DrayTek

VigorPhone 300

IP Phone

Your reliable networking solutions partner

User's Guide

V1.0

VigorPhone 300
User's Guide

Version: 1.0

Date: 05/03/2012

Copyright Information

Trademarks

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Safety Instructions and Approval

Safety Instructions

- Read the installation guide thoroughly before you set up the device.
- The router is a complicated electronic unit that may be repaired only be authorized and qualified personnel. Do not try to open or repair the router yourself.
- Do not place the device in a damp or humid place, e.g. a bathroom.
- The device should be used in a sheltered area, within a temperature range of +5 to +40 Celsius.
- Do not expose the device to direct sunlight or other heat sources. The housing and electronic components may be damaged by direct sunlight or heat sources.
- Do not deploy the cable for LAN connection outdoor to prevent electronic shock hazards.
- Keep the package out of reach of children.
- When you want to dispose of the device, please follow local regulations on conservation of the environment.

Warranty

We warrant to the original end user (purchaser) that the device will be free from any defects in workmanship or materials for a period of two (1) years from the date of purchase from the dealer. Please keep your purchase receipt in a safe place as it serves as proof of date of purchase. During the warranty period, and upon proof of purchase, should the product have indications of failure due to faulty workmanship and/or materials, we will, at our discretion, repair or replace the defective products or components, without charge for either parts or labor, to whatever extent we deem necessary tore-store the product to proper operating condition. Any replacement will consist of a new or re-manufactured functionally equivalent product of equal value, and will be offered solely at our discretion. This warranty will not apply if the product is modified, misused, tampered with, damaged by an act of God, or subjected to abnormal working conditions. The warranty does not cover the bundled or licensed software of other vendors. Defects which do not significantly affect the usability of the product will not be covered by the warranty. We reserve the right to revise the manual and online documentation and to make changes from time to time in the contents hereof without obligation to notify any person of such revision or changes.

Be a Registered Owner

Web registration is preferred. You can register your device via http://www.draytek.com.

Firmware & Tools Updates

Due to the continuous evolution of DrayTek technology, all devices will be regularly upgraded. Please consult the DrayTek web site for more information on newest firmware, tools and documents.

http://www.draytek.com

Regulatory Information

Federal Communication Commission Interference Statement

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the use is encouraged to try to correct the interference by one of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different form that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

This device complies with Part 15 of the FCC Rules. Operation is subject to the following two conditions:

(1) This device may not cause harmful interference, and

(2) This device may accept any interference received, including interference that may cause undesired operation.

Please visit http://www.draytek.com/user/AboutRegulatory.php.

CE Notice (European Union)

The symbol indicates compliance of this equipment to the EMC Directive and the Low Voltage Directive of the European Union. These markings indicate that this system meets the following technical standards:

- EN 55022 — “Limits and Methods of Measurement of Radio Interference Characteristics of Information Technology Equipment.”
- EN 55024 — “Information technology equipment - Immunity characteristics - Limits and methods of measurement.”
- EN 61000-3-2 — “Electromagnetic compatibility (EMC) - Part 3: Limits - Section 2: Limits for harmonic current emissions (Equipment input current up to and including 16 A per phase).”
- EN 61000-3-3 — “Electromagnetic compatibility (EMC) -Part 3: Limits - Section 3: Limitation of voltage fluctuations and flicker in low-voltage supply systems for equipment with rated current up to and including 16 A.”
- EN 60950 — “Safety of Information Technology Equipment.”

Table of Contents

Chapter 1 Overview

VigorPhone enables you to make phone calls through the IP network instead of calling through a tradition local PSTN line.

It is workable with VigorIPPBX series for auto provision capability. To manage various calling purposes, VigorPhone supports multi-sip registration with different accounts (up to 10) and support G.722 codec for promoting voice quality. The simple WEB UI based configuration allows you to operate VigorIPPBX with ease.

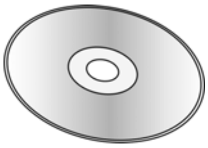
Read this user manual carefully to learn how to operate this product and take advantage of its features.

1.1 Package Contents

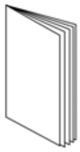
When unpacking the VoIP phone, ensure all the following items are present and undamaged. If anything appears to be missing or broken, contact your dealer for a replacement.

❶ IP Phone

❷ CD

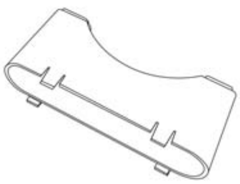


❸ Quick Start Guide

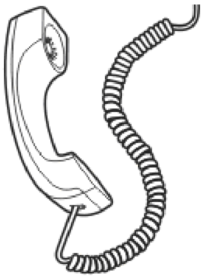


❹ RJ-45 Cable (Ethernet)

❺ Phone Stand /Bracket

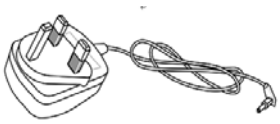


❻ Handset



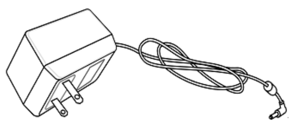
❼

UK-type power adapter



EU-type power adapter

USA/Taiwan-type power adapter

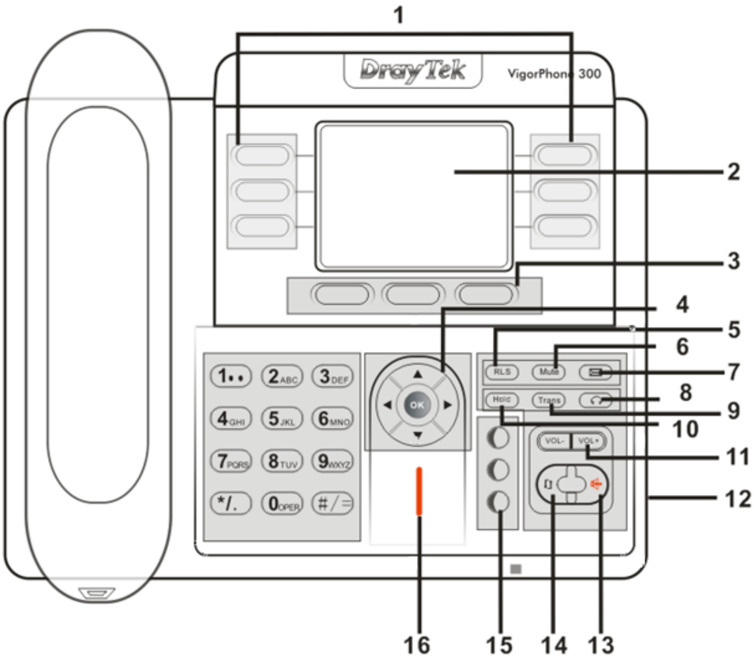


AU/NZ-type Power Adapter

1.2 Product Description

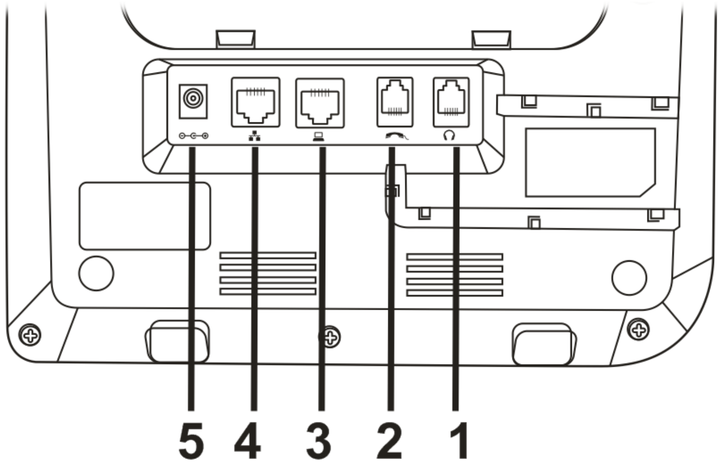
This VoIP Phone features based on SIP (RFC 3261). Please familiarize yourself with the functions of the VoIP phone.

1.2.1 Front View



Item	Name	Description
1	Memory key (1-6)	Users could store their commonly used number in these keys, and call for them as speed dial.
2	Display Screen	Displays calls and status information.
3	Soft key 1/2/3	Keys combination, include functions such as SMS / SDial /PBook /Answer /Conf /Enter /Save /Quit /Edit /Redial and so on.
4	Navigation	Left: Checking Incoming call / Up: Checking Missed Call Right: Checking line status / Down: Checking IP info OK: Enter into the phone's menu
5	Release key	Skip to stand-by mode.
6	Mute	Press this key in calling mode, you can hear the other side, and the other side can not hear you.
7	Envelope	LED inside, if blinks remind user have new voicemail.
8	HeadSet Button	Place and receive calls through an optionally connected headset.
9	Transfer	Use the key to realize blind transfer or attended transfer.
10	Hold	Temporarily hold the active call during the talking.
11	Volume -/+	Turn down or turn up the volume by pressing these two keys
12	Headset Jack	Allow to connect another headset optionally. (Port type: 3.5mm jack)
13	Hands-free	Make the phone into hands-free mode.
14	Redial	● In the hook off /hands-free mode, use the key to dial the last call number. ● In stand-by mode, it has a function to check the OUTGOING CALL.
15	Line1/2/3	Three SIP lines allow you to select any one to make the call, if it has been registered.
16	Indicator light	If the light blinking, indicate the phone has missed call(s).

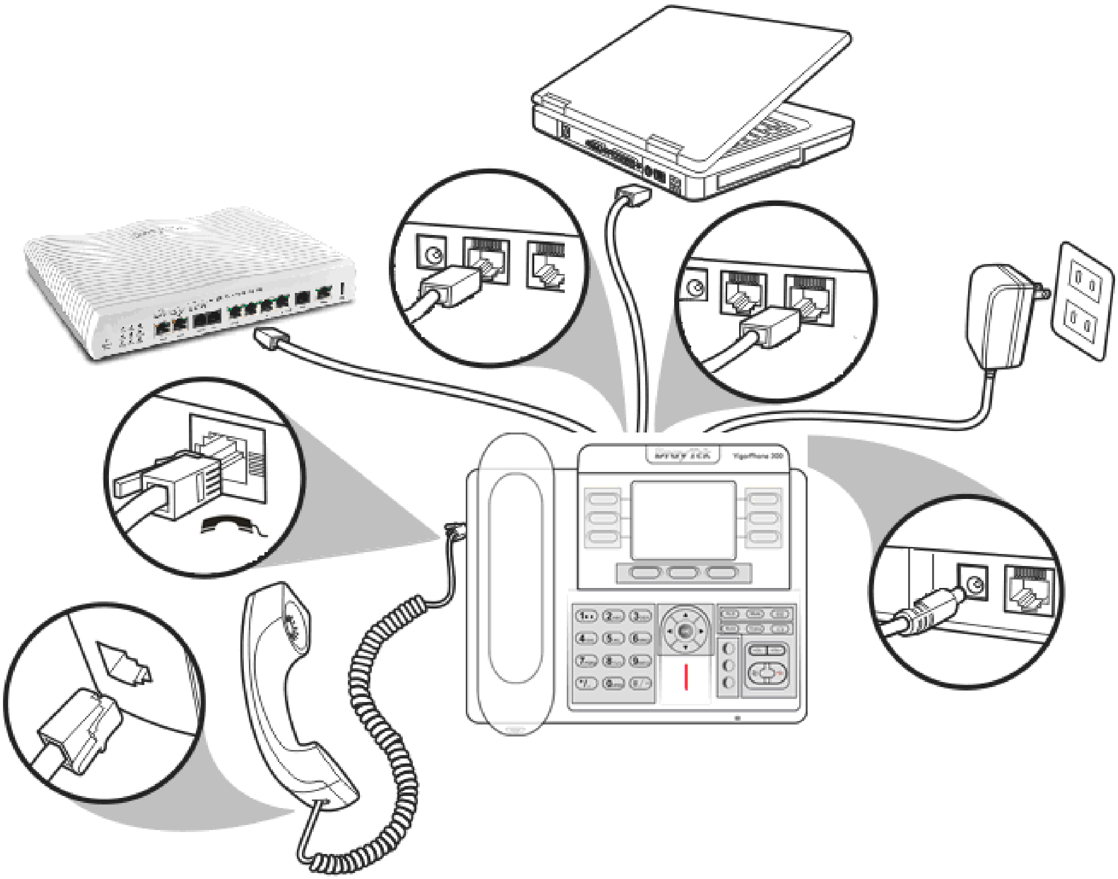
1.2.2 Back View



Item	Name	Description
1	Headset Jack	Connects to an external headset.(RJ-9)
2	Handset Jack	Connects to the phone.(RJ-9)
3	LAN/PC Port	Connects to PC. 10/100Mbps RJ-45 port for PC (downlink) connection. Connects to LAN cable.
4	WAN/ PoE Port	10/100Mbps RJ-45 port for LAN (uplink) connection. If you are using Power over Ethernet (PoE), the power to the phone is supplied when you connect the Ethernet cable. Draws power from either spare line or signal line.
5	Power Jack	Connects to AC power adapter. 5V AC power port.

1.3 Setting Up the Phone

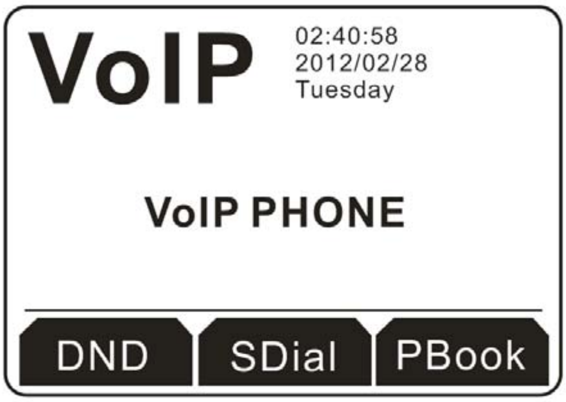
The following illustration shows how to connect the VoIP phone to power, LAN, WAN, and the handset or a headset.



Chapter 2 Display Screen Configuration

The menu directory enables you to setup the product configuration from Phone Settings, VoIP settings, and Network settings. Follow these steps to access the menu and the menu items.

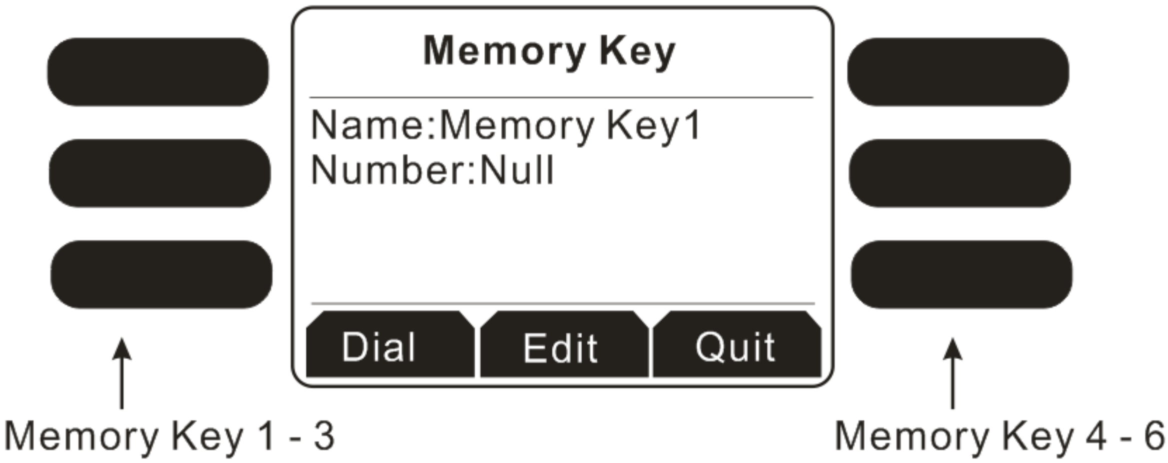
Below shows the LCD of VigorPhone (successful hardware connection):



2.1 Memory Key

Memory keys (also called function keys in the web configurator of Phone>>Function Key) can be set with specific type, value, line and other function parameters (speed dial, push to talk, DND and so on). You can go to Phone>>Function Key to configure the settings in details.

If you just want to edit the name and /or the number for each memory key, you can click one of the memory keys on the IP phone to change it.

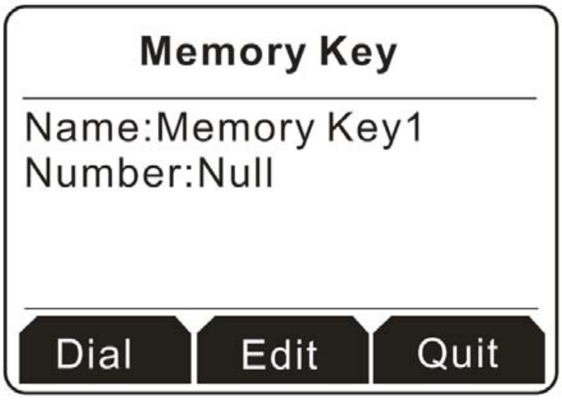


2.1.1 Dialing with Memory Key

Simply press the memory key (1 – 6) you want and click **Dial**.

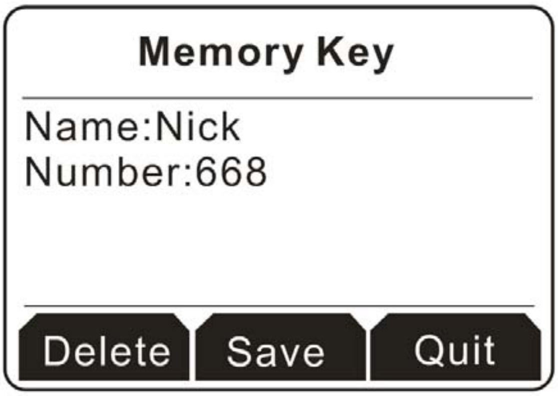
2.1.2 Edit the Memory Key

1. Click one of the memory keys you want. In default, all the telephone numbers will be displayed with **Null** if you haven't created any memory key.



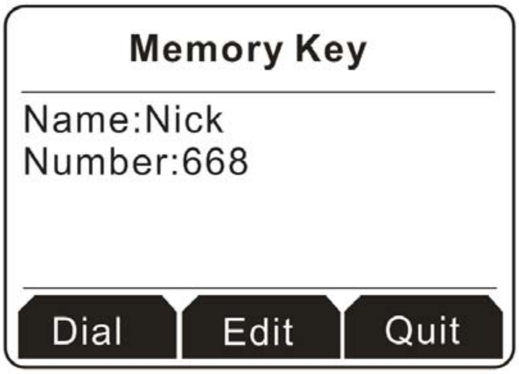
Button	Explanation
Dial	Have a phone call to the selected one.
Edit	Modify the information for the selected one.
Quit	Exit and return to previous page.

2. Click the soft key under **Edit**. The name and the number will be cleared and ask you to type new entries.
3. In the field of Name, please type **Nick**; and in the field of Number, please type **668**.



Button	Explanation
Delete	It allows you to remove the information you type.
Save	Save the information you type.
Quit	Exit and return to previous page.

4. Click the soft key under **Save** to store the settings. Now, memory key 1 has been changed with new name and number.

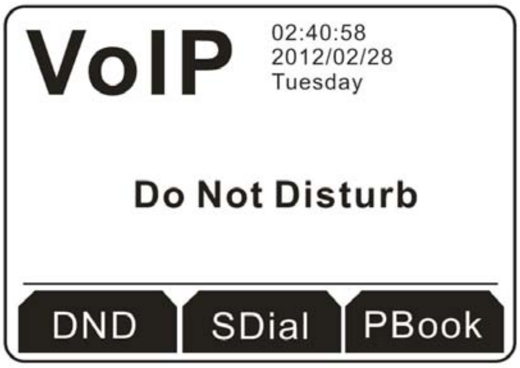


2.1.3 Exit the Memory Key

Simply press the soft key under **Quit** to exit the memory key and return to the home page.

2.2 Do Not Disturb

Simply press the soft key under DND button on the home page. The screen will be shown as below.

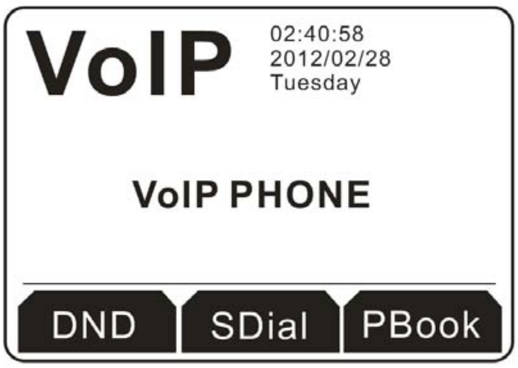


Now, you don't need to worry about the incoming phone calls to interrupt your work.

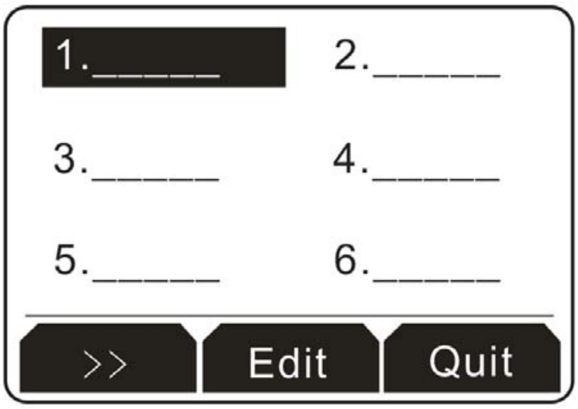
2.3 Speed Dial

Speed dial means user can make calls directly without hook off or using hands-free.

1. Press the soft key under **SDial** to access into the configuration page. There are 12 groups that you can set as speed dial numbers.



2. Use the **Navigation** keys to move up, down, left or right to choose the one you want. In this case, we choose #1 as an example.

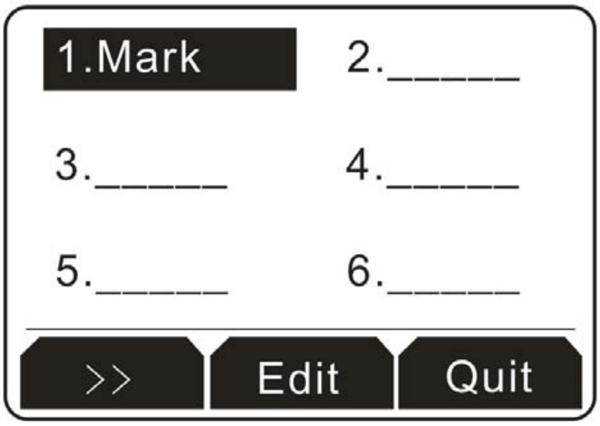


Button	Explanation
>>	Click it to access into next entry.
Edit	Modify the information for the selected one.
Quit	Exit and return to previous page.

3. Next, click the soft key under **Edit** to display the following screen**.** In the field of Name, please type **Mark**; and in the field of Tel, please type **667**.

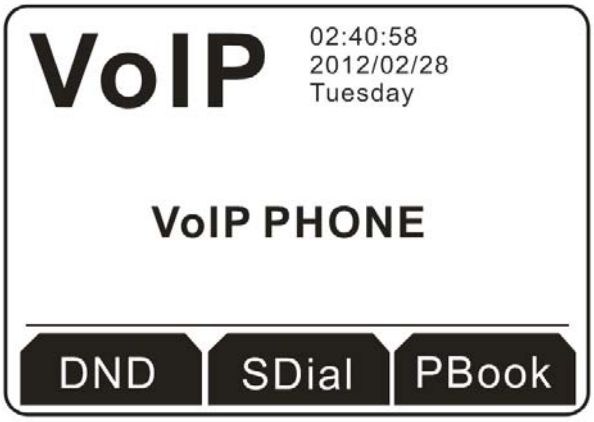
Button	Explanation
Delete	It allows you to remove the information you type.
Save	Save the information you type.
Quit	Exit and return to previous page.

4. Click the soft key under **Save** to store the settings. Now, speed dial # 1 has been changed with new name and number.

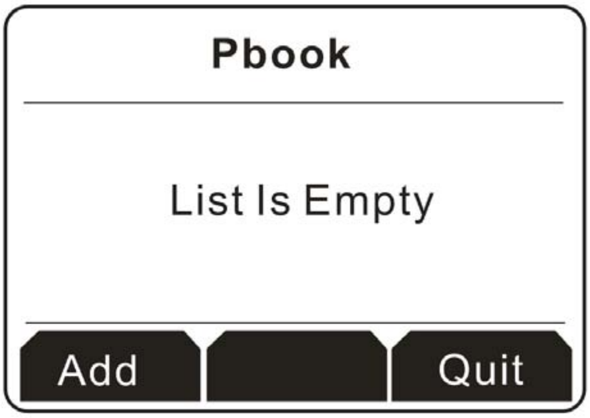


2.4 Phone Book

1. Press the soft key under **PBook** to access into the configuration page.

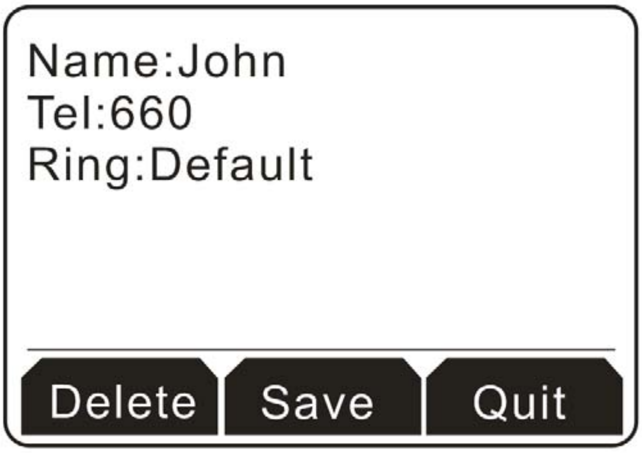


2. For there is no phone book created, the LCD displays the message of “List Is Empty”.



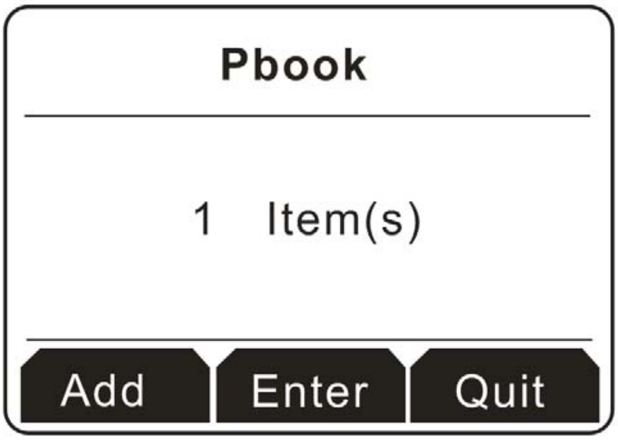
Button	Explanation
Add	It allows you to add a new name and telephone number to the phone book.
Enter	This button is available only when there is at least one item existed. If not, it will be blank.
Quit	Exit and return to previous page.

3. Click the soft key under **Add** to display the following screen. In the field of Name, please type **John**; and in the field of Tel, please type **660**.

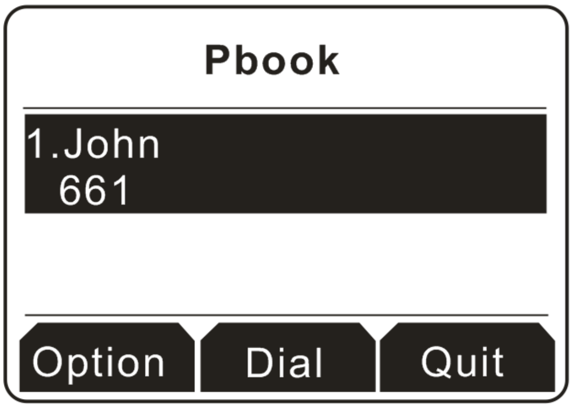


Button	Explanation
Delete	It allows you to remove the information you type.
Save	Save the information you type.
Quit	Exit and return to previous page.

4. Click the soft key under **Save**. When such item is created successfully, the screen will display as the figure below.



5. Click the soft key under **Quit**. You will find a new name with phone number has been created.

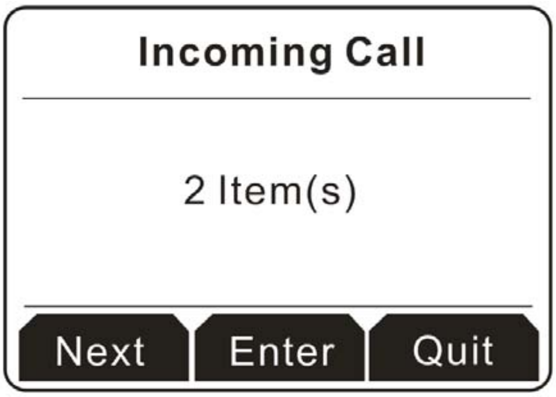


Button	Explanation
Option	It allows you to edit information, save the phone book, delete the phone book, send a message to other people and so on.
Dial	Have a phone call to the selected one.
Quit	Exit and return to previous page.

2.5 Incoming/Outgoing Call

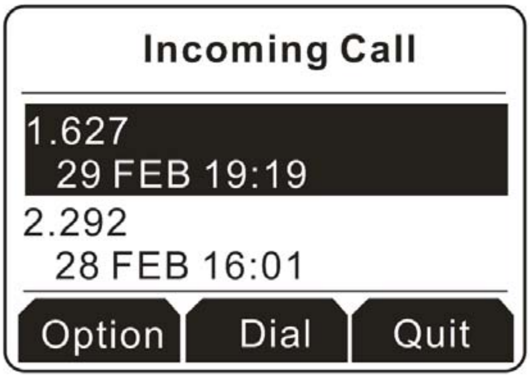
Later incoming/outgoing calls will be stored temporarily and be checked from the Display Screen.

1. Press the navigation key . You will see the incoming call records at the first. If there are many incoming call stored, please use scroll bard on the right side of the display screen to scroll up and down.



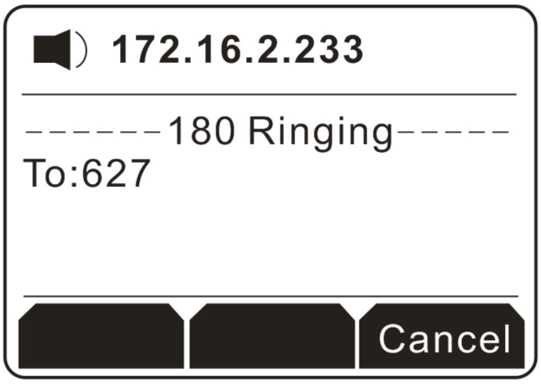
Button	Explanation
Next	Switch among the **incoming call**, **outgoing call** and **missed call**.
Enter	This button is available only when there is at least one item existed.
Quit	Exit and return to previous page.

2. Press the soft key under Enter to access into the next page of incoming call. See the figure below.



Button	Explanation
Option	It allows you to check detailed information for the missed call, save the missed call, delete the missed call, send a message to the missed call, and so on.
Dial	Call back for answering the incoming call.
Quit	Exit and return to previous page.

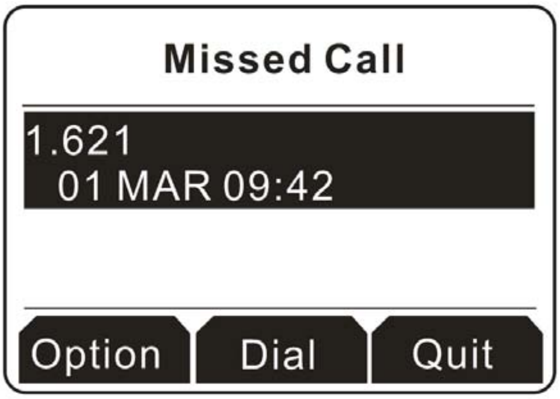
3. Click the soft key under Dial to have a phone call to the selected incoming call.



2.6 Missed Call and Indicator Light

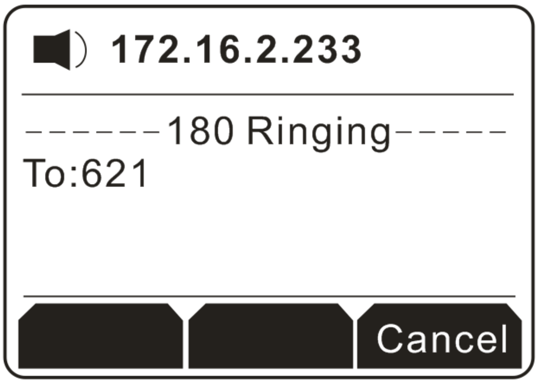
If the indicator light **always blinks**, it means there is a missed call that you have to process. Please do the following:

1. Press the navigation key  to open the missed call record. All the missed calls will be shown on the display screen. In this example, there is only one missed call.



Button	Explanation
Option	It allows you to check detailed information for the missed call, save the missed call, delete the missed call, send a message to the missed call, and so on.
Dial	Call back for answering the missed call.
Quit	Exit and return to previous page.

2. Click the soft key under Dial to have a phone call to the selected missed call.



Chapter 3 Web Configuration

This chapter contains important information to help you configure the settings for your VoIP phone from the web browser.

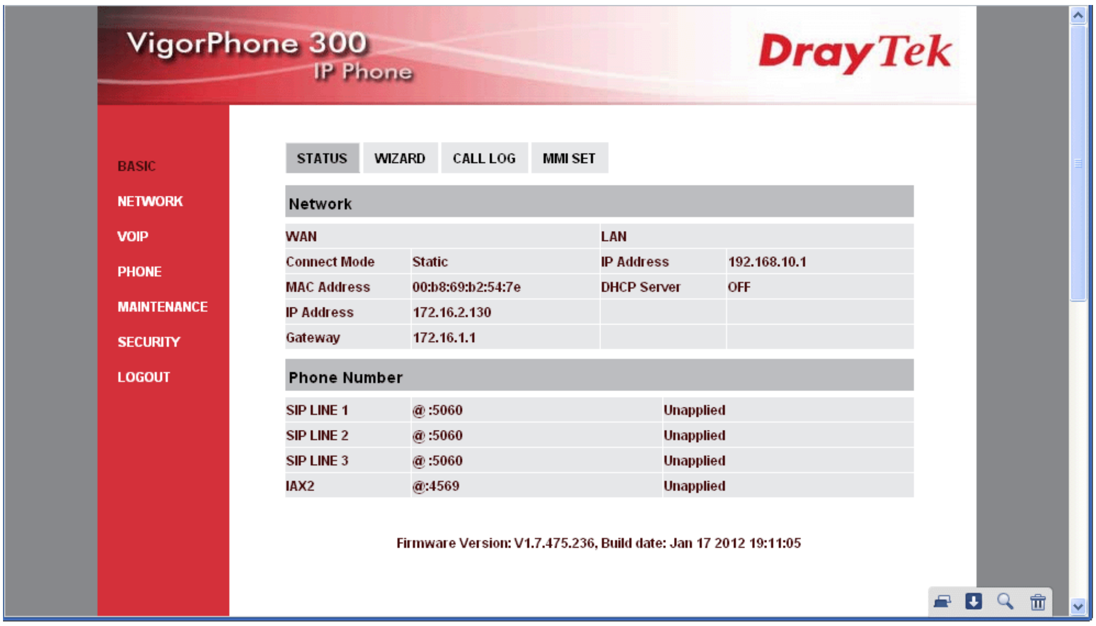
If your VoIP phone is using factory default, it sets LAN/PoE port as DHCP client and enable Bridge mode for PC port. To access the web configuration menu, do the following:

1. Connect one end of the Ethernet cable provided to the **LAN/PoE** port of your phone to your router with DHCP service enable.
2. Connect one end of the network cable to the PC port of your phone, connecting to your personal computer.
3. Plug in the power of the VoIP phone. Select the **Menu** soft key.
4. Select **Network, LAN Port Settings**, and then press the **Info** soft key.

You should be able to see the IP address displayed on the LCD screen.

Open your browser (such as Internet Explorer, Firefox, etc.) and type in the web address of the phone. For example, if the IP address you obtain in step 4 above is 192.168.1.2, enter the web address: http://192.168.1.2.

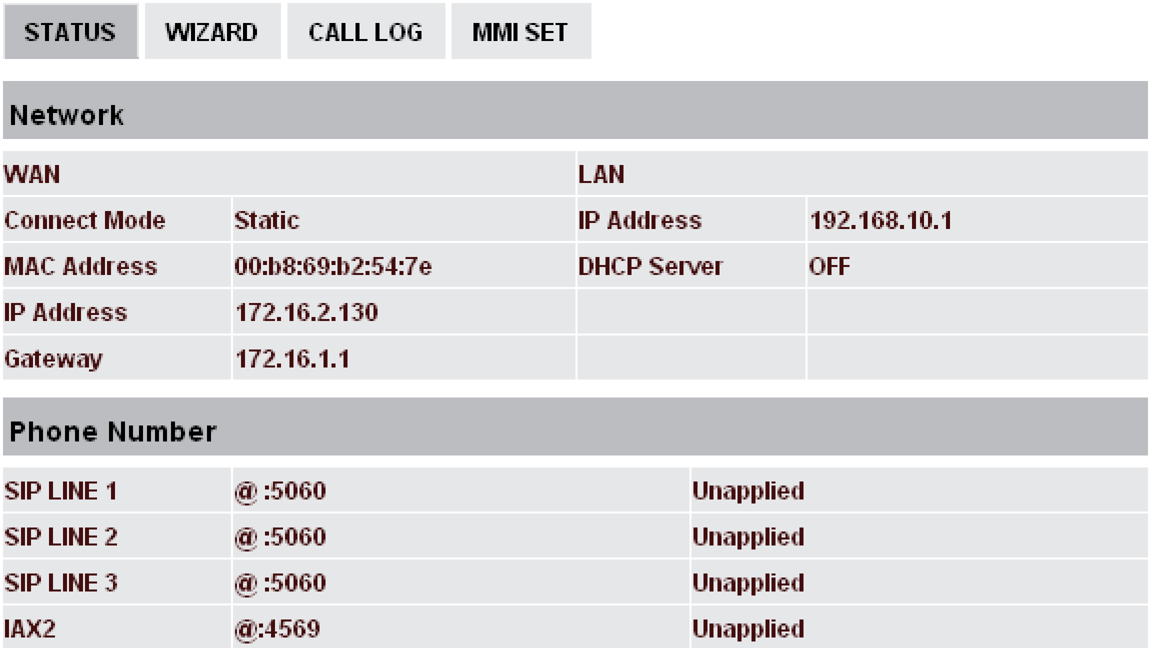
The Web login front page is displayed. Enter the user name (“**admin**”) and the password (“**admin**”) and click **Login**.



After you login, move the cursor over the menu items on the left navigation bar to access the dropdown menus.

3.1 Basic

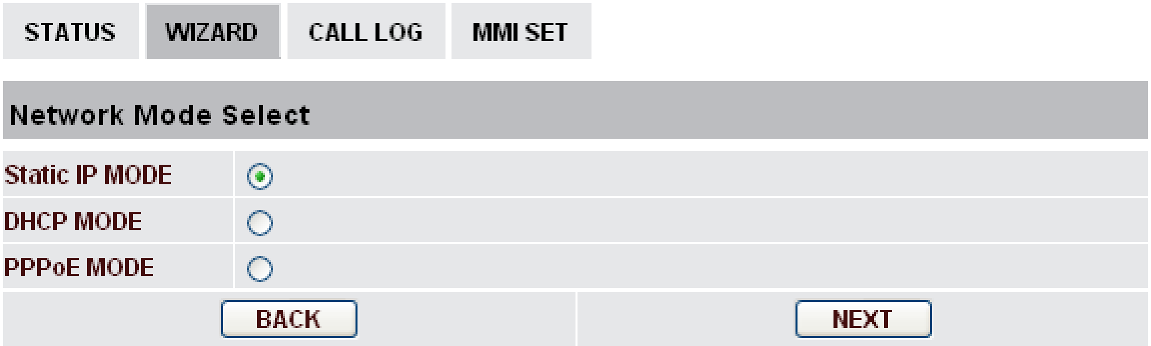
3.1.1 Status



Field name	Explanation
Network	Shows the configuration information on WAN and LAN port, including the connect mode of WAN port (Static, DHCP, PPPoE), MAC address, the IP address of WAN port and LAN port, ON or OFF of DHCP mode of LAN port.
Phone Number	Shows the phone numbers provided by the SIP LINE 1-3 servers and IAX2. The last line shows the version number and issued date.

3.1.2 Wizard

Please select the proper network mode according to the network condition. VigorPhone provides three different network settings.



Field name	Explanation
Static IP Mode	If your ISP server provides you the static IP address, please select this mode, then finish Static Mode setting. If you don't know about parameters of Static Mode setting, please ask your ISP for them.
DHCP	In this mode, you will get the information from the DHCP server automatically; need not to input this information artificially.
PPPoE	In this mode, your must input your ADSL account and password.

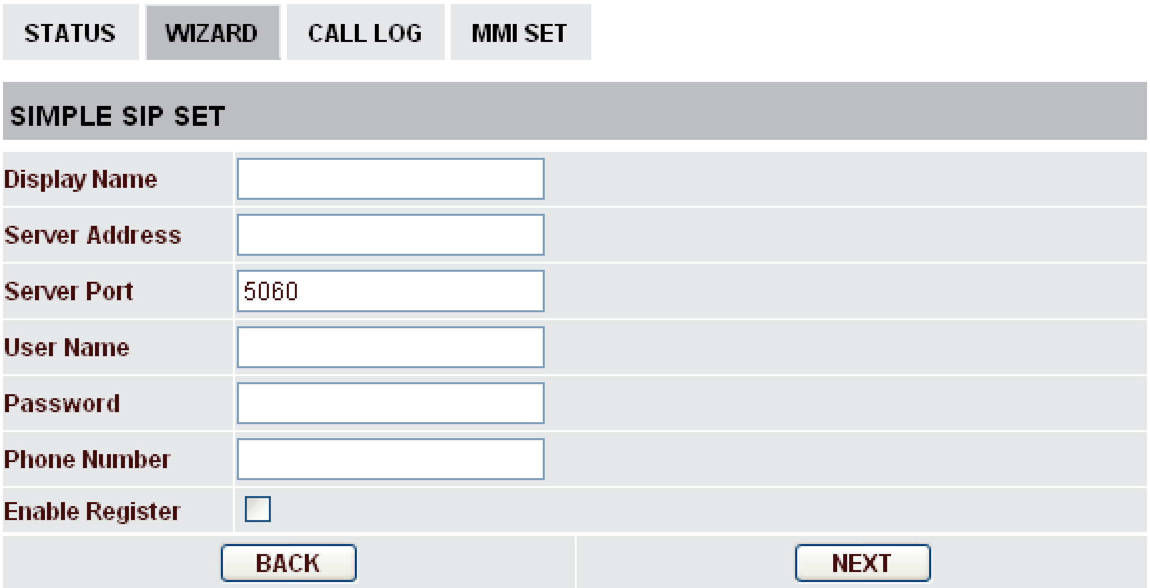
Static IP Mode

1. Choose **Static IP Mode** and click **Next**. You can get the following web page.

Field name	Explanation
Static IP Address	Input the IP address distributed to you.
Netmask	Input the Netmask distributed to you.

Gateway	Input the Gateway address distributed to you.
DNS Domain	Set DNS domain postfix. When the domain which you input can not be parsed, phone will automatically add this domain to the end of the domain which you input before and parse it again.
Primary DNS	Input your primary DNS server address.
Alter DNS	Input your standby DNS server address.
Back	Return to the last page.
Next	Get into the next page.

2. After finished the above settings, click **Next** to open the following page.



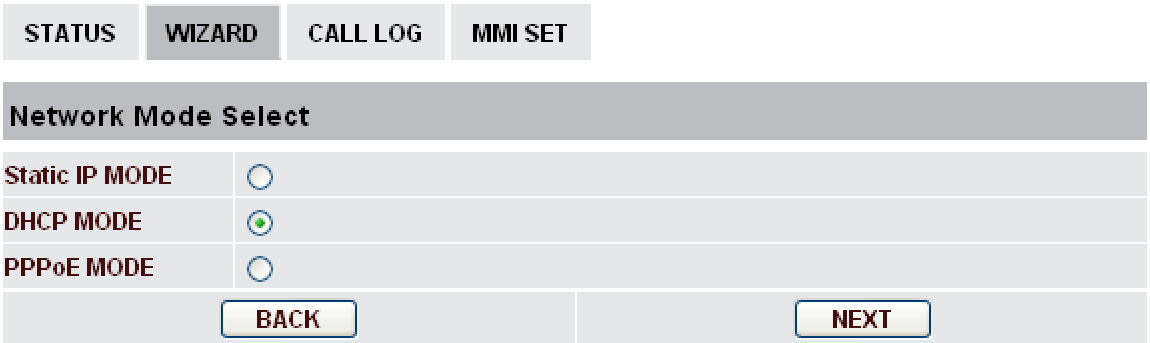
Field name	Explanation
Display Name	Set the display name.
Server Address	Input your SIP server address.
Server Port	Set your SIP server port.
User Name	Input your SIP register account name.
Password	Input your SIP register password.
Phone Number	Input the phone number assigned by your VOIP service provider.
Enable Register	Start to register or not by selecting it or not.
Back	Return to the last page.
Next	Get into the next page.

3. After finished the above settings, click **Next** to open the following page.

4. Click **Finish** to complete the configuration.

DHCP Mode

1. Choose **DHCP Mode** and click **Next**. You can get the following web page.



2. After finished the above settings, click **Next** to open the following page.

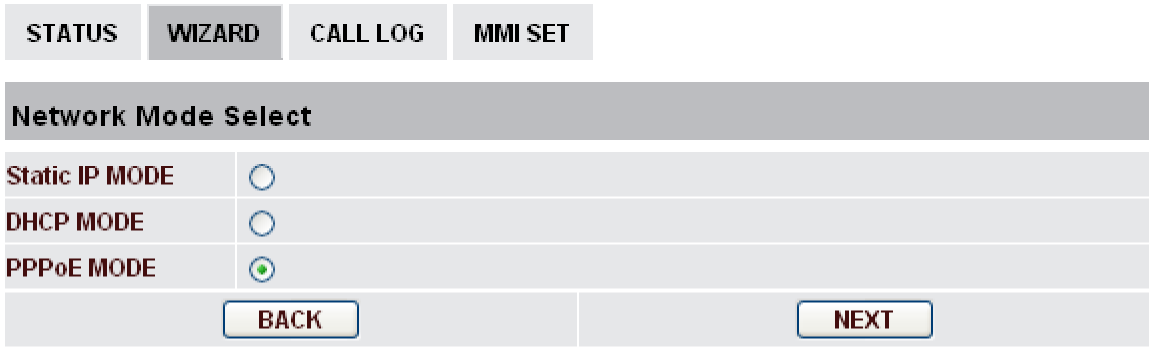
Field name	Explanation
Display Name	Set the display name.
Server Address	Input your SIP server address.
Server Port	Set your SIP server port.
User Name	Input your SIP register account name.
Password	Input your SIP register password.
Phone Number	Input the phone number assigned by your VOIP service provider.
Enable Register	Start to register or not by selecting it or not.

3. After finished the above settings, click **Next** to open the following page.

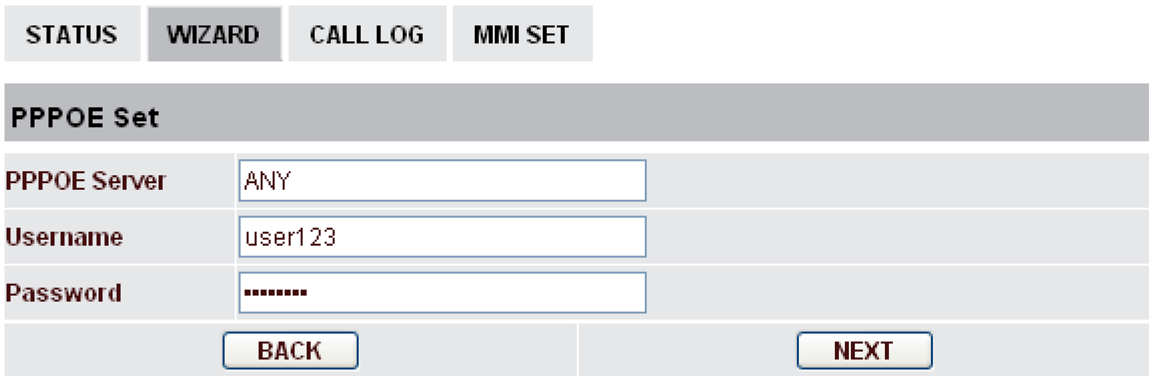
4. Click **Finish** to complete the configuration.

PPPoE Mode

1. Choose **PPPoE Mode** and click **Next**. You can get the following web page.

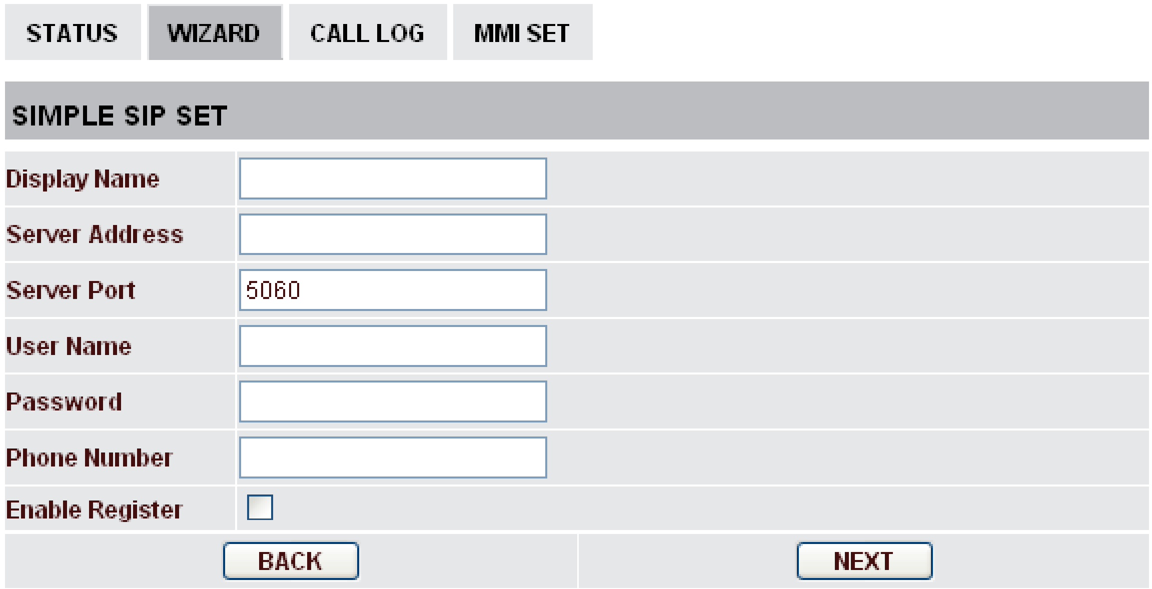


2. After finished the above settings, click **Next** to open the following page.



Field name	Explanation
PPPoE Server	It will be provided by ISP.
Username	Input your ADSL account.
Password	Input your ADSL password.

3. After finished the above settings, click **Next** to open the following page.



Field name	Explanation
Display Name	Set the display name.
Server Address	Input your SIP server address.
Server Port	Set your SIP server port.
User Name	Input your SIP register account name.
Password	Input your SIP register password.
Phone Number	Input the phone number assigned by your VOIP service provider.
Enable Register	Start to register or not by selecting it or not.

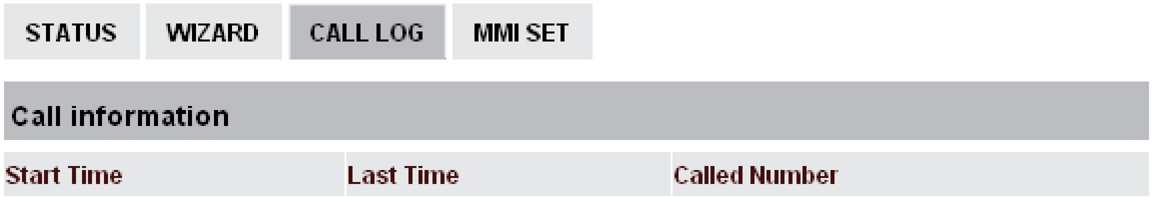
4. After finished the above settings, click **Next** to open the following page.

5. Click **Finish** to complete the configuration.

IP Phone will save the setting automatically and reboot. After reboot, you can dial by the SIP account.

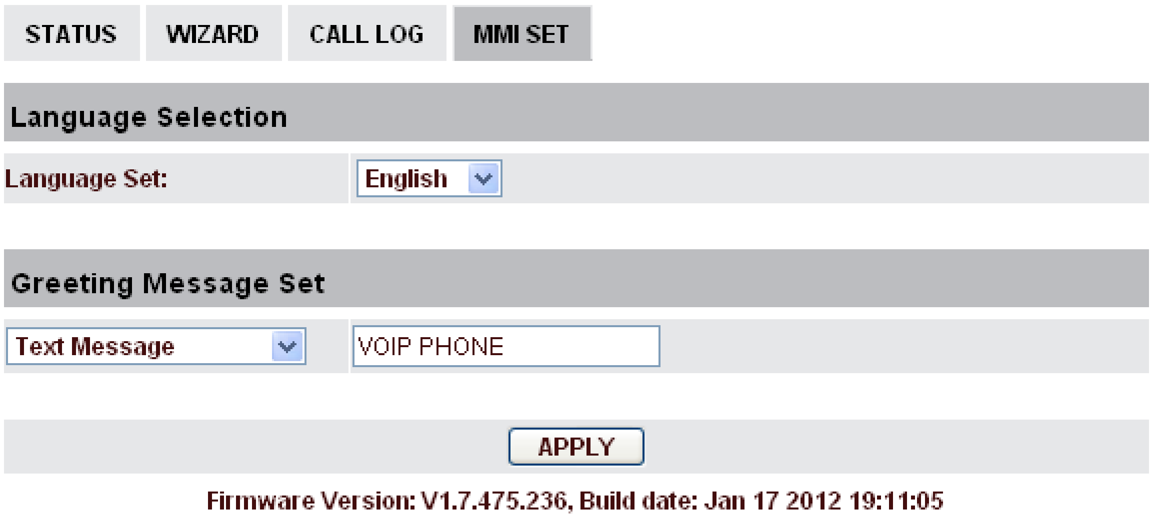
3.1.3 Call Log

You can query all the outgoing through this page.



Field name	Explanation
Start Time	Display the start time of the outgoing record.
Last Time	Display the conversation time of the outgoing record.
Called Number	Display the account/protocol/line of the outgoing record.

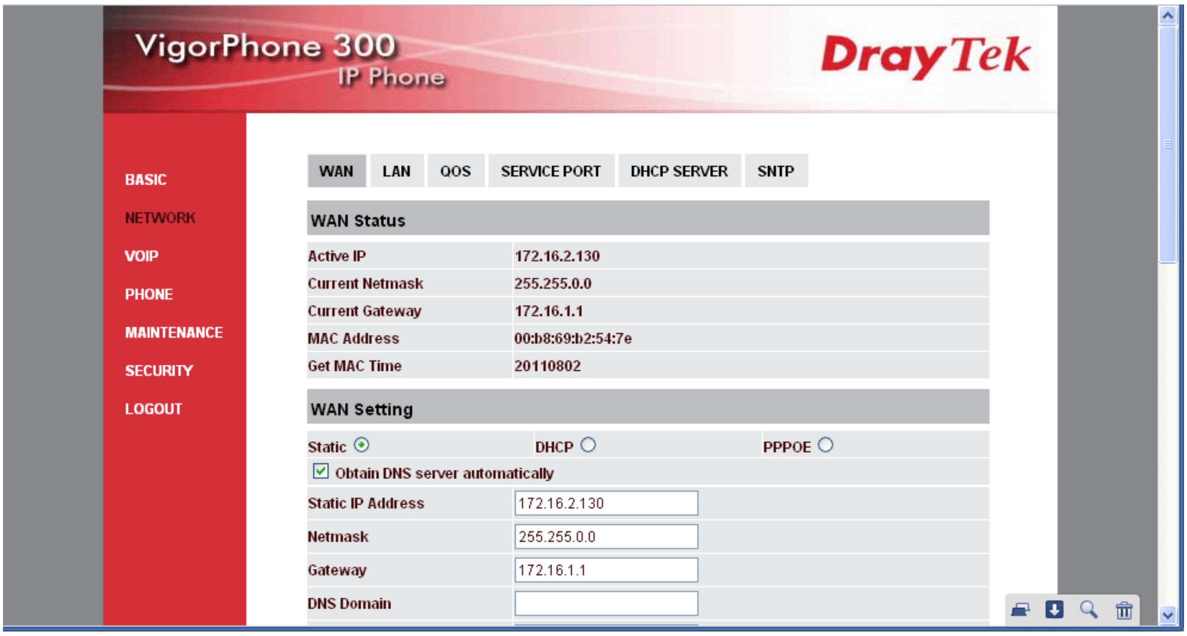
3.1.4 MMI Set



Field name	Explanation
Language Set	Set the language of phone, English is default. 
Text Message	The greeting message will display on LCD when phone is idle. It can support 16 chars. The default chars are “VOIP PHONE”.

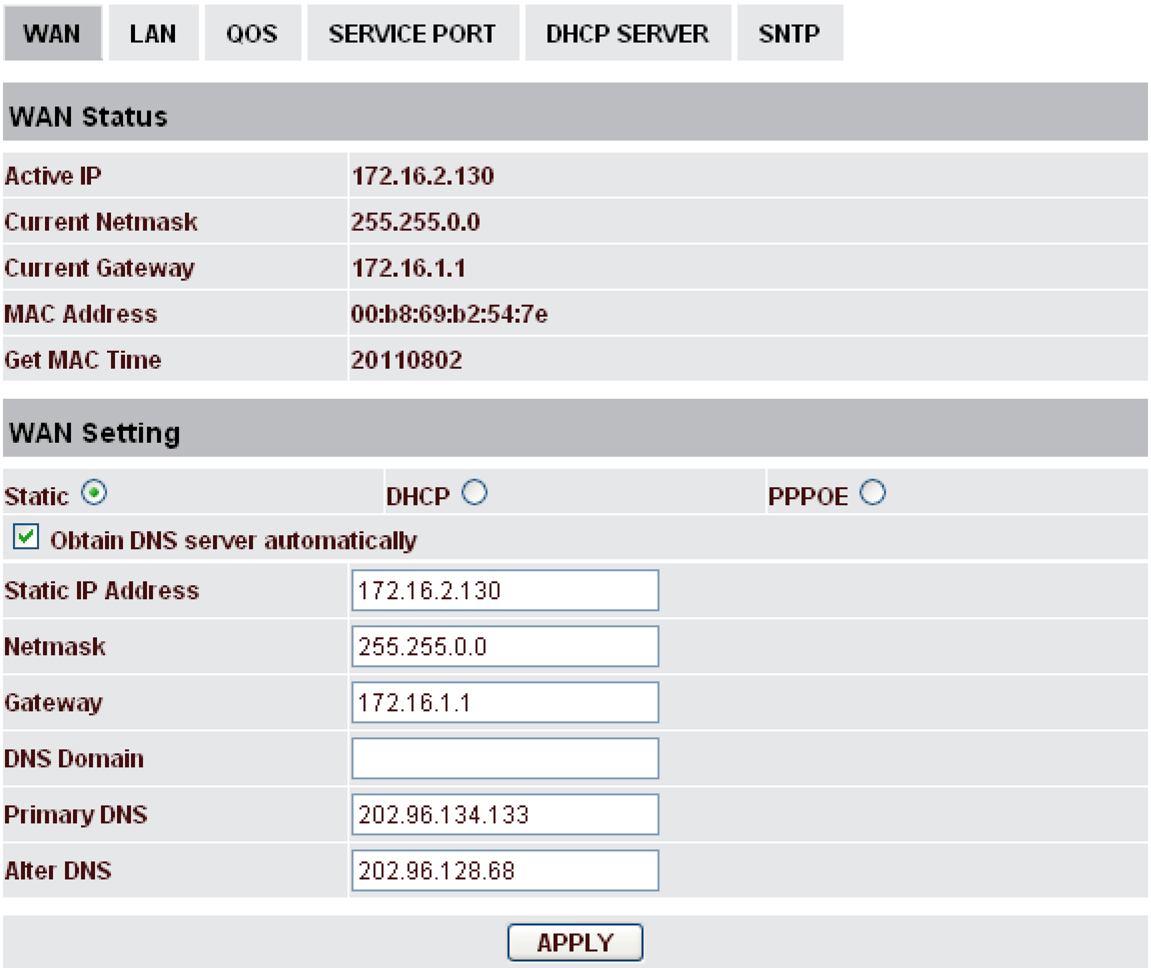
	
Line Info	In the standby screen showing the registration number of lines, when the time is displayed as NULL is not registered.

3.2 Network



3.2.1 WAN

Please select the proper network mode according to the network condition. Vigor router provides three different network settings (Static, DHCP and PPPoE).

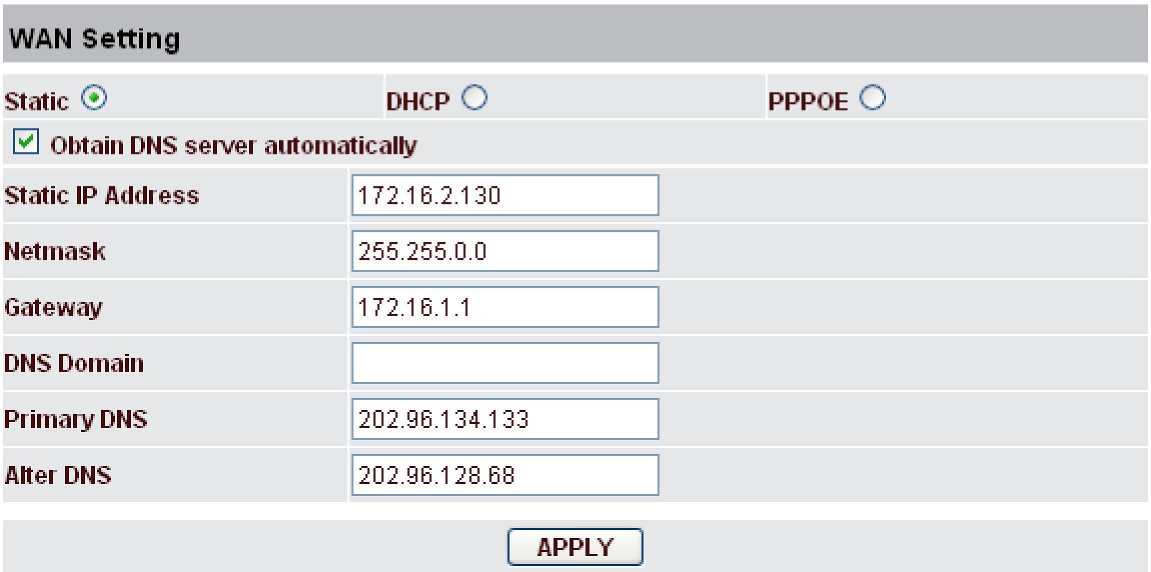


WAN Status

Field name	Explanation
Active IP	The current IP address of the phone.
Current Netmask	The current Netmask address.
MAC Address	The current MAC address of the phone.
Current Gateway	The current Gateway IP address.
Get MAC Time	Shows the time of getting MAC address

WAN Setting - Static

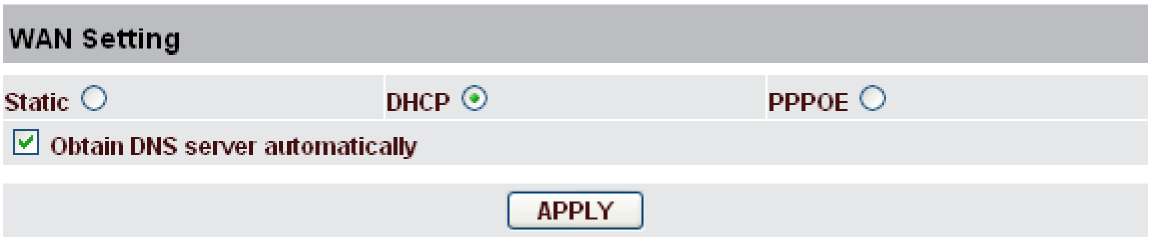
If your ISP server provides you the static IP address, please select **Static** and finish related setting. If you don’t know about parameters of Static Mode setting, please ask your ISP for them.



Field name	Explanation
Obtain DNS server automatically	Select it to use DHCP mode to get DNS address, if you don’t select it, you will use static DNS server. The default is selecting it.
IP Address	Input the IP address distributed to you.
Netmask	Input the Netmask distributed to you.
Gateway	Input the Gateway address distributed to you.
DNS Domain	Set DNS domain postfix. When the domain which you input can not be parsed, phone will automatically add this domain to the end of the domain which you input before and parse it again.
Primary DNS	Input your primary DNS server address.
Alter DNS	Input your standby DNS server address.
Apply	Save the settings.

WAN Setting - DHCP

If you uses DCHP mode, you will get the information from the DHCP server automatically. You don’t need to input this information artificially.



Field name	Explanation
Apply	Save the settings.

WAN Setting - PPPoE

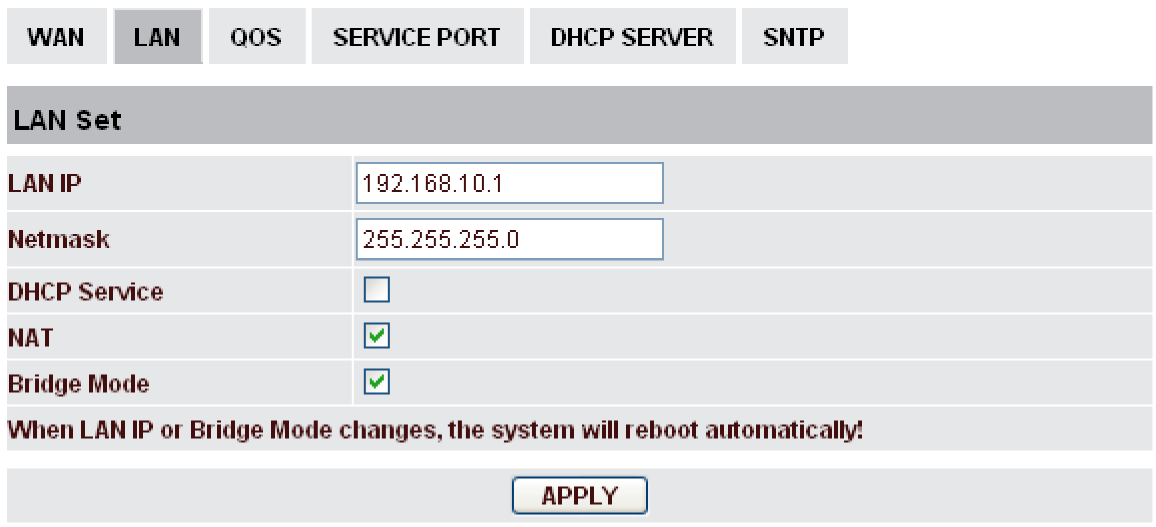
If you uses PPPoE mode, you need to make the following settings.

Field name	Explanation
PPPoE Server	It will be provided by ISP.
Username	Input your ADSL account.
Password	Input your ADSL password.
Apply	Save the settings.

Notice:

1. Click Apply button after finished your setting. IP Phone will save the setting automatically and new setting will take effect.
2 If you modify the IP address, the web page will not response by the old IP address. Your need input new IP address in the address column to logon in the phone.
3. If networks ID which is DHCP server distributed is same as network ID which is used by LAN of system, the system will use the DHCP IP to set WAN, and modify LAN’s networks ID(for example, system will change LAN IP from 192.168.10.1 to 192.168.11.1) when it uses DHCP client to get IP in startup. If the system uses DHCP client to get IP in running status and network ID is also same as LAN’s, the system will refuse to accept the IP to configure WAN. So WAN’s active IP will be 0.0.0.0.

3.2.2 LAN



Field name	Explanation
LAN IP	Specify LAN static IP.
Netmask	Specify LAN Netmask.
DHCP Service	Select the DHCP server of LAN port or not. After you modify the LAN IP address, phone will amend and adjust the DHCP Lease Table and save the result amended automatically according to the IP address and Netmask. You need restart the phone and the DHCP server setting will take effect.
NAT	Select NAT or not.
Bridge Mode	Select Bridge Mode or not. If you select Bridge Mode, the phone will no longer set IP address for LAN physical port. LAN and WAN will join in the same network. Click Apply, the phone will reboot. If you choose the bridge mode, the LAN configuration will be disabled.

3.2.3 QOS

The VOIP phone support 802.1Q/P protocol and DiffServ configuration. VLAN functionality can use different VLAN IDs by setting signal/voice VLAN and data VLAN. The VLAN application of this phone is very flexible.

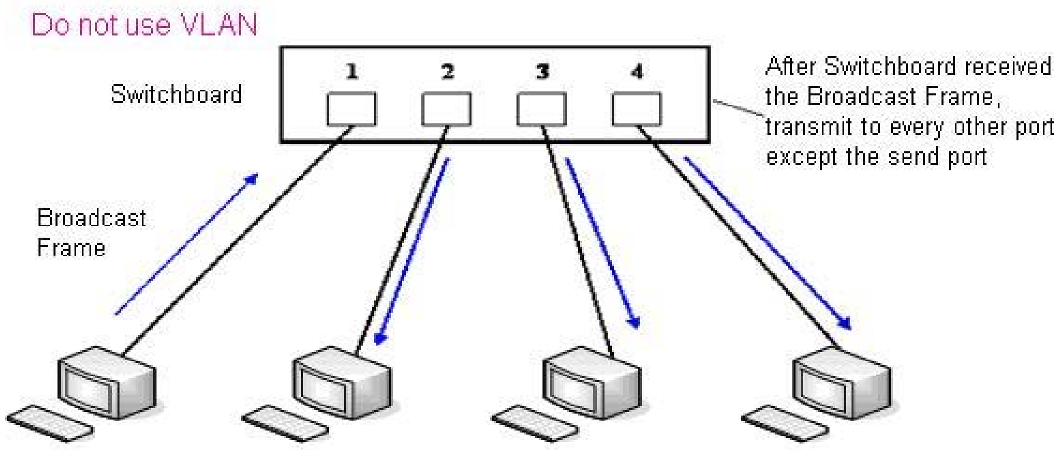


Chart 1

In chart 1, there is a switch without setting VLAN. Any broadcast frame will be transmitted to the other ports except the send port. For example, a broadcast information is sent out from port 1 then transmitted to port 2,3 and 4.

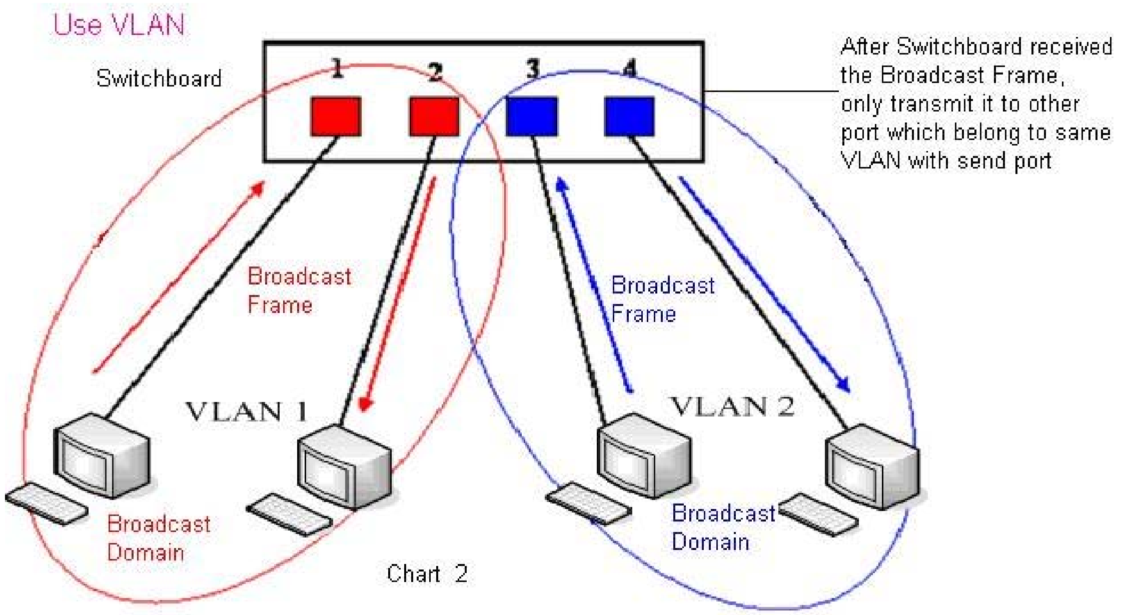
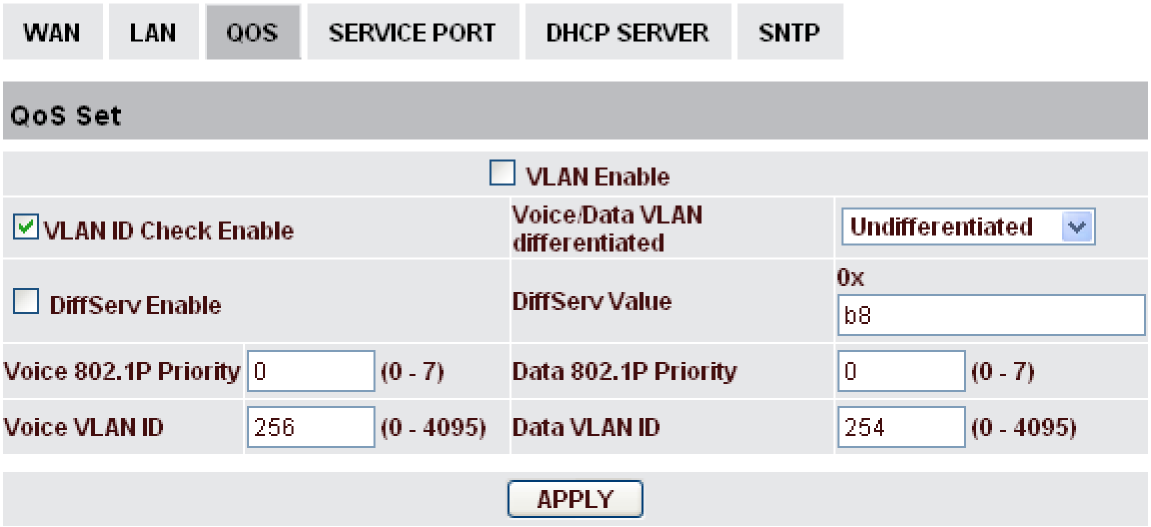


Chart 2

In chart 2, red and blue circles indicate two different VLANs in the switch, and port 1 and port 2 belong to red VLAN, port 3 and port 4 belong to blue VLAN. If a broadcast frame is sent out from port 1, the switch will transmit it to port 2, the other port in the red VLAN and not transmit it to port3 and port 4 in blue VLAN. By this means, VLAN divides the broadcast domain via restricting the range of broadcast frame transmission.

Note: Chart 2 uses red and blue to identify the different VLANs; but in practice, VLAN uses different VLAN IDs to identify them.



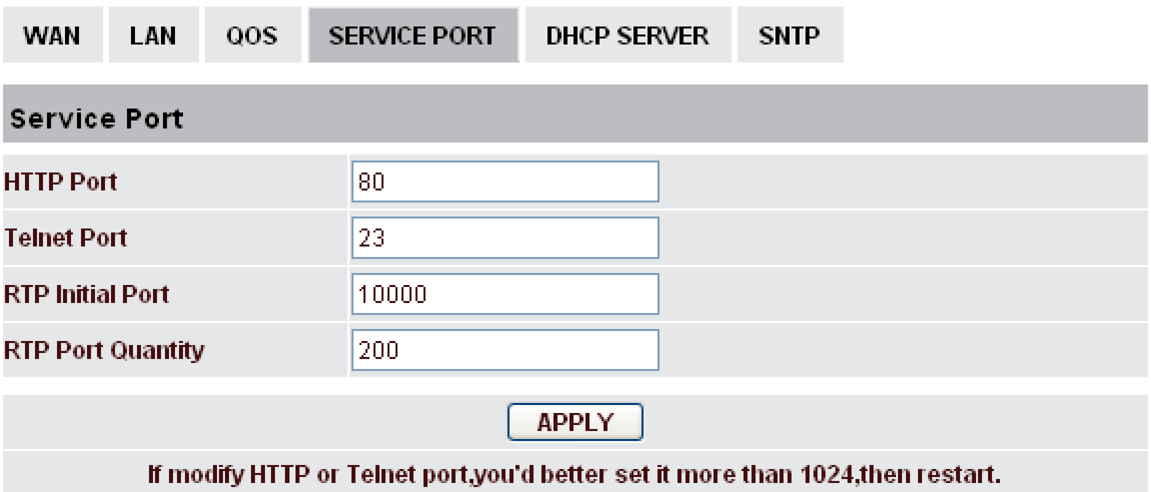
Field name	Explanation
VLAN Enable	Before select it to enable VLAN, you need enable Bridge mode in LAN configuration.
VLAN ID Check Enable	Enable VLAN ID check by selecting it. After enable VLAN ID check, if VLAN ID of a data package is not the same with the phone or a data package do not have VLAN ID, the data package will be discarded.
Voice/Data VLAN differentiated	After enable VLAN, system will set packets with different type of VLAN ID. Undifferentiated means after using VLAN, both VoIP packets and other data packets will use the voice VLAN ID; tag differentiated means after using VLAN, VoIP(signal and voice) packets will add voice VLAN ID, and other data packets will add data VLAN ID; data untagged means after using VLAN, only VoIP packets will add voice VLAN ID. Other data packets will not use VLAN. 
DiffServ Enable	Select it or not to Enable or disable DiffServ.
DiffServ Value	Set DiffServ value, the common value is 0x00.
Voice 802.1P Priority	Specify 802.1P Priority of voice/signal data package.
Data 802.1P Priority	Set 802.1p of data VLAN. Non-VoIP data (such as http, telnet, ping etc) will use this value to set VLAN package.
Voice VLAN ID	Set VLAN ID of voice/signal data package.
Data VLAN ID	Set 802.1q of data VLAN ID. Non-VoIP data (such as http, telnet, ping etc) will use this value to set VLAN package.
Apply	Save the settings.

NOTICE：

1. Startup VLAN, if set Voice/Data VLAN differentiated as Undifferentiated, all packets will use the Voice VLAN ID as the tag.
2. Startup VLAN, if set Voice/Data VLAN differentiated as tag differentiated and disables the DiffServ, then system will not distinguish the voice and data, all packets will use the Voice VLAN ID as the tag.
3. Startup VLAN, if set Voice/Data VLAN differentiated as tag differentiated and enables the DiffServ, then system will distinguish the voice and data and add the VLAN ID each other.
4. Startup VLAN, if set Voice/Data VLAN differentiated as data untagged, then the packet of the signal/voice will use the Voice VLAN ID as the tag, but the data packets will not take the VLAN tag.
5. If Disable the VLAN, regardless to set the Voice/Data VLAN differentiated or not, all packets will not take the VLAN tag; If enable the DiffServ, all packets will only take the DiffServ value.
6. One must to notice, enable the VLAN ID Check Enable that is default, If enable it, the phone will match the VLAN ID strictly. When others' VLAN ID does not match with us, the packets will discard. Contrarily, the phone will accept the packets with the distinct VLAN ID.
7. You must gain the IP with the Static mode when you set VLAN, otherwise can't gain the IP in the VLAN and also can not dial with point to point.

3.2.4 Service Port

You can set the port of telnet/HTTP/RTP by this page.



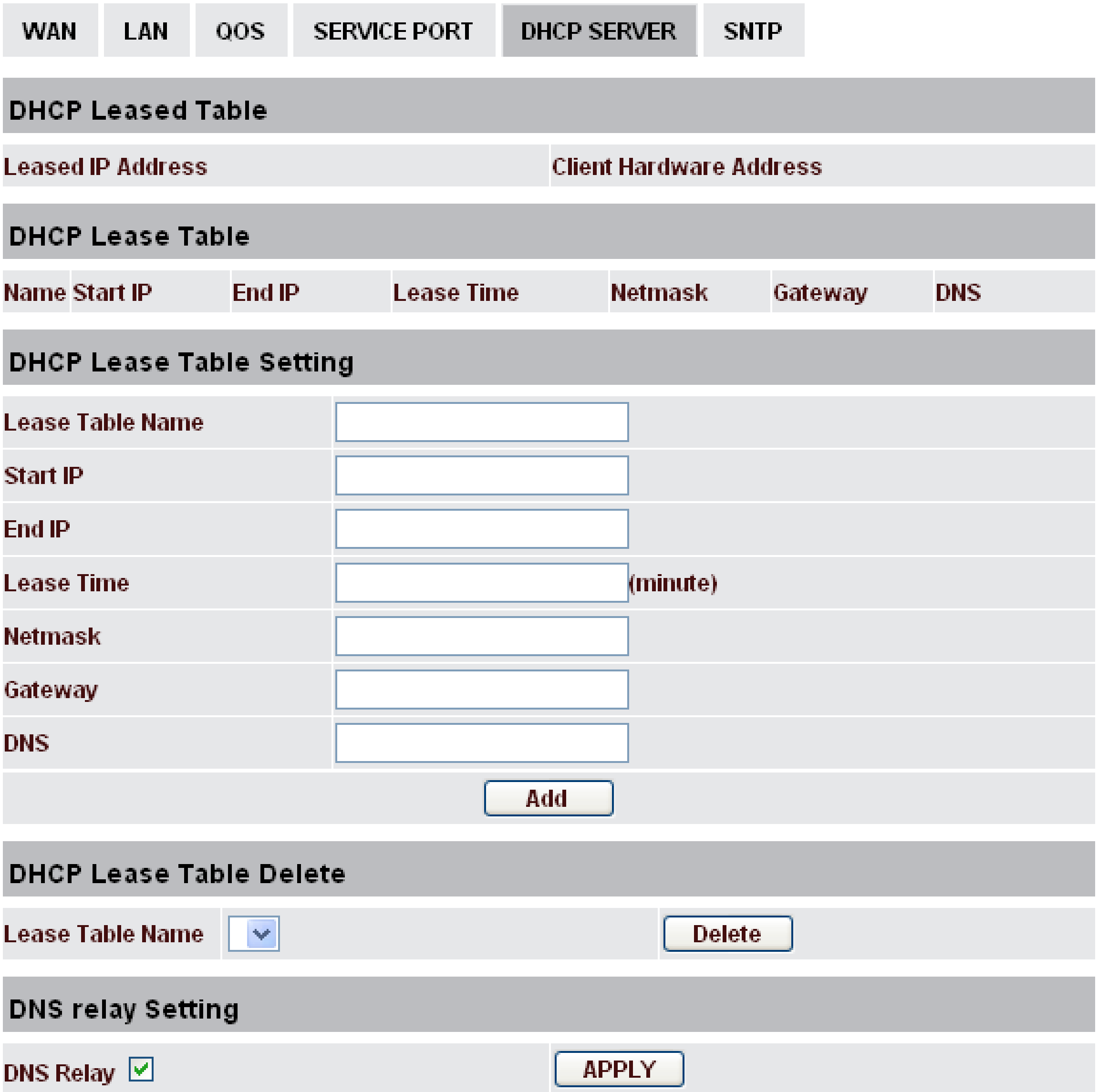
Field name	Explanation
HTTP Port	set web browse port, the default is 80 port，if you want to enhance system safety，you'd better change it into non-80 standard port； Example: The IP address is 192.168.1.70. and the port value is 8090, the accessing address is http://192.168.1.70:8090
Telnet Port	Set Telnet Port, the default is 23. You can change the value into others. Example: The IP address is 192.168.1.70. the telnet port value is 8023, the

	accessing address is telnet 192.168.1.70 8023
RTP Initial Port	Set the RTP Initial Port. It is dynamic allocation.
RTP Port Quantity	Set the maximum quantity of RTP Port, the default is 200.

Notice:

1. You need save the configuration and reboot the phone after set this page.
2. If you modify the port of Telnet and HTTP, you would better set the value more than 1024 because the port value less than 1024 is system port reserved.
3. If you set 0 for the HTTP port, it will disable HTTP service.

3.2.5 DHCP Server



Field name	Explanation
DHCP Leased Table	IP-MAC mapping table. If the LAN port of the phone connects to a device, this table will show the IP and MAC address of this device.

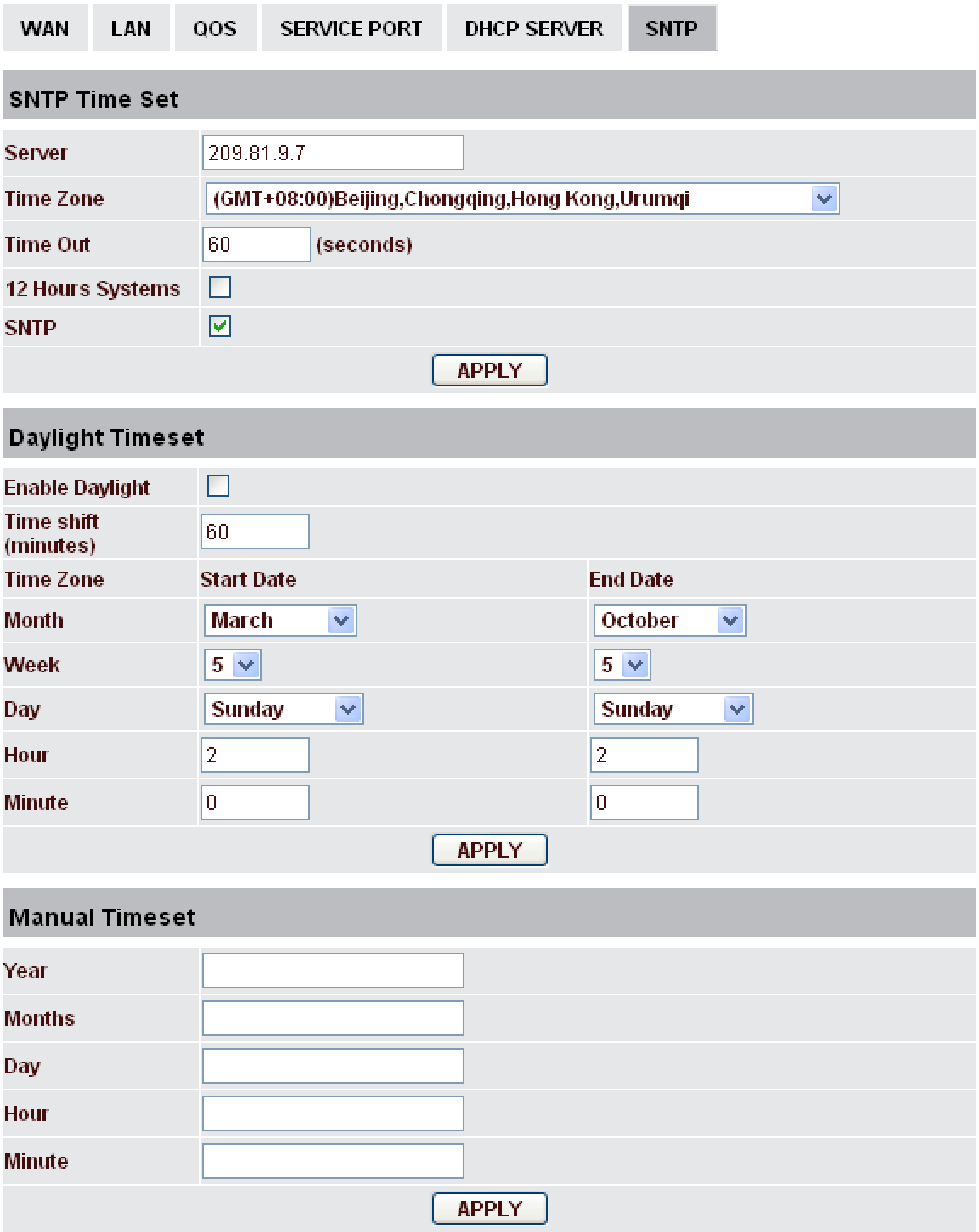
DHCP Lease Table	Shows the DHCP Lease Table, the unit of Lease time is Minute.
DHCP Lease Table Setting	Allow to set corresponding settings for DHCP lease settings
Lease Table Name	Specify the name of the lease table
Start IP	Set the start IP address of the lease table
End IP	Set the end IP address of the lease table, the network device connected to LAN port will get IP address between Start IP and End IP by DHCP.
Netmask	Set the Netmask of the lease table
Gateway	Set the Gateway of the lease table
Lease Time	Set the Lease Time of the lease table
DNS	Set the default DNS server IP of the lease table; Click the **Add** button to submit and add this lease table
DHCP Lease Table Delete	Lease Table Name - Select name of lease table, click the **Delete** button will delete the selected lease table from DHCP lease table.
DNS Relay	Select DNS Relay. The default is enabled. Click the **Apply** button to become effective.
Apply	Save the settings.

Notice:

1. The size of lease table can not be larger than the quantity of C network IP address. We recommend you to use the default lease table and not modify it.
2. If you modify the DHCP lease table, you need save the configuration and reboot.

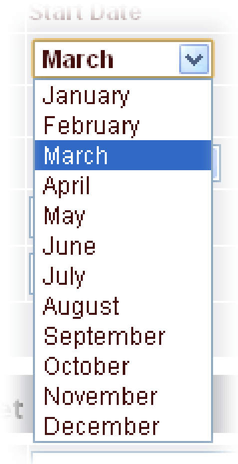
3.2.6 SNTP

Setting time zone and SNTP (Simple Network Time Protocol) server according to your location, you can also manually adjust date and time in this web page.

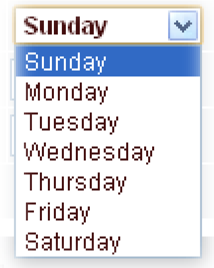


Field name	Explanation
Server	Set SNTP Server IP address.
Time Zone	Select the Time zone according to your location.
Time Out	Set the time out, the default is 60 seconds.

<table>
<tr><td>12 Hours Systems</td><td>Switch the time mechanism between 12 hours and 24 hours.
Default is 24 hours mode.</td></tr>
<tr><td>SNTP</td><td>Select the SNTP, and click Apply to make the SNTP Times effective.</td></tr>
<tr><td>Enable Daylight</td><td>Enable daylight saving time.</td></tr>
<tr><td>Time shift (minutes)</td><td>Setup the variety length.</td></tr>
<tr><td>Month</td><td>Setup stat and end month.



</td></tr>
<tr><td>Week</td><td>Setup start and end week.

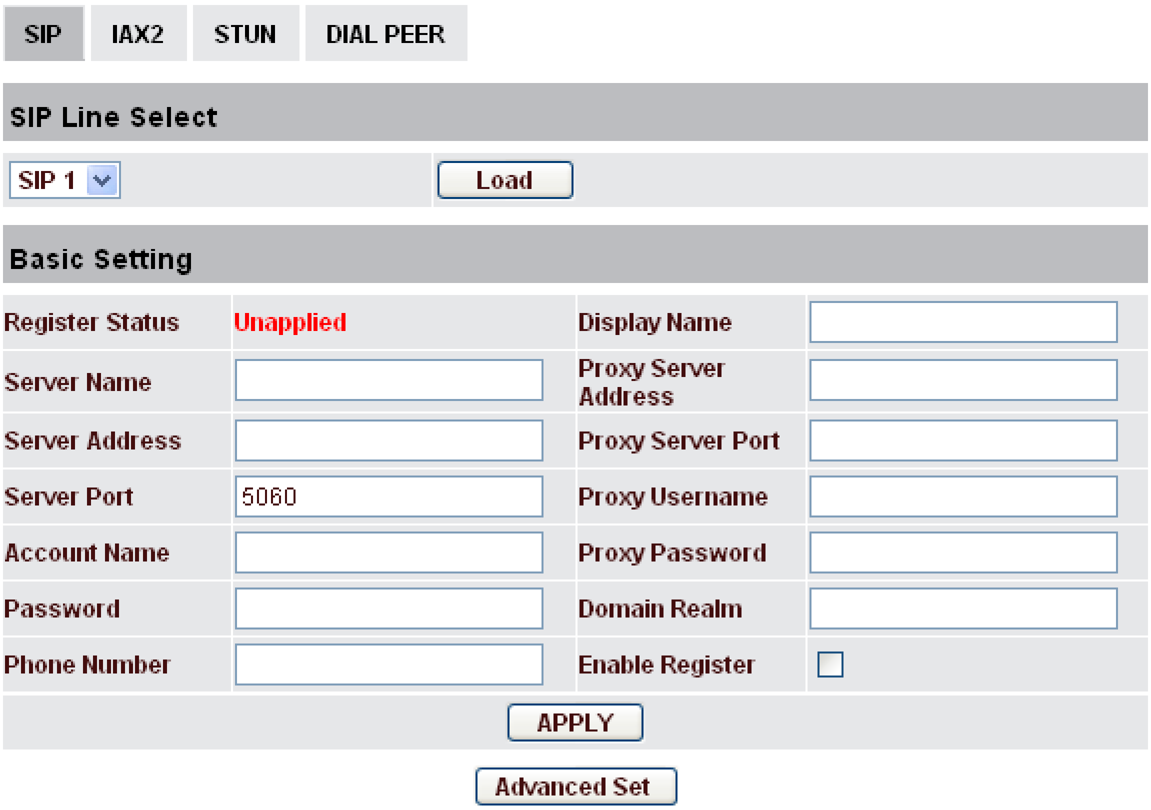


</td></tr>
<tr><td>Day</td><td>Setup start and end day.</td></tr>
<tr><td>Hour</td><td>Setup start and end hours.</td></tr>
<tr><td>Minute</td><td>Setup start and end minutes.</td></tr>
<tr><td>Manual Timeset</td><td>You need specify the all items.</td></tr>
<tr><td>Apply</td><td>Save the settings.</td></tr>
</table>

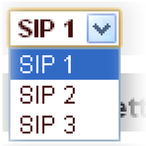
3.3 VoIP

3.3.1 SIP

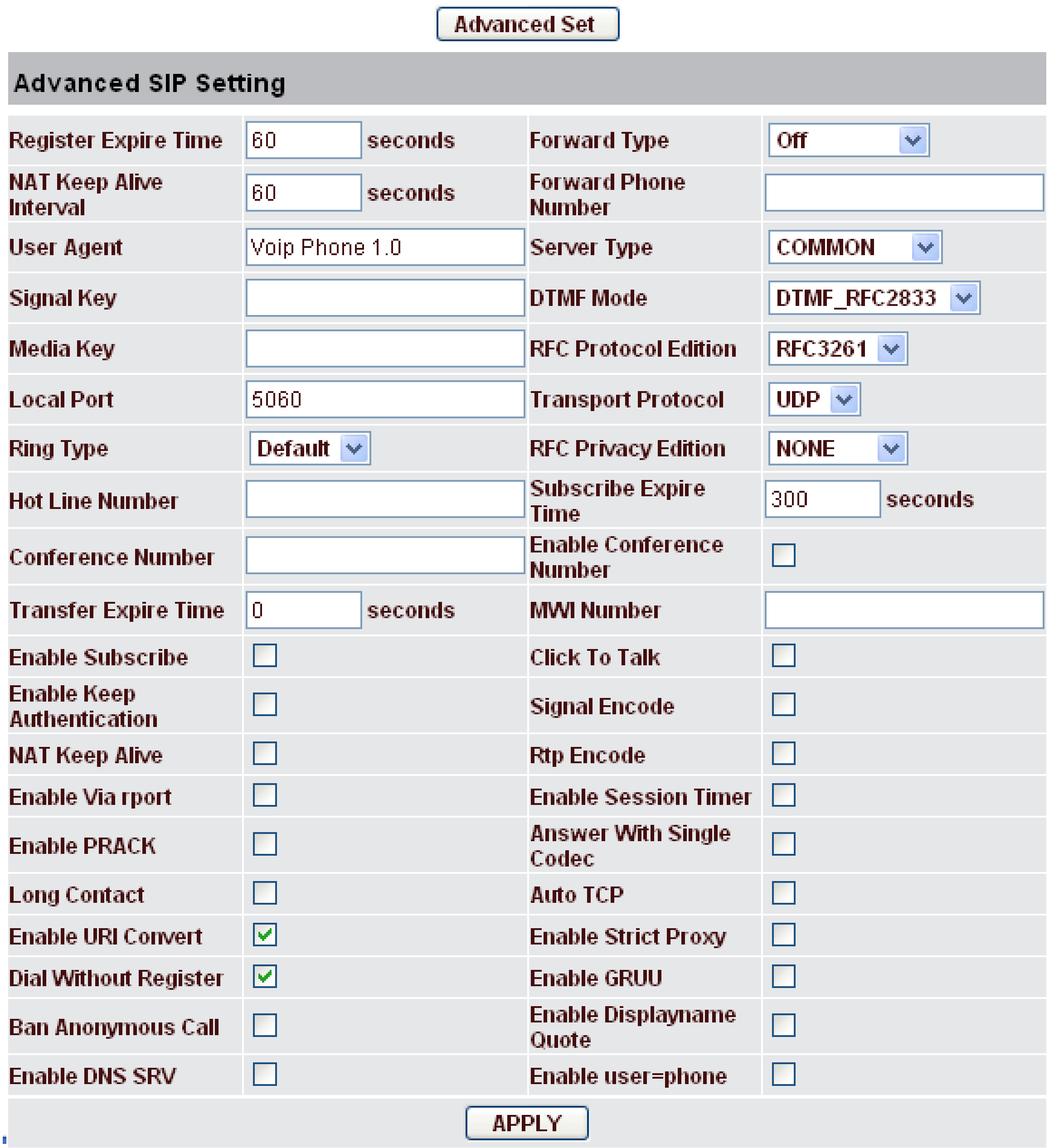
Set your SIP server in the following interface.



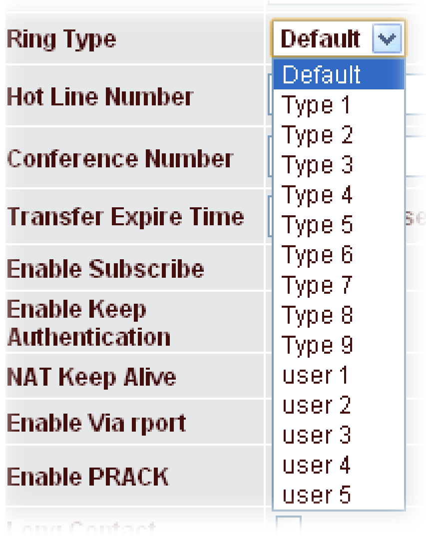
Field name	Explanation
SIP Line Select	Choose line to set info about SIP, there are 3 lines to choose. You can

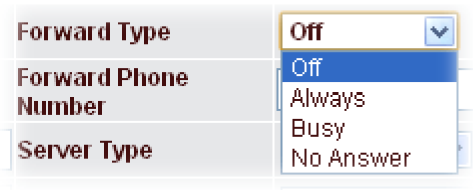
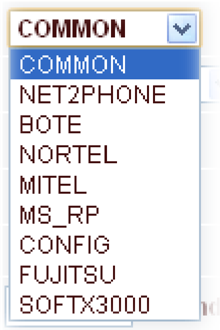
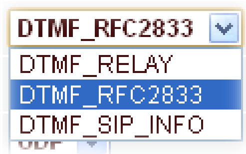
	switch by using the **Load** button.  Before configuring the basic settings, you have to load one SIP line first.
Register Status	Shows if the phone has been registered the SIP server or not; or so, show Unapplied;
Server Name	Set the server name.
Server Address	Input your SIP server address.
Server Port	Set your SIP server port.
Account Name	Input your SIP register account name.
Password	Input your SIP register password.
Phone Number	Input the phone number assigned by your VoIP service provider. Phone will not register if there is no phone number configured.
Display Name	Set the display name.
Proxy Server Address	Set proxy server IP address（Usually, Register SIP Server configuration is the same as Proxy SIP Server. But if your VoIP service provider give different configurations between Register SIP Server and Proxy SIP Server, you need make different settings.）
Proxy Server Port	Set your Proxy SIP server port.
Proxy Username	Input your Proxy SIP server account.
Proxy Password	Input your Proxy SIP server password.
Domain Realm	Set the sip domain if needed, otherwise this VoIP phone will use the Register server address as sip domain automatically. (Usually it is same with registered server and proxy server IP address).
Enable Register	Start to register or not by selecting it or not.

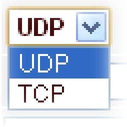
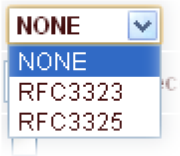
Click **Advanced Set** to get more detailed settings for SIP account.



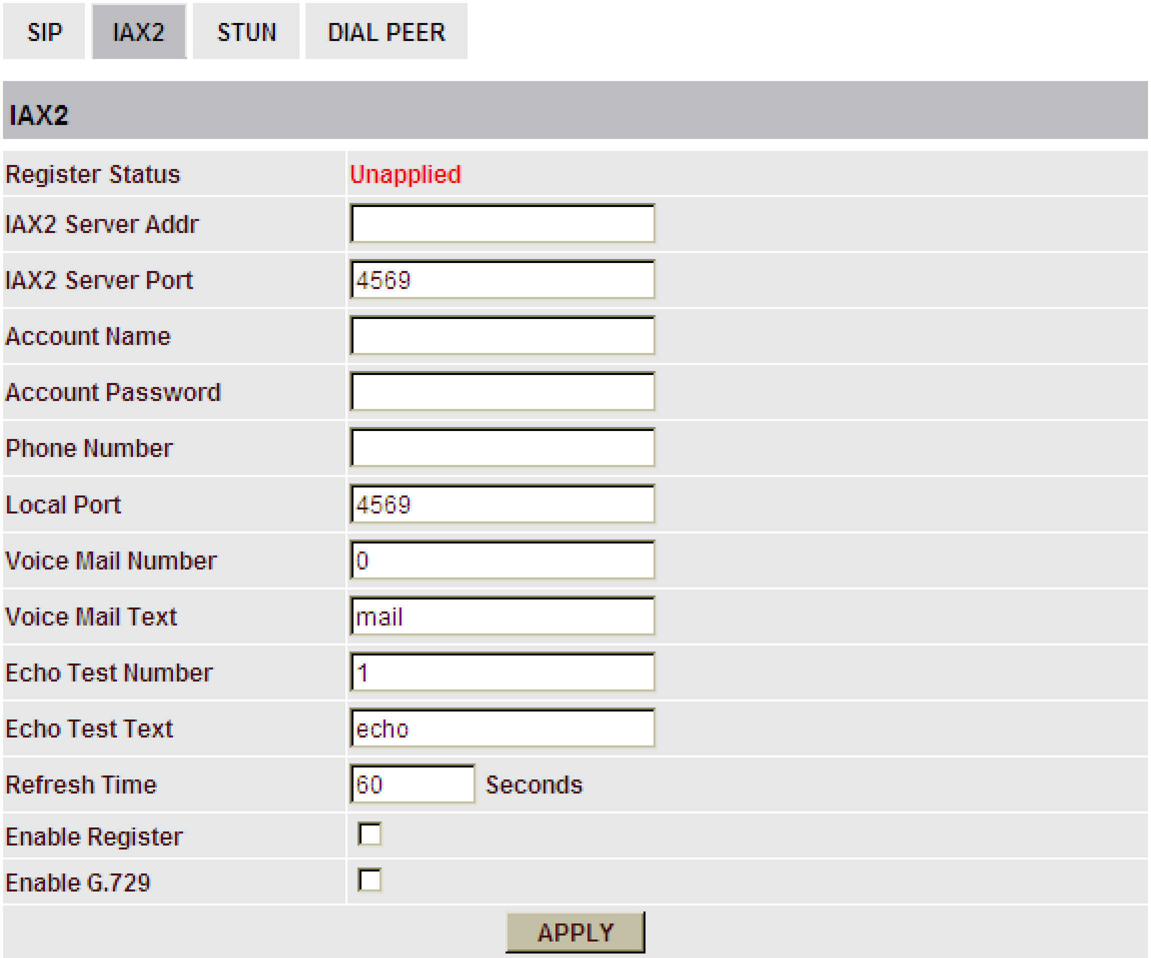
Field name	Explanation
Register Expire Time	Set expire time of SIP server register, default is 60 seconds. If the register time of the server requested is longer or shorter than the expire time set, the phone will change automatically the time into the time recommended by the server, and register again.
NAT Keep Alive Interval	Set examining interval of the server, default is 60 seconds.
User Agent	Set the user agent if have, the default is VoIP Phone 1.0.
Signal Key	Set the key for signal encryption.
Media Key	Set the key for RTP encryption.
Local port	Set sip port of each line.
Ring type	Set ring type of each line.

	
Hot line Number	Set hot line number of each line.
Conference Number	Configure conference number in server conference.
Transfer Expire Time	For the phone supports the transfer of certain special features server, set interval time between sending "bye" and hanging up after the phone transfers a call.
Enable subscribe	Enable the option, the phone will receive the notification from the server.
Enable Keep Authentication	Enable/Disable Keep Authentication System will take the last authentication field which is passed the authentication by server to the request packet. It will decrease the server's repeat authorization work, if it is enable.
NAT Keep Alive	Enable/Disable keeps NAT of SIP alive. If some server refuse to register with too short interval time, and has no packets sending to device in private network to keep NAT alive, user could set this function ON. It need set the keep alive interval time less than the NAT server's.
Enable Via rport	Enable/Disable system to support RFC3581. Via rport is special way to realize SIP NAT.
Enable PRACK	Enable or disable SIP PRACK function, suggest use the default configuration.
Long Contact	Set more parameters in contact field; connection with SEM server
Enable URI Convert	Convert # to %23 when send the URI.
Dial Without Register	Set call out by proxy without registration.
Ban Anonymous Call	Set to ban Anonymous Call.
Enable DNS SRV	Support DNS looking up with _sip.udp mode
Forward Type	Select call forward mode, the default is Off. ● Off: Close down calling forward ● Busy: If the phone is busy, incoming calls will be forwarded to the

	appointed phone. ● No answer: If there is no answer, incoming calls will be forwarded to the appointed phone. ● Always: Incoming calls will be forwarded to the appoint phone directly. The phone will Prompt the incoming while doing forward. 
Forward Phone Number	Appoint your forward phone number.
Server Type	Select the special type of server which is encrypted, or has some unique requirements or call flows. 
DTMF Mode	Select DTMF sending mode, there are three modes: ● DTMF_RELAY ● DTMF_RFC2833 ● DTMF_SIP_INFO Different VoIP Service providers may provide different modes. 
RFC Protocol Edition	Select SIP protocol version to adapt for the SIP server which uses the same version as you select. For example, if the server is CISCO5300, you need to change to RFC2543, else phone may not cancel call normally. System uses RFC3261 as default. 
Transport Protocol	Set transport protocols, TCP or UDP.

	
RFC Privacy Edition	Set Anonymous call out safely; Support RFC3323and RFC3325. 
Subscribe Expire Time	Overtime of resending subscribe packet. Suggest using the default configuration.
Enable Conference number	Set to use sever conference.
MWI Number	Input the number of the server's voice-mail box.
Click to Talk	Set click to Talk (need practical software support).
Signal Encode	Enable/Disable Signal Encrypt.
RTP Encode	Enable/Disable RTP Encrypt.
Enable Session Timer	Set Enable/Disable Session Timer, whether support RFC4028.It will refresh the SIP sessions.
Answer With Single Codec	Enable/Disable the function when call is incoming, phone replies SIP message with just one codec which phone supports.
Auto TCP	Set to use automatically TCP protocol to guarantee usability of transport as message is above 1300 byte.
Enable Strict Proxy	Support the special SIP server-when phone receives the packets sent from server， phone will use the source IP address, not the address in via field.
Enable GRUU	Set to support GRUU.
Enable Display name Quote	Set to make quotation mark to display name as the phone sends out signal, in order to be compatible with server.
Enable user=phone	It is just for satisfying the standard of SIP URI. If the SIP server or PSTN gateway does not have any request of SIP invite, you don’t need to enable this feature.

3.3.2 IAX2

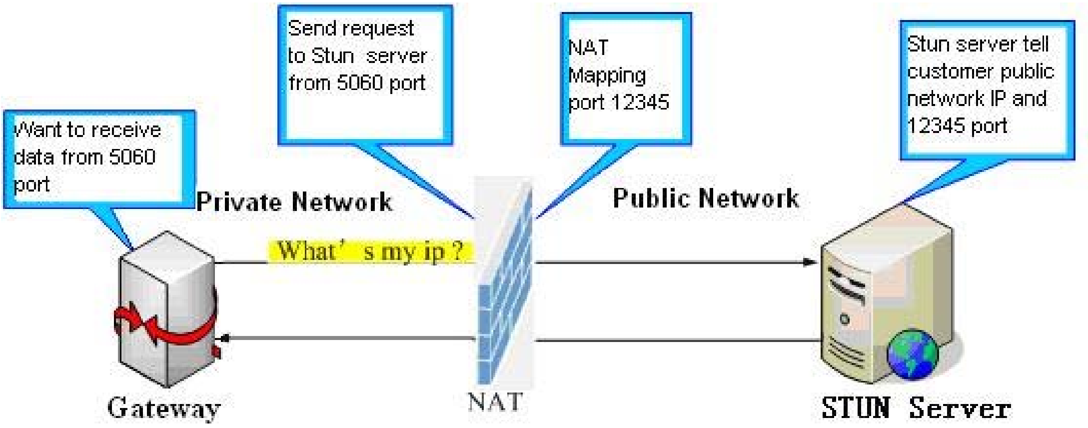


Field name	Explanation
Register Status	Shows if the phone has been registered the IAX2 server or not.
IAX2 Server Addr	Input your IAX2 server address.
IAX2 Server Port	Set your IAX2 server port, the default is 4569.
Account Name	Input your IAX2 register account name.
Account Password	Input your IAX2 register password.
Phone Number	Input your assigned phone number (usually it is same you're your IAX2 account name).
Local Port	Set your local sport，the default is 4569.
Voice Mail Number	Specify the voice mail's number.
Voice Mail Text	Specify the voice mail's name.
Echo Test Number	Set echo test number. If IAX2 server supports echo test, and echo test number is non- numeric, system could set an echo test number to replace the echo test text. So user can dial the numeric number to test echo voice test. This function is provided with server to make endpoint to test whether endpoint could talk through server normally.

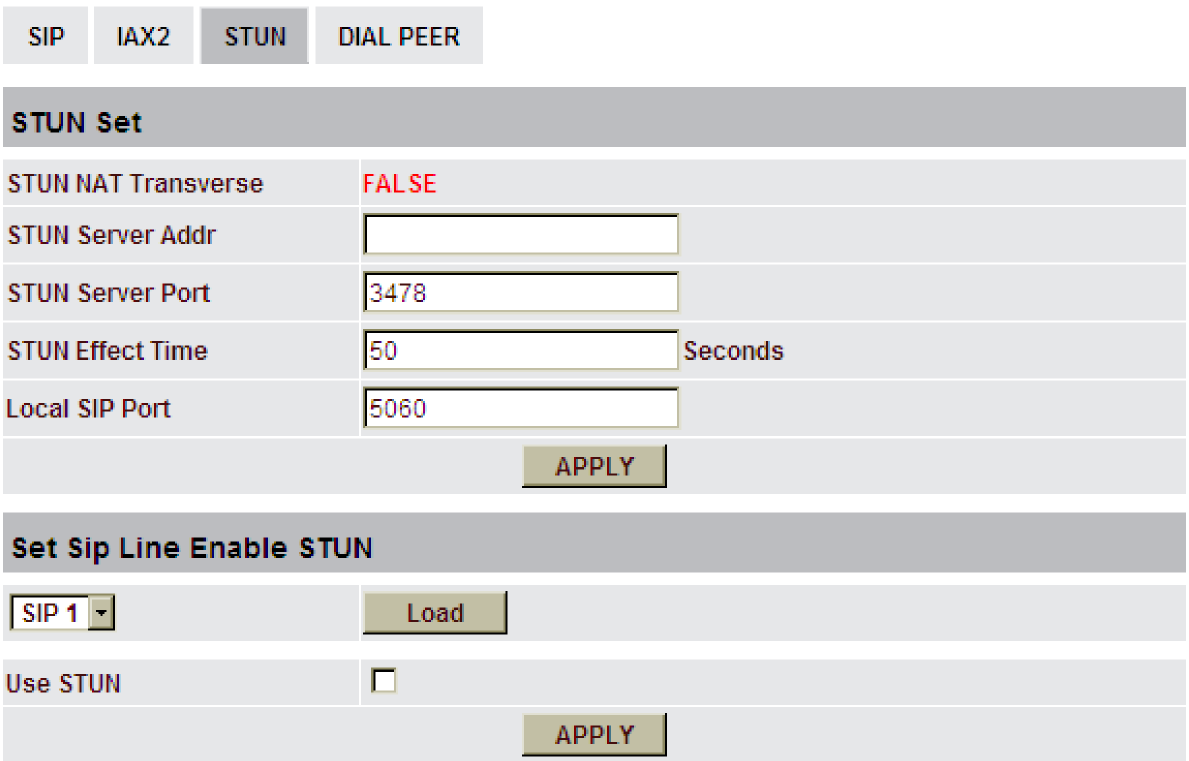
Echo Test Text	Specify echo test text's name.
Refresh Time	Set expire time of IAX2 server register, you can set it between 60 and 3600 seconds.
Enable Register	Start to register the IAX2 server or not by selecting it or not.
Enable G.729	Enable or disable code G.729 by selecting it or not

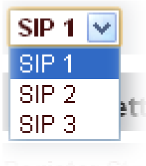
3.3.3 STUN

By STUN server, the phone in private network could know the type of NAT and the NAT mapping IP and port of SIP. The phone might register itself to SIP server with global IP and port to realize the device both calling and being called in private network.



In this web page, you can configure SIP STUN.



Field name	Explanation
STUN NAT Transverse	Shows STUN NAT Transverse estimation, true means STUN can penetrate NAT, while False means not.
STUN Server Addr	Set your SIP STUN Server IP address
STUN Server Port	Set your SIP STUN Server Port
STUN Effect Time	Set STUN Effective Time. If NAT server finds that a NAT mapping is idle after time out, it will release the mapping and the system need send a STUN packet to keep the mapping effective and alive.
Local SIP Port	Set the SIP port.
Set Sip Line Enable STUN	Choose line to set info about SIP, There are 3 lines to choose. You can switch by using the **Load** button. 
Use STUN	Enable/Disable SIP STUN.
Apply	Save the settings.

Notice: SIP STUN is used to realize SIP penetration to NAT. If your phone configures STUN Server IP and Port (default is 3478), and enable SIP Stun, you can use the ordinary SIP Server to realize penetration to NAT.

3.3.4 Dial Peer

This functionality offers you more flexible dial rule, you can refer to the following content to know how to use this dial rule.

- When you want to dial an IP address, the entry of IP addresses is very cumbersome, but by this functionality, you can set number 156 to replace 192.168.1.119 here.

SIP | IAX2 | STUN | DIAL PEER

Dial Peer Table

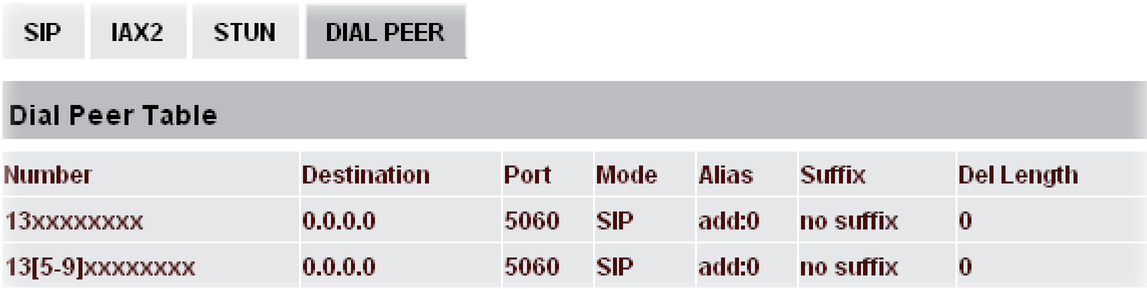
Number	Destination	Port	Mode	Alias	Suffix	Del Length
156	192.168.1.119	5060	SIP	no alias	no suffix	0

- When you want to dial a long distance call to Beijing, you need dial an area code 010 before local phone number, but you can also dial number 1 instead of 010 after we make a setting according to this dial rule. For example, you want to dial 01062213123, but you need dial only 162213123 to realize your long distance call after you make this setting.

Dial Peer Table

Number	Destination	Port	Mode	Alias	Suffix	Del Length
1T	0.0.0.0	5060	SIP	rep:010	no suffix	1

- To save the memory and avoid abundant input of user, add the follow functions:


SIP | IAX2 | STUN | DIAL PEER

Dial Peer Table

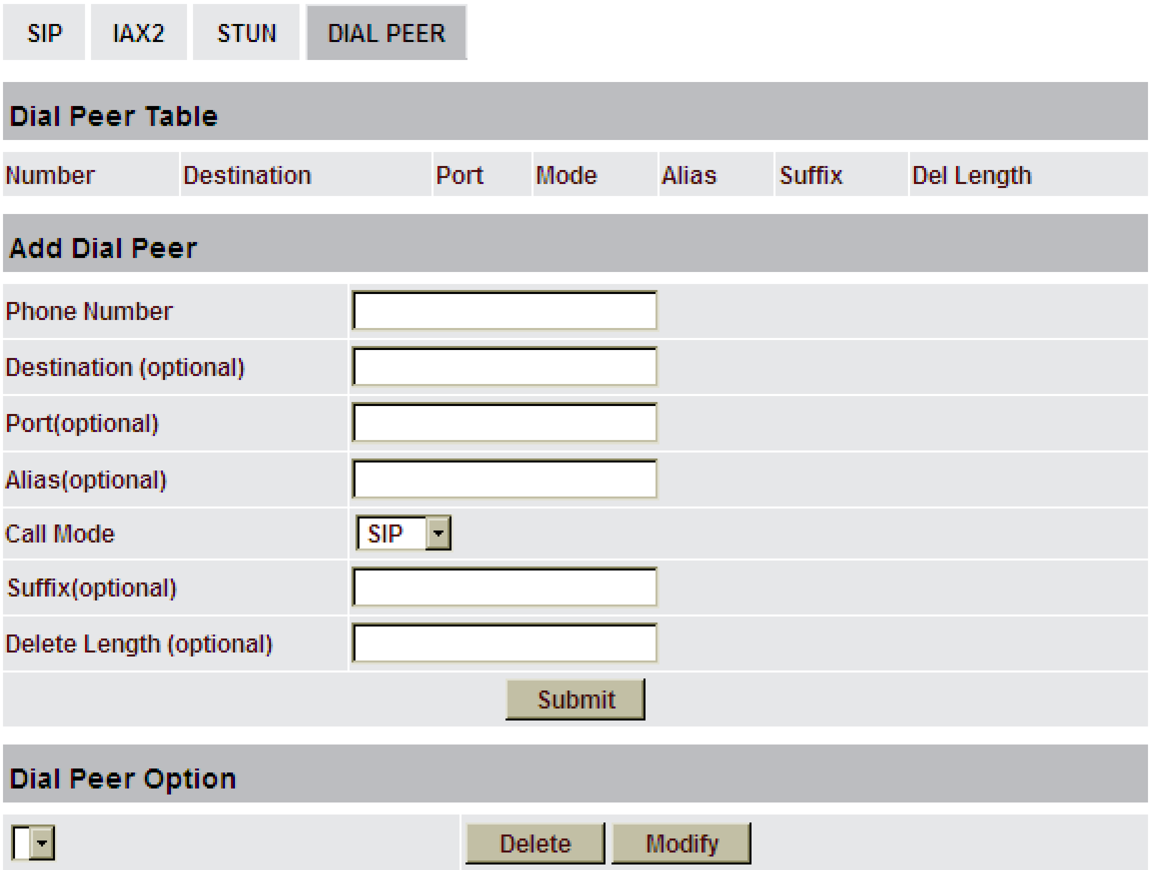
Number	Destination	Port	Mode	Alias	Suffix	Del Length
13xxxxxxxxx	0.0.0.0	5060	SIP	add:0	no suffix	0
13[5-9]xxxxxxxx	0.0.0.0	5060	SIP	add:0	no suffix	0

x matches any single digit that is dialed. If a user makes the above configuration, after he/she dials 11 digit numbers started with 13, the phone will send out 0 plus the dialed numbers automatically.

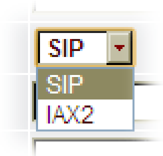
[ ] Specifies a range that will match digit. It may be a range, a list of ranges separated by commas, or a list of digits.

If a user makes the above configuration, after user dials 11 digit numbers started with from 135 to 139, the phone will send out 0 plus the dialed numbers automatically.

With this setting, you can realize dialing out via different lines without switch in web interface.



Field name	Explanation
Phone number	There are two types of matching conditions: one is full matching, the other is prefix matching. In the Full matching, you need input your desired phone number in this blank, and then you need dial the phone number to realize calling to what the phone number is mapped. In the prefix matching, you need input your desired prefix number and T; then

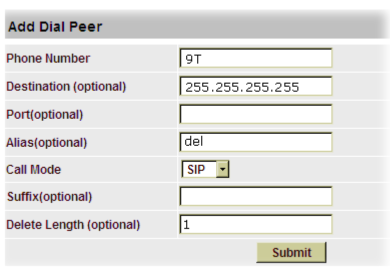
	dial the prefix and a phone number to realize calling to what your prefix number is mapped. The prefix number supports at most 30 digits.
Destination	Set Destination address. This is optional configuration item. If you want to set peer to peer call, please input destination IP address or domain name. If you want to use this dial rule on SIP2 line, you need input 255.255.255.255 or 0.0.0.2 in it.SIP3 into 0.0.0.3.
Port	Set the Signal port, the default is 5060 for SIP.
Alias	Set alias. This is optional configuration item. If you don't set Alias, it will show no alias.
Call Mode	Select different signal protocol, SIP or IAX2. 
Suffix(optional)	Set suffix, this is optional configuration item. It will show no suffix if you don't set it.
Delete Length (optional)	Set delete length. This is optional configuration item. For example: if the delete length is 3, the phone will delete the first 3 digits then send out the rest digits. You can refer to examples of different alias application to know how to set delete length.

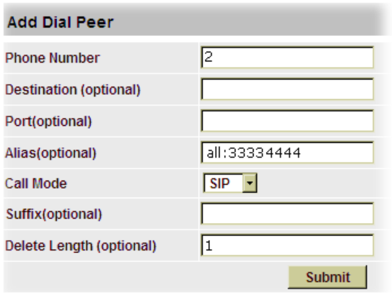
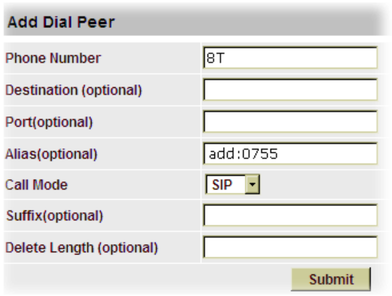
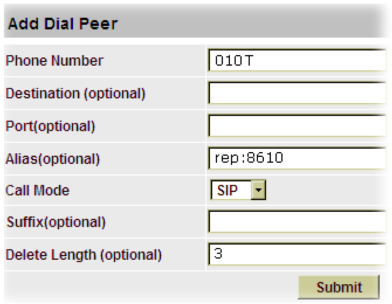
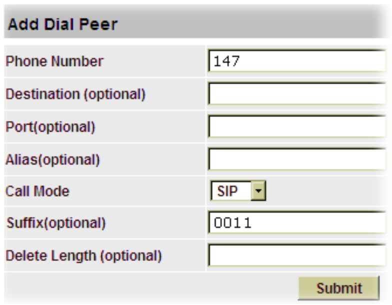
Note: There are four types of aliases.

- Add: xxx, it means that you need dial xxx in front of phone number, which will reduce dialing number length.
- All: xxx, it means that xxx will replace some phone number.
- Del: It means that phone will delete the number with length appointed.
- Rep: It means that phone will replace the number with length and number appointed.

You can refer to the following examples of different alias application to know more how to use different aliases and this dial rule.

Examples of different alias application

Set by web	Explanation	Example
	You need set phone number, Destination, Alias and Delete Length. Phone number is XXXT; Destination is 255.255.255.255 (0.0.0.2) and Alias is del. This means any phone No. that starts with your set phone number will be sent via SIP2 line after the first several digits of your dialed phone number are deleted according to delete length.	If you dial "93333", the SIP2 server will receive "3333"

	This setting will realize speed dial function, after you dialing the numeric key “2”, the number after all will be sent out.	When you dial “2”, the SIP1 server will receive 33334444
	The phone will automatically send out alias number adding your dialed number, if your dialed number starts with your set phone number.	When you dial “8309“, the SIP1 server will receive “07558309”
	You need set Phone Number, Alias and Delete Length. Phone number is XXXT and Alias is rep:xxx If your dialed phone number starts with your set phone number, the first digits same as your set phone number will be replaced by the alias number specified and New phone number will be send out.	When you dial “0106228”, the SIP1 server will receive “86106228”
	If your dialed phone number starts with your set phone number. The phone will send out your dialed phone number adding suffix number.	When you dial “147”, the SIP1 server will receive “1470011”

Introduction of how to set up dial-peer to implement switch between multi- SIP lines

SIP IAX2 STUN DIAL PEER

Dial Peer Table

Number	Destination	Port	Mode	Alias	Suffix	Del Length
9T	0.0.0.0	5060	SIP	del	no suffix	1
8T	0.0.0.2	5060	SIP	del	no suffix	1

9T mapping: If you have registered a SIP1 server and set dial-peer according to the above table, all calls will be sent via SIP1 server when you press the numeric key “9” in front of dialing destination phone numbers.

8T mapping: If you have registered a Private SIP2 server and set dial-peer according to the above table， all calls will be sent via SIP2 server when you press the numeric key “8” in front of dialing destination phone numbers.

SIP | IAX2 | STUN | DIAL PEER

Dial Peer Table

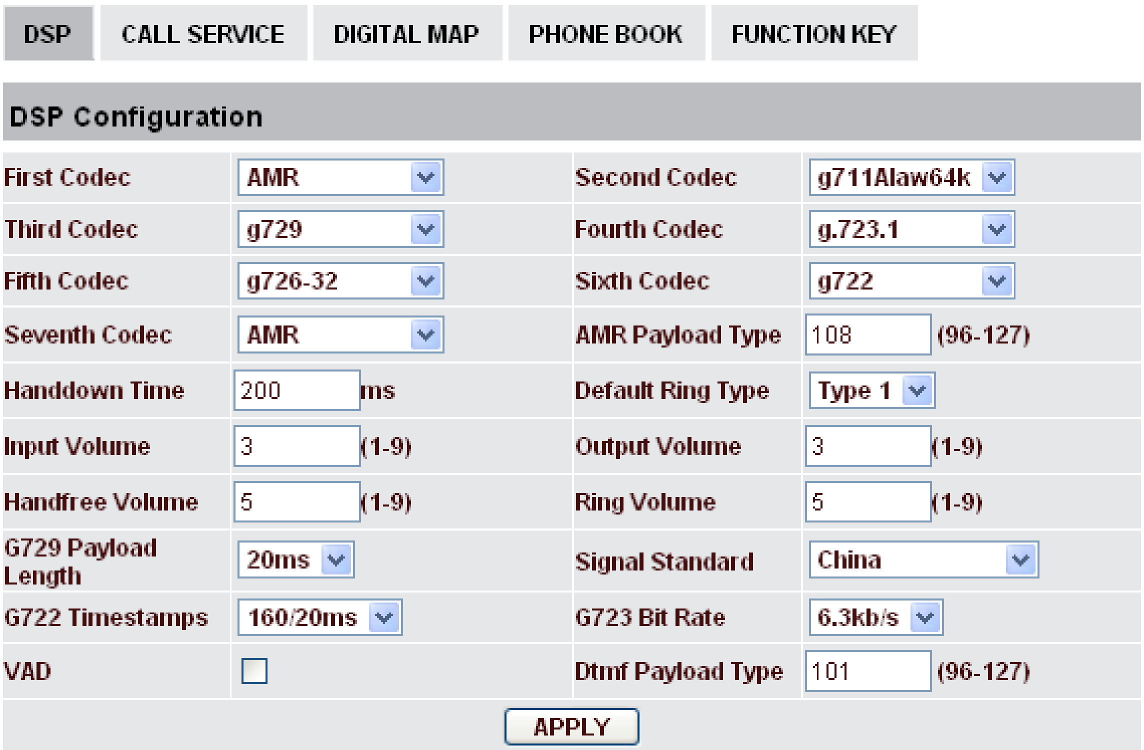
Number	Destination	Port	Mode	Alias	Suffix	Del Length
2T	0.0.0.0	4569	IAX2	del	no suffix	1

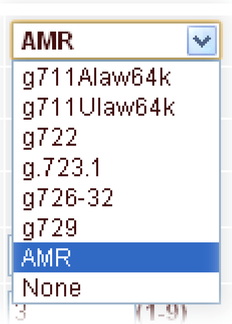
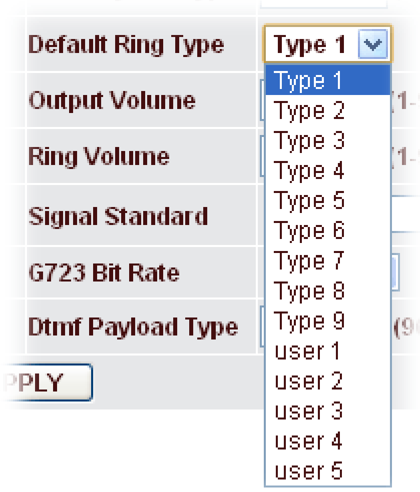
2T mapping: The rule of 2T means the user needs to dial the number with prefix 2 if he/she wants to dial via **IAX2** server.

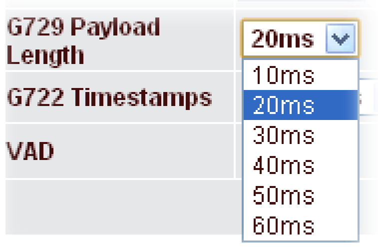
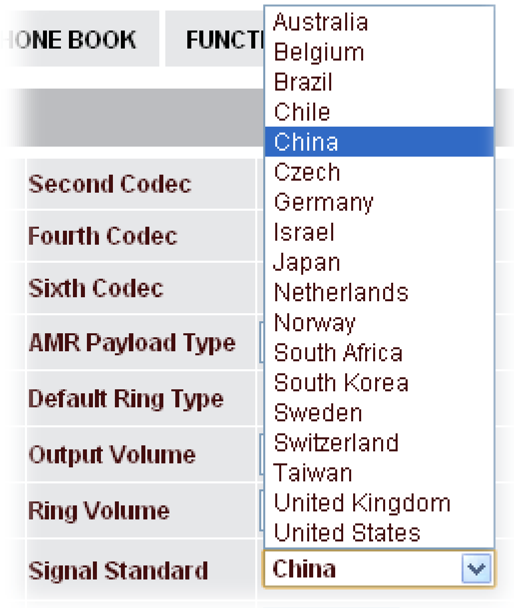
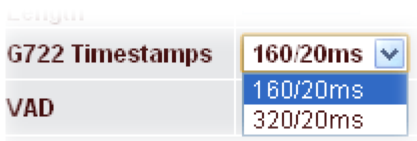
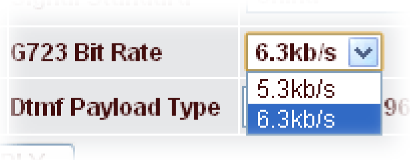
3.4 Phone

3.4.1 DSP

In this page, you can configure voice codec, input/output volume and so on.

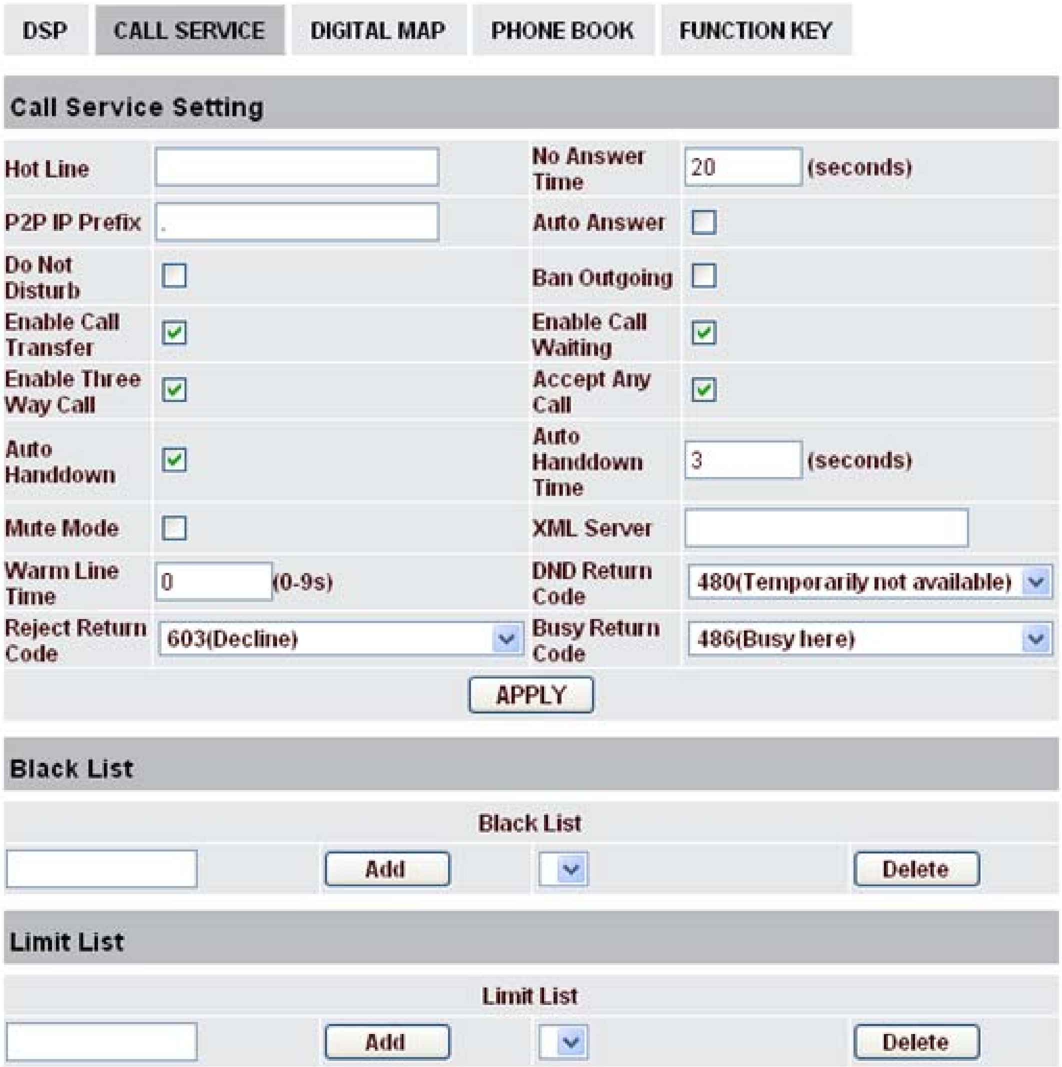


Field name	Explanation
First Codec	The fist preferential DSP codec: G.711A/u, G.722, G.723, G.729,G.726,AMR
Second Codec	The second preferential DSP codec: G.711A/u, G.722, G.723, G.729,G.726
Third Codec	The third preferential DSP codec: G.711A/u, G.722, G.723, G.729,G.726,AMR
Forth Codec	The forth preferential DSP codec: G.711A/u, G.722, G.723, G.729,G.726,AMR
Fifth Codec	The fifth preferential DSP codec: G.711A/u, G.722, G.723, G.729,G.726,AMR
Sixth Codec	The fifth preferential DSP codec: G.711A/u, G.722, G.723, G.729,G.726,AMR
Seventh Codec	The seventh preferential DSP codec: G.711A/u, G.722, G.723, G.729,G.726,AMR
AMR Payload Type	AMR Payload Type. 
Handdown Time	Specify the least reflection time of Handdown. The default is 200ms.
Default Ring Type	Set up the ring by default. 
Input Volume	Specify Input (MIC) Volume grade.
Output Volume	Specify Output (receiver) Volume grade.

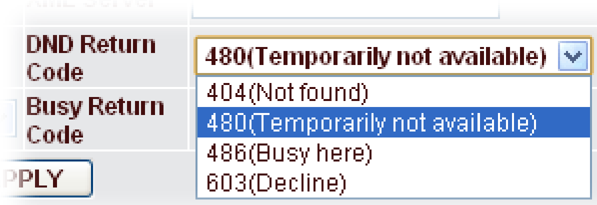
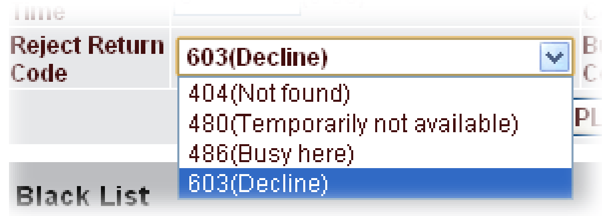
Hands-free Volume	Specify Hands-free Volume grade.
Ring Volume	Specify Ring Volume grade.
G729 Payload Length	Set G729 Payload Length. 
Signal Standard	Select Signal Standard. 
G722 Timestamps	160/20ms or 320/20ms is available. 
G723 Bit Rate	5.3kb/s or 6.3kb/s is available 
VAD	Select it or not to enable or disable VAD. If enable VAD, G729 Payload length could not be set over 20ms.
DTMF Payload Type	Set up DTMF payload type
Apply	Save the settings.

3.4.2 Call Service

In this web page, you can configure Hotline, Call Transfer, Call Waiting, 3 Ways Call, Black List, white list Limit List and so on.



Field name	Explanation
Hotline	Specify Hotline number. If you set the number, you can not dial any other numbers.
No Answer Time	Specify No Answer Time
P2P IP Prefix	Set Prefix in peer to peer IP call. For example: what you want to dial is 192.168.1.119, If you define P2P IP Prefix as 192.168.1., you dial only #119 to reach 192.168.1.119. Default is “.”. If there is no “.” Set, it means to disable dialing IP.
Auto Answer	If select it, the phone will auto answer when there is an incoming call.
Do Not Disturb	Select NO Disturb, the phone will reject any incoming call, the callers will be reminded by busy, but any outgoing call from the phone will work well.
Ban Outgoing	If you select Ban Outgoing to enable it, and you can not dial out any number.

Enable Call Transfer	Enable Call Transfer by selecting it.
Enable Call Waiting	Enable Call Waiting by selecting it.
Enable Three Way Call	Enable Three Way Call
Accept Any Call	If select it, the phone will accept the call even if the called number is not belong to the phone.
Auto Handdown	The phone will hang up and return to standby automatically at hands-free mode
Auto Handdown Time	Configuration automatically hang time, if it is hands-free mode, then more than auto handdown time, the phone automatically returns to standby mode, if the handle pattern, then more than auto handdown time, it automatically put a dial tone.
Mute Mode	Configuring the mute mode, if the mute mode, calls LCD will flash tips, but does not ring
XML Server	Xml configuration server address and the default xml file name
Warm Line Time	Warm line set timeout to set the time line when more than warm, it will automatically exhaled hotline number, if configured to 0, the hook immediately exhaled hotline number.
DND Return Code	When the status of the IP phone is "DND (do not disturb)", it will send a message to the server based on the code selected here. 
Reject Return Code	When the status of the IP phone is "Reject", it will send a message to the server based on the code selected here. 
Busy Return Code	When the status of the IP phone is "Busy", it will send a message to the server based on the code selected here.
Black List	Set Add/Delete Black list. If user does not want to answer some phone calls, add these phone numbers to the Black List, and these calls will be rejected. "x" and "." are wildcard. x means matching any single digit. for example, 4xxx expresses any number with prefix 4 which length is 4 will be forbidden to dialed out "." means matching any arbitrary number digit. For example, 6 expresses any number with prefix 6 will be forbidden to dialed out. If a user wants to allow a number or a series of number incoming,

	he/she may add the number(s) to the list as the white list rule. the configuration rule is -number, for example, -123456, or -1234xx. Black List -4119 . Means any incoming number is forbidden except for 4119 Note: End with “.” when set up the white list
Limit List	Set Add/Delete Limit List. Please input the prefix of those phone numbers which you forbid the phone to dial out. For example, if you want to forbid those phones of 001 as prefix to be dialed out, you need input 001 in the blank of limit list, and then you can not dial out any phone number whose prefix is 001. “x” and “.” are wildcard. x means matching any single digit. for example, 4xxx expresses any number with prefix 4 which length is 4 will be forbidden to dialed out “.” means matching any arbitrary number digit. For example, 6 expresses any number with prefix 6 will be forbidden to dialed out.

Notice: Black List and Limit List can record at most10 items respectively.

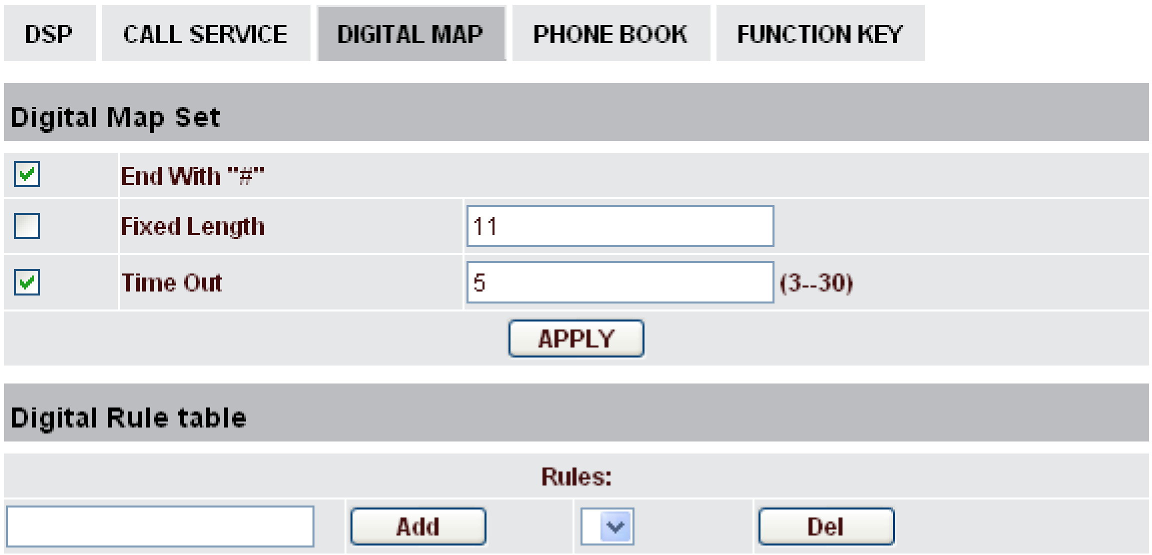
3.4.3 Digital Map

This system supports 4 dial modes:

- End with “#”: dial your desired number, and then press #.
- Fixed Length: the phone will intersect the number according to your specified length.
- Time Out: After you stop dialing and waiting time out, system will send the number collected.
- User defined: you can customize digital map rules to make dialing more flexible. It is realized by defining the prefix of phone number and number length of dialing.

In order to keep some users' secondary dialing manner when dialing the external line with PBX, the phone can be added a special rule to realize it so the user can dial a number as external line prefix and get the secondary dial tone to keep dialing the external number. After finishing dialing, phone will send the prefix and external number totally to the server.

For example, there is a rule 9, xxxxxxxx in the digital map table. After dialing 9, the phone will send the secondary dial tone, and the user may keep going for dialing. After finished, the phone will call the number which starts with 9; actually the number sent out is 9-digit with 9.



Field name	Explanation
End with "#"	Set Enable/Disable the phone ended with “#” dial.
Fixed Length	Specify the Fixed Length of phone ending with.
Time out	Set the timeout of the last dial digit. The call will be sent after timeout.
Digital Rule table	Set and display the user defined digital rules.

Below shows user-defined digital map rule:

[ ]: Specifies a range that will match digit. May be a range, a list of ranges separated by commas, or a list of digits.

x: Match any single digit that is dialed.

. : Match any arbitrary number of digits including none.

Tn: Indicates an additional time out period before digits are sent of n seconds in length. n is mandatory and can have a value of 0 to 9 seconds. Tn must be the last 2 characters of a dial plan. If Tn is not specified, it is assumed to be T0 by default on all dial plans.

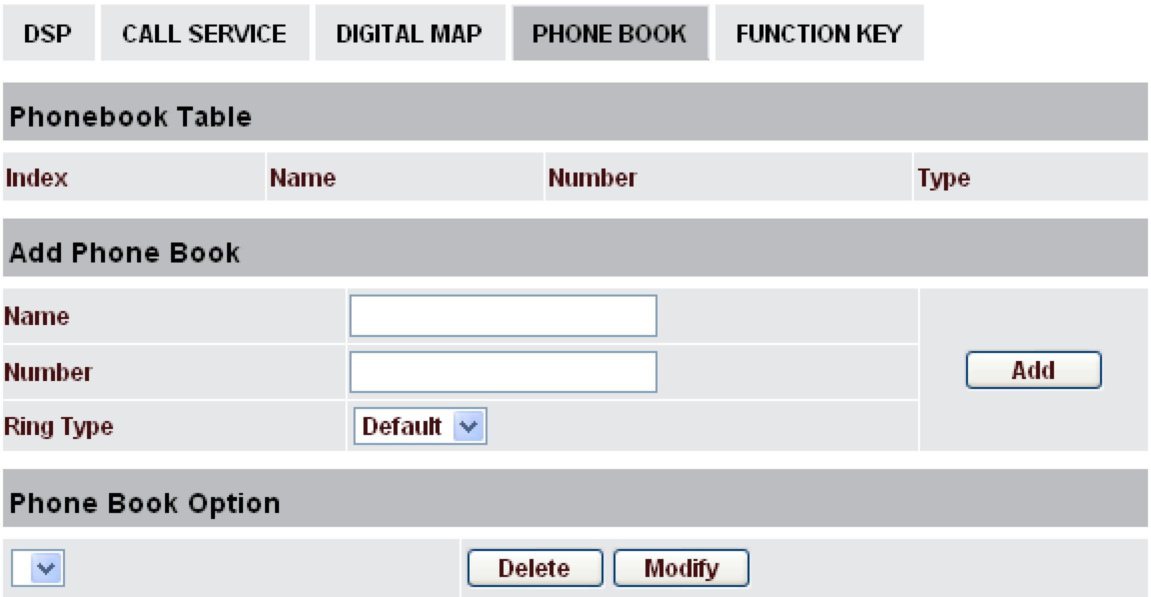
For example,

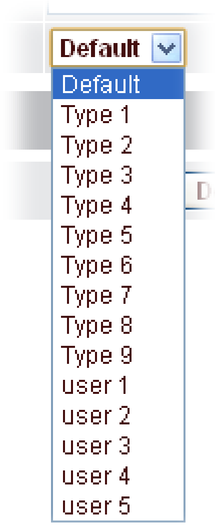
Rules	Explanation
[1-8]xxx	Cause extensions 1000-8999 to be dialed immediately.
9xxxxxx	Cause 8 digit numbers started with 9 to be dialed immediately.
911	Cause 911 to be dialed immediately after it is entered.
99T4	Cause 99 to be dialed after 4 seconds.
9911x, T4	Cause any number started with 9911 to be dialed 4 seconds after dialing ceases.

Notice: End with “#”, Fixed Length, Time out and Digital Map Table can be used simultaneously. System will stop dialing and send number according to your set rules.

3.4.4 Phone Book

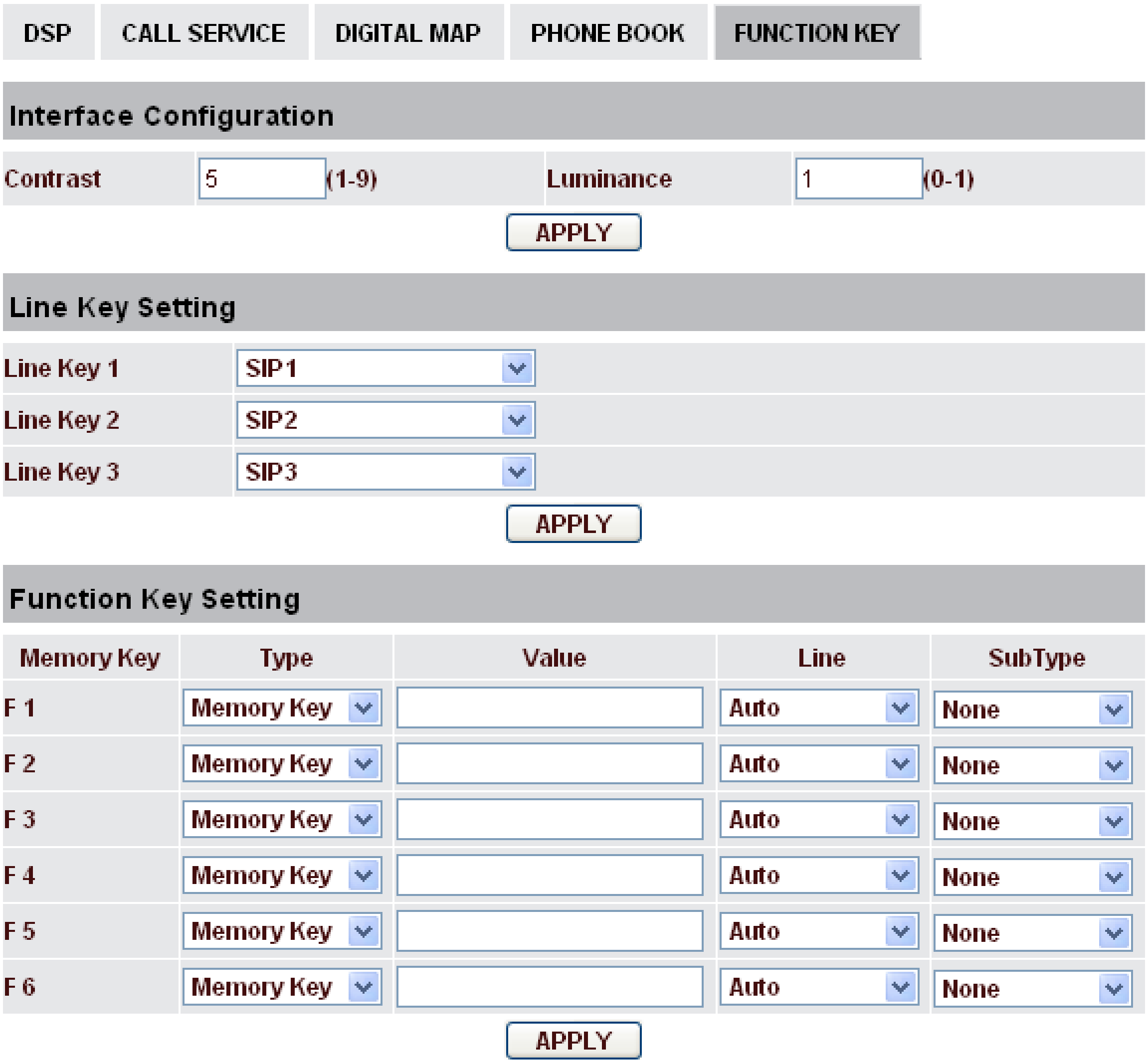
You can input the name, phone number and select ring type for each name here. The maximum capability of the phonebook is 500 items

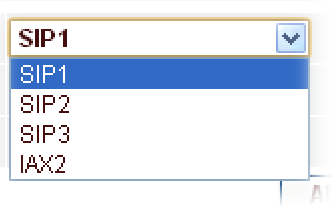


Field name	Explanation
Phonebook Table	Name - Shows the name corresponding to the phone number. Number - Shows the phone number.
Add Phone Book	Name – Type the name corresponding to the phone number. Number –Type the phone number. Add – Click it to add a new phone entry.
Ring Type	Choose one of the ring types for the incoming call. 
Delete/Modify	Click Modify to change the selected information and click the Delete to delete the selected record.

3.4.5 Function Key

This page allows you to configure function keys (also called memory keys in IP Phone) with specific type, value, line and other function parameters (speed dial, push to talk, DND and etc).



Field name	Explanation
Interface Configuration	Contrast - Set contrast of screen. Luminance - Set luminance of screen.
Line Key Setting	Select SIP1, SIP2, SIP3, Dial peer, or IAX2 in function key type. After you set it, you pick up handset or hands-free, press this function key, then you can use the corresponding IP line. 
Function Key Setting	Memory Key - Set the memory key's serial number. Type -

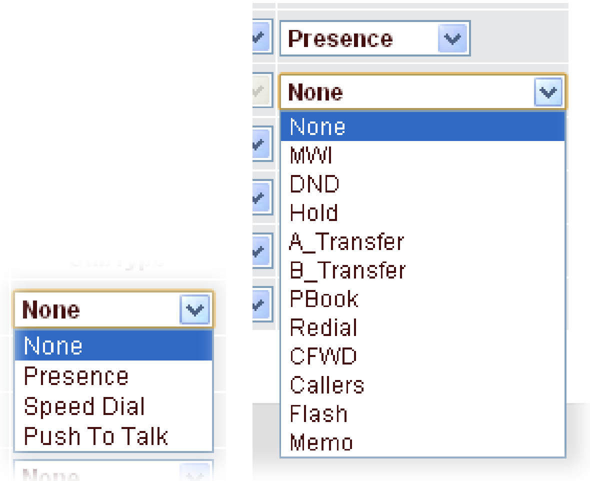
Memory Key: settings can be stored in key storage for each number, the standby or off-hook. Selecting the function keys on the keyboard can call this number.

DTMF：In the call, send DTMF.

Value –Set the type parameter values.

Line – Choose which lines to use this feature.

Subtype – Select the function parameters. Key Event and Memory Key will bring about different Subtype items.



Memory keys can be configured with the following type:

Speed Dial - through the configuration of the key corresponding to the number of ways as shown below:

The User can press the F1 key to allocate this number by line1 line.

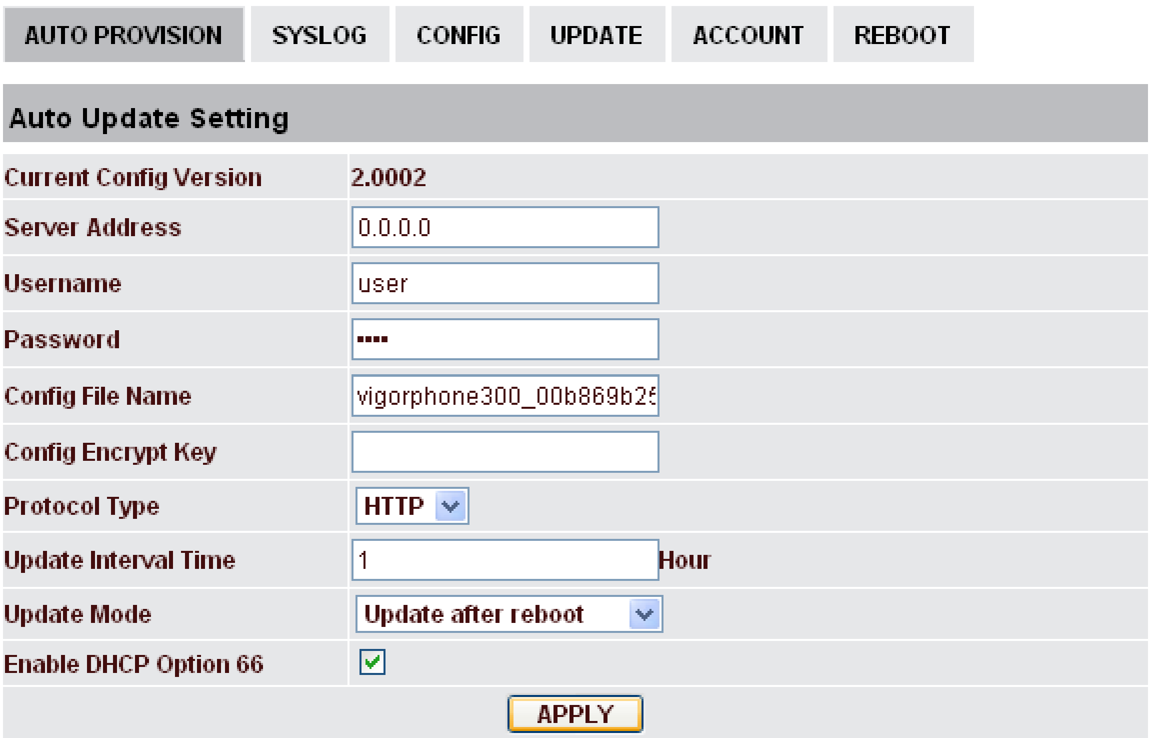
Push To Talk - you can press this key in standby to automatically answer the call and make each other:

The user can configure in accordance with the way of push to talk function. 4116 is the other number. Then press the standby button and make it automatically answering the call 4116.

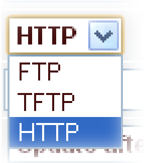
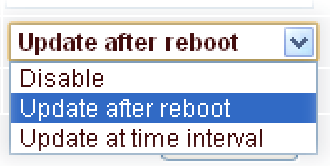
Key Event - key can be configured through certain event (e.g., DND).

3.5 Maintenance

3.5.1 Auto Provision



Field name	Explanation
Current Config Version	Show the current config file's version.

Server Address	Set FTP/TFTP/HTTP server IP address for auto update. The address can be IP address or Domain name with subdirectory.
Username	Set FTP server Username. System will use anonymous if username keep blank.
Password	Set FTP server Password.
Config File Name	Set configuration file's name which need to update. System will use MAC as config file name if config file name keep blank. For example, 000102030405.。
Config Encrypt Key	Input the Encrypt Key, if the configuration file is encrypted.
Protocol Type	Select the Protocol type FTP、TFTP or HTTP. 
Update Interval Time	Set update interval time, unit is hour.
Update Mode	Different update modes: 1. Disable: means no update 2. Update after reboot: means update after reboot. 3. Update at time interval: means periodic update. 
Enable DHCP Option 66	This option is enabled, TFTP server address defaults to the value of option 66

3.5.2 Syslog

Syslog is a protocol which is used to record the log messages with client/server mechanism. Syslog server receives the messages from clients, and classifies them based on priority and type. Then these messages will be written into log by some rules which administrator can configure. This is a better way for log management.

There are 8 levels in debug information:

Level 0---emergency: This is highest default debug info level. You system can not work.

Level 1---alert: Your system has deadly problem.

Level 2---critical: Your system has serious problem.

Level 3---error: The error will affect your system working.

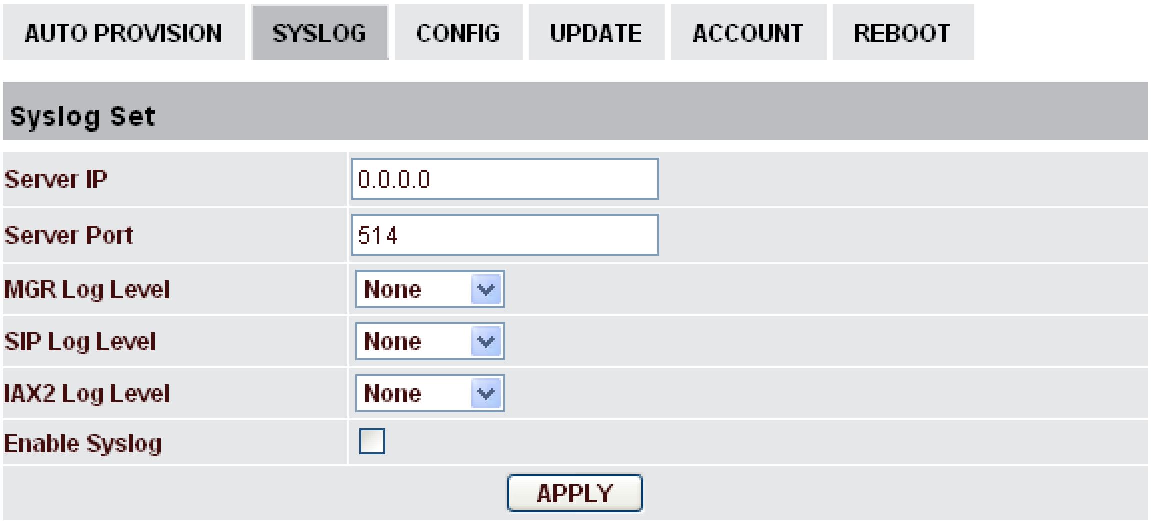
Level 4---warning: There are some potential dangers. But your system can work.

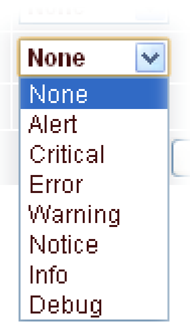
Level 5---notice: Your system works well in special condition, but you need to check its working environment and parameter.

Level 6---info: the daily debugging info.

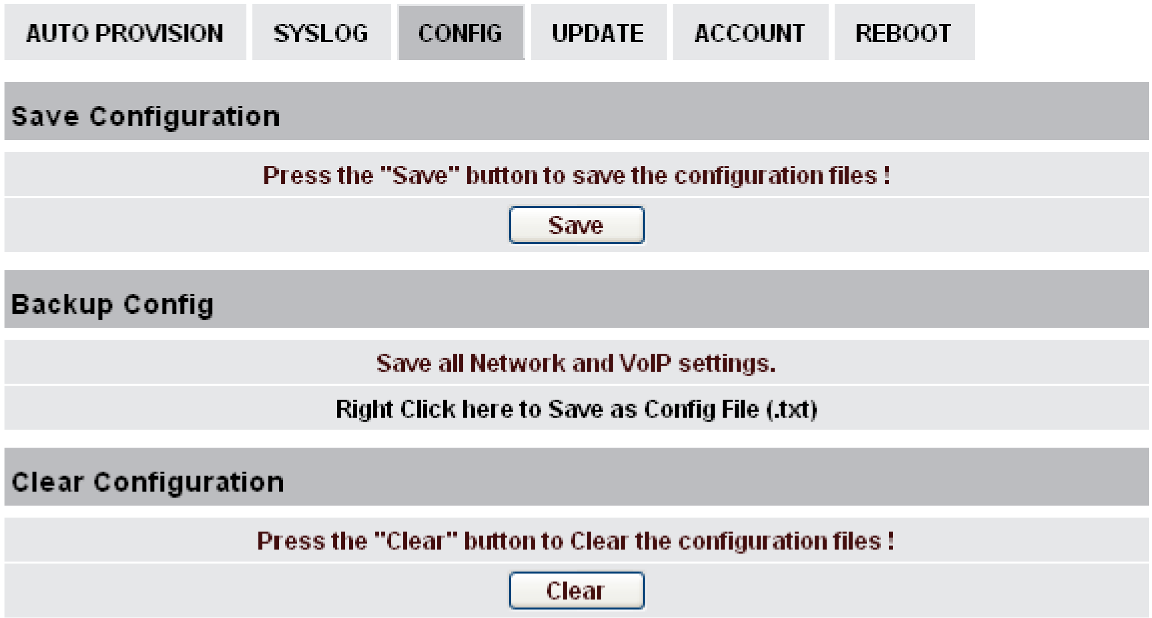
Level 7---debug: the lowest debug info. Professional debugging info from R&D person.

At present, the lowest level of debug information send to Syslog is info, debug level only can be displayed on telnet.



Field name	Explanation
Server IP	Set Syslog server IP address.
Server Port	Set Syslog server port.
MGR Log Level/ SIP Log Level/ IAX2 Log Level	Set the level of MGR log/ Set the level of SIP log/ Set the level of IAX2 log. 
Enable Syslog	Select it or not to enable or disable syslog.
Apply	Save the settings.

3.5.3 Config

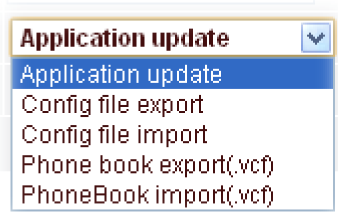
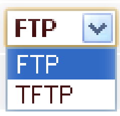


Field name	Explanation
Save Configuration	You can save all changes of configurations. Click the Save button, all changes of configuration will be saved, and be effective immediately. .
Backup Config	Right clicks on “Right click here…” and select “Save Target As….” then you will save the configuration file in .txt format
Clear Configuration	A user can restore factory default configuration and reboot the phone. If you login as Admin, the phone will reset all configurations and restore factory default; if you login as Guest, the phone will reset all configurations except for VoIP accounts (SIP1-2 and IAX2) and version number.

3.5.4 Update

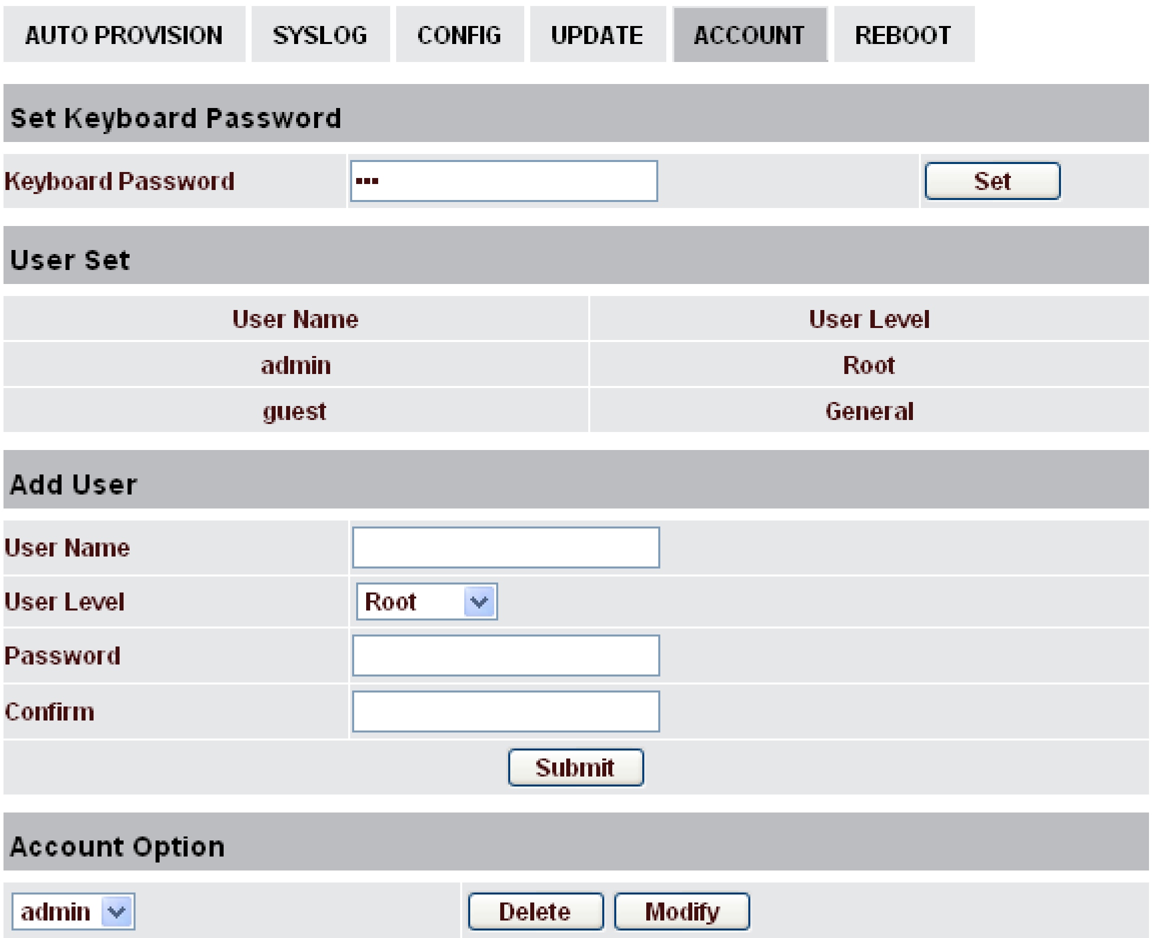
You can update your configuration with your configuration file in this web page.

Field name	Explanation
Web Update	Click the browse button, find out the configuration file saved before or provided by manufacturer, download it to the phone directly, press “Update” to save. You can also update downloaded update file, logo picture, ring file by web.
Server	Set the FTP/TFTP server address for download/upload. The address can be IP address or Domain name with subdirectory.
Username	Set the FTP server Username for download/upload.
Password	Set the FTP server password for download/upload.
File name	Set the name of update file or configuration file. The default name is the MAC of the phone, such as 000102030405.
Type	Action type that system want to execute： ● Application update: download system update file ● Configuration file export: Upload the configuration file to FTP/TFTP server, name and save it. ● Configuration file import: Download the configuration file to phone from FTP/TFTP server. The configuration will be effective after the phone is reset. ● Phone book export (.vcf): Upload the phonebook file to FTP/TFTP server, name and save it. ● PhoneBook import (.vcf): Download the phonebook file to phone

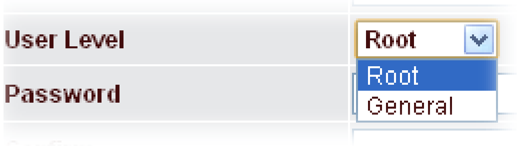
	from FTP/TFTP server. 
Protocol	Select FTP/TFTP server. 

3.5.5 Account

You can add or delete user account, and change the authority of each user account in this web page.

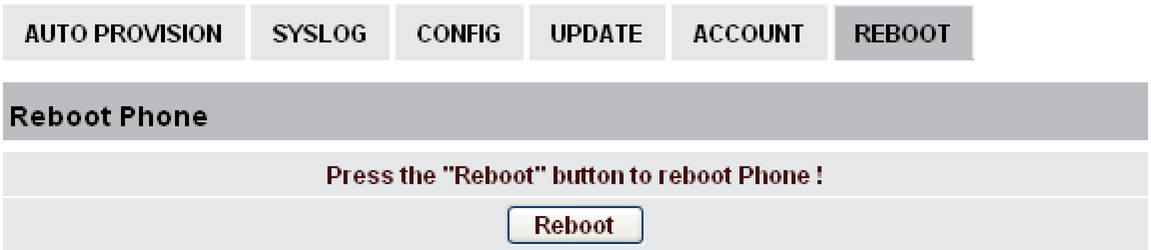


Field name	**Explanation**
Set Keyboard Password	Keyboard Password - Set the password for entering the setting menu of the phone by the phone‘s key board. The password is digit.
User Set	This table shows the current user existed.

Add User	User Name - Set account user name. User Level - Set user level, Root user has the right to modify configuration, General can only read. Password - Set the password.. Confirm - Confirm the password.  Submit – Save the settings.
Account Option	Select the account and click the **Modify** to modify the selected account, and click the **Delete** to delete the selected account. General user only can add the user whose level is General.

3.5.6 Reboot

If you modified some configurations which need the phone's reboot to be effective, you need click the Reboot button. Then the phone will reboot immediately.

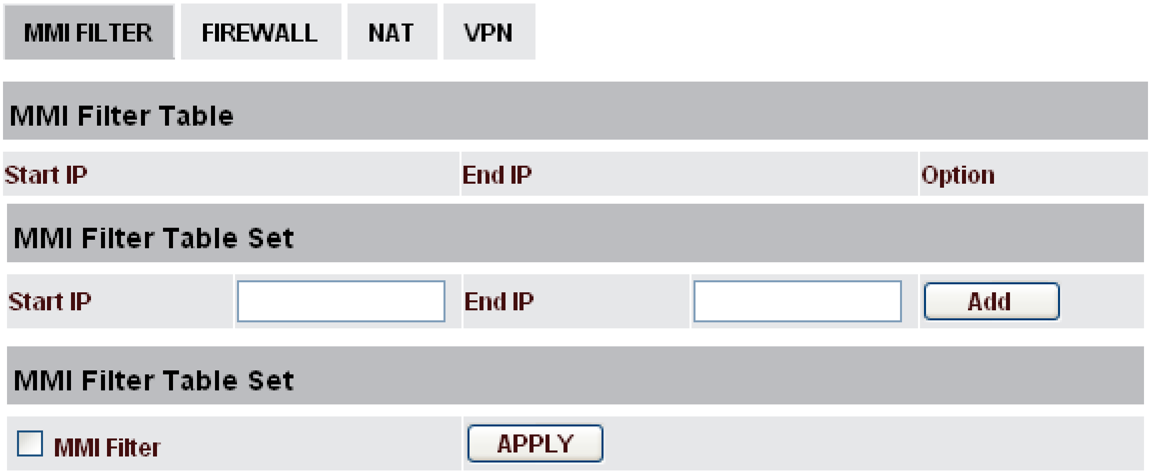


Notice: Before reboot, you need confirm that you have saved all configurations.

3.6 Security

3.6.1 MMI Filter

User could make some device own IP, which is pre-specified, access to the MMI of the phone to config and manage the phone.



Field name	Explanation
MMI Filter Table	MMI Filter IP Table list.
MMI Filter Table Set	Add or delete the IP address segments that access to the phone. Set initial IP address in the Start IP column, Set end IP address in the End IP column, and click Add to add this IP segment. You can also click Delete to delete the selected IP segment.
MMI Filter Table Set	MMI Filter - Select it or not to enable or disable MMI Filter. Click **Apply** to make it effective.

Do not set your visiting IP outside the MMI filter range, otherwise, you can not logon through the web.

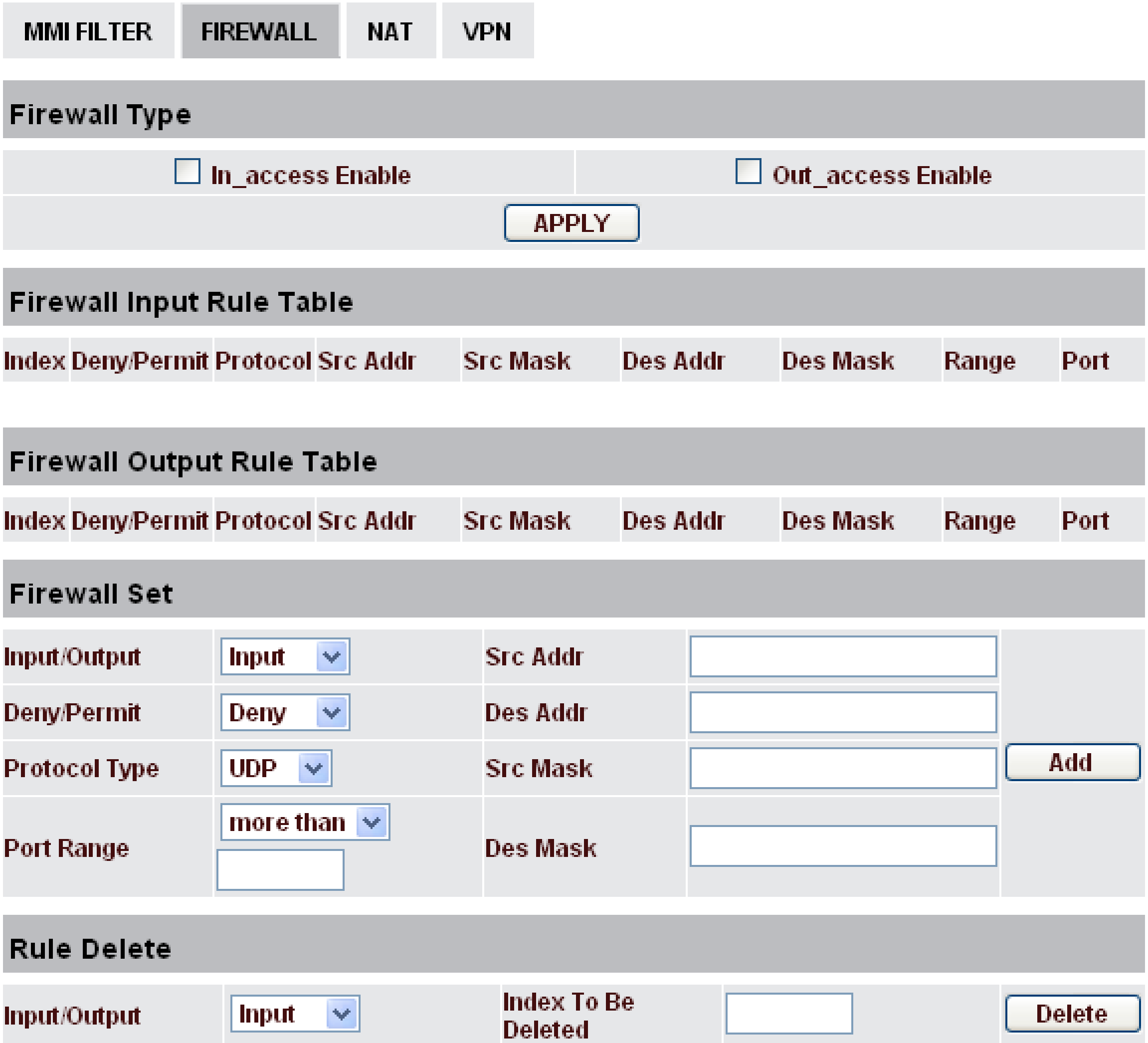
3.6.2 Firewall

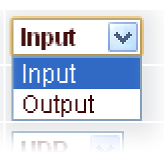
In this web interface, you can set up firewall to prevent unauthorized Internet users from accessing private networks connected to the Internet (input rule), or prevent unauthorized private network devices from accessing the Internet (output rule).

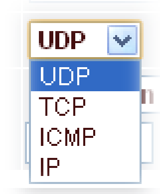
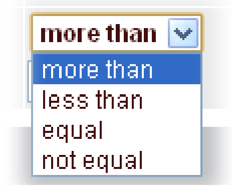
Firewall supports two types of rules: input access rule and output access rule. Each type supports at most 10 items.

Through this web page, you could set up and enable/disable firewall with input/output rules. System could prevent unauthorized access, or access other networks set in rules for security. Firewall, is also called access list, is a simple implementation of a Cisco-like access list (firewall). It supports two access lists: one for filtering input packets, and the other for filtering output packets. Each kind of list could be added 10 items.

We will give you an instance for your reference.



Field name	Explanation
In access enable	Select it to Enable in_ access rule
out access enable	Select it to Enable out_ access rule
Input/Output	Specify current adding rule by selecting input rule or output rule. 
Deny/Permit	Specify current adding rule by selecting Deny rule or Permit rule. 
Protocol Type	Filter protocol type. You can select TCP, UDP, ICMP, or IP.

	
Port Range	Set the filter Port range. 
Src Addr	Set source address. It can be single IP address, network address, complete address 0.0.0.0, or network address similar to *.*.*.0
Des Addr	Set the destination address. It can be IP address, network address, complete address 0.0.0.0, or network address similar to *.*.*.*
Src Mask	Set the source address' mask. For example, 255.255.255.255 means just point to one host; 255.255.255.0 means point to a network which network ID is C type.
Des Mask	Set the destination address' mask. For example, 255.255.255.255 means just point to one host; 255.255.255.0 means point to a network which network ID is C type.
Apply	Save the settings.
Delete	Delete the selected rule.

Click the **Add** button if you want to add a new output rule.

Firewall Input Rule Table

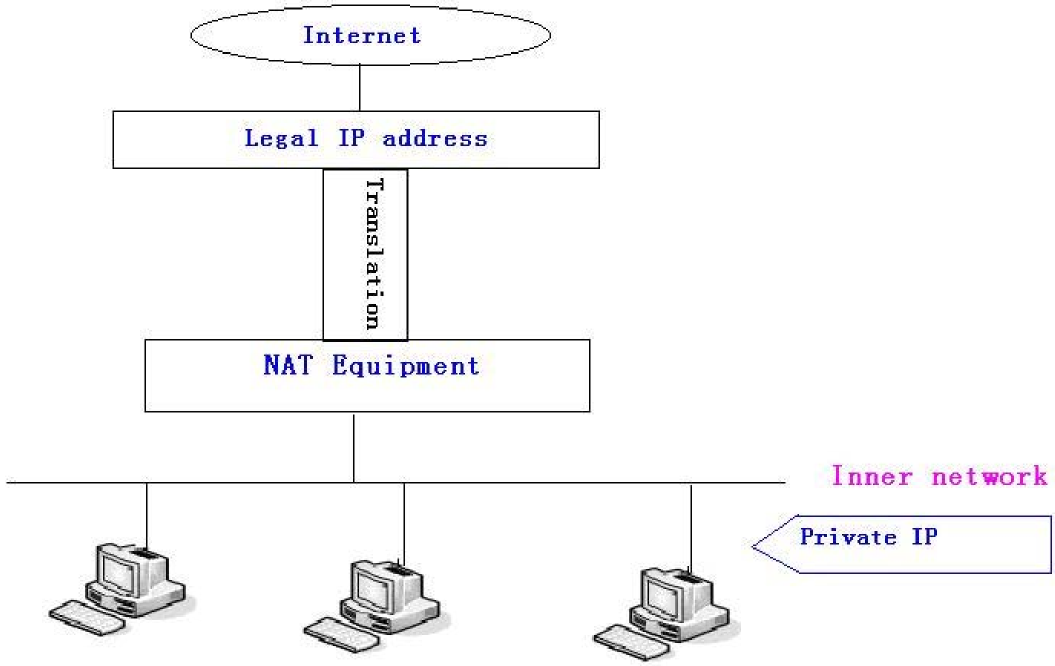
Index	Deny/Permit	Protocol	Src Addr	Src Mask	Des Addr	Des Mask	Range	Port
1	Deny	ICMP	192.168.1.14	255.255.255.0	192.168.1.118	255.255.255.0	More than	0

Then enable out access, and click the Apply button.

So when devices execute to ping 192.168.1.118, system will deny the request to send ICMP request to 192.168.1.118 for the out access rule. But if devices ping other devices which network ID is 192.168.1.0, it will be normal.

3.6.3 NAT

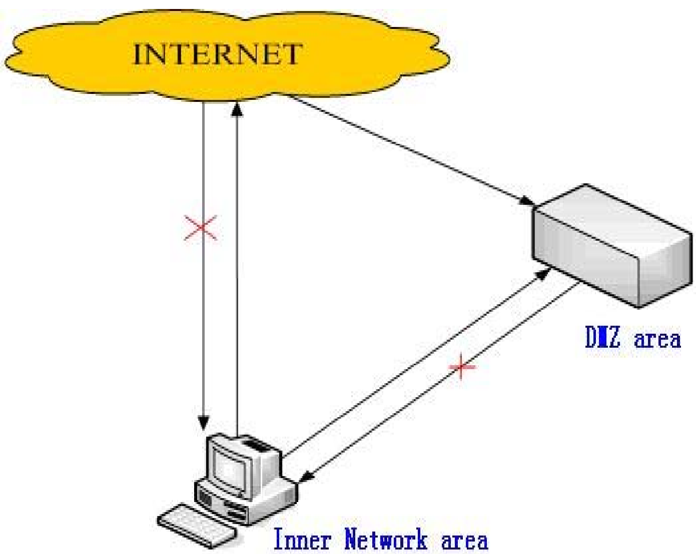
NAT is abbreviated from Net Address Translation; it's a protocol responsible for IP address translation. In other word, it is responsible for transforming IP and port of private network to public, also is the IP address mapping which we usually say.

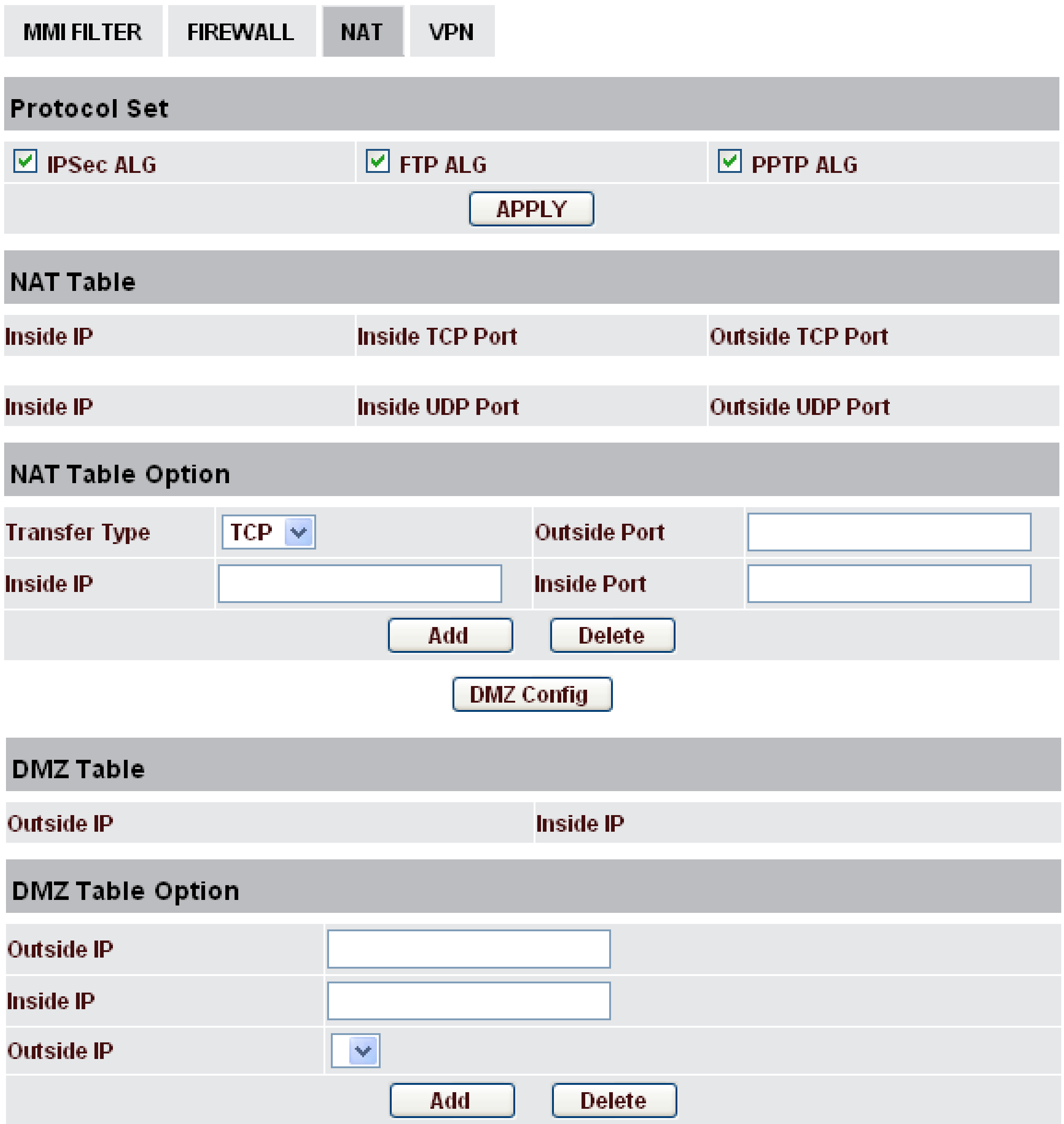


DMZ

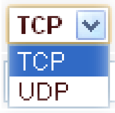
In order to make some intranet equipments support better service for extranet, and make internal network security more effectively, these equipments open to extranet need be separated from the other equipments not open to extranet by the corresponding isolation method according to different demands. We can provide the different security level protection in terms of the different resources by building a DMZ region which can provide the network level protection for the equipments environment, reduce the risk which is caused by providing service to distrust customer, and is the best position to put public information .

The following chart describes the network access control of DMZ:





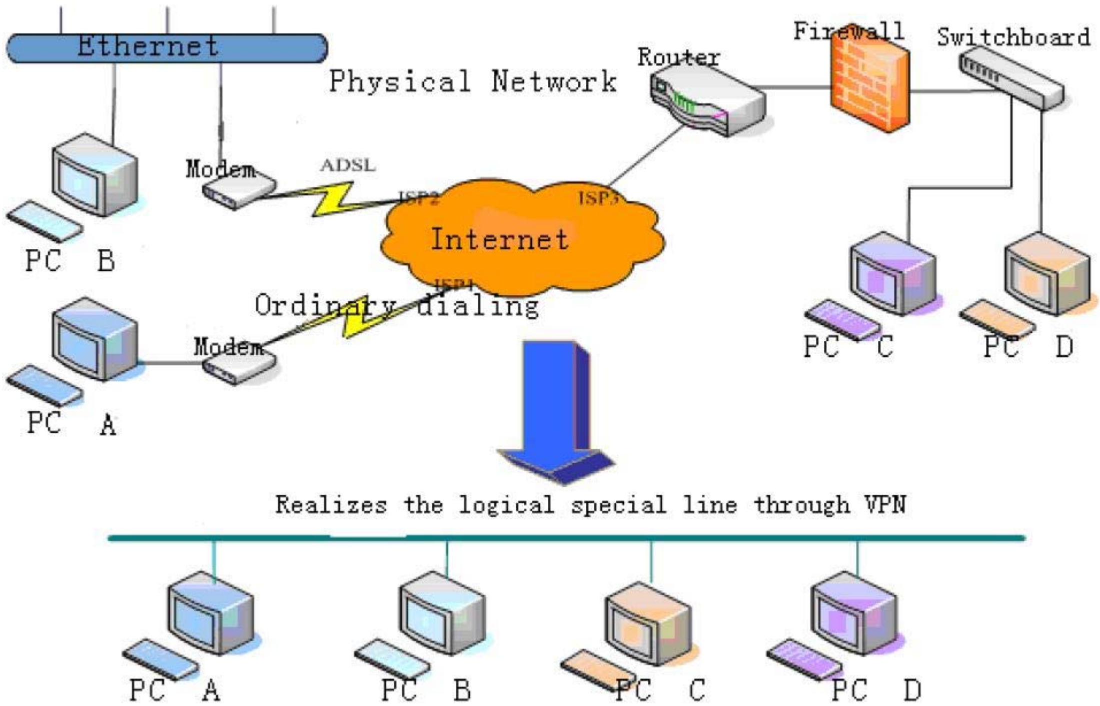
Field name	Explanation
Protocol Set	IPSec ALG - It is an encryption technology. Select it to enable IPSec ALG, the default is enabled. FTP ALG - FTP is a service of connection layer which can transform intranet IP into extranet IP when intranet IP is sending out packet. Select it to enable FTP ALG, the default is enabled. PPTP ALG - Select it enable PPTP ALG, the default is enabled.
NAT Table	Shows the NAT TC and UDP mapping table.
NAT Table Option	Transfer Type - Select the NAT mapping protocol style, TCP or UDP.

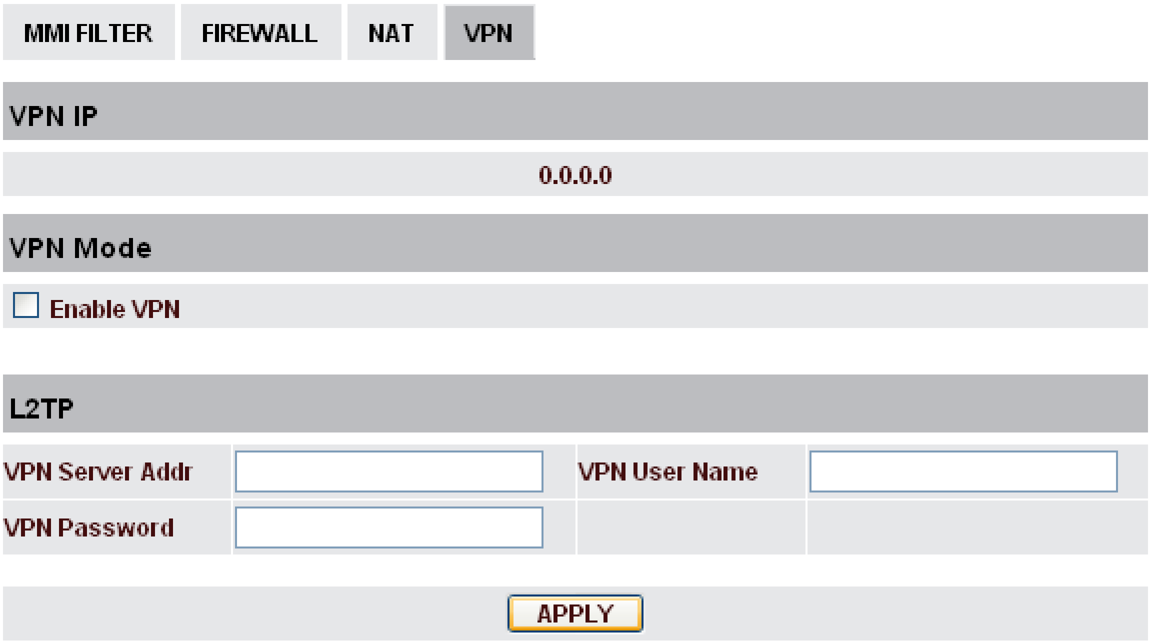
	 Outside Port - Set the WAN port of the NAT mapping. Inside IP - Set the IP address of device which is connected to LAN interface to do NAT mapping. Inside Port - Set the LAN port of the NAT mapping. **Notice:** After finish setting, click the Add button to add new mapping table; click the Delete button to delete the selected mapping table.
DMZ Table	Shows the outside WAN port IP address and the inside LAN port IP address.
DMZ Table Option	Outside IP - Set the outside Wan port IP address of DMZ. Inside IP- Set the inside LAN port IP address of DMZ. Click the **Add** button to add new table; click the **Delete** button to delete the selected mapping table.

Notice: 10M/100M adaptive means the network card, and other equipment physical consultations speed, testing speed under bridge mode near to 100M, in order to ensure the quality of voice and communications real-time performance, we made some sacrifices of NAT under the transmission performance. Transmit with full capability only when system is idle, so can not guarantee that the transmission speed reach to 100M.

3.6.4 VPN

This web page provides us a safe connect mode by which we can make remote access to enterprise inner network from public network. That is to say, you can set it to connect public networks in different areas into inner network via a special tunnel.

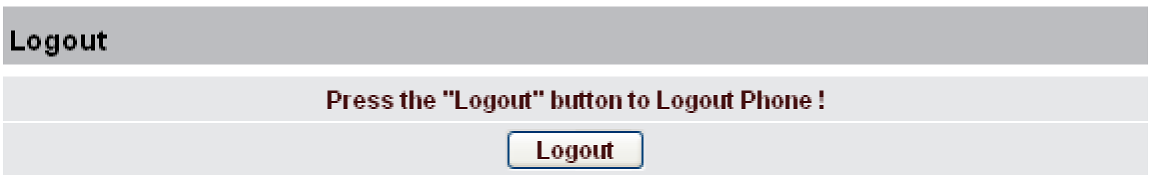




Field name	Explanation
VPN IP	Shows the current VPN IP address
VPN Mode	Enable VPN - Select it or not to enable or disable VPN.
L2TP	VPN Server Addr - Set VPN L2TP Server IP address. VPN User Name - Set User Name access to VPN L2TP Server. VPN Password - Set Password access to VPN L2TP Server.

3.7 Logout

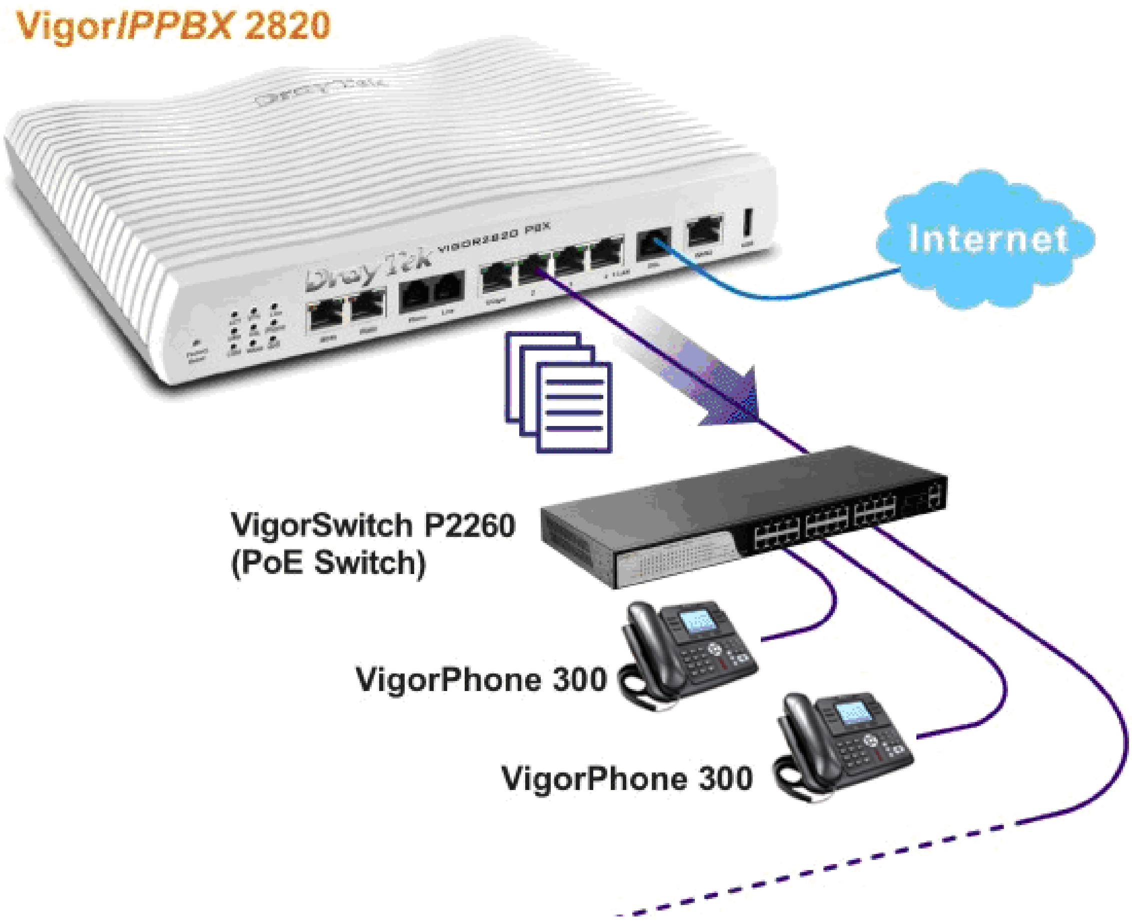
Click **Logout** and you will exit web page. If you want to enter it next time, you need input user name and password again.



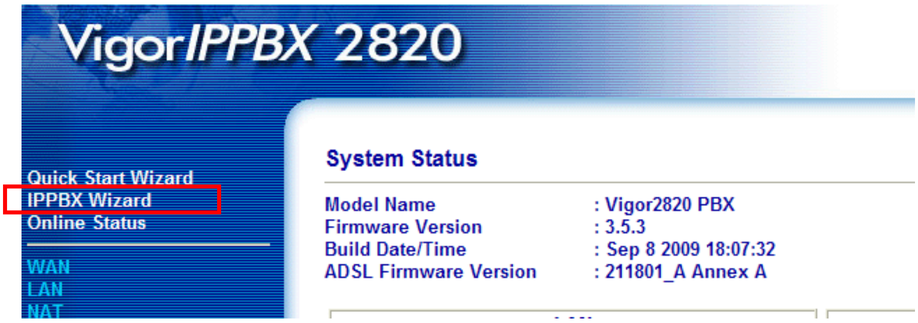
Chapter 4 Operation

4.1 Set up VigorPhone 300 with Vigor*IPPBX* Series

DrayTek Vigor*IPPBX* series supports the function of auto-provisioning. VigorPhone 300 is also capable of auto-provisioning, it can get a configuration text file from the Vigor*IPPBX* series. The configuration file contains SIP settings that the SIP devices can register with Vigor*IPPBX* series.



1. Configure the extension number and password for each IP phone on Vigor*IPPBX 2820*. You can configure extensions from IP PBX Wizard.



2. Click **IPPBX Wizard** to get the first screen as shown below:

IPPBX Wizard

Extension & Groups Setup : Index 1

Extension Group Name:	VigorPhone	(for example : sales)
Extension Group Number:	910	(for example : 100)
Start Number of the extension Group:	911	(for example : 101)
Number of extensions in this group:	10	(for example : 10, max = 20)
	OK	

Index	Group Name	Group Extension	Hunt List(Max 20 Extension)
1.	VigorPhone	910	911-920
2.			
3.			
4.			
5.			
6.			
7.			

3. Type the extension group name, group number, start number, and number of extension fields. Click **OK** to save them. The new added group will be displayed on the screen. Then click **Next** to access into next web page.

4. In the SIP Trunk Setup page, you can set up to six SIP profiles outside lines at one time.

IPPBX Wizard

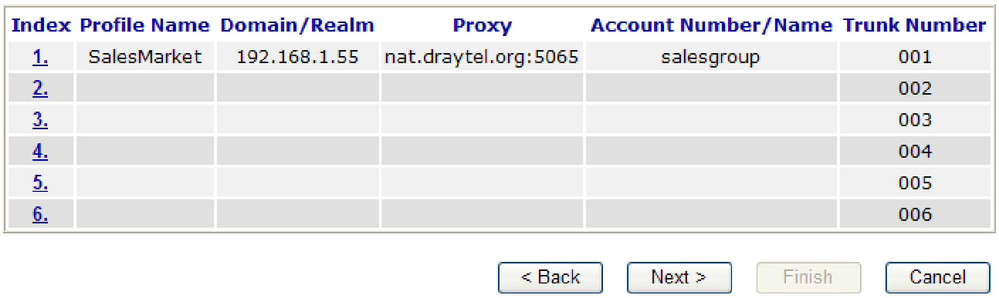
Sip Trunk Setup : Index 1

Profile Name:		(11 characters max.)
Domain/Realm:		(63 characters max.)
Proxy:		(63 characters max.)
Account Number/Name:		(63 characters max.)
Password:		(63 characters max.)
Trunk number:	001	(3 characters max.)
	OK	

Index	Profile Name	Domain/Realm	Proxy	Account Number/Name	Trunk Number
1.					001
2.					002
3.					003
4.					004
5.					005
6.					006

< Back | Next > | Finish | Cancel

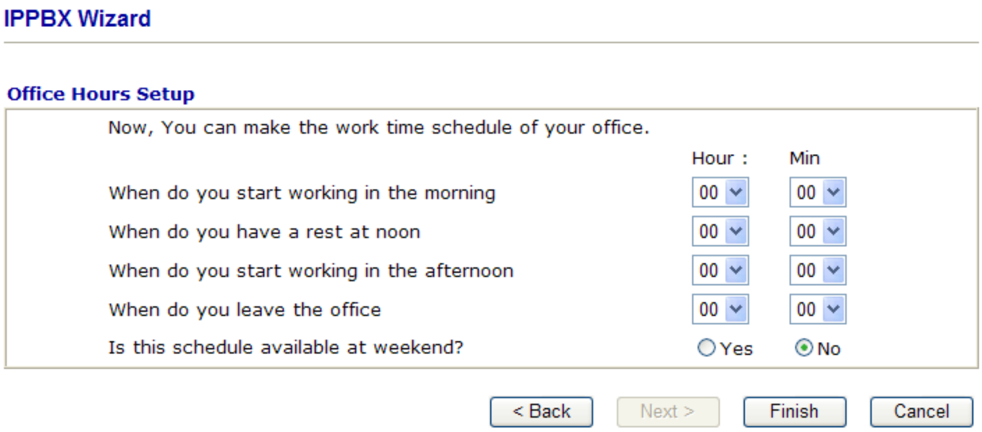
5. Type the profile name, domain/realm, proxy, account number/name, password and trunk number fields, then click **OK** to save them. The new added profile will be displayed on the screen.



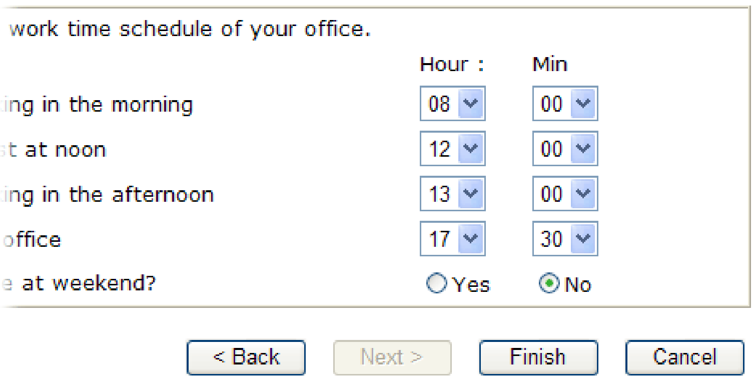
Index	Profile Name	Domain/Realm	Proxy	Account Number/Name	Trunk Number
1.	SalesMarket	192.168.1.55	nat.draytel.org:5065	salesgroup	001
2.					002
3.					003
4.					004
5.					005
6.					006

< Back   Next >   Finish   Cancel

6. Click **Next** to access into office hour setup page.

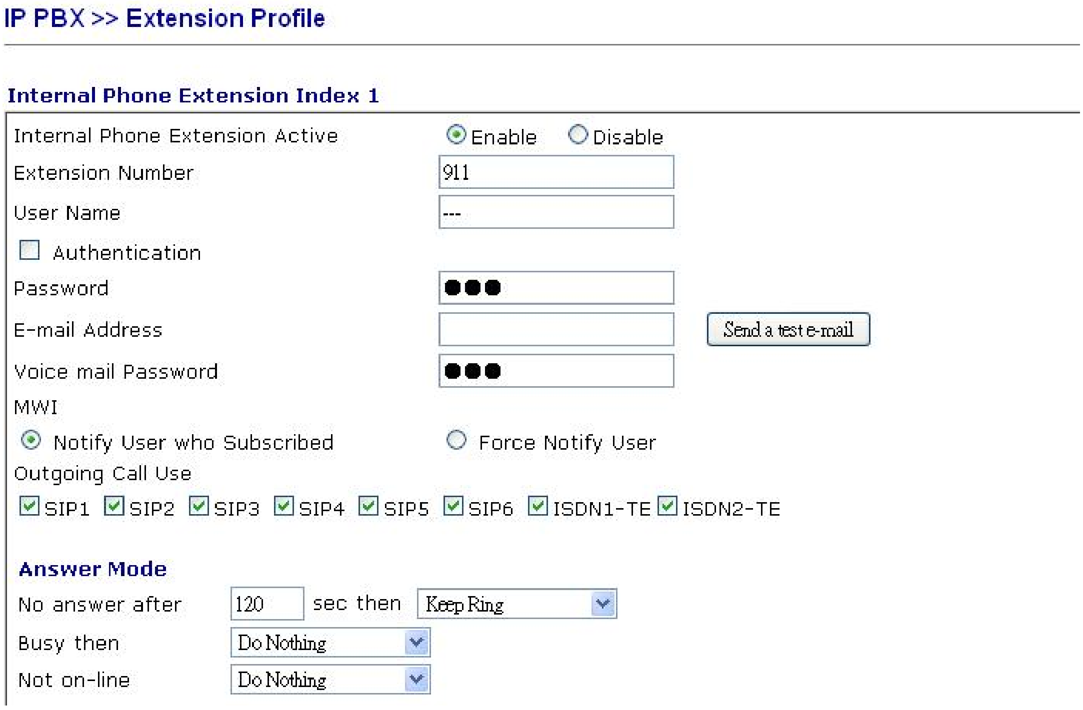


7. Please specify office hours including starting point and ending point on duty day(s).Then, click **Finish** to save the settings and exit the wizard.



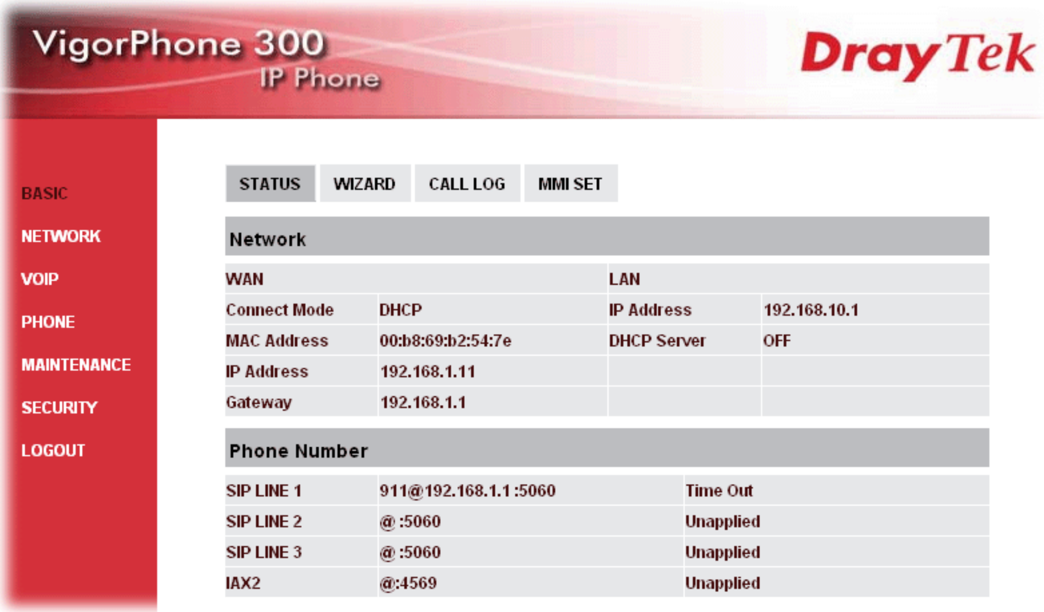
8. After finishing the Wizard, please go to **IPPBX>Extension** to configure the **Extension Number** and the **Password** settings. Click the index number 1.

9. Type in **Extension Number** and **Password.**

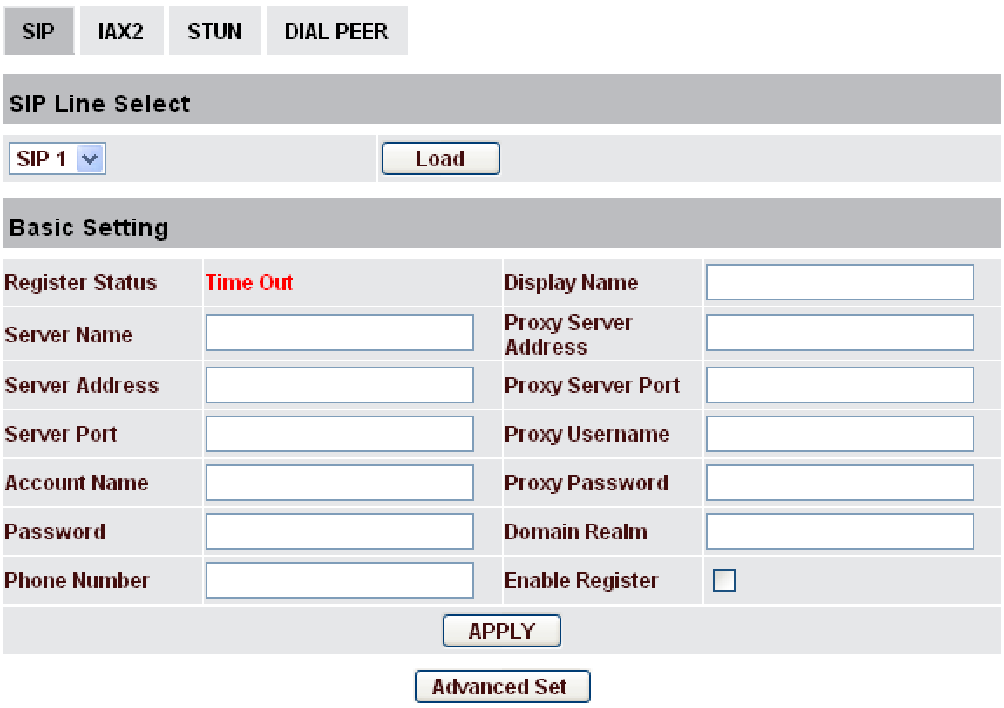


10. Then connect VigorPhone to the network. Each user of VigorPhone can get the extension number/password respectively.

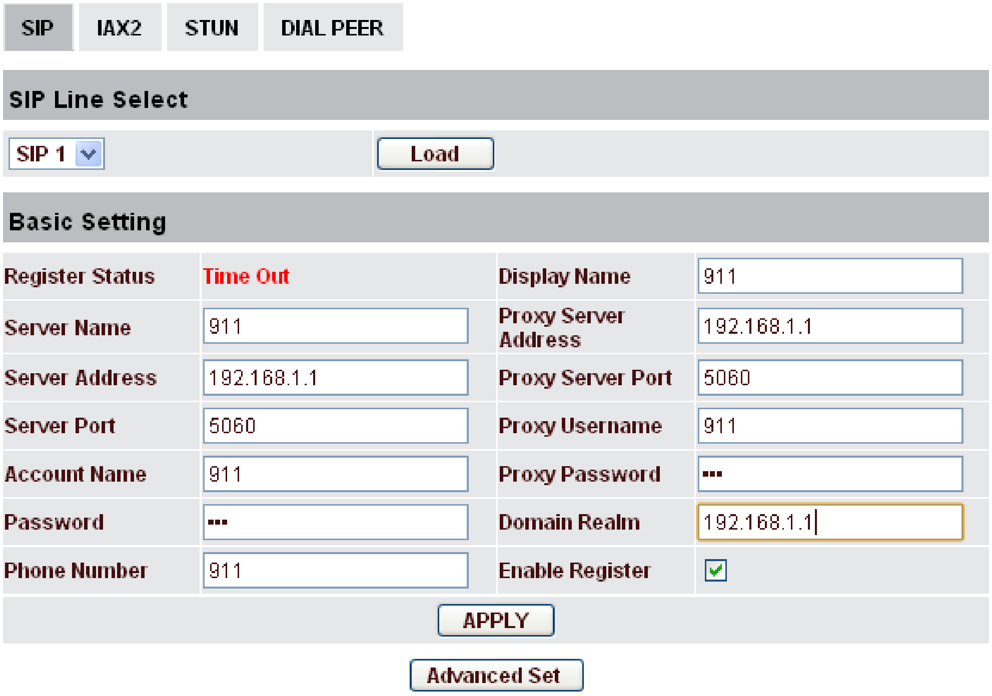
11. Access into the web configurator of VigorPhone 300 (e.g., 192.168.1.11).



12. Open **VoIP** and press the **SIP** tab to display the following page.

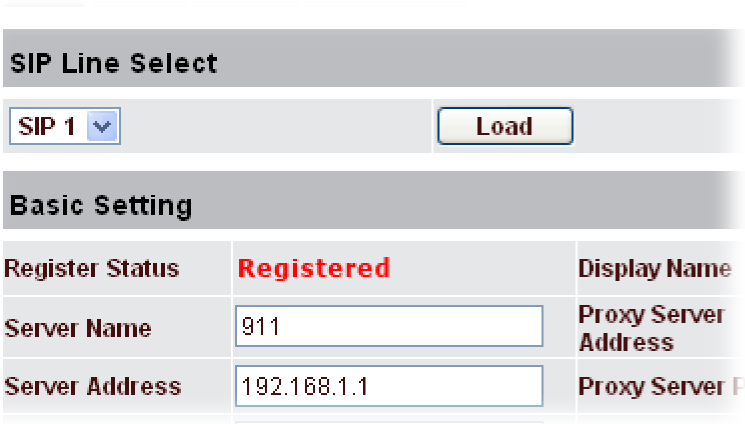


13. Fill in the information according to the settings (listed in Step 1 to Step 9) configured in VigorIPPBX series.



14. When you finished the settings, click **Apply** to save them. VigorPhone will try to register the number to VigorIPPBX series.

15. Later, if **Register Status** display "Registered", that means the extension number for VigorPhone has been registered successfully.



4.2 Answer Call

VigorPhone 300 will ring to indicate you when there is call incoming, below is the ways to answer call:

- **Answer with hook off**

  Take handset, you can talk directly. You can just hang up to finish talk.

- **Answer with the headset button**

  Press the **Headset** key  to answer the call, press the key again to finish talk.

- **Using handset instead of hands-free during a talk**

  Hook off the handset when you use hands-free and want to change to use handset. Just hook on to finish talk.

- **Using headset instead of hands-free during a talk**

  In the hands-free calls, press the **Headset** key . You can use the headset to call. After that, press the key again to hang up the call.

- **Using headset instead of handset during a talk**

  In the handset call, the **Headset** key , hang up the handset to continue using the headset call. After the call, press the key again to cut off the call.

- **Using handset instead of headset during a talk**

  In the headset call, hook off the handset after the call, just hook on to finish talk.

4.3 Place Calls

- **Using handset**

  Hook off (screen will show the current using line, or you could press key L1-L3 to select), after getting dialing tone, you could begin to dial number. After finishing it, press # and the IP phone will send the number and call the number. When you hear a ring-back tone and screen shows the callee's number, it shows that the person you called is ringing. If a callee answers the call, you can begin to talk and your phone will keep showing the callee's number and counting time. Just hang up to finish talk.

- **Using headset**

  Standby, press the **Headset** key (on screen display "Enter Number Pls") and hear the dialing tone, you can start dialing. After finishing it, press # or press the softkey2-Send.

  IP Phone can immediately begin connecting with each other. When you hear a ring-back tone and screen shows the callee's number, it shows that the person you called is ringing. If callee answers the call, you can begin to talk and your phone will keep showing callee's number and counting time. Just press [Headset] key to finish talk.

- **Using hands-free**

  Press the **Hands-free** key （screen will show the current using line, or you could press key L1-L3 to select), after getting dialing tone, you could begin to dial number. After finishing it, press # and the IP phone will send the number and call the number. When you hear the ringback tone and screen shows the callee's number, it shows that the person you called is ringing. If the callee answers the call, you can begin to talk and your phone will keep showing callee's number and counting time. Press again to finish talk.

- **Using directory**

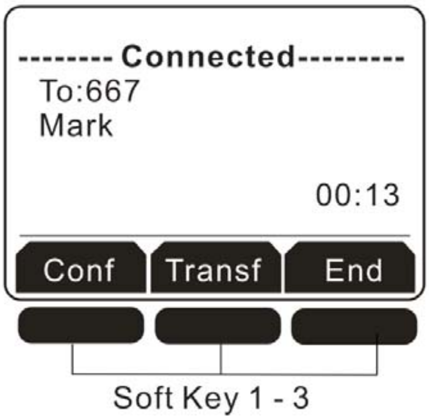
  Press Soft3 (PBook) in stand-by mode, you will access to phonebook. If there are many persons records stored in the directory, you can use navigation keys & to select number or press the first character of the name for searching the person which you want to contact. Press  to forward and press  to backward. Press Soft2 (Dial) to dial the current number shown on the screen.

- **Speed dial**

Speed dial means user can make calls directly without hook off or using hands-free. User can dial number in stand-by mode, but first, user need to add and edit SDial no. By pressing Soft2 (SDial) to edit and save the number to be an SDial number. In this way, user could make a call only press the number and Soft3 (Dial).

- **Multiple-way call**

If a user has 2 line calls and wants to invite the third party during the call, he/she can press Soft1 (Conf) or Soft2(Transf) “New CALL”, press Soft1(OK),enter the number ,then press Soft2(Send) and wait for the other party to answer. When the multiple-way calls, you can press the arrow keys to select a call.



4.4 End Calls

- **Hang up with handset hook on**

Hook on to finish talking.

- **Hang up with hands-free**

Press the **Hands-free** key  to finish talk when phone is in hands-free status.

- **Hang up with headset**

If you are in the headset call, press the **Headset** key  to end the call.

- **Hang up an active call with 2 calls**

When there are two calls，user might use Soft1(Switch)to switch to the call you want to hang up first. Then press Soft3 (Close) to finish talk, and phone will switch to the other call automatically.

4.5 Call Transfer

1. **Blind Transfer**

   During talk, press Trans or Soft2 (Transf), and then dial the number that you want to transfer to, and finished by "#". Phone will transfer the current call to the third party. After finishing transfer, the call you talk to will be hanged up.

2. **Attended Transfer**

   During talk, press Trans or Soft2 (Transf), then input the number that you want to transfer to and press Soft2 (Send). After that third party answers, then press Trans to complete the transfer. (You need enable call waiting and call transfer first). If there are two calls, you can just talk to one, and keep hold to the other one. The one who is keep hold can not speak to you or hear from you.

3. **Alert Transfer**

   During the talk, press Trans or Soft2 (Transf) firstly, then press Soft2 (Send) after inputting the number that you want to transfer. You are waiting for connection, now, press Trans or Soft2 (Transf) and the transfer will be done. (To use this feature, you need enable call waiting and call transfer first)

4.6 Call Hold

During talking, user could press Hold to hold the current call. Press Hold again to return the call or switch the call active.

4.7 3-way Conference Call

User can press Soft1 (Conf) to dial the line2 (press Soft1 (Answer) to answer the call directly if this call is from line2) during talking with line1. After line2 connect, user can press Soft2 (Conf) select another way into the three-way calling number, then press softkey1-OK to enter into conference mode. To back to line1 from conference, please press Soft1 (Split); to end the call, please press Soft3 (Close) or press RLS.

4.8 Call Records

The IP Phone supports 100 items of missed call, 100 items of incoming call, and 100 items of dialed call. If the records are full, the newest will replace the oldest. If phone's power cut or reboot, call records will be discarded.

- **Missed call**

  Press and screen displays "Missed Call" with the number and time of missed call. User can also use & to browse the missed call records, or press Soft1 (Option) to check the details of this record, then press Soft2 (EDial) again to change the current number. Pressing Soft2 (Dial) will call this number directly if user don't modify the number. If there is no missed call, screen will show "List Is Empty".

- **Incoming call**

Press and screen displays "Incoming Call", by pressing & to browse the records; or press Soft1 (Option) to check the details of this record, then press Soft2 (EDial) again to change the current number. Pressing Soft2 (Dial) will call this number directly if user don't modify the number. If there is no incoming call, screen will show "List Is Empty".

- **Dialed call**

Press , and use & to browse the dialed call records; or press Soft1 (Option) to check the details of this record, then press Soft2 (EDial) again to change the current number. Pressing Soft2 (Dial) will call this number directly if user don't modify the number. If there is no dialed call, screen will show "List Is Empty".

4.9 Special Keys

- **SMS function**

In the standby mode, press Softkey1-SMS, then press Soft1 (new) key. After inputting SMS content, press Soft2 (send) key to input callee's number, next, press Soft2 (OK) again to send SMS.

When user has new message, the phone will ring, there is a coin on the screen. Press softkey1-SMS, select inbox use up/down key, and then press softkey2-OK. When a number of text messages, users can use up/down key and press softkey2-Enter to select one to view. Press softkey2-Reply and input message content, finally, press Soft2(Send) again to reply this message. The phone can also send messages by phonebook.

Note: while user browses the message numbers, new messages will be marked by "new"; when a user edits message, press # key that to switch input method, e.g. ABC (uppercase English input), abc (lowercase English input), 123(digit input), Korean (Korean input(if your phone's firmware version supports Korean). PY (if your phone's firmware version supports Chinese) .

- **SpeedDial function**

User can pre-define numbers in these keys (numeric key 0-9). Hook off, press the defined numeric key, and then input "#". Your pre-defined numbers will send out.

Press softkey2-SDial to set speed dial in standby, a total of 12 numbers, users can select by memory key. Users can delete and press # key that to switch input method.

Note:

1. First 9 numbers corresponding digit key 1-9, 10th number corresponding digit key 0.
2. The first 10 set of numbers in standby mode press the corresponding number key and then press softkey3-Dial or key to exhale, but the first 11 groups and 12 group numbers without the corresponding number key is required to enter SDial menu to find the set of numbers by Corresponds memory key or softkey3-Dial button to exhaled.

- **Realize Secondary Dial by Dialing for only one time**

When you make secondary dial in off-hook/hands-free or standby pre-input mode, press [hold] button to postpone input, and screen display will show ^. One stands for 2 seconds. For example, you input 123^45, the phone will send DTMF (45) 2 seconds after the phone call 123. 123^^^45 will make phone send DTMF (45) at 6 seconds interval

- **Message waiting indication**

  After you set it, you can pick up or hands-free, then press  to listen to record in server when you have new voice message.

- **Phone book search function**

  In the Chinese version, users can be retrieved by the corresponding initials Chinese name, which simplifies the steps in the phone book to find contacts.

  For example, contact name is Zhang San, contact number is 123. When you enter the phonebook, you can press 9 key to select letter z, all the numbers of beginning with z will be displayed on the LCD. You can select the one you want to search by press up/down key.

4.10 Call Pickup

Call pickup is implemented by simulating pickup function of IPPBX. It's that, when A calls B, B rings but no answer. At this moment, C can hook off and input an appointed prefix plus B's number, pick up A's call and talk with A.

The following chart shows how to configure an appointed prefix in dial peer to have call pick up function. (Configuration in Dial Peer)

SIP	IAX2	STUN	DIAL PEER

Dial Peer Table

Number	Destination	Port	Mode	Alias	Suffix	Del Length
*1*T	0.0.0.0	5060	SIP	rep:pickup	no suffix	3

1 means appointed prefix code. After making the above configuration, C can dial *1* plus B's phone number to pick up A's call. User can set prefix in random, in the case of no affecting current dialing rules.

4.11 Join Call

When B is calling C, A can join in the existing call by inputting an appointed prefix numbers plus B or C number, if B or C also supports join call.

The following chart shows how to configure an appointed prefix in dial peer to have join call function. (Configuration in Dial Peer)

SIP	IAX2	STUN	DIAL PEER

Dial Peer Table

Number	Destination	Port	Mode	Alias	Suffix	Del Length
*2*T	0.0.0.0	5060	SIP	rep:joincall	no suffix	3

2 means appointed prefix code. After making the above configuration, A can dial *2* plus B or C number to join B and C's call. User can set prefix in random, in the case of no affecting current dialing rules.

4.12 Redial/Un-redial

If B is in busy line when A calls B, A will get notice: busy, please hang up. If A want to connect B as soon as B is in idle, he can use redial function at the moment and he can dials an appointed prefix number plus B’s number to realize redial function.

What is redial function? A can’t not build a call with B when B is in busy, then A will subscribe B’s calling mode at 60 second intervals. Once B is available, A will get reminder of rings to hook off, while A hooks off, A will call B automatically. If at this time A is occupied temporarily and unwilling to contact B, A also can cancel the redial function by dialing an appointed prefix plus B’s number before making the redial function.

SIP | IAX2 | STUN | DIAL PEER

Dial Peer Table

Number	Destination	Port	Mode	Alias	Suffix	Del Length
*3*T	0.0.0.0	5060	SIP	rep:redial	no suffix	3
*4*T	0.0.0.0	5060	SIP	rep:unredial	no suffix	3

3 is appointed prefix code. After making the above configuration, A can dial

3 plus B’s phone number to make the redial function.

4 is appointed prefix code. After configuration, A can dial *4* to cancel redial function.

User can set prefix in random, in the case of no affecting current dialing rules.

4.13 Click to Dial

When the user A browses in an appointed Web page, user A can click to call user B via a link, then user A’s phone will ring. After user A hooks off, the phone will dial to user B.

Appendix A Specifications

A.1 Specification

A.1.1 Hardware

Item		Description
Adapter (Input/Output)		Input: 100-240V Output: 5V 1A
port	WAN	10/100Base- T RJ-45 for LAN
	LAN	10/100Base- T RJ-45 for PC
Power Consumption		Idle: 2.5W/Active: 2.8W
LCD Size		128x96 53.5 x 70mm
Operation Temperature		0～40℃
Relative Humidity		10～65%
CPU		Broadcom
SDRAM		16MB
Flash		4MB
Dimension(L x W x H)		11.6×8×3 in.(295×205×75mm)
Weight		0.955kg

A.1.2 Voice features

- SIP supports 3 SIP servers
- Support SIP 2.0 (RFC3261) and correlative RFCs
- Codec: G.711A/u, G.723.1 high/low, G.729a/b，G.722， G.726
- Echo cancellation: G.168 Compliance in LEC, additional acoustic echo cancellation(AEC) can reach 96ms max filter length in hands-free mode
- Support Voice Gain Setting, VAD, CNG
- Support full duplex hands-free
- HD Voice
- SIP support SIP domain, SIP authentication(none basic, MD5), DNS name of server, Peer to Peer/ IP call
- Automatically select calling line, if one line can’t be connected, the phone can automatically switch to other line to call.
- DTMF Relay: support SIP info，DTMF Relay，RFC2833
- SIP application: SIP Call forward/transfer（blind/attended）/hold/waiting/3 way talking/sms /pickup /joincall /redial /unredial/multi line
- Call control features: Flexible dial map, hotline, empty calling No. reject service, black list for reject authenticated call, limit call, no disturb, caller ID, Flexible deer peer rule.
- Support phonebook 500 records，Incoming calls / outgoing calls / missing calls. Each supports 100 records
- Support IAX2
- Phonebook supports vcard standard
- 12/24 hours time display
- Support daylight saving time
- Support path

- Support SIP Privacy
- Support SMS
- Support WMI
- Support Speed dial
- Support XML

A.1.3 Network features

- WAN/LAN: support bridge and router model
- Support PPPoE for xDSL
- Support basic NAT and NAPT
- Support VLAN (optional: voice vlan/ data vlan)
- NAT Penetrate, Stun Penetrate
- Support DMZ
- Support VPN (L2TP) function
- Wan Port supports main DNS and secondary DNS server, can select dynamically to get DNS in DHCP mode or statically set DNS address.
- Support DHCP client on WAN
- Support DHCP server on LAN
- QoS with DiffServ
- Network tools in telnet server: including ping, trace route, telnet client

A.1.4 Maintenance and management

- Upgrade firmware through POST mode
- Web ,telnet and keypad management
- Management with different account right
- LCD and WEB configuration can be modified into requested language, and support multi-language dynamically shifted
- Upgrade firmware through HTTP, FTP or TFTP Telnet remote management/ upload/download setting file
- Support Syslog
- Support Auto Provisioning (upgrade firmware or configuration file)

A.1.5 Special features

- Support 3 softkeys, 6 memory keys, Navigation key.
- RLS,Pbook,MWI,HOLD,Trans,Mute,L1-L3,Vol -/+,Redial

A.2 Digit-character Map Table

Keypad	Character	Keypad	Character
	1 @		7 P Q R S p q r s
	2 A B C a b c		8 T U V t u v
	3 D E F d e f		9 W X Y Z w x y z

	4 G H I g h i		*/.
	5 J K L j k l		0
	6 M N O m n o		#/=