TW-VoIP-S2 User manual



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1. Features

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

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

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:

🕑 Control Panel File Edit View Favorites Tools Help 🔇 Back 🝷 🌔 🍷 彦 🔎 Search 🛛 🄂 Folders 🛛 🔢 🔻 Address 📴 Control Panel 🗙 🛃 Go P Xear El II 6 Ġ, -Z Control Panel Accessibility Add Hardware Add or Administrative Automatic CMI Audio Date and Time 🚱 Switch to Category View Options Remov... Tools Updates Config 90 3 **9** Ø So and 1 3 ۲ See Also Display Folder Options Fonts Game Internet Keyboard Mouse 🍓 Windows Update Controllers Options (2) Help and Support 4 3 Regional and Scanners and Network Setup Phone and Power Options Printers and Modem ... Faxes Wizard Cameras Language ... ۲ P **V** O, 8 Taskbar and User Accounts Start Menu Scheduled Sounds and Security System Speech Tasks Center Audio Devices ____)) e Wireless Windows Firewall Network Set...

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.

| Internet Protocol (TCP/IP) Properties | | | | |
|---|----------------|--|--|--|
| General | | | | |
| You can get IP settings assigned automatically if your network supports this capability. Otherwise, you need to ask your network administrator for the appropriate IP settings. | | | | |
| 🔿 Obtain an IP address automaticall | y | | | |
| Use the following IP address: — | | | | |
| IP address: | 192.168.10.110 | | | |
| Subnet mask: | 255.255.255.0 | | | |
| Default gateway: | 192.168.10.100 | | | |
| Obtain DNS server address autom | natically | | | |
| Our of the following DNS server add → Our of the following DNS server add | resses: | | | |
| Preferred DNS server: | 192.168.10.100 | | | |
| Alternate DNS server: | · · · | | | |
| Advanced | | | | |
| OK Cancel | | | | |

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 "

| 🕂 Local Area Connection Properties 🛛 🔹 💽 | | | | |
|---|--|--|--|--|
| General Authentication Advanced | | | | |
| Connect using: | | | | |
| SiS 900 PCI Fast Ethernet Adapter Configure | | | | |
| This connection uses the following items: | | | | |
| 🗹 💂 QoS Packet Scheduler 🔗 | | | | |
| ✓ 중 WPA Security Protocol (IEEE 802.1x) v1.5.1.65 | | | | |
| | | | | |
| | | | | |
| Install Uninstall Properties | | | | |
| Description | | | | |
| Transmission Control Protocol/Internet Protocol. The default wide area network protocol that provides communication across diverse interconnected networks. | | | | |
| Show icon in notification area when connected Notify me when this connection has limited or no connectivity | | | | |
| OK Cancel | | | | |

Step 5: Select " Obtain and IP Address automatically " and then click " OK "

| Internet Protocol (TCP/IP) Properties | | | | | |
|---|------------|---|--|--|--|
| General Alternate Configuration | | _ | | | |
| You can get IP settings assigned automatically if your network supports this capability. Otherwise, you need to ask your network administrator for the appropriate IP settings. | | | | | |
| Obtain an IP address automatic | ally | | | | |
| OUse the following IP address: - | | | | | |
| IP address: | | | | | |
| Subnet mask: | | | | | |
| Default gateway: | · · · · | | | | |
| Obtain DNS server address auto | omatically | | | | |
| Use the following DNS server a | ddresses: | | | | |
| Preferred DNS server: | | | | | |
| Alternate DNS server: | | | | | |
| Advanced | | | | | |
| OK Cancel | | | | | |

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.

| 🕹 Local Area Connection Properties 🛛 🔹 🛛 🛛 |
|--|
| General Authentication Advanced |
| Connect using: |
| B SiS 900 PCI Fast Ethernet Adapter Configure |
| This connection uses the following items: |
| Client for Microsoft Networks Client for Microsoft Networks Client for Microsoft Networks QoS Packet Scheduler WPA Security Protocol (IEEE 802.1x) v1.5.1.65 Install Uninstall Properties Description Allows your computer to access resources on a Microsoft network. |
| Show icon in notification area when connected Notify me when this connection has limited or no connectivity |
| Close Cancel |

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.

| 🚰 SIP VoIP Gateway Login - M | icrosoft Interne | t Explorer | | | | | |
|------------------------------|------------------|----------------|-----------------|---------|--------|---------|-----|
| File Edit View Favorites Too | ols Help | | | | | | _ |
| 🕲 Back 👻 🕑 👻 😫 | Sear | ch 🤺 Favorites | 🚱 🔗 🌺 | w • 📙 🎬 | - 28 | | |
| Address 🙆 http://10.0.0.2/ | | | | | 💌 🄁 Go | Links » | 🔁 • |
| | | | | | | | ~ |
| | | | | | | | |
| | | | | | | | |
| | | | | | | | |
| | | D.O. (| | | | | |
| 4 | ogin SIP voi | P Gateway | | | | | |
| | Enter your use | rname and pa | ssword to login | | | | |
| | | SIP VolP | Gateway | | | | |
| | Username | admin | | | | | |
| | Password | ••••• | | | | | |
| | | | | | | | |
| | | Login | Clear | | | | |
| | | | | | | | |
| | | | | | | | |
| | | | | | | | |
| | | | | | | | |
| E Done | | | | | 🌍 Inte | rnet | |

Step 2:

Now you could configure the VoIP Gateway in detail.



3.5.2 VoIP Configuration

Step 1:

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

| Gelew all* | SIP VoIP Gateway | €rele Well® |
|---|--|--------------------|
| Status > System Info > Network Status > VoIP Status | VoIP Configuration | |
| <u>Quick Setup</u> | | |
| Configuration VMAN LAN VIOIP DDNS VLAN VILAN VITUAL Server VITUAL Server SETE Settings Alarm Settings Alarm Settings System Authority | SIP Settings SIP Service Provider Port Settings Codec ID Settings DTMF Settings RPort Settings QoS | |
| Save Settings/Reudor System Aeset factory default Firmware Update Auto Update | Phone Book Phone Book | |
| | Call Forward | |
| | Volume Settings | |
| | DND Settings | |
| | Caller ID Settings | |
| | Dial Plan Settings | |
| | Flash Time Settings | |
| | Call waiting Settings | |
| | Hot line Settings | |
| | | |
| | Others | |
| | Auto Configuration Settings | |
| | FXS Impedence Settings | |
| | MAC Clone Settings | |
| | Tones Settings | |
| | Advanced Settings | |
| | | |

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 ".

SIP Service Provider

You could set information of SIP service provider in this page.

| Service Provider | |
|--------------------|-------------------|
| Active: | ⊙ On ◯ Off |
| Domain Server: | : |
| Proxy Server: | |
| Outbound Proxy: | |
| Display Name: | 1234 |
| User Name: | 1234 |
| Register Name: | 1234 |
| Register Password: | •••• |
| Status: | Not Registered |
| | |
| | Submit Reset Back |

Step 3:

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

Information

This page inform user important information.

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.

Save Settings / Reboot

| You have to save settings & reboot to effect them. | |
|--|---------------|
| Save Settings and reboot: | Save & Reboot |
| You could press the reboot button to restart the sys | stem. |
| Reboot system without saving settings: | Reboot Only |

Step 5:

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

Information

This page inform user important information.

System will reboot automaitcally 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 "

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 |

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

Information

This page inform user important information.



Step 3:

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

Save Settings / Reboot

| You have to save settings & reboot to effect them. | |
|--|---------------|
| Save Settings and reboot: | Save & Reboot |
| You could press the reboot button to restart the syste | em. |
| Reboot system without saving settings: | Reboot Only |

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.

Information

This page inform user important information.

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

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.

| 🕘 SIP VoIP Gateway Login - Mi | icrosoft Internet | Explorer | | | | | |
|-------------------------------|-------------------|--------------|----------------|---------|--------|---------|-----|
| File Edit View Favorites Too | ols Help | | | | | | - 🥂 |
| 🕞 Back 👻 🐑 👻 🛃 | Search | Favorites | 🚱 🔗 🌺 | w • 📃 🕯 | 1 48 | | |
| Address 🕘 http://10.0.0.2/ | | | | | 💙 🄁 Go | Links » | 🔁 🔹 |
| | | | | | | | ~ |
| | | | | | | | |
| | | | | | | | |
| | | | | | | | |
| | | | | | 1 | | |
| 4 | ogin SIP Volf. | P Gateway | | | | | |
| E | Enter your useri | name and pas | sword to login | | | | |
| | | SIP VoIP | Gateway | | | | |
| L | Username 🦉 | admin | | | | | |
| F | Password | | | | | | |
| | | | | | | | |
| | | Login | Clear | | | | |
| | | | | | | | |
| | | | | | | | |
| | | | | | | | |
| | | | | | | | |
| e Done | | | | | 🥥 Inte | rnet | |

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

WAN Settings

You could configure the WAN settings in this page.

LAN Mode: OBridge ONAT

| WAN Setting | |
|--------------|----------------------------------|
| IP Type: | ○ Fixed IP ○ DHCP Client ⊙ PPPoE |
| IP: | |
| Mask: | ; |
| Gateway: | |
| DNS Server1: | |
| DNS Server2: | |
| MAC: | |

| PPPoE Setting | | |
|---------------|---|--|
| User Name: | - | |
| Password: | | |

| Submit | Reset |
|--------|-------|
|--------|-------|

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

Information

This page inform user important information.

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.

| Save Settings / Reboot | | |
|---|---------------|--|
| You have to save settings & reboot to effect them. | | |
| Save Settings and reboot: | Save & Reboot | |
| You could press the reboot button to restart the syst | em. | |
| Reboot system without saving settings: | Reboot Only | |

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.

Information

This page inform user important information.

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

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 "

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 |

Step 3:

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

Information

This page inform user important information.

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.

Save Settings / Reboot

| You have to save settings & reboot to effect them. | |
|---|---------------|
| Save Settings and reboot: | Save & Reboot |
| You could press the reboot button to restart the syst | em. |
| Reboot system without saving settings: | Reboot Only |

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.

Information

This page inform user important information.

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

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) <u>http://10.0.0.2/</u> 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.

| 🚳 SIP VoIP Gateway Login - Microsoft Ir | ternet Explorer | |
|---|----------------------------------|----------------------|
| File Edit View Favorites Tools Help | | At 1 |
| Ġ Back 🝷 🐑 👻 🛃 🎸 | Search 👷 Favorites 🚱 🔗 + 🌺 👿 🔹 🧾 | 11 - 8 |
| Address 🗃 http://10.0.0.2/ | | 🔽 🄁 Go 🛛 Links 🎽 📆 🕶 |
| | | |
| | | |
| | | |
| | | |
| Login SI | | |
| | | |
| Enter you | r username and password to login | |
| | Sir Voir Gateway | |
| Oseman | | |
| Passwor | ********* | |
| | | |
| | Login | |
| | | |
| | | |
| | | |
| | | × |
| E Done | | Internet |

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

System Information

This page illustrate the system related information.

| Model Name: | SIP VoIP Gateway |
|-------------------|---------------------------------|
| Firmware Version: | Wed May 30 10:11:00 2007 |
| Codec Version: | Thu Apr 19 14:04:07 2007. |
| Software Version: | RM0S2_70425_TeleWell_04 (70530) |

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 | |
|---------------|---------------|
| Туре: | DHCP Client |
| IP: | 192.168 |
| Mask: | 255.255.255.0 |
| Gateway: | 0.0.0.0 |
| DNS Server 1: | 168 |
| DNS Server 2: | 168. |

| LAN Status | |
|---------------|---------------|
| Туре: | DHCP Server |
| IP: | 10.0.0.2 |
| Mask: | 255.255.255.0 |
| Gateway: | 10.0.0.2 |
| DNS Server 1: | 168 100 |
| DNS Server 2: | 168 2012 |

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 |
| | 400 |
| DNS Server 2: | 168. |

| LAN Status | |
|---------------|---------------|
| Туре: | DHCP Server |
| IP: | 10.0.0.2 |
| Mask: | 255.255.255.0 |
| Gateway: | 10.0.0.2 |
| DNS Server 1: | 168 1 |
| DNS Server 2: | 168 512 1 |

4.2 Configuration Page

4.2.1 WAN Configuration Page

You could configure the WAN settings in this page.

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 |

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.

LAN Settings

You could configure the LAN settings in this page.

| LAN Setting | |
|-------------|---------------|
| IP: | 10.0.0.2 |
| Mask: | 255.255.255.0 |
| MAC: | 000296aabbcd |

| DHCP Server | |
|--------------|---------------|
| DHCP Server: | ⊙ On ◯ Off |
| Start IP: | 150 |
| End IP: | 200 |
| Lease Time: | 1 : 0 (dd:hh) |



4.2.3 VoIP Gateway Configuration Page

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

VoIP Configuration

You could configure the VoIP settings, please click the hyperlink.

| SIP Settings |
|-----------------------------|
| SIP Service Provider |
| Port Settings |
| Codec Settings |
| Codec ID Settings |
| DTMF Settings |
| RPort Settings |
| QoS |
| |
| Phone Book |
| Phone Book |
| |
| Phone Settings |
| Call Forward |
| Volume Settings |
| DND Settings |
| Caller ID Settings |
| Dial Plan Settings |
| Flash Time Settings |
| Call Waiting Settings |
| T.38(FAX) Settings |
| Hot line Settings |
| |
| Others |
| Auto Configuration Settings |

FXS Impedence Settings

STUN Settings

MAC Clone Settings

Tones Settings

Advanced Settings

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 Settings, 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.

| SIP Settings |
|----------------------|
| SIP Service Provider |
| Port Settings |
| Codec Settings |
| Codec ID Settings |
| DTMF Settings |
| RPort Settings |
| <u>QoS</u> |

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

You could set information of service domains in this page.

| Service Provider | |
|--------------------|------------|
| Active: | ⊙ On Off |
| Domain Server: | |
| Proxy Server: | • |
| Outbound Proxy: | |
| Display Name: | 666666 |
| User Name: | 666666 |
| Register Name: | 666666 |
| Register Password: | ••••• |
| Subscribe for MWI: | On ⊙Off |
| Status: | Registered |
| | |



| SIP Service Provider | |
|---------------------------|--|
| Active | First you need click On to enable the Service Domain, then |
| | you can input the following items: |
| Domain | For example, in test@domain.com, the domain is |
| Server | "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 | If your VoIP service provider has an outbound proxy address |
| Proxy | 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 volP service |
| Derister | Provided by the VelD Service Provider |
| Register | Provided by the volp service provider. |
| Password Subscribe for | When set to On a Subseribe for Message Waiting Indiantion |
| | when set to on a subscribe for message waiting indication |
| NIVI Dogistor | Vou can see the Degister Status in the Status item. If the |
| Status | itom shows "Pogistorod", then your VolD Cateway is |
| Status | registered to the ISD you can make a phone call directly |
| Submit Button | When you finished the setting please click the Submit |
| | when you minished the setting, please thick the Submit |
| Reset Rutton | You can reset the configured parameters before you submit |
| Back Button | Go back to the previous web page |
| | |

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.

Port Settings

You could set the port number in this page.

| SIP Port: | 5060 (10~65533) |
|-----------|-------------------|
| RTP Port: | 60000 (10~65533) |
| | Submit Reset Back |
| | For 1 FXS Port |

Local Port Settings

You could set the port number in this page.

| Local Port of Phone1 | | |
|----------------------|-------|------------|
| SIP Port of Phone1: | 5060 | (10~65533) |
| RTP Port of Phone1: | 41000 | (10~65533) |
| | | |
| Local Port of Phone2 | | |
| SIP Port of Phone2: | 5062 | (10~65533) |
| RTP Port of Phone2: | 60100 | (10~65533) |
| | | |
| Submit Reset Back | | |
| For 2 FXS Port | | |

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.

Codec Settings

You could set the codec settings in this page.

| Codec Priority | | |
|-------------------|-------------|---|
| Codec Priority 1: | G.711 a-law | * |
| Codec Priority 2: | G.711 u-law | ~ |
| Codec Priority 3: | G.723 | * |
| Codec Priority 4: | G.729 | ~ |
| Codec Priority 5: | G.726 - 16 | * |
| Codec Priority 6: | G.726-24 | ~ |
| Codec Priority 7: | G.726 - 32 | ~ |
| Codec Priority 8: | G.726 - 40 | * |
| Codec Priority 9: | GSM | ~ |
| | | |
| RTP Packet Length | | |
| G.711 & G.729: | 20 ms 🐱 | |
| G.723: | 30 ms 💌 👘 | |
| | | |
| G.723 5.3K | | |
| G.723 5.3K: | 🔘 On 💿 Off | |
| | | |
| Voice VAD | | |
| Voice VAD: | 🔘 On 💿 Off | |
| | | |

Reset

Back

Submit

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

Codec ID Settings

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

| Codec Type | ID | Default Value |
|--------------|-------------------|---------------|
| G726-16 ID: | 23 (95~255) | 23 |
| G726-24 ID: | 22 (95~255) | 22 |
| G726-32 ID: | 2 (95~255) | 2 |
| G726-40 ID: | 21 (95~255) | 21 |
| RFC 2833 ID: | 101 (95~255) | ☑ 101 |
| | | |
| | Submit Reset Back | |

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.

DTMF Setting

You could set the DTMF setting in this page.

| DTMF | |
|----------------------|-------------------|
| ⊙ 2833 | |
| OInband DTMF | |
| ○ Send DTMF SIP Info | |
| | |
| | Submit Reset Back |

• 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.

RPort Setting

| You could enable/disable the RPort setting in this page. | | |
|--|--|--|
| ⊙On ○Off | | |
| Submit Reset Back | | |
| For 1 FXS Port | | |
| | | |
| | | |

RPort Settings

You could enable/disable the RPort setting in this page.

| RPort of Phone1: | ⊙ On ◯ Off |
|------------------|-------------------|
| RPort of Phone2: | ⊙ On ◯ Off |
| | Submit Reset Back |
| | Eor 2 EXS Port |
| | |

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.

QoS

You could set QoS settings in this page.

| Hold by RFC of Phone1: | On ⊙Off |
|------------------------|-------------------|
| Hold by RFC of Phone2: | On ⊙Off |
| | |
| Voice QoS (Diff-Serv): | 40 (0~63) |
| SIP QoS (Diff-Serv): | 40 (0~63) |
| SIP Expire Time: | 60 (15~86400 sec) |
| Use DNS SRV: | ⊙ On ◯ Off |
| | |
| | Submit Reset Back |

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

You could add/delete items in current phone book.

Phone Book Page: page 1 💌

| Position | Name | Number | URL | Select |
|----------|----------|------------|------------|--------|
| 0 | | | | |
| 1 | | | | |
| 2 | | | | |
| 3 | | | | |
| 4 | | | | |
| 5 | | | | |
| 6 | | | | |
| 7 | | | | |
| 8 | | | | |
| 9 | | | | |
| Delete S | Selected | Delete All | Reset Back | |

Add New Phone

| Position: | (0~139) |
|-----------|---------|
| Name: | |
| Number: | |
| URL: | |
| Add Phon | e Reset |

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

| 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 | Add the new Phone which you configured | | |
| [Button] | | | |
| 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.

| Phone Settings |
|-----------------------|
| Call Forward |
| Volume Settings |
| DND Settings |
| Auto Answer |
| Caller ID |
| Dial Plan Settings |
| Flash Time Settings |
| Call Waiting Settings |
| T.38(FAX) Settings |
| Hot line Settings |

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.

| Forward Setting | | | | | | |
|--|-------|-------|----------|------------|---|--|
| You could set the forward number of your phone in this page. | | | | | | |
| All Forward: | ⊙ Off | OIP | ○ PSTN | | | |
| Busy Forward: | 💽 Off | OIP | | | | |
| No Answer Forward: | 💽 Off | OIP | ○ PSTN | | | |
| | | | | | | |
| | | Name | | URL/Number | | |
| All Fwd No.: | | | | |] | |
| Busy Fwd No.: | | | | |] | |
| No Answer Fwd No.: | | | | |] | |
| | | | | | | |
| No Answer Fwd Time Out: 3 (2~8 Ring) | | | | | | |
| | | - | | | | |
| | Submi | t Res | set Back | | | |

| 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 | If you can not answer the phone after a specific ring you | | | | | |
| Forward | configured, the incoming call will forward to the | | | | | |
| | URL/number you configured. | | | | | |
| Off | Disable call forward. | | | | | |
| IP | Enable call forward for URL/number. | | | | | |
| PSTN | Enable call forward for PSTN phone number. | | | | | |
| (Optional) | 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 | You can set the Time Out time for system to start to forward |
| Fwd Time Out | 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 Forward: | 🔘 Off | 💿 IP | ○ PSTN |
|--------------------|-------|------|--------|
| Busy Forward: | 💽 Off | OIP | |
| No Answer Forward: | 💽 Off | OIP | ○ PSTN |

| | Name | | URL/Number | |
|--------------------|------|--|------------|--|
| All Fwd No.: | 7777 | | 7777 | |
| Busy Fwd No.: | | | | |
| No Answer Fwd No.: | | | | |

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 Forward: | | 00 | ff | 💿 IP | ○ PSTN | |
|-----------------|---------------|--------------|-----|----|------|------------|--|
| | Busy Forward: | | 0 📀 | ff | ΟIP | | |
| | No Answ | ver Forward: | 0 📀 | ff | OIP | ○ PSTN | |
| | | Name | | | | URL/Number | |
| All Fwd No.: | | | | | | | |
| Busy Fwd No.: | | | | | | | |
| No Answer Fwd N | lo.: | | | | | | |

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

Example 3: All Forward: PSTN, Name.: 888888888, URL/Number: 888888888

| | All Forward: | | 00 |)ff | O IP | PSTN | |
|------------------|--------------------|----------|--------------|-----|-------|------------|--|
| | Busy Forward: | | 0 C |)ff | ΟIP | | |
| | No Answer Forward: | | ⊙ (⊙ |)ff | OIP | ○ PSTN | |
| | | Name | | | | URL/Number | |
| All Fwd No.: | | 88888888 | | 888 | 88888 | | |
| Busy Fwd No.: | | | | | | | |
| No Answer Fwd No | 0.1 | | | | | | |

All incoming call will forward to the PSTN phone number 888888888.

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

| All Forward: | 💿 Off | OIP | ○ PSTN |
|--------------------|-------|------|--------|
| Busy Forward: | 🔘 Off | 💿 IP | |
| No Answer Forward: | 💿 Off | OIP | ○ PSTN |

| | Name | URL/Number |
|--------------------|------|------------|
| All Fwd No.: | | |
| Busy Fwd No.: | 7777 | 7777 |
| No Answer Fwd No.: | | |

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

| | All Forward: Busy Forward: | | 0 📀 | ff | OIP | ○ PSTN | |
|-----------------|-------------------------------|--------------|------|----|------|------------|---|
| | | | 00 | ff | 📀 IP | | |
| | No Answ | ver Forward: | 0 () | ff | OIP | ○ PSTN | |
| | | Name | | | | URL/Number | |
| All Fwd No.: | | | | | | |] |
| Busy Fwd No.: | | | | • | | |] |
| No Answer Fwd N | No.: | | | | | | |

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.

| xample 6: All Forward: IP, Name.: 7777, URL/Number: 7777 | | | | | | |
|--|--------------|------------|-----|------|------------|--|
| All Form | /ard: | 0 | Dff | ΟIP | ○ PSTN | |
| Busy Fo | orward: | 0 C | Off | ΟIP | | |
| No Ans | wer Forward: | 00 | Off | ⊙ IP | ○ PSTN | |
| | Name | | | | URL/Number | |
| All Fwd No.: | | | | | | |
| Busy Fwd No.: | | | | | | |
| No Answer Fwd No.: | 7777 | | 777 | 7 | | |
| | | | | | | |
| No Answer Fwd Time Out: | 3 (2~8 Ring |) | | | | |

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

| All For | ward: | 💿 Of | fOIP | ○ PSTN | |
|------------------------|----------------|------|--------|------------|--|
| Busy F | Forward: | 💿 Of | fOIP | | |
| No An | swer Forward: | 🔿 Of | f 💿 IP | ○ PSTN | |
| | Name | | | URL/Number | |
| All Fwd No.: | | | | | |
| Busy Fwd No.: | | | | | |
| No Answer Fwd No.: | | | | | |
| | | | | | |
| No Answer Fwd Time Out | : 3 (2~8 Ring) |) | | | |

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.: 888888888, URL/Number: 888888888

| All Forward: | | 💿 C | ff | ΟIP | ○ PSTN | |
|-------------------------|--------------|-----|-----|-------|------------|--|
| Busy Forward: | | 💿 C | ff | ΟIP | | |
| No Answer Forward: | | OC | ff | OIP | ⊙ PSTN | |
| | Name | | | | URL/Number | |
| All Fwd No.: | | | | | | |
| Busy Fwd No.: | | | | | | |
| No Answer Fwd No.: | 88888888 | | 888 | 88888 | | |
| | | | | | | |
| No Answer Fwd Time Out: | 3 (2~8 Ring) | | | | | |

If you can not answer the phone after 3 rings, the incoming call will forward to the PSTN phone number 888888888.

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.

Volume Setting

You could set the volume of your phone in this page.

| PSTN-Out | |
|------------------|-------------------|
| Handset Volume: | 10 (0~12) |
| PSTN-Out Volume: | 10 (0~12) |
| | |
| PSTN-In | |
| Handset Gain: | 10 (0~15) |
| PSTN-In Gain: | 10 (0~15) |
| | Submit Reset Back |

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 Settings

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

| DND Always: | ◯ On ⊙ Off |
|-------------|-----------------|
| | |
| | |
| DND Period: | On ⊙Off |
| From: | 00 :00 (hh:mm) |
| | |
| To: | 00 : 00 (hh:mm) |



| 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) |
| То | Input the end time of the time period. (24 hours format, |
| | hh:mm) |
| Submit | When you finished the setting, please click the Submit |
| Button | 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

You could enable/disable the auto answer in this page.

| Auto Answer: | On ⊙Off |
|----------------------|-------------------|
| Auto Answer Counter: | 03 (0~8) |
| | |
| | |
| PIN Code Enabled: | On ⊙Off |
| PIN Code: | |
| | |
| | Submit Reset Back |

| 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 | Default is Off (disable). When it was On (enable), It'll |
| Enabled | 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

Auto Answer

You could enable/disable the auto answer in this page.

| Auto Answer: | ⊙ On ◯ Off |
|----------------------|-------------------|
| Auto Answer Counter: | 3 (0~8) |
| | |
| | |
| PIN Code Enabled: | On ⊙Off |
| PIN Code: | |
| | |
| | Submit Reset Back |

How to Use - PSTN to VoIP Call:

- 1. Call in via PSTN.
- $\ensuremath{\mathbf{2}}$. 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 VolP.

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

Auto Answer

You could enable/disable the auto answer in this page.

| Auto Answer: | ⊙On Off |
|----------------------|-------------------|
| Auto Answer Counter: | 3 (0~8) |
| | |
| | |
| PIN Code Enabled: | ⊙On Off |
| PIN Code: | 1234 |
| | |
| | Submit Reset Back |

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 VolP.

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.

Caller ID Settings

You could enable/disable the caller ID setting in this page.

| Caller ID: Caller ID after 1st Ring (FSK) Single Caller ID: ○ Yes ⊙ No CID Without Time: ○ Yes ⊙ No CID Type 2: ○ Yes ⊙ No | | | |
|--|-------------------|--------------------------------|---|
| Single Caller ID: ○ Yes ⊙ No CID Without Time: ○ Yes ⊙ No CID Type 2: ○ Yes ⊙ No | Caller ID: | Caller ID after 1st Ring (FSK) | * |
| CID Without Time: O Yes O No CID Type 2: O Yes O No | Single Caller ID: | 🔿 Yes 💿 No | |
| CID Type 2: O Yes No | CID Without Time: | 🔿 Yes 💿 No | |
| | CID Type 2: | 🔿 Yes 💿 No | |

Submit Reset Back

| Single Caller | Default is Off (disable). When it was Yes (enable), It'll |
|---------------|---|
| ID | detect the Singel Caller ID. |
| CID Without | Default is Off (disable). When it was Yes (enable), It'll |
| Time | 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.

Dial Plan Settings

You could the set the dial plan in this page.

| Drop prefix : | ⊖Yes ⊙No |
|-----------------------|-------------------------------|
| Replace rule 1: | 002 + 8613+8662 |
| Drop prefix : | ⊙ Yes ⊃ No |
| Replace rule 2: | 006 + 002+003+004+005+007+009 |
| Drop prefix : | ⊖Yes ⊙No |
| Replace rule 3: | 009 + 12 |
| Drop prefix : | ⊖Yes ⊙No |
| Replace rule 4: | 007 + 6xxx+21xx |
| | |
| Dial now: | ×× |
| Auto Dial Time: | 5 (3~9 sec) |
| Use # as send key: | ⊙ Yes ONo |
| Use * for IP dialing: | ⊙ Yes ◯ No |



| Drop Prefix | Default is NO (Add | the Prefix). When it was Yes (Drop the | | |
|---------------------|---|--|--|--|
| | Pref | ix), 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 | x): When it meets the rule which you | | |
| | configured, it'll drop | b the prefix and replace the number | | |
| | which you configure | d. Maximum input digits are 31. | | |
| Replace rule1 | There are | total 4 replace rules for use. | | |
| Replace rule2 | +: or | | | |
| Replace rule3 | xxx: Define the leng | th of digits. | | |
| Replace rule4 | | | | |
| Dial now | If the numbers which | n you dialed met this rule, it will dial out | | |
| | with i | ts 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 inv | alid due to the first digit is 0. | | |
| Auto Dial | Default is 5 (Second | ds). How long the phone number will be | | |
| Time | dialed out a | fter finishing dialing the digits. | | |
| Use # as send | Default is Yes. When it was No, It'll wait for the setting of | | | |
| key | Auto Dial Time and then dial out after dialing the phone | | | |
| | numbers. | | | |
| Use * for IP | Default is Yes. Wher | n it was No, the * key will not be as . for | | |
| dialing | IP Dialing. | | | |
| Submit | When you finished the setting, please click the Submit | | | |
| Button | button. | | | |
| Reset Button | You can reset the configured parameters before you submit | | | |
| Back Button | Go bac | k to the previous web page | | |
| Symbol explain |): | | | |
| > | k or X | 0,1,2,3,4,5,6,7,8,9 | | |
| + | | or | | |

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

| Drop prefix : | ⊖Yes ⊙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

| Drop prefix : | 💿 Yes i 🔘 N | Νο |
|-----------------|-------------|---------------------------|
| 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 **0045555** and the digits **004** will be

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

| Drop prefix : | ⊖Yes ⊙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

| Drop prefix : |
|-----------------|
| Replace rule 4: |

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

| Dial now: |
|-----------|
|-----------|

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

Auto Dial Time function

| Auto Dial Time: | 5 | (3~9 sec) |
|-----------------|---|-----------|
|-----------------|---|-----------|

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

Use # as send key: 💿 Yes 🔵 No

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 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.

Flash Time Setting

You could set the flash time in this page.

| FXO Flash Time | |
|-----------------|---------------------|
| Flash Time: | 5 (3~200, 1->10ms) |
| | |
| SLIC Flash Time | |
| Max Flash Time: | 60 (4~255, 1->10ms) |
| | |
| | Submit Reset Back |

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.

| Call Wa | iting Settings |
|------------------|---|
| You could enable | /disable the call waiting setting in this page. |
| Call Waiting: | ⊙ On Off |
| | Submit Reset Back |

4.2.3.3.9 T.38 (FAX) Setting

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

T.38 (FAX) Settings

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

| T.38 (FAX): | On ⊙Off |
|-------------|--------------------|
| T.38 Port: | 60000 (1024~65533) |
| | |
| | Submit Reset Back |

For 1 FXS Port

T.38 (FAX) Settings

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

| T.38 (FAX): | ⊙On ○Off |
|----------------------|---|
| T.38 Port of Phone1: | 60000 (Only support one port at a time) |
| T.38 Port of Phone2: | 60100 (1024~65533) |
| | |

Submit Reset Back

For 2 FXS Port

| T.38 (FAX) | When it was On (Enable) , It'll enable the T.38 Fax function. |
|---------------------|--|
| T.38 Port/ | Default is 60000. (Only support one port at a time) |
| T.38 Port of | |
| Phone1 | |
| T.38 Port of | Default is 60100. (Only support one port at a time) |
| Phone2 | |
| Submit | When you finished the setting, please click the Submit |
| Button | 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.

Hot line Settings

| You could set the hot line in this page. | | |
|--|----------|------------|
| Use Hot Line : | ○ Enable | ⊙ Disable |
| | | |
| Hot line number: | | |
| | Submit | Reset Back |

| Use Hot Line | Default is Disable. When it was Enable, It'll enable the Hot |
|---------------------|--|
| | Line function. |
| Hot Line | Enter the URL, VoIP Phone Number, Remote WAN IP Address |
| Number | 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

| Use Hot Line : | 💿 Enable | ◯ Disable | |
|------------------|----------|------------|--|
| | | | |
| Hot line number: | | | |
| | Submit | Reset Back | |

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

| Use Hot Line : | 💿 Enable i 🔘 Disable |
|------------------|----------------------|
| | |
| Hot line number: | voiptest |
| | Submit Reset Back |

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

| Use Hot Line : | 💽 Enable | ○ Disable |
|------------------|----------|------------|
| | | |
| Hot line number: | | |
| | Submit | Reset Back |

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

Others
Auto Configuration Settings
FXS Impedence Settings
STUN Settings
MAC Clone Settings
Tones Settings
Advanced Settings

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 Settings

| You could enable/disable the auto configuration setting in this page. | | | | | |
|---|-------|----------|---------|-----------------|--|
| Auto Configuration: | ⊙ Off | ○ TFTP | O FTP | OHTTP | |
| | | | | | |
| TFTP Server: | | | | | |
| HTTP Server: | | | | · · · · · · · | |
| HTTP File Path: | | | | Exp. /download/ | |
| | | | | | |
| FTP Server: | | | | | |
| FTP Username: | | | | | |
| FTP Password: | | | | | |
| FTP File Path: | | | | Exp. /file/load | |
| | Subm | nit Rese | et Back |] | |

| Auto | Default is Off(Disable). When it was Enable, there are 3 |
|---------------------|---|
| Configuration | 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: /

| Auto Configuration: | 🔘 Off | ○ TFTP | 🔘 FTP | 💽 HTTP | |
|---------------------|-------|----------|--------|--------|------------|
| | | | | | |
| | | | | | |
| TFTP Server: | | | | | |
| HTTP Server: | | | | | · · · · |
| HTTP File Path: | 1 | | | Exp. | /download/ |
| | | | | | |
| FTP Server: | | | | | |
| FTP Username: | | | | | |
| FTP Password: | | | | | |
| FTP File Path: | | | | Exp. | /file/load |
| | | | | | |
| | Subr | nit Rese | t Back |] | |

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

| Auto Configuration: | Off | TFTP | FTP | ○ HTTP |
|---------------------|------|---------|--------|-----------------|
| | | | | |
| | _ | | | |
| TFTP Server: | | | | |
| HTTP Server: | | | | |
| HTTP File Path: | | | | Exp. /download/ |
| | | | | |
| FTP Server: | | | | |
| FTP Username: | | | | |
| FTP Password: | | | | |
| FTP File Path: | | | | Exp. /file/load |
| | | | | |
| | Subm | it Rese | t Back |] |

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: /

| Auto Configuration: | 🔘 Off | ○ TFTP | FTP | ○ HTTP |
|---------------------|-------|--------|-----|-----------------|
| | | | | |
| | | | | |
| TFTP Server: | | | | |
| HTTP Server: | | | | · · · · · |
| HTTP File Path: | | | | Exp. /download/ |
| | | | | |
| FTP Server: | • | | | . · · · - |
| FTP Username: | 1234 | | | |
| FTP Password: | •••• | | | |
| FTP File Path: | 1 | | | Exp. /file/load |



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 & FXS Impedence Setting

You could select the FXO & FXS impedence of the analog telephone by different country in this page.

| FXO Port: | USA | * |
|-----------|--------|------------|
| FXS Port: | USA | ~ |
| | | |
| | Submit | Reset Back |

| 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 | When you finished the setting, please click the Submit |
| Button | 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 Setting

| You could set the IP of STUN server in this page. | | | |
|---|-------------------|--|--|
| STUN: | ◯ On ⊙ Off | | |
| | | | |
| STUN Server: | stun.xten.com | | |
| STUN Port: | 3478 (1024~65535) | | |
| | Submit Reset Back | | |

| 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 <i>stun.xten.com</i> . 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 | When you finished the setting, please click the Submit | | | | | | |
| Button | 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 Settings

You could enable/disable the MAC clone setting in this page.

| MAC Clone: | 🔘 On 💿 Off |
|------------|-------------------|
| | Submit Reset Back |

| MAC Clone | Default is Off (disable). When it was On (enable). The VolP |
|---------------------|---|
| | Gateway clones the computer's MAC that was originally set |
| | up for such an ISP. |
| Submit | When you finished the setting, please click the Submit |
| Button | 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: | | V | | | V | ✓ |
| 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.

Advanced Settings

You could change advanced setting in this page.

| ICMP Not Echo: | ⊖Yes ⊙No |
|----------------------|------------------|
| Send Anonymous CID: | ⊖Yes ⊙No |
| Management from WAN: | ⊖Yes ⊙No |
| Billing Signal: | Disabled 🗸 |
| CPC Delay: | 2 (2~5 Seconds) |
| CPC Duration: | 0 x 10MS (0~120) |
| Send Flash event: | Disabled 🗸 |
| SIP Encrypt: | Disabled 👻 |
| PPPoE retry period: | 5 Seconds |
| System Log Server: | |
| System Log Type: | None |
| | |

Submit

Reset

Back

| ICMP Not Echo | Default is Off (disable). When it was On (enable). The VoIP |
|---------------|--|
| | Gateway will not echo the ICMP request. |
| Send | The <i>Anonymous</i> Caller ID to display when you make a call |
| Anonymous | 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 | Default is Disable . There are two types of Flash event: | | |
| 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 <i>GX</i> . | | |
| PPPoE retry | Default is 5 seconds. The range is 5 to 255. When PPPoE | | |
| period (*) | failed to connect to ISP, it will wait for the period which | | |
| | you configured to redial. | | |
| System Log | To upload the system log on the specified Server | | |
| Server | | | |
| System Log | Default is None. There are 7 types: Call Statistics, General | | |
| Туре | 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

ICMP Not Echo: 💿 Yes 🔿 No

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

Example 2: Send Anonymous CID: Yes

Send Anonymous CID: 💿 Yes 🔾 No

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

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

Example 4:

| Send Flash event | DTMF | FVFNT |
|------------------|------|-------|
| | | |

| Send Flash event: | DTMF EVENT 💌 | |
|-------------------------|--------------|--|
| the DTME EVENT as Elach | ovent | |

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.

DDNS Settings

You could set the configuration of DDNS in this page.

| DDNS: | ⊙ On ◯ Off |
|-------------------|--------------|
| | |
| Host Name: | |
| User Name: | |
| Password: | |
| E-mail Address: | |
| | |
| DDNS Server: | |
| DDNS Server List: | User Input |
| Туре: | dyndns 💌 |
| Wild Card: | on 💌 |
| | |
| BACKMX: | On ⊙Off |
| Off Line: | ⊖ On ⊙ Off |
| | Submit Reset |

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: 💦 🔘 I

🔘 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.

DDNS Settings

You could set the configuration of DDNS in this page.

| DDNS: | ⊙ On ◯ Off |
|-------------------|----------------------|
| | |
| Host Name: | |
| User Name: | |
| Password: | |
| E-mail Address: | • |
| | |
| DDNS Server: | |
| DDNS Server List: | members.dyndns.org 💌 |
| Туре: | dyndns 💌 |
| Wild Card: | off 🗸 |
| | |
| BACKMX: | On ⊙Off |
| Off Line: | On ⊙Off |
| | Submit Reset |

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

| ø | SIP VoIP Gatewa | y Login - Microsoft Internet Explorer 📃 🗖 | |
|----|------------------------|---|----------|
| F | ile Edit View Fa | avorites Tools Help | <u> </u> |
| (| 🕃 Back 🔹 🕥 | 💌 😰 🏠 🔎 Search 👷 Favorites 🪱 | » |
| Ac | ldress 🙆 http://willia | amcheng.dyndns.org/ 💽 🄁 Go 🛛 Links 🎽 😨 | • |
| | Login SIP Vol | P Gateway | ^ |
| | Enter your use | rname and password to login | |
| | | SIP VoIP Gateway | |
| | Username | | |
| | Password | | |
| | | Login Clear | |
| | | | * |
| e | Done | 🥥 Internet | : |

4.2.5 VLAN Settings Page

You could set the VLAN settings in this page.

VLAN Settings

You could set the VLAN settings in this page.

| /LAN Packets: ○ On ○ Off /ID (802.1Q/TAG): 136 (2 ~ 4094) /Iser Priority (802.1P): 0 (0 ~ 7) (E1: 1 (0 ~ 1) | | |
|---|-------------------------|----------------|
| /ID (802.1Q/TAG): 136 (2 ~ 4094) Jser Priority (802.1P): 0 (0 ~ 7) (E1: 1 (0 ~ 1) | VLAN Packets: | 🔘 On 🛛 Off |
| User Priority (802.1P): 0 (0 \sim 7) | VID (802.1Q/TAG): | 136 (2 ~ 4094) |
| (E) $(0 - 1)$ | User Priority (802.1P): | 0 (0 ~ 7) |
| | CFI: | 1 (0 ~ 1) |



| VLAN Packets | Default is Off(Disable). When it was On(Enable), It'll |
|---------------|--|
| | enable to receive VLAN Packets function. |
| VID | Default is 136. Configure the Virtual LAN ID (VLAN ID or |
| (802.1Q/TAG) | 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 | Default is 0 . Configure user priority. |
| (802.1P) | 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 <i>Canonical Format Indicator</i> 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.

DMZ Settings

You could configure your demilitarized zone setting in this page.

| DMZ: | ◯ On ⊙ Off |
|--------------|--------------|
| | |
| DMZ Host IP: | 0.0.0.0 |
| | Submit Reset |

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 Settings

You could set your virtual servers in this page.

The usual port numbers are WEB [TCP 80], FTP(Control) [TCP 21], FTP(Data) [TCP 20], E-mail(POP3) [TCP 110], E-mail(SMTP) [TCP 25], DNS [UDP 53] and Telent [TCP 23].

| Virtual Server Page: page 1 💌 | | | | | | |
|--|--------|----------|------------|---------|-----------|--------|
| Num | Enable | Protocol | In Port | Ex Port | Server IP | Select |
| 0 | | | | | | |
| 1 | | | | | | |
| 2 | | | | | | |
| 3 | | | | | | |
| 4 | | | | | | |
| 5 | | | | | | |
| 6 | | | | | | |
| 7 | | | | | | |
| Enable Selected Delete Selected Delete All Reset | | | | | | |
| Add Virtual Server | | | | | | |
| Num: (0~23) | | | | | | |
| Server IP: | | | | | | |
| Protocol: TCP 💌 | | | | | | |
| Internal P | ort: | | External P | ort: | | |
| Add S | erver | Reset | | | | |

| | Virtual Server Page |
|-----------------|--|
| Virtual Server | Default page is Page1. There are total 3 pages from Page 1 |
| Page | 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 | Display the Internal Port that you configured |
| (Internal Port) | |
| Ex Port | Display the External Port that you configured |
| (External | |
| Port) | |
| Server IP | Display the private network IP address for the particular |
| | server. |
| Select | Select the item of the Virtual Server |
| Enable | Enable selected item |
| Selected | |
| [Button] | |
| Delete | Delete selected item |
| Selected | |
| [Button] | |
| Delete All | Delete all items |
| [Button] | |
| Reset | Reset selected item |
| [Button] | |

| 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 | Add the new Server which you configured |
| [Button] | |
| 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

| Add Virtual Server | | | | | | |
|--------------------|-----------|-------------|-----------|---------|--------------|--------|
| | Nu | um: | 0 | (0~23) | | |
| | Se | erver IP: | 10.0.0.15 | 0 | | |
| | Pr | otocol: | TCP 🔽 |] | | |
| | Int | ernal Port: | 21 | Extern | nal Port: 21 |] |
| Virtual Se | rver Page | Add Server | Reset | | | |
| Num | Enable | Protocol | In Port | Ex Port | Server IP | Select |
| 0 | V | TCP | 21 | 21 | 10.0.0.150 | |

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.

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

Table 4-3. Well Known TCP/UDP Ports

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

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

You could set the PPTP server in this page.

| PPTP: | ◯ On ⊙ Off |
|----------------|------------|
| | |
| PPTP Server: | |
| PPTP Username: | |
| PPTP Password: | |
| | |

| 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 | Enter the Username of PPTP client. |
| Username | |
| PPTP | Enter the Pasword of PPTP client. |
| Password | |
| 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.

| SNTP Set | tings |
|-----------------------|----------------------------|
| You could set the SNT | P servers in this page. |
| SNTP: | ⊙ On Off |
| Duintama Canaam | |
| Primary Server: | |
| Secondary Server: | |
| Time Zone: | GMT + 💙 08 💌: 00 💙 (hh:mm) |
| Sync. Time: | 1 : 0 : 0 (dd:hh:mm) |
| | Submit Reset |
| Туре: | dyndns 💙 |
| Wild Card: | off 💌 |
| BACKMX: | OΩn ⊙Ωff |
| Off Line: | On ⊙Off |
| | Submit Reset |

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 Settings

You could set the alarm time in this page.

| Alarm: | ○ ON ⊙ OFF |
|---------------|------------------|
| | |
| Alarm Time: | 0 : 0 (hh:mm) |
| | |
| Current time: | 2005-01-01-08:11 |
| | Submit Reset |

96

| 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 | When you finished the setting, please click the Submit | |
| Button | button. | |
| Reset Button | You can reset the configured parameters before you submit | |

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

| Alarm: | ⊙ ON ○ OFF |
|---------------|------------------|
| | |
| Alarm Time: | 8 : 1 (hh:mm) |
| | |
| Current time: | 2005-01-01 08:00 |
| | Submit Reset |

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

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

| Alarm: | ⊙ ON ○ OFF | |
|---------------|------------------|--|
| | | |
| Alarm Time: | 23 : 31 (hh:mm) | |
| | | |
| Current time: | 2006-03-05 23:29 | |
| | | |
| | Submit Reset | |

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.

System Authority

You could change the login username/password in this page.

| New username: | |
|---------------------|--------------|
| New password: | |
| Confirmed password: | |
| | |
| | Submit Reset |

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.

| Save Settings / Reboot | |
|--|---------------|
| You have to save settings & reboot to effect them. | |
| Save Settings and reboot: | Save & Reboot |
| You could press the reboot button to restart the sys | stem. |
| Reboot system without saving settings: | Reboot Only |

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.

| Reset to Factory Default | |
|---|--|
| You could click the restore button to restore the factory settings. | |
| | |
| Restore default settings: Restore | |

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.

Firmware Update

| You could update the newest firmware in this page. | | |
|--|----------|--------|
| | | |
| | | |
| File Location: | | Browse |
| | Update F | Reset |

4.3.3 Auto Update Page

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

Auto Update Settings

You could set auto update settings in this page.

| Update via: | |
|-----------------------|--|
| | |
| TFTP Server: | |
| HTTP Server: | a manual and a second |
| HTTP File Path: | Exp. /download/ |
| | |
| FTP Server: | |
| FTP Username: | |
| FTP Password: | |
| FTP File Path: | Exp. /file/load |
| | |
| Check new firmware: | Power ON O Scheduling |
| Scheduling (Date): | 14 (1~30 days) |
| Scheduling (Time): | AM 00:00- 05:59 💌 |
| Automatic Update: | ⊙ Notify only ○ Automatic (Scheduling) |
| Firmware File Prefix: | TA2S |
| | |
| Next update time: | 2005-01-16 00:31 |
| | |
| | Submit Reset |

| 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 | Power ON: It'll check if there is a new firmware on the |
| firmware | 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 | Default is 14. It'll check if there is a new firmware on the |
| (Date) | TFTP/FTP/HTTP Server periodically. The range of the |
| | Scheduling Date is 1 - 30. |
| Scheduling | Default is AM 00:00- 05:59. It'll check if there is new |
| (Time) | 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 | Notify only: When there is a newer firmware, it will only |
| Update | 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 | The file prefix of the firmware |
| Prefix Next undate | It a the payt undate or sheek time |
| | It's the next update of check time. |
| LIME Submit Button | When you finished the setting please click the Submit |
| | button |
| Posat Rutton | Dutton. You can reset the configured parameters before you submit |
| | יטע כמה ופזפנ נוופ כטוווועטופט אמו מווופנפו ז מפוטו פ זטע זעטווונ |

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

Auto Update Settings

You could set auto update settings in this page.

| Update via: | Off | ○ TFTP | 🔘 FTP | ⊙ HTTP | | |
|-----------------------|---|------------|------------|-----------------|--|--|
| | | | | | | |
| TFTP Server: | | | | | | |
| HTTP Server: | | | | | | |
| HTTP File Path: | 1 | | | Exp. /download/ | | |
| | | | | | | |
| FTP Server: | | | | ing the | | |
| FTP Username: | | | | | | |
| FTP Password: | | | | | | |
| FTP File Path: | | | | Exp. /file/load | | |
| | | | | | | |
| Check new firmware: | O Pow | ver ON (| Scheduling | g | | |
| Scheduling (Date): | 14 | (1~30 days | ;) | | | |
| Scheduling (Time): | AM 00:00- 05:59 🗸 | | | | | |
| Automatic Update: | Notify only Automatic (Scheduling) | | | | | |
| Firmware File Prefix: | TA2S | | | | | |
| | | | | | | |
| Next update time: | 2005- | 01-16 00:3 | 1 | | | |
| | | | | | | |
| | Subr | nit Res | et | | | |

RULE of AUTO UPDATE:

Every time when you power on the VoIP Gateway, it'll notify you by "*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:

- 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

Auto Update Settings

You could set auto update settings in this page.

| Update via: | Off | ● TFTP | 🔘 FTP | OHTTP |
|-----------------------|---------|------------|------------|-----------------|
| | | | | |
| TFTP Server: | | | | |
| HTTP Server: | | | | Exp. |
| HTTP File Path: | | | | Exp. /download/ |
| | | | | |
| FTP Server: | | | | Exp |
| FTP Username: | | | | |
| FTP Password: | | | | |
| FTP File Path: | | | | Exp. /file/load |
| | _ | | | |
| Check new firmware: | O Power | er ON 🛛 🤇 | Scheduling | 3 |
| Scheduling (Date): | 1 (| 1~30 days) |) | |
| Scheduling (Time): | AM 00 | :00- 05:59 | * | |
| Automatic Update: | 💿 Notif | y only 🛛 🤇 | Automatic | (Scheduling) |
| Firmware File Prefix: | TA2S | | | |
| | | | | |
| Next update time: | | | | |
| | | | | |
| | Subm | nit Res | et | |

RULE of AUTO UPDATE:

Every time when you power on the VoIP Gateway, it'll notify you by "*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:

- 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

Auto Update Settings

You could set auto update settings in this page.

| Update via: | OOff OTFTP ⊙ FTP (| OHTTP | | | | |
|-----------------------|---|-----------------|--|--|--|--|
| | | | | | | |
| TFTP Server: | | | | | | |
| HTTP Server: | | Exp. | | | | |
| HTTP File Path: | | Exp. /download/ | | | | |
| | | | | | | |
| FTP Server: | | Exp. | | | | |
| FTP Username: | 1234 | | | | | |
| FTP Password: | •••• | | | | | |
| FTP File Path: | 1 | Exp. /file/load | | | | |
| | | | | | | |
| Check new firmware: | | | | | | |
| Scheduling (Date): | 1 (1~30 days) | | | | | |
| Scheduling (Time): | AM 00:00- 05:59 💌 | | | | | |
| Automatic Update: | Notify only Automatic (Scheduling) | | | | | |
| Firmware File Prefix: | TA2S | | | | | |
| | | | | | | |
| Next update time: | | | | | | |
| | | | | | | |
| | Submit Reset | | | | | |

RULE of AUTO UPDATE:

Every time when you power on the VoIP Gateway, it'll notify you by "*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:

- 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
Auto Update Settings

You could set auto update settings in this page.

| Update via: | OOff O | TFTP | FTP | OHTTP |
|-----------------------|--------------------------------|---------|------------|-----------------|
| | | | | |
| TFTP Server: | | | | |
| HTTP Server: | | | | Exp. |
| HTTP File Path: | | | | Exp. /download/ |
| | | | | |
| FTP Server: | | | | Exp |
| FTP Username: | 1234 | | | |
| FTP Password: | •••• | | | |
| FTP File Path: | 1 | | | Exp. /file/load |
| | | | | |
| Check new firmware: | O Power ON | V 📀: | Scheduling | L |
| Scheduling (Date): | 1 (1~30 | I days) | | |
| Scheduling (Time): | AM 06:00-1 | 11:59 🔽 | | |
| Automatic Update: | Notify onl | y 🔘 | Automatic | (Scheduling) |
| Firmware File Prefix: | TA2S | | | |
| | | | | |
| Next update time: | | | | |
| | | | | |
| | Submit | Reset | t | |

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

Auto Update Settings

You could set auto update settings in this page.

| Update via: | |
|-----------------------|-------------------------|
| | |
| TFTP Server: | |
| HTTP Server: | Exp (258080) |
| HTTP File Path: | / Exp. /download/ |
| | |
| FTP Server: | Ex¢ |
| FTP Username: | |
| FTP Password: | |
| FTP File Path: | Exp. /file/load |
| | |
| Check new firmware: | ○ Power ON Scheduling |
| Scheduling (Date): | 1 (1~30 days) |
| Scheduling (Time): | AM 00:00- 05:59 👻 |
| Automatic Update: | ○ Notify only |
| Firmware File Prefix: | TA2S |
| | |
| Next update time: | 2007-02-23 01:48 |
| | |
| | Submit Reset |

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

Auto Update Settings

Update via: Off ○ ● TFTP ○ FTP O HTTP TFTP Server: HTTP Server: Ехμ HTTP File Path: Exp. /download/ FTP Server: Ελμ. FTP Username: FTP Password: FTP File Path: Exp. /file/load Check new firmware: O Power ON Scheduling Scheduling (Date): 1 (1~30 days) Scheduling (Time): AM 00:00- 05:59 🔽 Automatic Update: O Notify only O Automatic (Scheduling) Firmware File Prefix: TA2S Next update time: 2005-01-03 00:07 Submit Reset

You could set auto update settings in this page.

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

Auto Update Settings

You could set auto update settings in this page.

| Update via: | OOff OTF1 | 'P 💿 FTP | ◯HTTP |
|-----------------------|----------------|-------------------------------|-----------------|
| | | | |
| TFTP Server: | | | |
| HTTP Server: | | | Exp. |
| HTTP File Path: | | | Exp. /download/ |
| | | | |
| FTP Server: | | | Exp |
| FTP Username: | 1234 | | |
| FTP Password: | •••• | | |
| FTP File Path: | 1 | | Exp. /file/load |
| | | | |
| Check new firmware: | O Power ON | 📀 Scheduling | |
| Scheduling (Date): | 1 (1~30 da | iys) | |
| Scheduling (Time): | AM 00:00- 05:5 | i9 🗸 | |
| Automatic Update: | 🔘 Notify only | Automatic | (Scheduling) |
| Firmware File Prefix: | TA2S | | |
| | | | |
| Next update time: | 2007-02-23 01 | :48 | |
| | | | |
| | Submit R | eset | |

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 | Parameter(s) | Notes: |
|----------|-------------------|----------|--------------|--|
| | | Choice | | |
| 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 | #122# | None | IVR will report current in use VoIP number |
| | Number | | | |
| Info | Check Network | #123# | None | IVR will report the WAN Port network mask |
| | Mask | | | |
| Info | Check Gateway IP | #124# | None | IVR will announce the current gateway IP address |
| | Address | | | of the VoIP Gateway |
| Info | Check Primary DNS | #125# | None | IVR will announce the current setting in the |
| | Server Setting | | | Primary DNS field. |
| Info | Check IP Address | #126# | None | IVR will report the WAN port IP address |
| Info | Check Firmware | #128# | None | IVR will announce the version of the firmware |
| | Version | | | 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 <u>www.freeworlddialup.com</u>; 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. <i>That Open Loop</i> <i>Disconnect Timer is very important!</i> |
| 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 | Üser 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 | A business that provides a VoIP service, allowing a user to |
| Provider | 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 | |
|--------|--|