Public Switch Telephone Network. The traditional land-line phone network.
PTM	Packet Telephony Module RTP Real-time Transport Protocol
RFC	Request for Comment. A document that describes an aspect of an internet technology. An RFC may be a proposed, draft or full internet standard.
RTP	Real-time Transport Protocol. A protocol for delivering the media portion of a data transmission over an IP network. SRTP is another media protocol.
Signaling	In a VoIP phone call, the information in a call that deals with establishing and controling the connection, and managing the network. The non-signaling portion of the call is the Media.
SIP	Session Initiation Protocol. The signaling protocol followed by VoIP Gateway for handling phone calls.
SIP account	An account that provides the user the ability to make VoIP phone calls. The account encapsulates the rules and functions the user can access.
SIP address	The address used to connect to a SIP endpoint. In other words, the “phone number” used in a VoIP phone call. For example, sip:test@domainA.com.
STUN	Simple Transversal of UDP through NAT
TCP	Transmission Control Protocol. A transport protocol for

	delivering data over an IP network. Other transport protocols are TLS and UDP.
TLS	Transport Layer Security. A transport protocol for delivering data over an IP network. TLS is a secure transport protocol, which means that all the data being transmitted (signaling and media) is encrypted. Other transport protocols are TCP and UDP.
UA	User Agent
UDP	User Datagram Protocol. A transport protocol for delivering data over an IP network. Other transport protocols are TCP and TLS.
URI	URI Uniform Resource Identifier. A name or address that identifies a location on the world wide web. A SIP address is a type of URI.
URL	Uniform Resource Locator. A URI that both identifies a name or address and indicates how to locate it.
VoIP	Voice over Internet Protocol. A variation of IP used for sending voice data over the internet, in other words, used for making phone calls over the internet.
VoIP Service Provider	A business that provides a VoIP service, allowing a user to connect to the internet in order to make VoIP phone calls using VoIP Gateway. The VoIP service provider sets up a SIP account for the user.

A.2 Keyword and Definitions

Caller	Call Originating End is called the Caller
Callee	The Call Terminating End is called the Callee
Transferor	The End transferring the call
Transferee	The End being transferred
Transfer	The End to whom the transferee is being transferred

Target	